

Mercury Core V5 Gateway Configuration Editor Guide

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About this Guide

The Mercury Configuration Guide explains how to use Gateway, the Configuration Editor to design, build and deploy Mercury configurations.

While this guide is not intended to teach how to build a specific configuration, it does provide information about the configuration of all of the Mercury system components and the individual settings which may be changed to provide the required functionality.

Please take some time to read through this guide since it will help you understand and use Gateway.

Intended Audience

The information presented in this guide is written for designated Mercury System Administrators with responsibility for planning and configuring a Mercury communication system. Skills required include a good knowledge of IT and networks, plus an appreciation of the communication needs of the organisation.



Acronyms and Abbreviations

AGC Audio Expansion Board
AGC Automatic Gain Control

API Application Programming Interface

BMP Windows Bitmap

COR Carrier Operated Relay

CTCSS Continuous Tone-Coded Squelch System

DDI Direct Dial In

DEP Data Execution Prevention

DHCP Dynamic Host Configuration Protocol

DNS Domain Name System/Server
DSCP Differentiated Services Code Point

DSP Digital Signal Processor / Digital Signal Processing

DTMF Dual-Tone Multi-Frequency

E&M Ear and Mouth **EC** Echo Cancellation

EDHS Enterprise Domain Host Subscriber

FXO Foreign Exchange Office
FXS Foreign Exchange Station
GPI General Purpose Input

GPIO General Purpose Input Output
GPO General Purpose Output
GUI Graphical User Interface
HTTP Hypertext Transfer Protocol

ICO Icon file
ID Identity

IEC International Electrotechnical Commission
IEEE Institute of Electrical and Electronics Engineers

IFB Interruptible FoldBacks

IGMP Internet Group Management Protocol

IP Internet Protocol
LAN Local Area Network
MAC Media Access Control
MCF Mercury Configuration File
MIU Mercury Interface Unit
MTBF Mean Time Between Failure

NAT/PAT Network Address Translation /Port Address Translation

NetBIOS Network Basic Input/Output System

OCC Open Collector
OFH Off Hook

PABX Private Automatic Branch Exchange

PBX Private Branch Exchange

PCI Peripheral Component Interconnect

PHB Per Hop Behaviour

POTS Plain Old Telephone Service
PSTN Public Switch Telephone Network

PSU Power Supply Unit



PTT Push To Talk

QoS Quality of Service

RAM Random Access Memory

RH Relative Humidity

RIB Radio Interoperability Board

RJ11 Registered Jack 11
RJ45 Registered Jack 45
RLL Reliable Link Layer
ROM Read-Only Memory

RTP Real-time Transport Protocol
SDK Software Development Kit
SFST Safety of Flight Sidetone
SIP Session Initiation Protocol
SVGA Super Video Graphics Array

TBC Talkback Controller

TCP Transmission Control Protocol

TCP/IP Transmission Control Protocol / Internet Protocol

TEB Telephone Expansion Board
TFTP Trivial File Transfer Protocol
TPS Trilogy (GPS) Positioning System

TTL Transistor-Transistor Logic

UDP User Datagram Protocol

UPS Uninterruptible Power Supply

URI Uniform Resource Identifier

URL Uniform Resource Locator

USB Universal Serial Bus

VAD Voice Activity Detector/Detection

VID VLAN ID

VIF Voice Information Field
VLAN Virtual Local Area Network
VoIP Voice Over Internet Protocol
VOX Voice Operated Transmission

VPAN Virtual Panel

VPN Virtual Private Network
WAN Wide Area Network
WMF Windows Metafile



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Chapter 1 - Introduction

Overview

The Configuration Editor is used to model and implement the structure of your Mercury system. To begin defining the structure and connectivity of the system, you need to put in place all of the required domains, groups, conferences, Interruptible FoldBacks (IFBs), Session Initiation Protocol (SIP) connections zones, areas, and tones that are required. This is done using the Enterprise Editor.

A unique addressing system is used in the Mercury system. This is known as the Enterprise Domain Host Subscriber (EDHS) system. The host is the main element of the system and subscribers make use of the host's facilities to provide audio communications. Hosts are grouped together into logical groups, called "domains". These are predominantly groupings of convenience, allowing a system to be segregated into separate areas. There is a close analogy between this definition of domains and Internet Protocol (IP) network domains, since it may be appropriate to run each area, or building, on a different subnet.

The Enterprise level sits above the domain level, allowing domains to be grouped into a larger entity. Every host in the system has its own unique EDHS address which is assigned by Gateway and entered during host installation. This makes it possible to cross-reference an EDHS address to an IP address; while the IP address may change, the EDHS remains the same. User interfaces can be configured to allow communications between EDHS entities. Refer to the Hardware Panels and Virtual Panels chapters in the Mercury Implementation Guide for details.

Once all of basic objects have been defined in your Enterprise, you must then configure the domains and add hosts using the Domain Editor. Subscribers and panels can then be added to the defined hosts using the Host Editor, while the host settings, including the IP addresses of the audio card and the host itself, can be defined. Routes are added using the Routes Editor. These routes can then be used when defining General Purpose Input Output (GPIO) processes. Subscribers are configured using the Panel Configuration Editor. Here the Administrator allocates keys corresponding to the calls that you wish to make and receive.

The Conference Editor is used to specify members of a conference and to define all of the hosts that may need to take part in a conference at some time. The Groups Editor is used to define group membership. Zone control can be applied to your configuration, if required. Implementing zones and areas is a way to control access to sources.

Once the complete configuration has been defined and saved, it is transferred to the hosts and the subscribers on the system using open database connections. Uploading the settings sends them to their destinations where they are cached. To make these changes active (i.e. replace the working files with the cached version), you must instate or activate the changes. A configuration can be changed at any time to reflect changes in the Enterprise, and transferred to the hosts before being instated or activated.



Next Steps

For information on the installation of Gateway, please refer to the Mercury Implementation Guide.

Before you start creating your configuration and defining the Enterprise, it is important that you fully understand the structure of the Enterprise that you want to model using Gateway. Taking time before you start, avoids problems later on.



NOTE: You can create your configuration in any order but we would recommend that you create the physical structure first (domains - hosts - subscribers) and then add in virtual entities such as groups, conferences etc.

- Define the basic elements of your Mercury configuration using the Enterprise Editor. See the "Enterprises and the Enterprise Editor" chapter in this Configuration Guide.
- Define operators and permission sets using the Operators Editor and Permission Set Editor. See the "Operators and the Operators Editor" and "Permissions and the Permission Set Editor" chapters in this Configuration Guide.
- Define Mercury hosts and the gateway IP address using the **Domain Editor**. See the "Domains and the Domain Editor" chapter in this Configuration Guide.
- Define the subscribers and the host settings using the Host Editor.
 See the "Hosts and the Host Editor" chapter in this Configuration Guide.
- Add members to groups using the Groups Editor. See the "Groups and the Groups Editor" chapter in this Configuration Guide.
- Add sources to a conference using the Conference Editor. See the "Conferences and the Conference Editor" chapter in this Configuration Guide.
- Add routes to the Mercury system using the Routes Editor. See the "Routes and the Routes Editor" chapter in this Configuration Guide.
- Define routes and triggers using the **IFB Editor**. See the "IFBs and the IFB Editor" chapter in this Configuration Guide.
- Define GPIO processes using the GPIO Processes Editor. See the "GPIO and the GPIO Processes Editor" chapter in this Configuration Guide.
- Add frequently used phone numbers that can be used locally on a panel using the **Phone Number Editor**. See the "Phone Numbers and the Phone Number Editor" chapter in this Configuration Guide.
- Phone sets can also be defined using the Phone Number Editor and changed using the Phone Sets Editor. See the "Phone Sets and the Phone Sets Editor" chapter in this Configuration Guide.
- Add phone controls to SIP connections using the SIP Connection Editor. See the "SIP and the SIP Connection Editor" chapter in this Configuration Guide.
- Define panel sources and configure subscribers using the Panel Configuration Editor. See the "Panel and Subscriber Configuration" chapter in this Configuration Guide.



Chapter 2 - Gateway Basics

Overview

Gateway, the Configuration Editor is a Windows application used to manage the configuration details in the Mercury database. Use it to model and implement the structure of your Mercury system and to upload configuration data to Mercury hosts on the network.

To begin, create a new empty configuration and then use the individual editors to define and configure your Mercury system. For details, refer to the relevant chapter in this Configuration Guide.

To safeguard your work, you can backup a configuration at any time. Previous configurations can then be loaded back into Gateway as required. Once a configuration is complete, you can use Gateway to upload it to the Mercury hosts that you have defined.

Before You Start

Typically one computer is designated as the configuration management computer. Any changes that are made to the configuration of the Enterprise using this computer are then transferred to all other hosts within the Enterprise.

Refer to the **Installing the Database** chapter in the Implementation Guide for more information on database configuration.



NOTE: While it is possible to install Gateway on more than one computer within the Enterprise, this is not recommended initially, since it makes fault finding and tracking difficult.

Unless a separate PC or server is specified to host the Mercury database, the configuration software and database run on the same machine. The configuration software can, at a later date, be installed on other machines in the Enterprise, if required.

Once Gateway is installed, you should always ensure that the Database Supervisor software is running on the master machine. This is installed as a service within Windows and should therefore start automatically. You can check that the Database Supervisor is running in one of two ways:

- Look for the **Database Supervisor GUI** icon in the system tray. If the service is running you will see in the bottom right of your screen. Note that the management GUI is an optional component, so the second method provides a more definitive result.
- Open the Windows Services window (accessed via Administrative Tools from the Control Panel) and ensure that Mercury DB Server service is started.



Creating a Configuration

Gateway provides you with a view on the underlying database tables and allows you to add, delete and maintain records in the database.

To create a configuration that models your Enterprise:

- Make sure you understand the structure of your Enterprise in terms
 of domains so you can create a plan that includes details of the hosts
 and subscribers in the Enterprise.
- Input the information from your plan using Gateway. Gateway provides a logical tree view of the items that you need to configure.
- We recommend that you start at the top level and work logically down through the various elements, such as domains and groups, which together make up a configuration.
- Transfer the configuration to all destinations within your Enterprise.
 The configuration can be tested and deployed incrementally, if required.

Using Gateway

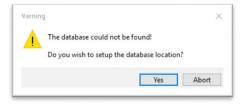
Starting Gateway

To start Gateway, select it from the **Program** list, available on your Windows Start menu: (Trilogy\Trilogy V5\Configuration Editor).

We recommend that you also place a shortcut to Gateway on the desktop of the computer.

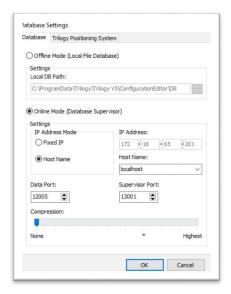


When you start Gateway, you must ensure that the Mercury DB Supervisor is running and the location of the database is correct. If not, you will see a warning message.



If the service is running, click the **Yes** button to specify the location of the database.





If the database is running and the settings are correct, you will be prompted to login.



Enter the operator name and password and click the \mathbf{OK} button.

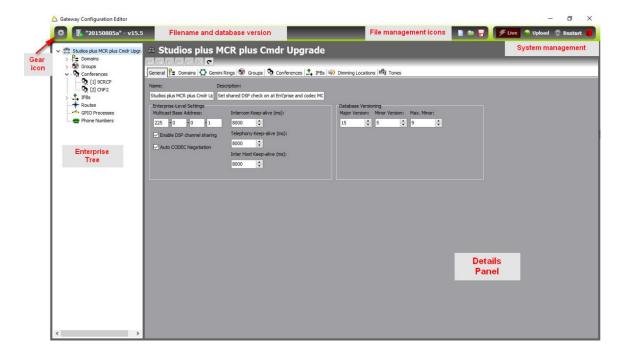


NOTE: The default operator name is **Admin** and the default password for the Admin operator is **trilogy**. You may change this after you have logged in. The password is case- sensitive. Refer to the Roles and Administrators chapter for further information.



Gateway Interface

Gateway is divided into main areas as shown below:



Filename and database version

The left side of the toolbar carries text showing the filename of the configuration currently being edited. If the configuration has been started from scratch and not yet saved, the text <NEW CONFIGURATION – NOT SAVED> will be displayed and the version will revert to 1.0.

File Management

This area provides access to options to:

- Create a new configuration.
- Load a configuration.
- Backup [save] a configuration.

The menu options are described below.



Button	Description
	Starts a new (empty) configuration. The current configuration will be lost unless you save it before creating a new configuration. A warning is given and you need to click Yes to continue.
	Loads a previously saved configuration (archive with file extension .trilogy). The current configuration will be overwritten. A warning is given and you need to click Yes
	Creates a backup of the current configuration to an archive. This can be loaded back into Gateway later, if required. As with all computer based work, it is good practice to take frequent backups. Pressing the button will immediately overwrite the previous file, without warning. To avoid this and to create incremental backups, use the Save As Option from within the menus.

System Management and Status

This area provides access to options to:

- Upload the current database to the connected system
- Restart the connected system using the data just uploaded
- Switch from online mode (the default: upload followed by restart) to "live" mode whereby minor system changes can be implemented without requiring a restart to take place. Live mode is an option.

The drum icon on the right of this area (shown red) provides a visual indication of the database content compared to the current system state. Full details are in the following table.

Icon	Description
8	The configuration has been changed. The database now differs from the connected system.
a	A configuration upload is in progress.
8	Changes have been uploaded to the system but the system has not yet restarted.
9	The system is now in the process of restarting.
8	The restart is complete: database and system are now back in step.
->	Starts a configuration update. Uploading the settings sends them to their destinations where they are cached. Upload the settings when you have completed or modified a configuration.
- >	Instates (activates) configuration changes. The working files on each host are replaced with the cached versions. Note that this button may not be enabled if you have chosen to automatically instate settings (see "Preferences" on page 2-12).



Enterprise Tree

The Enterprise tree shows an expandable view of the container objects that make up the configuration as a whole.

A right-pointing arrow next to an item indicates that it can be expanded to show more items; a down arrow indicates that it can be collapsed.

The Enterprise tree is the primary means of navigating the different sections of Gateway. Normally, two levels of editing are provided – "basic" and "advanced". For each level of the tree, the basic editor is opened with a single click – the advanced editor with double-click. This operation can be changed in Preferences. The design concept of Gateway software is that most day-to-day tasks can be carried out using the basic editors – the advanced editors should only be required during initial creation.

Details Panel

If you select an item within the Enterprise tree, the correct editor for that item will be shown in the Details panel. As explained above, a single click will display the Basic editor which should be sufficient for most tasks.



Menu Options (Gear Icon)

The menu options accessed from the gear icon at top left of the main Configuration Editor Window are detailed in the following section.

File Menu

Option	Description
New	Starts a new (empty) configuration. The current configuration will be lost unless you save it before creating a new configuration.
Load	Loads a previously saved configuration (.trilogy archive). The current configuration will be replaced after a warning.
Save	Immediately saves the current configuration to an archive file This follows the convention established by Microsoft and the last opened archive will be overwritten without warning. For a newly created configuration, a file name and location dialogue is shown. The archive can be loaded back into Gateway later, if required.
Save as	A file name and location dialogue is shown and the current configuration is written to a new archive. This can be loaded back into Gateway later, if required.
Switch to Live Mode	Switches from the default "online" mode of operation (upload followed by restart) to a "live" mode which allows minor changes to be made without the need for a restart. This is a paid for option, enabled by keycode within Gateway.
Start upload	Recent changes to the database are transferred to the connected system, where they are locally cached prior to a restart.
Restart	On completion of an upload, the data previously cached is transferred to active RAM on connected system hosts. Each host will restart, taking between 30 – 60 seconds, during which time the system is out of service.
Exit	Closes Gateway.



View Menu

Option	Description
Active Clients	Opens the Active Clients' Status window. This provides an intuitive view of all clients connected to the database and gives simple granular control over the upload and restart process.
Expand View	Expands all branches of the Enterprise tree.
Collapse View	Collapse all branches of the Enterprise tree.
Refresh View	Refresh all branches of the Enterprise tree.
Configuration Search	

Setup Menu

Option	Description
Database Settings	Opens the Database Settings dialog box. Here you can change network settings, including the IP address of the Database Supervisor.
Preferences	Opens the Preferences dialog box. Here you can change some of the settings that relate to the operation of Gateway.
Entity Filter	To simplify the visual appearance of what can become a complex GUI, unused entities may be hidden from view. For example, control panel types which have not been purchased for your project.
Enter Software keys	Features within Gateway are unlocked using software keys, purchased from Trilogy. To enable a feature not currently activated, purchase an additional key and enter it here.

Tools Menu

Option	Description
Recalculate all Conference best profiles	Automatically sets the Codec Profile for all Conferences in the Enterprise.
Change Password	Opens the Change Password dialog boxes.
Launch Access Management console	The access management console allows the creation of additional operators with roles which may be tailored to permit access to specified sections of a large system.
Gemini Upgrade	Allows for bulk upgrades of multi host systems (Gemini only).



Window Menu

Option	Description
Close	Closes the currently open editor and returns the application to the opening screen.

Help Menu

Option	Description
About	The Help dialogue carries version information relating to both Gateway and the overall installation package, plus details of the software activation keys currently in place.

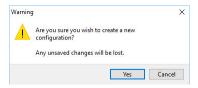


Creating a Configuration

You can create a new configuration at any time. However, if you do not save (archive) your current configuration, all changes will be **lost** when you create a new configuration. If you want to keep a configuration that you are working on (and possibly come back to later), ensure that you backup the configuration before you create a new one.

To create a new configuration:

- 1. Click the Start a new (empty) configuration button $\crit{f \bot}$
- 2. A message box is displayed asking whether you are sure that you want to create a new configuration.



3. Click the Yes button to confirm that you want to create a new configuration. Click the Cancel button if you want to save or keep the current configuration.



NOTE: A default configuration (or Enterprise) is created with a single domain, two groups, two conferences and one IFB. The Enterprise tree shows an expandable view on the configuration that has been created. The name of the Enterprise includes the creation date. The configuration may be changed to match your own specific requirements.

Saving and Loading a Configuration

It is good practice to backup a configuration, with the date and time that it was saved in the filename, at various stages of its development. If you save a configuration regularly, you can restore if you need to.



NOTE: The configuration data is important, and we would recommend that you either save it to removable media or ensure that it is backed up with your other data.

To save a configuration:





- 5. If the configuration has **not** previously been saved, you will be prompted to enter a filename and location. Enter a name for the configuration file and select the folder into which you want to save the configuration. Click the **Save** button.
- 6. If the configuration has previously been saved, or was loaded from an earlier archive file, then that file, named as shown on the toolbar, will immediately be overwritten.
- 7. The save icon (disk) colour changes to red to indicate that database edits have taken place which have not yet been backed up.

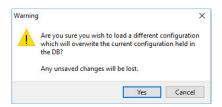


To load a saved configuration:

1. Click the **Open a previously saved configuration** button it to load an existing configuration from an archive.



2. If you have not backed up the current database (i.e. the save disk icon is red) a warning box is displayed asking you to confirm that you wish to load the different configuration. Your existing configuration will be overwritten if you select Yes.



- 3. If you are sure that you want to load a saved configuration, click **Yes**. Alternatively, click Cancel and backup the current data before continuing.
- 4. Browse for and select the configuration that you want to load.
- 5. Click the **Open** button to load the configuration.



Updating a connected system with a new configuration

Once you have completed developing your configuration, or have completed the details for a particular host, you can send the changes to the connected hosts in your system, and subsequently to each subscriber.

This is a two stage process.

- The first stage is a background task and is non disruptive. *Uploading* the data sends it to the appropriate destination hosts where it is cached.
- To make these changes active (i.e. replace the working files with the cached version), you must activate the changes. This second phase requires a *restart* of each host and is disruptive.

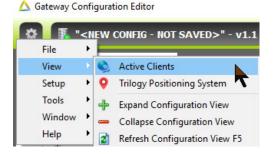
There are two methods available to achieve this:

- Using the "Active Clients" dialogue. This gives a detailed view of all connected hosts and their current status. The Active Clients form may be accessed from the View menu, or by pressing the status drum icon on the toolbar, top-right. You may need to change a setting to allow the drum to operate in this way – explained below.
- 2. Using the "upload" and "restart" buttons on the toolbar, top-right of screen. This gives limited information and is normally only recommended if you have a simple, single host system.

Method 1 is therefore to be preferred.

Showing Active Clients

1. Either, from the View menu (gear icon), click "Active Clients".



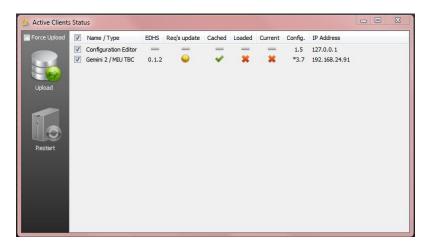
2. Or, click the status drum icon on the toolbar, shown here red.





Stage 1: Uploading a Configuration

 Show the Active Clients form as described above. Note that the Restart button is currently disabled. The status flags for the connected host show that an update is now required and has not yet commenced.

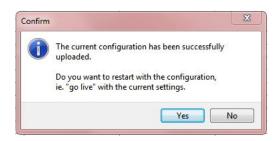


 Press the Upload button: data is now transferred and locally cached on all connected hosts. Depending on the size and complexity of the configuration, and the type of host, this can take between 15 and 60 seconds.



NOTE: By default, data is uploaded to all currently connected hosts. Use the checkboxes to de-select individual hosts.

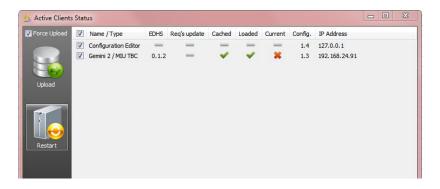
3. When the new data has uploaded, the message shown below will appear. You can choose whether to activate the configuration immediately, or postpone until later. If you click the No button, you can activate the changes later by clicking the restart button which will now be enabled.



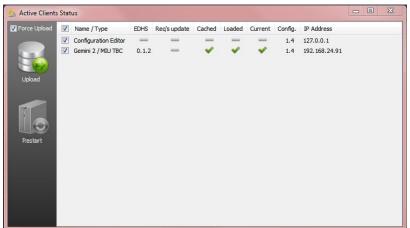


Stage 2: Restarting the connected system hosts

 Press the Yes button (above) or press the Restart button on the Active Clients form. If you have opted to postpone the restart, simply return to the Active Clients form later to resume the process. Note that the Restart button is now available. By comparison with the previous screen image, the new configuration is "loaded" but not yet "current".



2. When the restart is complete, Active Clients will look like this:





NOTE: The host status now updates to "current" and the configuration version (1.4) matches the Gateway editor.



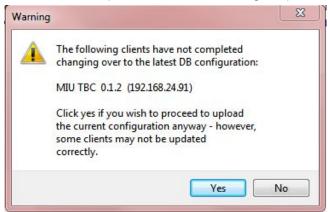
Configuration Update Messages

When you are updating a configuration the following message boxes may be displayed.

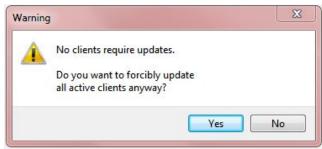
No active clients are running therefore update cannot take place.



Changes have been uploaded but not restarted. Additional changes have been made and uploaded without restarting the previous changes.



The current configuration is active on all clients and no update is required. If you click the **Yes** button, all Mercury hosts will be restarted. This ensures that all versions are synchronised.





Understanding the Status of a Configuration

The database drum icon which is shown in the top-right of the Gateway screen shows the status of the database with respect to the connected system.

Icon	Description
•	The configuration has been changed. The database now differs from the connected system.
a	A configuration upload is in progress.
9	Changes have been uploaded to the system but the system has not yet restarted.
9	The system is now in the process of restarting.
8	The restart is complete: database and system are now back in step.



Gateway Configuration Editor Setup

In broad terms this area of the application can be divided into:

- Settings which are essential for correct operation
- Settings which while not essential, may be appropriate for a particular system or convenient for an operator. These are termed "Preferences"

Changing Database Settings

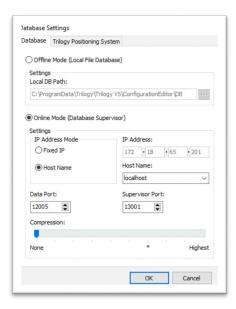
Once you have connected to the database, you will not usually have to change the database settings provided that the IP address of the database machine does not change. If this is allocated by Dynamic Host Control Protocol (DHCP), you may need to change the settings and specify the name of the host PC should the IP address change. For this reason, a static or fixed address is recommended. Database location by hostname is supported and requires DNS settings to be correctly specified. Support for this will vary according to your local IT structure.



NOTE: You can specify 127.0.0.1 in the IP address field or localhost in the name field if you want to use the Loopback Address. This is the normal situation when database and editor co-exist on the same PC.

To change database settings:

1. On the Setup menu, open the Database Settings dialog box.



- Make the required changes to the Online Mode settings. The default ports should not need to be changed unless these ports are in use elsewhere on the network.
- 3. Click OK to save your settings.
- 4. If the settings are not correct, an error message will be displayed informing you that the database cannot be found.





5. Click the OK button and review the database settings. If you still cannot connect to the database, ensure that the Database Supervisor is running and that the IP address or host name is correct. Also check that no hardware or software firewall is preventing connection.



The *Offline Mode (local file)* section of the form carries settings which allow Gateway to operate in isolation from the served database. This can be useful during training, or when developing changes to an already deployed system. During this time the served database is maintained, thus allowing the system to continue undisturbed.

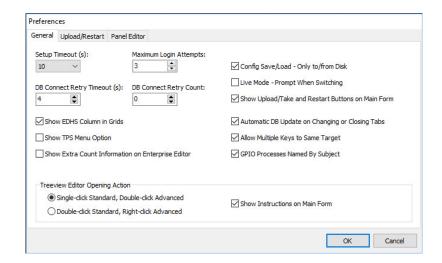


MORE INFORMATION: Refer to the "Config. Editor Settings - Database Settings tab" for a description of the fields on the Config. Editor Settings - Database Settings tab.

Changing Preferences

The preferences can be changed, if required, although in a typical working environment you may not need to do this. On the Setup menu, select Preferences... to open the Preferences dialog box. Preferences are arranged across three tabs of the dialogue.

Preferences - General:



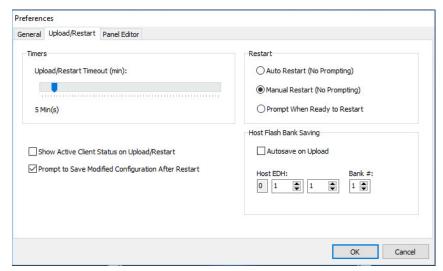
You may need to change the timeout values if the database is located on another machine on the network.



MORE INFORMATION: Refer to the "Preferences - General tab" for a description of the fields.



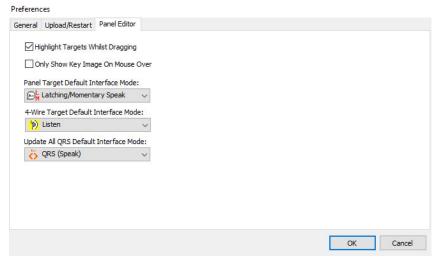
Preferences – Upload/Restart





MORE INFORMATION: Refer to the "Preferences – Upload/Restart tab" for a description of the fields.

Preferences - Panel Editor





MORE INFORMATION: Refer to the "Preferences – Panel Editor tab" for a description of the fields.

Make the required changes to the preferences. Click the OK button to save the changes.



Tools

There are a number of system tools provided, as described below.

Change Password

This is used to change the password of the currently logged in operator.



NOTE: The default password for the *Admin* operator is *trilogy*.

To change the operator password:

- 1. On the **Tools** menu, click **Change Password**. The Enter OLD Password dialog box is displayed.
- 2. Enter the old password and click OK. The Enter NEW Password dialog box is displayed.
- 3. Enter the new password and click OK. The Re-Enter NEW Password dialog box is displayed.
- 4. Re-enter the new password and click the OK button.



IMPORTANT: If you change the Admin password and subsequently cannot remember the new one and have not defined any other operators with appropriate access rights, please contact Trilogy Technical Support.

Access Management Console

The access management console allows the creation of additional administrators with roles which may be tailored to permit access to specified sections of a large system. For example, within a large multi-site system, additional administrators may be defined with full access to their local site, plus read-only access to other sites.



MORE INFORMATION: Refer to "Roles and Administrators" on page 54.

Gemini Upgrade

This tool provides the facility to bulk update Gemini firmware from a single console. This avoids the need to log into each Gemini separately, thus simplifying the operation and saving time.



Active Clients

The Active Clients option, on the View menu, displays a list of hosts that are currently connected to the database. These hosts will receive configuration updates if you run a configuration update (see "Uploading a Configuration" on page **Error! Bookmark not defined.**).



NOTE: This option is not available if you are using a local file system. See "Using a Local Database".

Viewing active clients is also a useful troubleshooting tool since it shows:

- Which hosts are connected to the system.
- Which configuration version the attached hosts are running.
- Whether the configuration is uploaded and restarted.
- The EDHS and IP address of each connected host.



Configuration Editor – Setup - Field Definitions

Setup - Database Settings dialogue

Field/Display Item	Description
Local Database	Click this radio button to use a local file system database rather than connect to a served database.
Served/Remote Database	Click this radio button to connect to a database on the host machine or network. You can use a local database rather than connecting to a database on the network. The same local database must be specified in the Supervisor Settings for the TBC if it is required to operate with this file system configuration.
IP Addr	The IP address of the PC hosting the database. Host name can be used if the IP address is assigned by DHCP.
Host Name	The name of the PC hosting the database. It provides an alternative to an IP address that is assigned by DHCP.
Data Port	The IP port that the Database Master (or Slave) uses for data connections. Changing the value changes the IP port number on which the database accepts incoming connections. It needs to be changed if:
	IP Port 12005 is in use elsewhere on the network by another application.
	Network security deems that another port should be used, or Network Address Translation /Port Address Translation (NAT/PAT) requires that another port is used. Changing this value also means that all clients (including slaves) must be configured to connect to the new port value. Default = 12005
Super. Port	The IP port on which the Database Master (or Slave) listens for connection control data. Changing the value changes the IP port number on which the database accepts incoming connections. This needs to be changed if:
	IP Port 13001 is in use elsewhere on the network by another application.
	Network security deems that another port should be used, or NAT/PAT requires that another port is used. Changing this value also means that all clients (including slaves) must be configured to connect to the new port value. Default = 13001
Compression	Compresses each transaction exchanged between server and clients by an adjustable amount. This is useful on networks with limited bandwidth. The asterisk on the scale is a suggested starting point, giving a good compromise between CPU usage (to compress the data) and bandwidth saved.



Setup - Preferences - General tab

Field/Display Item	Description
Setup Timeout	The time, in seconds, in which Gateway will
	timeout on a connection to the Database Supervisor.
	Default = Never.
	Options are:
	Never Windows Transport Control Protocol (TCP) timeout.
	Numerical Value in seconds Override Windows timeout.
DB Connect Retry Timeout	Specifies the database connection retry timeout, in seconds.
	Default = 1 second.
	Maximum = 5 seconds.
Maximum Login Attempts	Specifies the maximum number of unsuccessful login attempts before Gateway automatically closes. Default = 3. Range = 1–10.
DB Connect Retry	Specifies the number of times a database connection will be retried if not successful.
	Default = 1 second.
	Maximum = 5 seconds.
Show EDHS Column in grids	
Show TPS Menu Option	
Show extra count information on Enterprise Editor	
Config save/load only to/from disk	
Live mode – prompt when switching	
Show Upload/Take and Restart buttons on main form	
Automatic DB update on changing or closing tabs	



Field/Display Item	Description
Allow multiple keys to same target	If selected, then it is possible to configure more than one key on a panel to the same destination.
	If left unselected it is only possible to have one key to any given subscriber on a panel.
	Default = Selected.
Name GPIO processes by subject	Relates to GPIO process screens and causes the GPIO routing statement name to be derived from the statement itself, instead of using a default, potentially unfriendly, name.
	Default = Selected.
Treeview Editor Opening Action	2 option radio buttons:
Show Instructions on Main Form	

Setup - Preferences - Upload/Restart tab

Field/Display Item	Description
Upload/Restart Timeout (min)	Default = 5 minutes.
Show active client status on upload/ restart	Displays the Active Clients' Status window when updating Mercury hosts with a new or modified configuration. This in turn displays the progress of the delivery of the configuration to all hosts, and when complete instates the new version. Default = Selected.
Prompt to save modified configuration after restart	
Restart	3 option radio buttons:
Host Flash Bank Saving	Automatic on upload Host EDH Bank #



Setup - Preferences - Panel Editor tab

Field/Display Item	Description
Highlight Targets while Dragging	
Only show key image on mouse over	
Panel target – default key mode	
4-wire target – default key mode	
Update All QRS – default key mode	



Chapter 3 - Enterprises and the Enterprise Editor

Overview

The Enterprise Editor is one of a series of Editors that you will use to define the structure and components of your Mercury system. It is the 'top-level Editor that is used to define the overall structure of your Enterprise.

Use it to add and delete domains, conferences, groups, IFBs, SIP connections, zones, areas and ring tones.

Once these have been created, use the relevant Editor to configure the various Mercury objects. Refer to the appropriate chapter of this Configuration Guide for full configuration details:

- Domains
- Groups
- Conferences
- SIP connections
- IFBs
- Roles and Administrators



NOTE: You can create your configuration in any order but we would recommend that you create the physical structure first (domains - hosts - subscribers) and then add in virtual entities such as groups, conferences etc.

Before you start creating your configuration and defining the Enterprise, it is important that you fully understand the structure of the Enterprise that you want to model using Gateway. Taking time before you start, avoids problems later on.



TIP: Mercury objects are generally created (and deleted) either one level above in the Enterprise tree, or on the overview page for that entity. For example, domains are created and deleted using the Enterprise Editor but configured using the Domain Editor.



Using the Enterprise Editor

Opening the Enterprise Editor

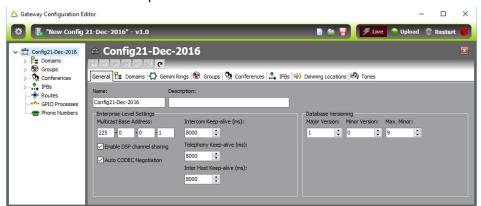
To open the Basic Enterprise Editor:

- 1. Single-click on the Enterprise name to open the basic editor.
- This offers fields to edit the Enterprise name and to adjust the database version numbering plan.



To open the Advanced Enterprise Editor:

- 1. Double-click on the Enterprise name to open the basic editor.
- 2. This offers multiple tabs as described below.



The Enterprise Editor Tabs

The Enterprise Editor has the following tabs which are used to define the basic structure of your Enterprise:

- **Details** tab, to change basic details relating to the Enterprise.
- Domains tab, to add and configure domains.
- Gemini Rings tab, to manage Gemini HSL connections
- Groups tab, to add groups to the Enterprise.
- Conferences tab, to add conferences to the Enterprise.
- IFBs tab, to add IFBs to the Enterprise.
- SIP Con's tab, to add SIP connections to the Enterprise.
- Tones tab, to define custom ring tones for use in Gateway.



Editor Basics

Once you have opened the Enterprise Editor, you will notice a series of buttons at the top of the window. You will see the same set of buttons on every Editor window in the system. Some of these buttons may be unavailable (for example if there is only one record in a list) and some may only be enabled when you make changes to the details.

The functions of the basic set of buttons on an Editor window are explained below:

Table 3.1 Editor navigation buttons

Button	Usage
H	Moves to the first record in a list.
•	Moves to the record prior to the selected one.
•	Moves to the next record.
►I	Moves to the last record in the list.
~	Posts (writes) the changes to the database.
×	Cancels the changes made to the current record since the last post. If you have added a new record, this record will be removed.
C	Refreshes the view and posts any changes.
	Moves the selected item up. Not available in the Enterprise Editor.
•	Moves the selected item down. Not available in the Enterprise Editor.

Using Images

Images can be used to identify callers on Virtual Panels. The following image formats are accepted by Gateway and are then converted internally to JPG format: BMP, JPG, WMF and ICO.

You can either select individual records in a list or select more than one record at a time.

Selecting Records in Lists

- To select more than one record, hold down the Ctrl key and click the records that you want to select.
- To select records that are listed consecutively, click on the first one, hold down the **Shift** key and click the last one.

Deleting Records

Records can be deleted from the database, if required. When you delete records from the database, you can either delete the required records individually (using the Delete button) or delete them all at the same time (using the Clear All button).



IMPORTANT: When you click the Clear All button, all records in the list will be deleted from the configuration, not just the selected one. Use this button with care.



Enterprise Editor - Details

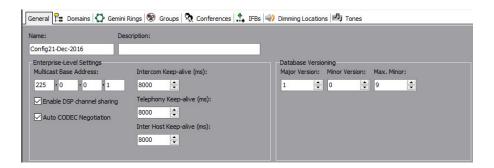
The Enterprise Editor - Details tab lets you specify basic details about an Enterprise, including its name, description and the first multicast base address.

Changing Enterprise Details

You can change the basic details relating to your Enterprise, as required. For example, you may wish to rename the Enterprise or enter a description.

To change Enterprise details:

1. Open the Enterprise Editor. The **General** tab is selected by default.



- 2. Make the required changes to the Enterprise details.
- 3. Click the $\stackrel{\checkmark}{}$ button to post the changes to the database.



MORE INFORMATION: Refer to the "Enterprise Editor - Details tab" on page 3-26 for a description of the fields on the Enterprise Editor - Details tab.



Enterprise Editor - Domains

A domain is a collection of hosts that share a common IP network (network ID, subnet mask and default gateway).

The Enterprise Editor - Domains tab is used to add domains to your Enterprise, make changes to existing domains and also to delete any domains that are no longer part of the Enterprise.

Buttons at the top of the Domains tab let you:

- Add a new domain. (Hosts and subscribers will need to be manually added).
- Delete a selected domain (after confirmation).

Adding a Domain

A domain is a logical grouping of hosts. Domains are predominantly groupings of convenience, allowing a system to be segregated into separate areas. There are two ways to add a new domain: you can either add a blank domain or duplicate an existing domain.



NOTE: Your default configuration will contain one domain. More can be added, as required.

To add a domain:

1. Open the Enterprise Editor and click the **Domains** tab.

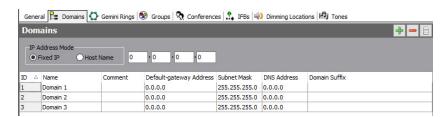


2. Click the Add button. The Enter Number dialog box is displayed.



Specify how many domains you want to add then click the OK button to create them.

New domains are given a default name which can be changed, if required, to match the structure of your network.





Duplicating a Domain

Duplicating a domain adds the domain and also automatically duplicates the hosts and subscribers under the selected domain.

To duplicate a domain:

- Open the Enterprise Editor and click the **Domains** tab.
- Select that domain that you want to duplicate.
- Right click and select Duplicate from the context menu.
- Specify how many domains you want to add (based on the selected domain).
- Click the **OK** button to duplicate the domain and its hosts and subscribers.

Changing a Domain

You can change the name of a domain and also add and change comments which can be used to describe the domain in more detail. The gateway address cannot be changed. This is specified using the Domain Editor. See "Domain Settings" on page 6-11.

To change domain details:

- Open the Enterprise Editor and click the **Domains** tab.
- Change the name of the domain and add or change comments, as required.
- Click the button to post the changes to the database.



MORE INFORMATION: Refer to the "Enterprise Editor - Domains tab" on page 3-27 for a description of the fields on the Enterprise Editor - Domains tab.

Deleting a Domain

Domains can be deleted if they are no longer required. It is advisable to take a copy of the configuration before deleting a domain so that you can load in the original configuration should this be required at a later date.

You can delete selected domains (using the Delete button) or delete all domains at the same time (using the Clear All button).



IMPORTANT: Deleting a domain will also delete all of the hosts that have been added to that domain.



Enterprise Editor – Gemini Rings

A Gemini Ring is used to define the interconnection of multiple Gemini hosts using the HSL (High Speed Link) connection. HSL allows Gemini to provide programme quality audio to hosts connected in this way. The ring should be circular and closed, so the last host in the chain must connect back to the first in the chain. Actual physical sequence of connection is not important.

Gemini hosts are assigned to rings using the Host editor: the Enterprise merely defines the existence of any rings.

Mercury series hosts cannot be added to rings.

The Enterprise Editor – Gemini Rings tab is used to add rings to your Enterprise, modify existing rings and also to delete any rings that are no longer part of the Enterprise.

Buttons at the top of the rings tab let you:

- Add a new Gemini Ring.
- Delete a selected Gemini Ring (after confirmation).



Managing Gemini Rings

By default, an Enterprise will not contain any Gemini Rings. They can be added, as needed.

Follow the same process as outlined previously to add, delete, or edit the details of Gemini Rings.

Post your changes to the database by pressing the "tick" button.



IMPORTANT: When you delete Gemini Rings, any Gemini hosts which had been assigned to that ring will now be independent.



Enterprise Editor - Groups

A group is a collection of user-defined subscribers. The Enterprise Editor - Groups tab is used to add groups to your Enterprise and also to delete any groups that are no longer part of the Enterprise.

Buttons at the top of the Groups tab let you:

- Add a new group.
- Delete a selected group (after confirmation).



Managing Groups

By default, an Enterprise will contain two groups which are added when the configuration is created. More can be added, as needed.

Follow the same process as outlined in the previous section to add, delete, or edit the details of Groups.

Post your changes to the database by pressing the "tick" button.



IMPORTANT: When you delete groups, all references to the deleted groups will be lost. Any keys targeting deleted groups will be automatically removed from panels.



Enterprise Editor - Conferences

A conference is a special type of group. It sets up communication between a group of subscribers where each subscriber has the choice of whether to join in, listen only, or switch off.

Panels and 4-wires can be included in a conference. Multicasting is used to minimise bandwidth usage.

The maximum number of active speaking conference members may be limited to further control bandwidth and IP channel usage or for other operational reasons.

Panel members are simply added by targeting a key at the conference.

Each speaking Mercury host will multicast audio and each listening host can receive multiple audio streams if many people speak.



NOTE: Conferences use multicast audio whereas groups do not. However, not all networks support multicast. Mercury conferences will not function on networks that do not support multicast.

The Enterprise Editor - Conferences tab is used to add conferences to your Enterprise, modify existing conferences and also to delete any conferences that are no longer required.

Buttons at the top of the Conferences tab let you:

- Add a new conference.
- Delete a selected conference (after confirmation).



Managing Conferences

By default, an Enterprise will contain two conferences which are added when the configuration is created. More can be added, as needed.

Follow the same process as outlined previously to add, delete, or edit the details of Conferences.

Post your changes to the database by pressing the "tick" button.



IMPORTANT: When you delete conferences, all references to the deleted conferences will be lost. Any keys targeting deleted conferences will be automatically removed from panels.



Enterprise Editor - IFBs

An IFB is an audio route that can be broken into or interrupted by a third party, replacing the original audio with the third party panel microphone. It comprises the IFB destination (usually a 4-wire port) and the IFB interrupt that generates the control signal and interrupting audio. There is one additional audio signal which is the standing feed which is present before the interrupt takes place. This is referred to as the IFB Source.

The Enterprise Editor - IFBs tab is used to add IFBs to your Enterprise, modify existing IFBs and also to delete any IFBs that are no longer part of the Enterprise.

Buttons at the top of the IFBs tab let you:

- Add a new IFB.
- Delete a selected IFB (after confirmation).



Managing IFBs

By default, an Enterprise will contain one IFB which is added when the configuration is created. More can be added, as needed.

Follow the same process as outlined previously to add, delete, or edit the details of IFBs.

Post your changes to the database by pressing the "tick" button.



IMPORTANT: When you delete IFBs, all references to the deleted IFBs will be lost. Any keys targeting deleted IFBs will be automatically removed from panels.



Enterprise Editor - SIP Connections

SIP is an open, standards based protocol for negotiating voice and multimedia calls over a network. The addition of SIP provides Mercury with the capability to make and receive SIP VoIP calls to any configured, basic SIP devices that may be present on the network.

A SIP connection is an object used to identify a SIP entity that exists outside of Mercury.

Buttons at the top of the SIP Con's tab let you:

- Add a new SIP connection.
- Delete a selected SIP connection (after confirmation).

Managing SIP Connections

By default, an Enterprise will not contain any SIP connections. If you intend to use SIP, you must add the required number of connections using the Enterprise Editor.



IMPORTANT: When you delete SIP connections, all references to the deleted conferences will be lost. Any keys targeting deleted SIP connections will be automatically removed from panels.



Enterprise Editor - Tones

Use the Enterprise Editor - Tones tab to define custom ring tones for use in Gateway. Tones are a combination of none, one, two or four different tones with a configurable up and down period (amplitude). Up to 16 tones can be defined.

The requirements are:

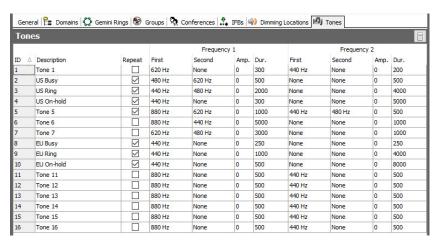
- Sequences have two phases (frequencies) and a repeat/continuous attribute as follows:
 - Phase1. Zero, one or two tones from the four available tones can be selected. The amplitude of each tone can be set separately, between +12 dB and -60 dB. The selected tones can be gated on for a configurable period, between 0 and 20,000 ms.
 - Phase2. Zero, one or two tones from the four available tones can be selected. The amplitude of each tone can be set separately, between +12 dB and -60 dB. The selected tones can be gated on for a configurable period, between 0 and 20,000 ms.
- 2. The sequence can be set to repeat until turned off (as used, for example, for phone ringing) or as a once-off (as used, for example, as a call alert; e.g. a "bing-bong").

Changing a Tone

Tones can be changed, as required. They cannot be added or deleted.

To change a tone:

1. Open the Enterprise Editor and click the **Tones** tab.



- 2. Make the required changes to the frequency settings.
- 3. Click the button to post the changes to the database.



Enterprise Editor - Field Definitions

Enterprise Editor - Details tab

Field/Display Item	Description
Name	A default name is given to the Enterprise when it is created (this name is based on the date the configuration is created). The name can be changed to something more descriptive, if required. The name is for information only.
Description	Maximum length for the name is 54 characters.
Description	A description of the Enterprise. This description is for information only.
	Maximum length for the description is 70 characters.
Multicast base address	Multicast addresses are used so devices can transmit audio traffic via a single multicast address to multiple hosts, rather than via individual IP connections. They can be used by conferences.
	Multicast addresses fall in the range of 224.0.0.0 to 239.255.255.255. The base address must be entered in this field, which is used by the first Mercury host. Each subsequent host has a value 32 times higher in the last IP octet than the preceding host. The multicast base address can be set on a per-host basis in the range 224.0.1.0 to 238.0.0.0.
Major version	Configuration version numbers have the following format: X.Y
	X is the major version of the current configuration. This usually starts at 1 and increments each time the minor revision (Y) reaches a certain value (see below).
	Using version control for a particular configuration allows a reference, or a backup configuration that is known to be "good" to be identified. It can also be used to catalogue changes.
	The major version can be changed as required.
Minor version	The minor version is incremented each time a configuration update is sent to the TBCs running on the Mercury hosts.
Inc. Major if Minor exceeds	When the minor version reaches the value specified here it is reset to zero and the major version is incremented.
Intercom Keep-alive (ms)	The period in which all hosts in the Enterprise will contact each other with a "heartbeat" network message. A setting of zero disables this functionality.
	Recommended setting and default = 8000 ms. Minimum = 5000 ms, maximum = 15000 ms.
Telephony Keep-alive (ms)	Similar to Intercom keep-alive but used for calls to FXO, FXS and E&M devices, rather than for intercom calls.
	Recommended setting and default = 8000 ms.
	Minimum = 5000 ms, maximum = 15000 ms.



Field/Display Item	Description
Enable Auto-codec Neg.	When selected, automatically negotiates the best codec to use. Default = Selected.
Fast Inhibits	An Enterprise-level setting that directs all hosts to use Fast Inhibits. This feature forces all hosts to inhibit DSP audio routes without contacting the remote peer. This can result in a significant increase in responsiveness to route inhibit requests, but the side effect of this is that DSP channels and IP resources are not released when a call is suspended. The recommended setting is not to use Fast Inhibits.

Enterprise Editor - Domains tab

Field/Display Item	Description
ID	A number to identify the domain. This cannot be changed.
Name	The name of the domain. Defaults to Domain 1, 2, etc. but can be changed. A descriptive domain name is recommended; names do not have to be unique.
	Maximum length is 20 characters.
Comment	Comments relating to the selected domain. To add comments, select the domain (click once on the line), click in this field and enter your comments.
	Maximum length for comments is 255 characters.
IP Gateway	The IP gateway address, if this has been entered for the selected domain. To set or edit the IP gateway address, highlight the row, then enter the address in the edit boxes alongside the radio buttons.
Subnet Mask	The subnet mask set for this domain. This may be entered here, at Enterprise level, or using the Domain Editor.
DNS Address	The IP address of a Domain Name Server assigned for this domain. This may be entered here, at Enterprise level, or using the Domain Editor.



Enterprise Editor - Groups tab

Field/Display Item	Description
ID	A number to identify the group. This cannot be changed.
Name	The name of the group. Defaults to GRP1, GRP2, etc. but can be changed. The name does not have to be unique.
	Maximum length for the group name is 20 characters.
Long Name	An alternative name which, if set, will be used on control panels with displays supporting 8 characters.
Comment	Comments relating to a selected group. To add comments, select the group (click once on the line), click in this field and enter your comments. Maximum length for comments is 255 characters.
	Maximum length for comments is 255 characters.

Enterprise Editor - Conferences tab

Field/Display Item	Description
ID	A number to identify the conference. This cannot be changed.
Name	The name of the conference. Defaults to CNF1, CNF2, etc. but can be changed. The name does not have to be unique.
	Maximum length for a conference name is 20 characters.
Long Name	An alternative name which, if set, will be used on control panels with displays supporting 8 characters.
Comment	Comments relating to a selected conference. To add comments, select the conference (click once on the line), click in this field and enter your comments.
	Maximum length for comments is 255 characters.
EDHS	Shows the EDHS address of the conference. This is allocated automatically.
	Addresses 2xxx are reserved for conferences.



Enterprise Editor - IFBs tab

Field/Display Item	Description
ID	A number to identify the IFB. This cannot be changed.
Name	The name of the IFB. Defaults to IFB1, IFB2, etc. but can be changed. The IFB name does not have to be unique.
	Maximum length for the IFB name is 20 characters.
Long Name	An alternative name which, if set, will be used on control panels with displays supporting 8 characters.
Comment	Comments relating to a selected IFB. To add comments, select the IFB (click once on the line), click in this field and enter your comments.
	Maximum length for comments is 255 characters.
EDHS	Shows the EDHS address of the IFB. This is allocated automatically.
	Addresses 4xxx are reserved for IFBs.

Enterprise Editor – SIP Con's tab

Field/Display Item	Description
ID	A number to identify the SIP connection. This cannot be changed.
Name	The name of the SIP connection. Defaults to SIP1, SIP2, etc., but can be changed. The SIP name does not have to be unique.
	Maximum length for the SIP name is 20 characters.
Comment	Comments relating to a selected SIP connection. To add comments, select the SIP (click once on the line), click in this field and enter your comments.
EDHS	Shows the EDHS address of the SIP connection. This is allocated automatically. Addresses 6xxx are reserved for SIP connections.



Enterprise Editor - Tones tab

Field/Display Item	Description
ID	An ID used to identify the tone. This cannot be changed.
Description	User description for the tone.
	Maximum length for the description is 32 characters.
Repeat	Repeat tone or a single "bing-bong".
Frequency 1	First First tone for phase 1.
	Second Second tone for phase 1.
	Amp. The amplitude of the tone.
	Duration The duration of the tone.
Frequency 2	First First tone for phase 1.
	Second Second tone for phase 1.
	Amp. The amplitude of the tone.
	Duration The duration of the tone.



Chapter 4 - Access Management Console – Roles and Administrators

Overview

To use Gateway, you must login using a valid operator name and password. There is one built in top level Administrator (Admin) but others can be added as required via the Access Management console.



NOTE: To use the Access Management console you must login with the top level built in Administrator account.

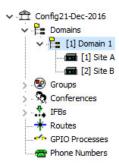
The access management console allows the creation of additional administrators with roles which may be tailored to permit access to specified sections of a large system. For example, within a large multi-site system, additional administrators may be defined with full access to their local site, but just read-only access to other sites.



Launching the Access Management console

When the top level administrator has logged in, an additional "lock" icon is displayed on the toolbar at the top right corner of the screen. A shortcut is also provided on the Tools menu. As explained in earlier sections, the credentials for the top level administrator are Admin | trilogy. When Gateway is first installed, only the top level system administrator is provided: no roles or additional administrators are present.

Our first task is to create a number of roles based on a system encompassing two studios. The example below shows a new configuration with two hosts named Site A and Site B.





Using the Roles and Administrators Editor

Creating Roles

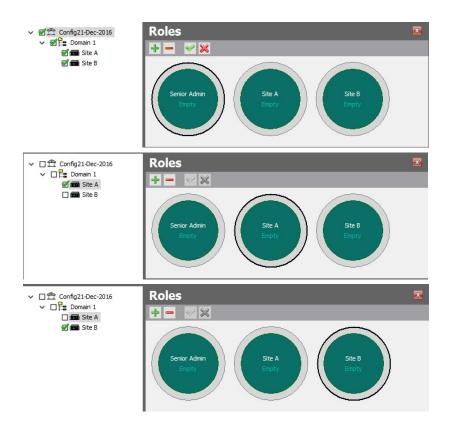
Firstly we will create three new roles:

- Senior Administrator
- Site A administrator
- Site B administrator

After logging in as the top level administrator, switch to the Roles and Administrators editor by pressing the lock icon on the toolbar. Click the green "+" icon to create a new role. Highlight this new role by clicking on the name in the centre of the circle, so the outer circle is bold and then edit the name of the role within the text box by double-clicking in the top row.



As Roles are added, they are by default granted access to all parts of the Enterprise. Next, highlight the Roles in turn and uncheck the boxes now displayed in the Enterprise tree as appropriate. See screenshots for each, below.

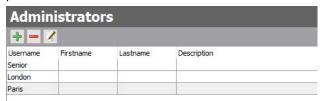




Creating Administrators

The next stage is to create a number of administrators and then to assign them to specific roles. Each administrator is simply a set of login credentials plus a description but they have no authority or permissions until assigned to a role. To create a new administrator, click on the green "+" icon in the lower pane of the Editor. Enter the details of each new administrator in the edit box and press the tick when complete. To edit the details of an administrator, highlight the list entry and click the edit button on the Administrators toolbar.

The image below shows three new Administrators: only the username and password have been entered.



Currently these accounts have no authority – the next step is to attach Administrators to Roles.

Associating Roles and Administrators

The final step is to assign the operators to specific roles. To do this, simply highlight an administrator from the list and drag the cursor into the appropriate role "circle".

The following rules and conditions apply:

- each role may have one or more administrators
- each administrator may be assigned to one or more roles
- a role without any administrators serves no purpose
- an administrator who is not assigned to any roles also serves no purpose

The screen below shows the administrators who have now been assigned to specific roles in the configuration of the network. The number in the centre of each roll circle indicates the number of administrators currently assigned to that role. Position the mouse cursor over the centre of the circle to see a list of the names of current administrators in that role.



NOTE: if additional hosts are added later, they are automatically added to every current role with the permissions set to on. Although these settings may be appropriate for the overall system administrator, minor roles such as Site A and Site B (above) must be edited to remove these new hosts as appropriate.



Using Roles and Administrators

It is important to realise that although multiple administrators may be created for a system, they cannot log-in simultaneously. To avoid data conflicts, only one administrator may log-in at a specific point in time. If a second administrator attempts to log-in, the following will result:

- The person already logged in will see a pop-up message to advise that someone else is attempting to connect. He can choose to block the second person, or allow him to connect. If he chooses to allow the other person, he must then exit Gateway. If he ignores this message, he will be forcibly disconnected after 30 seconds. This allows for the situation where a PC has been left un-attended.
- The second person attempting to connect will see a pop-up message to advise that another operator is already logged in. He can then choose to abandon his attempt to log-in, or wait for the other person to respond. If there is no response from the other party, then the second person will be allowed to connect around 30 seconds later.



Chapter 5 - Domains and the Domain Editor

Overview

A domain is a logical grouping of hosts. These are predominantly groupings of convenience, allowing a system to be segregated into separate areas. Mercury hosts on different IP networks must reside on different Mercury domains to each other.

Domains are added to an Enterprise using the Enterprise Editor or on the overview page for all domains.

Once domains have been added, use the Domain Editor to configure each domain to add or delete hosts, and also specify the IP address of the network gateway if you have more than one IP domain.

As explained earlier, the basic domain editor should provide sufficient control for most day-to-day tasks: the advanced editor is used initially and will be described here.



Using the Domain Editor

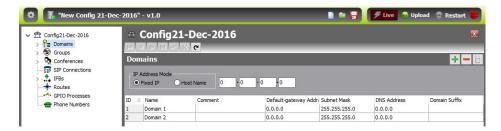
Viewing a List of Domains

All of the domains that are added using the Enterprise Editor are grouped together under 'Domains' in the Enterprise tree.

To view a list of domains:

- 1. Expand the Enterprise tree in Gateway.
- 2. Single-click on **Domains**.

The details panel now lists all of the domains that have been defined and allows some low level tasks to be performed.



Viewing a Summary of Hosts within a Domain

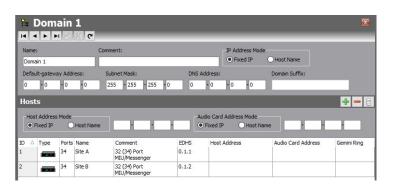
Hosts are added to a domain using the Domain Editor. You can quickly view a list of hosts in a domain by selecting the domain in the Enterprise tree.

To view a list of hosts in a domain:

- Click the "right arrow" next to **Domains** in the Enterprise tree to list the available domains.
- 2. Single-click on Domain 1.

The basic editor lists all of the hosts within the selected domain and allows most routine tasks to be performed. In this instance there are two 32 Port MIU/Messenger hosts.





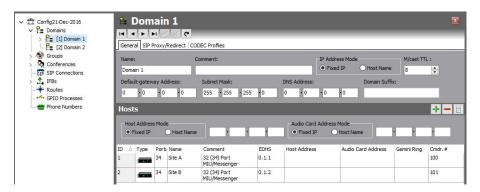


Opening the Domain Editor

Domains are maintained using the Domain Editor.

To open the Domain Editor:

- 1. Expand **Domains** in the Enterprise tree
- 2. Double-click on the required domain to open the advanced Domain Editor. By default, the General tab is displayed.



The Domain Editor Tabs

The Domain Editor has three tabs which are outlined below.

General tab

The General tab shows the name of the domain, as specified when the domain was added to the Enterprise. This name can be changed and a comment can also be added, if required. Edit as required then press the tick to post changes to the database.

The General tab is also used to add and maintain hosts in the selected domain. Hosts can also be swapped, adopted and duplicated. Any hosts that are not required can be deleted.

The General tab is also used to set the IP address of the IP gateway when multiple domains are present in the Enterprise.

SIP Proxy/Redirect tab

The SIP Proxy/Redirect tab is used to add and maintain SIP Proxy/Redirect Servers. Up to five servers may be defined. Servers that are no longer required can be deleted.

CODEC Profiles tab

The CODEC Profiles tab is used to make changes to CODEC profiles used by hosts in the domain.



Domain - General

The Domain Editor - General tab shows the name of the selected domain, as specified using the Enterprise Editor, and any comments that have already been entered for that domain.

Changing Domain details

- The name of a domain can be changed, as required. Comments can also be added. Change the name of the domain, or add a comment if required.
- 2. Click the button to post the changes to the database.

Managing Hosts

The Domain Editor - General tab shows a list of all hosts currently defined for the selected domain.

Buttons at the top of the Hosts tab let you:

- · Add hosts of specified types to the domain.
- · Delete one or more hosts from the domain.

The right-click context menu allows you to:

- Swap the EDHS address of the selected host with that of another host in the domain.
- Adopt a host from another domain. This removes the host from the first domain and adds it to the second domain. The EDHS address is automatically updated when a host is adopted. Use the Adopt function when you are migrating hosts.
- Duplicate a host that is already part of the domain. Duplicate hosts when you are expanding the configuration and adding new hardware.

Adding Hosts to a Domain

Hosts can be added to a domain by adding a new host record, by adopting a host from another domain or by duplicating a host that already exists.



NOTE: Duplicating or adopting a host will include any subscribers already defined on that host in the new one.

To add a host to the domain:

- 1. Open the Domain Editor and click the General tab.
- 2. Click the green + button and select the host type from the pop-up list.







Specify the number of hosts of the selected type that you want to add in the Enter Number dialog box.

- 3. Click the **OK** button to add the host(s).
- 4. You can change the host name or add a comment, if required.
- 5. Click the 🖊 button to post any changes you make.

Adopting Hosts

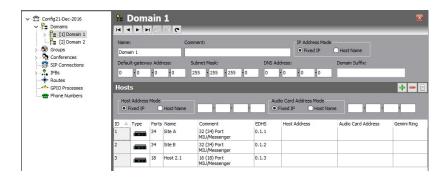
Hosts can also be adopted from another domain. When you adopt a host, it is removed from another domain and added to the current domain. All host settings are retained when a host is adopted.

To adopt a host:

- 1. Open the Domain Editor and click the **General** tab.
- Right-click in the free space below the list of hosts, to display the context menu.
- 3. Select "Adopt" and a list of hosts currently available for adoption will be displayed. This is a list of hosts in other domains.



- 4. Select the required host on the list and click the **Apply** button at the bottom of the list box. The EDHS address of the new host is automatically corrected to reflect its new domain. The name however has not changed and should be edited to avoid confusion.
- 5. Domain 1 now contains 3 hosts.





Duplicating Hosts

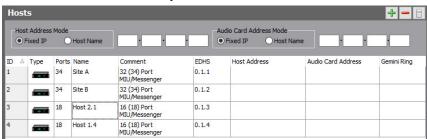
Another way to add a new host to your configuration is to duplicate a host that is already in the domain.



NOTE: The following are also duplicated when you duplicate a host: port types, subscribers and subscriber settings, port settings, virtual ports, telephony board termination settings and audio trunking ports.

To duplicate a host:

- 1. Open the Domain Editor and click the **General** tab.
- 2. Right-click on the host you wish to duplicate and select the **Duplicate** option from the context menu.
- 3. In the "enter number" dialogue box, choose the number of duplicate copies you wish to make, and press OK.
- 4. EDHS numbers are corrected automatically.
- 5. The example below shows one additional duplicate copy of Host 2.1, which has been automatically named as Host 1.4.



Changing Host Details

You can change host details, as required. The information that you can change is: the host name, comments, the Host (Windows) IP address, audio card IP address, and the Commander/Orator system address.

To change host details:

- 1. Open the Domain Editor and click the **Hosts** tab.
- 2. Make the required changes.
- 3. To change the Commander/Orator address, select the required number from the drop-down list.
- To change IP addresses, highlight the host and then enter new addresses in the edit boxes.
- 5. Click the button to post any changes you make.

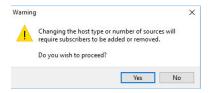


Changing Host Type

The host type can be changed, but be aware that settings that you have made in the Host Editor may be lost.

To change host type:

- 1. Open the Domain Editor and click the **General** tab.
- 2. Click in the **Type** field, click the "down" arrow and then select the required host type.
- 3. Make any other changes, as required.
- 4. Click the button to post the changes. A warning box is displayed.



Click the **Yes** button to confirm that you want to change the host type.



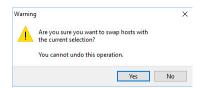
NOTE: If you change the host type, any subscribers that have been allocated to ports that are not available on the new host type will be lost. You may also lose host-specific configuration settings, GPIO settings and trunking. For example, a subscriber on port two on an MIU will still be on port two if the host is changed to a USB adapter. However, if the subscriber is on port three on the MIU, it will be deleted if the host is changed to a USB adapter. After changing a host type, always check all related settings carefully.

Swapping Host EDHS

You can swap the EDHS address of one host with another in the domain.

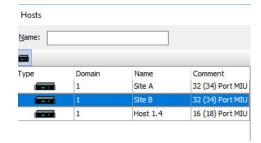
To swap a host:

- 1. Open the Domain Editor and click the Hosts tab.
- 2. Highlight the host that you want to swap with another host in the system.
- Right-click to display the context menu and select Swap. A warning message is displayed.



4. Click the Yes button to continue and open the Hosts dialog box. This shows a list of all hosts, except the host initially highlighted in step 2 above, including their domain number, ID, name and any comments that have been entered.





5. Select the required host on the Hosts list and click the **Apply** button.

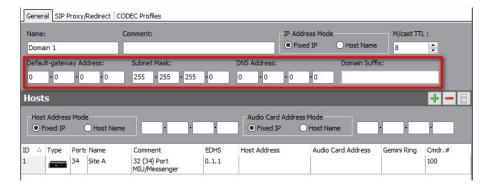
If you swap with a host that is in another domain, the EDHS address is automatically corrected to reflect its new domain.

Deleting Hosts

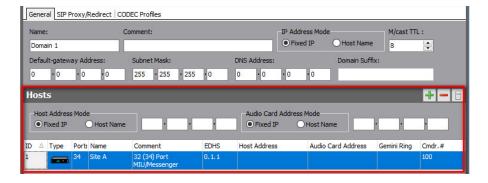
Any hosts that are no longer required in your configuration can be deleted. To delete a host, select it on the Hosts tab and click the red - button.

IP Address parameters

The Domain Editor - General tab is used to specify the IP address and subnet mask of the default gateway used to connect to other domains within the enterprise. Edit values in the area highlighted below and post any changes to the database.



The same form is used to set the IP addresses of each host within the domain. Highlight the host in question and edit values in the area shown below in red. When complete, post edits to the database.





NOTE: Remember that Mercury host types require 2 IP addresses – the first for the Windows OS, the second exclusively for the Mercury Audio Card.



Domain - SIP Proxy and Redirect

A SIP Proxy Server receives "call" requests from a User Agent and forwards these requests to one or more addresses to locate the destination of the call. Messages forwarded by the Proxy Server may be modified and any responses are returned to the calling User Agent as appropriate.

A SIP Redirect Server receives "call" requests from a User Agent, locates the destination and returns a "redirect" response to the User Agent with the destination address.

Adding SIP Proxy/Redirect Servers to a Domain

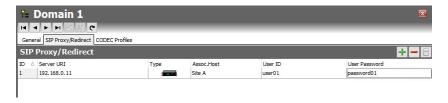
One SIP Proxy/Redirect Server per IP address (per Proxy) or DNS address may be defined. You must provide a server IP address or name, user ID and password.



NOTE: You can select a host for each Proxy/Redirect Server. If you do this, the specified user ID and password will be used by that host to register with the Proxy/Redirect Server. Different hosts can therefore register with the server using different credentials using this approach. If you do not specify a host, all hosts in the domain will connect using the same user ID and password.

To add a SIP Proxy/Redirect Server to the domain:

- 1. Open the Domain Editor and click the SIP Proxy/Redirect tab.
- 2. Click the green + button to add a new row to the server list.
- 3. Click in **Server URI** and enter the name or IP address of the server.
- To specify a host for the Proxy/Redirect Server, click in the Assoc.
 Host field, click the button, select the required host and click the Apply button.
- 5. The **Type** field icon is automatically filled in.
- 6. Click in **User ID** and enter the user name for Proxy/Redirect authentication.
- 7. Click in **User Password** and enter the password for Proxy/Redirect authentication.
- 8. Click the button to post any changes you make.





Changing SIP Proxy/Redirect Server Details

You can change the IP address or name of a SIP Proxy/Redirect Server, as required. User ID and user password can also be changed. An associated host can be specified, changed or deleted depending on requirements.

To change SIP Proxy/Redirect server details:

- 1. Open the Domain Editor and click the SIP Proxy/Redirect tab.
- Make the required changes to the Server URI, User ID and/or User Password.
- 3. Specify, change or delete the associated host, as required.
- 4. Click the button to save the changes.

Deleting SIP Proxy/Redirect Servers

Any SIP Proxy/Redirect Servers that are no longer required can be deleted. You can delete selected servers (using the Delete button).



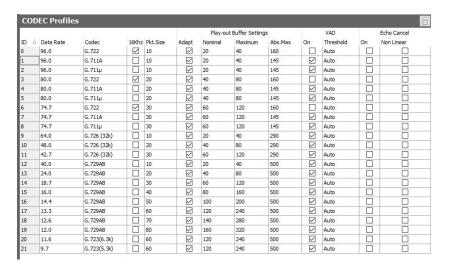
CODEC Profiles

The Domain Editor - CODEC Profiles tab is used to make changes to CODEC profiles used by audio, intercom and telephony hosts in the domain.

Changing CODEC Profiles

To change a CODEC profile:

5. Open the Domain Editor and click the **CODEC Profiles** tab.



- Make the required changes to the supplied default settings. You cannot change the Data Rate, Codec name, Frequency or Packet Size.
- 7. Click the button to post the changes to the database.
- 8. To change the settings back to their defaults, select a profile, right click to show the context menu and select the **Default** option.



Domain Editor - Field Definitions

Domain Editor - General tab

Field/Display Item	Description
Name	The name of the domain. This is usually defined when the domain is first added to the Enterprise using the Enterprise Editor, but can be changed here, if required. The name does not have to be unique. Maximum length for a domain name is 20
	characters.
Comment	Comments relating to a selected domain. To add comments, select the domain (click once on the line), click in this field and enter your comments. Maximum length for comments is 255 characters.
ID	A numerical identifier for the hosts that have been defined. This is set by the system and cannot be changed manually.
Туре	Shows an icon representing the host type.
	To change the type, click on this field, click the button and select the required host type from the list.
Ports	The number of ports on the selected host type. Cannot be changed without changing the host type.
Name	The default name for the host. This can be changed, if required and does not have to be unique. The name specified appears on the dynamic call selection list.
O a manus a mat	Maximum length for a host name is 20 characters.
Comment	Any comments that have been added relating to this host. To add a new comment, click in the field and enter your comment.
	Maximum length for comments is 255 characters.
EDHS	The EDHS address assigned to the host by the Mercury system.
	The EDHS address cannot be changed.
MIU Address	The IP address of the audio card. This is defined using the Host Editor (Audio Card tab).
Card Address	Shows the IP address (or host name) of the host. This is defined using the Host Editor (PC Settings tab).
Cmdr#	The Commander address, which is required when networking with Commander/Orator systems. The address can be changed by clicking in the field and selecting the required address from the drop-down list. If Commander/Orator networking is not being used this value has no effect.
IP Address Mode	Fixed IP must be used.
Gateway IP Host Name	DO NOT USE THIS OPTION.
	Host name selection will only work if the name is listed in a local host lookup table.
	This functionality is only partly supported at present.



Field/Display Item	Description
Gateway IP Address	The IP address of the gateway.
	On a Mercury network with more than one domain, this is the IP address of the router in the current domain which provides access to other domains.
	It may be left as 0.0.0.0 in a single domain network.

Domain Editor - SIP Proxy/Redirect tab

Field/Display Item	Description
Server URI	The IP address or name of the SIP Proxy/Redirect Server.
Туре	Displays an icon which identifies the type of associated host.
Associated Host	If this field is left blank, the user ID and password specified for the Proxy/Redirect Server is used for all hosts in the domain.
	If a host is selected, the specified user ID and password will be used by that host to register with the Proxy/Redirect Server. Different hosts can therefore register with the Proxy/Redirect Server using different credentials.
User ID	The user name for Proxy/Redirect Server authentication.
Password	The password (corresponding to the entered user name) for Proxy/Redirect Server authentication.

Domain Editor - CODEC Profiles tab

Field/Display Item	Description
ID	A number to identify the profile.
	The ID cannot be changed.
Data Rate	Shows the bandwidth of the profile.
	The Data Rate cannot be changed.
Codec	Shows the codec name.
	The name cannot be changed.
16 kHz	Shows whether the profile is a 16 kHz profile.
	This setting cannot be changed.
Pkt. Size	Shows the packet size.
	This setting cannot be changed.



Field/Display Item	Description
Play-out Buffer	Adapt
Settings	When selected, (which is the default setting) this forces the Mercury card to create an input and output buffer for handling audio traffic. This introduces a fail-safe buffer into streaming audio data over an IP network at the expense of latency.
	Nominal
	A dynamic input buffer that controls the throughput of packet transfers. Like a flow control buffer, it sits in the middle of the buffer (1/2 size of maximum delay) and moves up/down depending on incoming packet data.
	Maximum
	An input buffer - the maximum number of voice packets stored in the DSP before send.
	Abs. Max
	The absolute maximum input buffer.
VAD	On
	Enables Voice Activity Detection (VAD).
	Threshold
	A system-wide threshold setting used to differentiate a speaking voice from background noise. This saves system resources by preventing background noise from consuming those resources. If the input level is higher than the configured value, input is then presented on a given input port.
	By default this is set to Auto, which provides control of the value to an internal system algorithm. Range -30 to 0 dB, or Auto.
Echo	On
	Cancels any electrical echoes resulting from mismatched line impedances on the analog port using the IP channel. Typically this would be needed if an external telephone interface was in use.
	Cannot be enabled for 16 kHz profiles.
	Non Linear
	Employs an alternate algorithm to prevent system generated sidetone at user stations for all ports on this host.



Chapter 6 - Hosts and the Host Editor

Overview

A host is a constituent part of a domain. It is the focal point for all user input. A host can be a dedicated piece of hardware, such as an MIU, or a PC with hardware added either internally (PCI card) or externally (USB adapter).

Hosts are added to domains using the Domain Editor. See the "Domains and the Domain Editor" chapter in this Configuration Guide for details.

Once hosts have been added, you must add the panels and subscribers to the correct port on the host and specify the various host settings using the Host Editor.

Panels and subscribers are then configured using the Panel Configuration Editor. See the "Panel and Subscriber Configuration" chapter in this Configuration Guide.



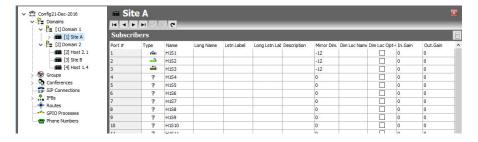
Using the Host Editor

Viewing a List of Subscribers

Locate the host you wish to configure (Site A) in the Enterprise tree.

To view a list of subscribers:

- 1. Expand the Enterprise tree in Gateway.
- 2. Click the "right arrow" next to the selected domain until any hosts contained within the domain are visible in the tree
- 3. Single-click on the relevant host.
- 4. The basic Host Editor is now displayed.

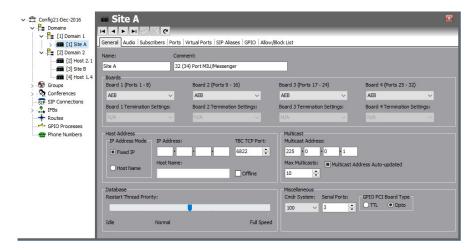


In the example above, three panel subscribers have been added to the Mercury host "Site A". The basic editor allows configuration of most day to day tasks and corresponds to the Subscribers tab of the Advanced Editor which is described below.

Opening the Advanced Host Editor

To open the Host Editor:

- 1. Expand the Enterprise tree until the "Site A" host is visible.
- 2. Double-click on the host to open the Advanced Host Editor.



The Advanced Editor opens at the General tab.



The Host Editor tabs

The Host Editor tabs are used to maintain basic details about the host, to work with subscribers and to specify subscriber and host settings.

General tab

The Details tab shows the name of the host (specified when the host is added to the domain), which can be changed. A comment can also be added, if required. This tab is also used to specify which expansion boards have been fitted into an MIU.

Audio tab

The Audio Card tab is used to specify the audio settings for the selected Mercury host.

Subscribers tab

The Subscribers tab is used to add and maintain the panels and subscribers that are connected to the ports on the selected host.

Ports tab

The Port Settings tab is used to specify the settings for the ports on the selected host. Note that the number and types of ports varies according to the type of host selected.

Virtual Ports tab

The Virtual Ports tab is used to assign targets to virtual ports.

SIP Aliases tab

The SIP Aliases tab is used to associate a SIP connection with a URL (or alias). When a caller attempts to connect to that URL (or alias) on a host, the SIP connection corresponding to the alias is 'activated'. Once activated, a SIP connection key on any panel connected to the host can be targeted to answer the call.

GPIO tab

The GPIO tab is used to define GPIOs on the selected host. These are used elsewhere in the system. GPIOs are control signals received from, or sent to third-party equipment.

Allowed/Blocked List tab

The Allowed/Blocked List tab permits Administrators to set either an exclusive list of IP addresses that a host will accept IP connections from (Allowed) or a specific list of IP addresses that will not be allowed to connect (Blocked). Note that only one of these lists should be populated per host. If the Allowed list contains at least one entry, all IP connection attempts to the relevant host will not be accepted unless the source IP address is on the Allowed list.



Host Editor - General

The Host Editor - Details tab shows the name of the selected host and any comments that have been entered.

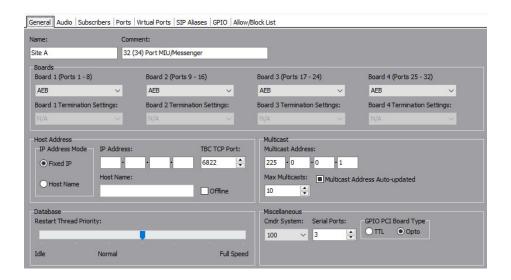
If you are using an MIU, you must also specify which expansion boards have been installed. Ports are labelled on the back of the MIU:

- 1-8: Bottom Board, Slot 1
- 9–16: 2nd Board from the Bottom, Slot 2
- 17–24: 3rd Board from the Bottom, Slot 3
- 25-32: Top Board

Boards that may be fitted into the MIU are:

- Audio Expansion Board (AEB).
- RIB.
- RIB Panel.
- Foreign Exchange Office (FXO).
- Foreign Exchange Station (FXS).
- Ear and Mouth (E&M).
- E1.

It is important that the information is setup accurately under the Host Details tab, so that Digital Signal Processing (DSP) channels are assigned properly.





NOTE: An MIU has a default of three serial ports and eight DSP channels. More DSP can be obtained with an add-in module for the MIU and full-length PCI card.



Changing Host Details

You can change most parameters related to the host.

If you are configuring an MIU, you **must** specify the types of boards that you have installed in the MIU.



IMPORTANT: There is no hardware confirmation check performed by the MIU. If the hardware configuration does not match the system configuration database, unpredictable behaviour will result including the Mercury host not connecting to the network. Also, if you change existing board types, a warning message will be displayed.



IMPORTANT: The host IP address(es) would normally be set at Domain level when the host was added: if not, you should do it now. If you do not complete this step, a warning message will be shown. The system cannot operate correctly until IP settings are in place.



WARNING: If you selected a RIB and subsequently change to another board type and save the changes, you will receive the following warning alerting you that VOX GPIO records automatically created for the RIB will be deleted if you proceed.

Click to post the changes to the database.

If you have selected a TEB, a warning message will advise that the first suitable coding profile will be automatically selected. Click the OK button to continue.



MORE INFORMATION: Refer to the "Host Editor - Details tab" for a description of the fields on this tab.



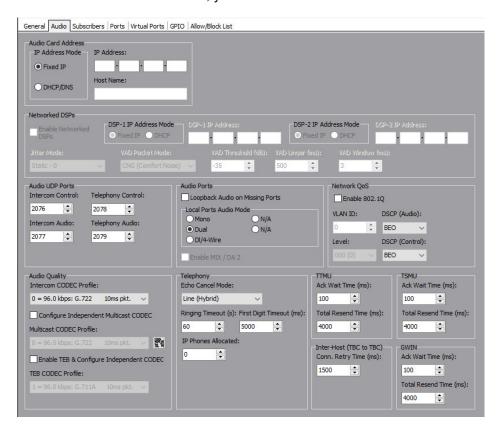
Host Editor - Audio

The Host Editor - Audio tab is used to specify the IP and audio settings for the Mercury hardware. The CODEC profiles for intercom and multicast will default to 0 unless you have installed a TEB or E1 board into an MIU host, in which case they will default to 1.

The tab is also used to set the DSCP values on the host if Quality of Service is implemented at the domain level.

The tab is used to specify the number of DSP channels that are allocated to telephony on the selected host. The CODEC profile used for telephony is also set on this tab if you have a TEB installed in the MIU that you are configuring.

The other key setting is the IP address of the host's Mercury card. This must be a fixed IP address. This will normally have been set when the host was added at domain level but if not, you should do it now.





Changing Audio Settings

Once the IP address of the Mercury card is set, you can change other audio card settings, as required. The configuration of the local ports is set normally to dual channel. Audio may be looped back on any unconnected ports, which is useful for testing the system before it is installed, or for advanced routing. A bandwidth profile can be picked to suit the conditions. It should be noted that not all settings give good results, so care should be taken if this is to be changed. If in doubt use the default.

To change audio settings:

- 1. Open the Host Editor and click the **Audio** tab. The audio settings for the subscribers are displayed.
- 2. Make the required changes.
- 3. Click the button to post the changes to the database.



Host Editor - Subscribers

The Host Editor - Subscribers tab is used to add and maintain the subscribers that are connected to the host. Subscribers include:

- 4-wire.
- Desktop Panels.
- Virtual Panels.
- Telephone Hybrids.
- Telephone Expansion Boards (TEBs).
- Radios and radio phones.
- Virtual 4-wire.
- Patch Panels.



NOTE: If you are using a full-length PCI card host, the subscribers are limited to 4-wire, Patch Panels and Virtual Panels. Radio and radio phone ports can only be declared if a RIB is installed in an MIU. FXO, FXS and E&M ports can only be declared if the relevant board is installed in an MIU.

Adding a Subscriber

When adding subscribers to the host, you must take note of where each subscriber will be physically attached to the ports on the host.

A Virtual Panel is typically "attached" to the last port on the host, although it can be attached anywhere. The location must be specified in the Virtual Panel Settings.

An MIU can have 8/16/24/32 + 2 ports, and can address up to 32 panels, provided that 4 AEBs are installed. Port 33 is not generally used. Port 34 can be used for a Virtual Panel. Any expansion board can be installed in any slot.

To add a subscriber:

- 1. Open the Host Editor and click the **Subscribers** tab.
- 2. Click in the **Type** field and click the button to show a pop-up list of subscriber types.



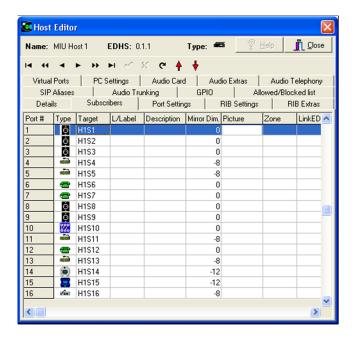
Note that the available types will depend on the host and port selected. Radio port and radio phone are only available for a RIB.

If you select an E&M, FXS or FXO, you can only add the relevant telephone subscriber.





- 3. Select the required subscriber type. If you have installed a TEB, you can only declare four ports on the board.
- 4. Complete the other fields for each port, as required. **Name** defaults to the name of the panel or subscriber and may be altered.
- To add a picture to identify the subscriber on a Virtual Panel, click in the
- 6. **Picture** field and then click the button to browse for and select an image file.
- 7. To place the port under zone control, click in the **Zone** field and click the button to open the Zones Table. Select the required zone and click the **Apply** button.
- 8. Click the button to save the changes.
- 9. Add additional subscribers, as required.





MORE INFORMATION: Refer to the "Host Editor - Subscribers tab" for a description of the fields on this tab.

Changing Subscriber Details

Subscriber details can be changed. For example, you can change or add a listen label for a port or place a port under zone control.

To change subscriber details:

- 1. Open the Host Editor and click the **Subscribers** tab.
- 2. Make the required changes to the details for each subscriber.
- 3. To add or change a picture to identify the subscriber on a Virtual Panel, click in the **Picture** field and then click the button to browse for and select an image file.



- 4. To place the port under zone control, click in the Zone field and click the <u>w</u> button to open the Zones Table. Select the required zone and click the Apply button.
- 5. Click the button to save the changes.

Changing Subscriber Type

You can add additional subscribers to the free ports on your host or change the subscribers that you have previously added.



NOTE: If you have added key assignments to a panel subscriber and then change the subscriber type, your key assignments may be affected.

To change subscriber type:

- 1. Open the Host Editor and click the **Subscribers** tab.
- 2. Locate the subscriber in the list, click in the **Type** field and click the button.
- 3. Select the required subscriber type from the subscriber list.
- 4. Make any other changes, as required.
- 5. Click the button to save the changes. A warning box is displayed.



6. Click the **Yes** button to confirm that you want to change the subscriber type.

Deleting a Subscriber

Subscribers can be removed from ports, as required.

To delete a subscriber:

- 1. Open the Host Editor and click the **Subscribers** tab.
- 2. Locate the subscriber in the list, click in the **Type** field and then click the button to show a list of subscriber types.
- 3. Select None.
- 4. Click the button to save the changes. A warning box is displayed.



Click the **Yes** button to confirm that you want to remove the subscriber.



Host Editor - Ports

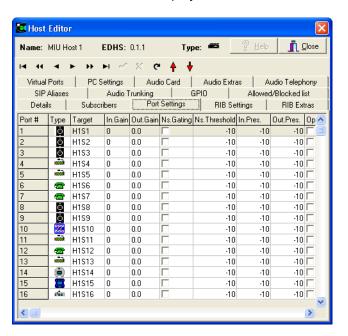
The Host Editor – Ports tab is used to maintain the settings for a particular audio or telephone port.

Changing Port Settings

The settings for each port on a host can be changed, as required. For example, you can change the gain settings, audio levels and thresholds and enable operator recording. The settings for each port can be different.

To change port settings:

 Open the Host Editor and click the **Ports** tab. The settings for the defined subscribers are displayed.



- 2. Select the required port and make the required changes.
- To enable operator recording, select the Op. Rec check box and then select the channel that you want to use for recording from the M/Cast Addr drop- down list. A third-party external recorder is required; please contact Trilogy for more details.
- 4. Click the button to save the changes.



MORE INFORMATION: Refer to the "Host Editor - Port Settings tab" for a description of the fields on the Host Editor - Port Settings tab.



Host Editor - Virtual Ports

Virtual ports allow audio to be routed to an IP channel directly, or for an IP channel to be routed to subscribers or conferences directly. A virtual 4-wire port might be included in a conference and by this means all active participants of the conference will be mixed and sent as outgoing unicast audio on the virtual 4- wire's IP channel. If a virtual 4-wire is then declared on another host, and included in a different conference the two conferences may be interconnected (using a unicast IP channel rather than the multicast channel conferences utilise) by setting up a fixed route with the virtual 4-wire in the first conference as a source and the virtual 4-wire in the second conference as the destination.

The following port numbers can be assigned as virtual 4-wire ports:

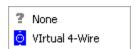
- MIU and full-length PCI card: ports 35 to 66.
- USB and half-length PCI card: ports 35 to 42.

Adding a Virtual 4-Wire Port

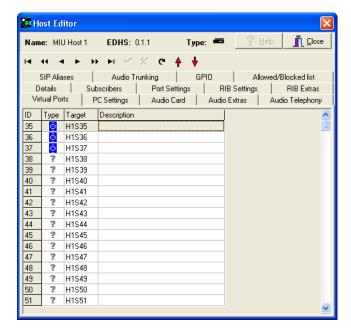
Virtual 4-wire ports can be added to the host, as required. The number of ports that can be declared depends on the host type.

To add a virtual 4-wire port:

- 1. Open the Host Editor and click the Virtual Ports tab.
- 2. Click in the **Type** field and click the button to show a pop-up list of subscriber types.



- 3. Select Virtual 4-Wire.
- 4. Add a description of the virtual port, if required.
- 5. Click the button to post the changes.







MORE INFORMATION: Refer to the "Host Editor - Virtual Ports tab" for a description of the fields on the Host Editor - Virtual Ports tab.

Deleting a Virtual 4-Wire Port

You can delete a virtual 4-wire port by declaring that there is no subscriber attached to a particular virtual port.

To delete a virtual 4-wire port:

- 1. Open the Host Editor and click the Virtual Ports tab.
- 2. Locate the port in the list, click in the **Type** field and then click the button to show a list of subscriber types.
- 3. Select None.
- 4. Click the button to save the changes.



Host Editor - SIP Aliases

SIP is an open, standards-based protocol for negotiating voice and multimedia calls over a network. The addition of SIP provides Mercury with the capability to make and receive SIP VoIP calls to any configured, basic SIP devices that may be present on the network.

A SIP connection is an object used to identify a SIP entity that exists outside of Mercury. SIP connections are added at Enterprise level using the Enterprise Editor. SIP connections are associated with a URL (or alias) using the Host Editor. When a caller attempts to connect to that URL (or alias) on a host, the SIP connection corresponding to the alias is 'activated'. Once activated, a SIP connection key on any panel connected to the host can be targeted to answer the call.



NOTE: SIP aliases are listed on the Host Aliases tab in the SIP Connection Editor.

Buttons at the top of the SIP Aliases tab let you:

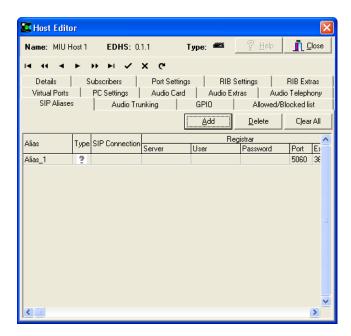
- Add a SIP alias.
- Delete a selected SIP alias.
- Delete all SIP aliases at the same time.

Adding a SIP Alias

You can define an alias for a SIP connection that has been defined using the Enterprise Editor. Required entries are the SIP connection for which you want to create an alias. Registrar and challenge details can be added if these are required for the selected connection.

To add a SIP alias:

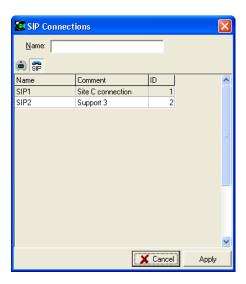
- 1. Open the Host Editor and click the SIP Aliases tab.
- 2. Click the Add button to add a new alias row.



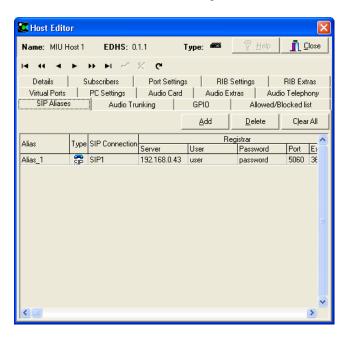
3. Click in the **SIP Connection** field and click the button to show a list of SIP connections. These are added at Enterprise level using the Enterprise Editor. Click the **Panel Sources** button to show a list of



FXO and FXS sources.



- 4. Select the required SIP connection, FXO or FXS source and click Apply.
- 5. Complete the Registrar and Challenge details, as required. To do this click in the required field and type the server name, user name or password. Port and Expires are set using the spin control which appears when you click in the field.
- 6. Click button to save the changes.





MORE INFORMATION: Refer to the "Host Editor - SIP Aliases tab" for a description of the fields on the Host Editor - SIP Aliases tab.



Changing a SIP Alias

Once a SIP alias has been defined, you can change or add registrar and challenge details and also change the listen port and expiry time for registration.

To change SIP alias details:

- 1. Open the Host Editor and click the SIP Aliases tab.
- 2. Make the required changes to the alias details.
- 3. Click the button to post the changes to the database.

Deleting a SIP Alias

SIP aliases can be deleted if they are not required. You can either delete a selected alias (using the Delete button) or delete all aliases at the same time (using the Clear All button).



Host Editor - GPIO

The Host Editor - GPIO tab is used to physically assign all GP inputs and outputs for the selected host. The types that can be defined depend on the host type:

MIU

- Up to 16 expansion GP inputs and up to 16 outputs (an Advantech option board must be fitted in the MIU).
- Up to 34 audio (VOX) inputs and up to 34 outputs.
- Two standard GP inputs and two outputs.

PCI Card

- Two audio (VOX) inputs and two outputs.
- Two standard GP inputs and two outputs.

USB Adapter

- Two audio (VOX) inputs and two outputs.
- Two standard GP inputs and two outputs.



NOTE: If a board is declared as a RIB, the relevant GPIO database entries are created automatically and cannot be changed. RIB GPIOs are not visible in Gateway. If an imported configuration has defined RIB GPIOs the configuration will be updated to declare a RIB as being fitted in the associated expansion board slot. To change parameters relating to RIB card setup and GPIO attack/hold times, etc., see "RIB Settings" on page 7-11.

Buttons at the top of the GPIO tab let you:

- Add a GPIO of the selected type.
- Delete a selected GPIO.
- Delete all GPIOs of the selected type.

Adding a GPIO

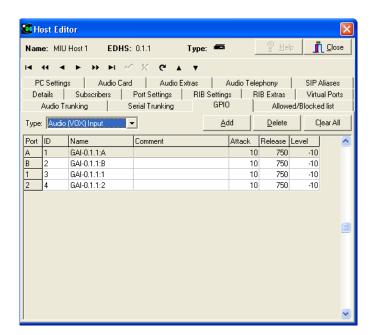
The number and type of GPIO that can be added depends on the host type. Ensure that you select the correct type of GPIO for the selected host type.

The types of GPIO that can be added are: Expansion GPI/GPO, Audio (VOX) Input/Output, Standard GPI/GPO.



To add a GPIO:

- Open the Host Editor and click the GPIO tab to display the currently defined inputs/outputs. You can change the input/output type to show inputs/outputs of different types.
- Select the type of input/output that you want to add from the **Type** drop-down list.
- 3. Click the **Add** button to open the Enter Number dialog box. The maximum number of inputs/outputs of the selected type that you can add is shown.
- Specify how many inputs/outputs you wish to add and click the OK button.
- 5. Make any changes to the settings for the input/output, if required. The settings that can be changed depend on the input/output type.
- 6. Click the 🖊 button to save the changes.





MORE INFORMATION: Refer to the "Host Editor - GPIO tab" for a description of the fields on the Host Editor - GPIO tab.

Deleting a GPIO

Any GPIOs that are no longer required or used can be removed from the host. You can either remove a selected GPIO (using the Delete button) or remove all GPIOs at the same time (using the Clear All button). To remove a GPIO, you must first select the type of GPIO and then select the GPIO itself.



IMPORTANT: Deleting GPIOs will automatically delete any GPIO processes that are based on the deleted GPIO.



Host Editor - Allow/Blocked Lists

The Host Editor - Allowed/Blocked List tab is used to define allowed and blocked IP addresses. At the hardware level, you can set a list of IP addresses that are filtered to prevent or allow calls being received from certain IP addresses. This applies to all calls received at an MIU regardless of whether the destination is a panel, 4-wire, radio, telephone, etc.



IMPORTANT: Call blocking will only work if you have defined a static host IP address rather than a host name.

Call blocking using allow or block lists works independently of any other control mechanism that might otherwise permit or disallow audio routes, such as: zones, areas, access levels, GPI functions, PTT; etc. Blocking is achieved at the IP stack level of the card firmware.

- Allow lists are the most prohibitive and secure. They define the IP addresses from which the host will accept IP traffic. The card will discard any packets from IP addresses not on the allow list.
- Block lists define the IP addresses from which the host will not accept IP traffic. The card will discard any packets from IP addresses on the block list.
- If block list and allow list are both empty, then every IP address is allowed.
- If an IP address is on both lists, the block list takes precedence and blocks rather than allows the address.

Buttons at the top of the Allowed/Blocked List tab let you:

- Add an IP address to an allowed or blocked list.
- Delete an IP address from an allowed or blocked list.
- Delete all IP addresses from an allowed or blocked list.



Adding an IP Address to the Allowed/Blocked List

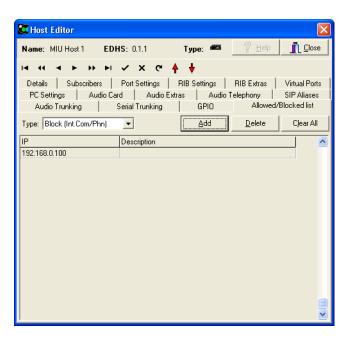
IP addresses can be added to an allowed or blocked list as required. When adding an address, you must choose whether you want to allow or block intercom and phone traffic or SIP traffic.

To add an address:

- 1. Open the Host Editor and click the **Allowed/Blocked List** tab to display any addresses that have already been defined.
- 2. From the Type drop-down list, select Block (Int.Com/Phn), ONLY Allow (Int.Com/Phn), Block (SIP), ONLY Allow SIP.
- Click the Add button to display the Enter An IP Address dialogue box. Enter the IP Address that is to be allowed or blocked and then click OK.



- 4. Enter a description of the IP address, if required.
- 5. Click the button to post the changes.



Deleting an Allowed or Blocked IP Address

IP addresses can be removed from allowed or blocked lists, as required. You can either remove selected addresses (using the Delete button) or remove all addresses at the same time (using the Clear All button). To remove an address, you must first select the type of list and then select the address itself.



Host Editor - Field Definitions

Host Editor - General tab

Field/Display Item	Description
Name	The name of the host. This is usually defined when the host is added to the enterprise using the Domain Editor. See the "Domains and the Domain Editor" chapter in this Configuration Guide for details. The host name can be changed using the Host Editor, if required. The host name does not have to be unique. Maximum length for the host name is 20 characters.
Comment	Comments relating to a selected host. To add
	comments, click in this field and type your comments.
	Maximum length for comments is 255 characters.
Commander System #	A Commander ID is needed only if Commander networking is used. Gateway automatically creates an ID for each Mercury host. This ID is required by the Pathfinder software, which operates as the Configuration Editor for Commander/Orator systems. Cannot be blank. If Commander/Orator networking is not being used this value has no effect.
Board 1: (Ports (1-8)	The expansion board type the TBC is to expect at ports 1-8. Expansion boards must be installed in order, from one to four. If there is only one board installed in an MIU, it must occupy the Board 1 position, if there are two, they must occupy positions 1 and 2, etc. ONLY AVAILABLE FOR AN MIU.
Board 2: (Ports (9-16)	This field is used to specify what expansion board type the TBC is to expect at ports 9-16. ONLY AVAILABLE FOR AN MIU.
Board 3: (Ports (17-24)	This field is used to specify what expansion board type the TBC is to expect at ports 17-24. ONLY AVAILABLE FOR AN MIU.
Board 4: (Ports (25-32)	This field is used to specify what expansion board type the TBC is to expect at ports 25-32. ONLY AVAILABLE FOR AN MIU.
IP Address Mode	Fixed IP The IP address is fixed and should be entered in the IP address field.
	Host Name Mercury peers contact this host using the host name entered here.
Host Name	Specifies the host name of the computer as an alternative to its IP address.
IP Address	Specifies the IP address of the host if Fixed IP is selected.
Multicast Address	The multicast base address. The first multicast used in a pool of addresses is defined by the total number of DSP channels available for conferences in the system. Change this to suit the network environment in which the MIUs are used.



Field/Display Item	Description
Multicast address auto-update	Select this check box for all host multicast addresses to be automatically updated whenever the base address changes. Default = Not selected.
Max. Multicasts	The maximum number of multicast allowed.
Wax. Wullicasts	Default = 10
TBC TCP Port	The TCP listen port used to accept TBC-to-TBC peer communications.
	Default 6822, minimum 1, maximum 65535. Must be the same on all hosts.
Serial Ports	The total number of serial ports available for use by Commander networking.
	Range 0-3.
DSP Channels	Specifies the total number of DSP channels on the Mercury host. This value must be manually set here and is not auto-detected.



Host Editor - Audio tab

Field/Display Item	Description
IP Address Mode	The IP address mode. This is set to Fixed IP .
Inside IP Address	The IP address of the audio card. Enter the relevant address in the space provided.
Subnet Mask	The subnet mask. This must match the network ID of the default gateway.
Local ports audio- mode	Defines the behaviour of the two audio ports built into the USB adapter/Mercury PCI card (analogue ports A and B). These local ports are only available for audio use if defined in the system configuration. It is possible to customise the behaviour of the headset port on a card in the following ways. Mono Analogue ports A and B are joined together.
	Dual Analogue ports A and B are separated.
	DI/4 wire Overrides microphone with Aux line. This is known as "Trunk mode".
	Record (PTT) Routes microphone audio when there are active, outgoing routes, plus received audio for that port at all times, to the 4-wire port allowing an analogue voice recorder to be connected to the 4-wire port (USB and half-length PCI cards only).
	Record (AII) Routes microphone audio plus received audio, at all times, for that port to the 4-wire port allowing an analogue voice recorder to be connected to the 4-wire port (USB and half-length PCI cards only).
Loopback audio on missing ports	When selected, causes the routing engine to automatically loop back any audio routed to configured ports that are not physically fitted to the host.
	This is done when testing and also in some advanced configurations to enable routing of sources to address other sources in the configuration that may not have physical
CODEC profiles	Intercom - the bandwith profile used for intercom.
	Multicast - the bandwidth profile used for multicast. The default bandwidth profile is 0. This gives the highest quality available. Other settings allow for bandwidth restriction with the parameters being chosen to suit the path degradation or to conserve bandwidth if many channels are required. It is possible to use different coding profiles on hosts within the Enterprise. However the codecs must be within the same family and within the same talkback network. Codec profiles are maintained using the Domain Editor. If using telephony, G722 codecs may NOT be used. G722 codecs use 16 kHz, the other profiles use 8 kHz.
	NOTE: The telephony modules can only support 8 kHz sample rate.



Field/Display Item	Description
DSCP (Audio)	Differential Services Codepoints (RFC2474) may be independently set for voice data, on a per-host basis.
	Select the required DCSP setting for audio from the drop-down list.
DSCP (Ctrl)	Differential Services Codepoints (RFC2474) may be independently set for control data, on a per-host basis.
	Select the required DCSP setting from the drop-down list.
Min. IP Phones	The number of DSP channels that are allocated for panels to talk to telephones (FXO) on this host only (i.e. outgoing calls). DSP channels must be reserved for this operation.
Telephony Support	Select this check box to enable telephone support. The CODEC profile can then be changed. Telephone support must be enabled if an FXO card is installed and activated on other hosts that inter- operate with FXO hosts.
Codec Profile	The coding profile used by the main DSP to control telephony ports. Using TEB support forces the system into 8 kHz operation. The G722 codec cannot be used at 8 kHz.
Board 1 – 4 Termination Settings	These settings only apply to FXO and FXS. E&M termination settings are specified in the Panel Editor for the specific E&M port. Available options when enabled are Japan, CTR21, 600-R, 900-R.
First Digit Timeout	Default is CTR21. Determines how long the TBC will wait for valid
(ms)	phone control digits when receiving a call on a TEB port. After this timeout period the call is "forwarded" to any panels with keys targeting that TEB (i.e. the yellow tally flashes indicating an incoming call). Default = 5000 ms. Minimum = 1000 ms, Maximum = 30000 ms.
Associate aliases with outgoing SIP	When selected, no phone number lookup is necessary (provided an alias for the number is defined within the Registrar/Proxy SIP Server) when placing a call. This option is useful when a Registrar Server also acts as a Proxy/Redirect Server. When a call is made, the domain of the SIP Proxy Server is used. For example, calling 111 will be sent as 111@proxyserver-domain.com Default = Not selected.



Host Editor - Subscribers tab

Field/Display Item	Description
Port #	Shows the port number on the host. The number and type of port varies according to the type of expansion board and host selected. This cannot be changed.
Туре	Shows an icon corresponding to the panel/ subscriber type. To specify or change the panel/
	subscriber, click in this field and click the button. Select the required subscriber from the pop-up menu. The list of available panels/subscribers depends on the host and expansion board type fitted in the MIU.
Name	The name of the panel or subscriber that is configured on the relevant port. The name can be changed.
	It is used by default by all panels on keys that target that given subscriber. The name is also used to construct names for routes defined in the Route Editor.
	Maximum length for the target name is 20 characters.
Long Name	If text is entered, this will be used as an alternate to the Name field, on panels with displays supporting 8 characters. If left blank, the Name field will always be used.
L/Label	The listen label is the name displayed on a panel that is listening to the given subscriber. This label overrides the target name, but only when the key targeting the subscriber is a listen-only key type. Maximum length for the label is 20 characters.
Description	Descriptive comments relating to a particular subscriber.
	Maximum length for the description is 255 characters.
Mirror Dim	If two-way (Speak & Listen) communication is open between this and another subscriber, then the panel speaker is dimmed by the amount specified here (in dB). This helps prevent howl around. For this to function at its best, it is advisable to enable it on both subscribers. The default value depends on the subscriber attached to the port.
Picture	Assigns a picture to a particular subscriber so that Virtual Panel Desktop or Square users see that picture when speaking or listening to the subscriber, which can help identify the caller.
	Click the button to browse for and select an image file.
Dim Loc Name	Dimming Location Name is used to select a suitable dimming location, if defined and according to the physical placement of hardware panels.



Field/Display Item	Description
Dim Loc Opt-out	Dimming Location Opt-out checkbox is used to allow specific subscribers to opt-out (i.e. not follow) the rules generated by location based dimming.
CmdrEDHS	Shows the EDHS of the destination trunk. A trunk is a static route that is established when the associated hosts boot, and audio is permanently routed from one 'trunked' port to the other.
	This field also shows the the EDHS of the Commander/Orator host, if the port has been selected as a Commander/Orator port.
CmdrName	Shows the name of the destination trunk.
	This field also shows the the name of the Commander/Orator host, if the port has been selected as a Commander/Orator port.
Owner Host	When making a route from a panel an "owner" is set, usually the panel itself, which then locks the route from being changed by any other panel or TBC.
	To allow two panels to change a route an owner host must be specified. Two panels with the same owner host can control the same route (because route requests effectively appear to come from the same owner).



Host Editor - Ports tab

Field/Display Item	Description
Port #	The number of the port on the selected host. The number and type of port varies according to the type of host selected. This cannot be changed.
Туре	Shows an icon corresponding to the subscriber type. To specify or change the subscriber, click in
	this field and click the button. Select the required subscriber.
Name	Shows the port's name. The name can be changed, if required.
In.Gain	A fixed gain (in dB) that is applied to all incoming audio on this port, regardless of subscriber type.
	Default = 0 dB. Range 12.0 to -34.5 dB, Mute available.
	Click and select the required value from the drop-down list.
Out.Gain	A fixed gain (in dB) that is applied to all outgoing audio on this port, regardless of subscriber type.
	Default = 0 dB. Range 12.0 to -34.5 dB, Mute available.
	Click and select the required value from the drop-down list.
Ns.Gating	Noise gating will mute the input audio if the input audio level is below the configured level (Ns.Threshold) for ≥5 seconds, and unmute again immediately if the input audio level exceeds the set threshold. The level needs to be set according to the requirements of the specific installation.
	To enable noise gating, select the check box.
Ns.Threshold	The audio level below which input audio is muted if Ns. Gating has been enabled.
	Default = -10 dB. Range -10 dB to -50 dB.
In.Pres	This is the audio threshold used to indicate input audio presence on a panel key, when that panel key is configured to display input audio presence (Signal Presence).
Out Proc	Default = -10 dB. Range 0dB to -50dB.
Out.Pres	This is the audio threshold used to indicate output audio presence on a panel key, when that panel key is configured to display output audio presence (Signal Presence). Default = -10 dB. Range 0 dB to -50 dB
Op. Rec	Select the check box to enable recording. By
•	default, recording is disabled for all ports.
	When enabled on a port, audio outgoing to the port, and incoming from the port will be mixed and made available as a single record multicast stream. Operator recording can function on all 34 ports of an MIU simultaneously.
Op. Rec Type	Controls whether Speak, Listen or Both Directions are sent to the multicast stream to be recorded.



Field/Display Item	Description
M/Cast Addr	A multicast channel used for recording operator audio from this Mercury host port. Note that an external recording device is required.
	By allocating the same multicast address to multiple ports, audio to and from those ports will be mixed and streamed on the single multicast address.
	When multicast addresses are selected for recording, they will no longer be available for use by conferences.

Host Editor - RIB - Standard and Advanced

RIB Settings are provided here, using the Host Editor, and also at Subscriber level. More advanced settings are available from the Subscriber Editor and any changes made in either place are replicated in the other.

Please refer to Chapter 12 - Radio Port and Radio Phone- Subscriber Configuration on page 152 for more information.



Host Editor - Virtual Ports tab

Field/Display Item	Description
ID	Shows the ID of the port. The total number of IDs will be equal to the maximum number of IP channels on the host. The ID cannot be changed.
Туре	Shows an icon corresponding to the subscriber type. To specify or change the subscriber, click in
	this field and click the button. Select Virtual 4- wire or None from the pop-up menu.
Name	The name of the Virtual 4-wire that is configured on the relevant port. The name can be changed.
	It is used by default by all panels on keys that target that given subscriber. The name is also used to construct names for routes defined in the Route Editor.
	Maximum length for the target name is 20 characters.
Description	Descriptive comments relating to a particular Virtual 4-wire.
	Maximum length for the description is 255 characters.

Host Editor - SIP Aliases tab

Field/Display Item	Description
Alias	Shows the SIP alias. The alias name can be changed, as required.
Туре	Shows an icon representing a SIP connection.
Registrar	The details used by the Mercury SIP endpoint when registering with a SIP Registrar Server. Server The IP or DNS server name.
	User The username (defaults to the alias name) required to authenticate.
	Password the password required to authenticate (optional).
	Port The port that the SIP Registrar Server is listening on.
	Expires The expiry time for registration (seconds).
Challenge	Used for incoming call authentication. When a SIP Proxy or SIP device (e.g. telephone) attempts to call this SIP endpoint a challenge username/password is required by the calling device.
	User The username used for challenge.
	Password The password used for challenge.



Host Editor - GPIO tab

Field/Display Item	Description
Туре	
ID	A numerical identifier for each port.
Name	The name of the GPI or GPO. By default when GPIOs are added, a distinguishable name is automatically given that describes the GPIO type and port number. The name does not have to be unique. The maximum length of the name is 20 characters.
	A & B are the local PCI audio ports commonly referred to as onboard.
	GII/GIO A & B The onboard GPIO ports included on the full-length PCI and half-length PCI cards.
	GPI/GPO 1-16 The ports on the 16 port expansion card.
	GAI/GAO 1-32 GPI/GPO fired on voice activity detection 'VOX'.
Comment	Any comment relating to this port. Add a new comment in the space provided.
Attack	The wait time before a GPIO is said to have occurred, i.e., setup time (debounce).
	Minimum value is 0 ms, Maximum is 5000 ms, in steps of 10 ms.
Release	The wait time before a GPIO is said to have been released i.e., hold time.
	Minimum value is 0 ms, Maximum is 5000 ms, in steps of 10 ms.
Level	The level above which a GPIO will consider the GPIO is related to voice rather than noise.
	(Audio Input and Output only.)

Host Editor - Allow/Blocked List tab

Field/Display Item	Description
Туре	Select the type of IP address that you want to allow or block.
IP	The IP address that will be allowed or blocked.
Description	The description of the IP address.



DCSP Details

A Differentiated Services Codepoint (DSCP) is comprised of the upper six bits of the ToS byte in IP packets. In the switch default QoS configuration, some codepoints are configured with default 802.1p priority settings for Assured- Forwarding and Expedited Forwarding, while others are unused and listed with no-override as the priority. There are 64 possible codepoints.

DSCP Name	DSCP Number	PHB Name	PHB Number	Drop Probability Name	Drop Probability Number
BE0	DSCP0	BE	000	None	00
BE1	DSCP2	BE	000	Low	01
BE2	DSCP4	BE	000	Medium	10
BE3	DSCP6	BE	000	High	11
AF10	DSCP8	AF1	001	None	00
AF11	DSCP10	AF1	001	Low	01
AF12	DSCP12	AF1	001	Medium	10
AF13	DSCP14	AF1	001	High	11
AF20	DSCP16	AF2	010	None	00
AF21	DSCP18	AF2	010	Low	01
AF22	DSCP20	AF2	010	Medium	10
AF23	DSCP22	AF2	010	High	11
AF30	DSCP24	AF3	011	None	00
AF31	DSCP26	AF3	011	Low	01
AF32	DSCP28	AF3	011	Medium	10
AF33	DSCP30	AF3	011	High	11
AF40	DSCP32	AF4	100	None	00
AF41	DSCP34	AF4	100	Low	01
AF42	DSCP36	AF4	100	Medium	10
AF43	DSCP38	AF4	100	High	11
EF0	DSCP40	EF	101	None	00
EF1	DSCP42	EF	101	Low	01
EF2	DSCP44	EF	101	Medium	10
EF3	DSCP46	EF	101	High	11
IC0	DSCP48	IC	110	None	00
IC1	DSCP50	IC	110	Low	01
IC2	DSCP52	IC	110	Medium	10
IC3	DSCP54	IC	110	High	11
NC0	DSCP56	NC	111	None	00
NC1	DSCP58	NC	111	Low	01
NC2	DSCP60	NC	111	Medium	10
NC3	DSCP62	NC	111	High	11



Chapter 7 - Subscriber Configuration

Overview

The Mercury system can be exposed to users by a variety of Virtual Panels (PC users) or Hardware Panels (non-PC users). These panels can be provided to individual users anywhere on the network.

Subscribers are added to a host using the Host Editor. Once they have been added, you must configure the panel type subscribers, define the targets of the panel keys and specify settings. To do this, use the Panel Configuration Editor.

Configuring a panel or subscriber allocates targets to each key on the selected panel. Some subscribers, for example a 4-wire source, require no configuration, whereas others require you to input details of the targets, key modes and key settings.

Depending on the type of expansion board fitted to an MIU, different subscriber types are supported such as radios or telephones. The subscriber editor appearance and complexity varies greatly according to type and each is dealt with in later chapters.

If you have defined globally-available phone numbers using the Phone Number Editor, you can allocate selected numbers to a panel for recall later. See "Phone Numbers and the Phone Number Editor" for more information on phone numbers.



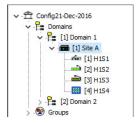
Using the Subscriber Configuration Editor

Viewing a Summary of Subscribers on a Host

You can view a list of all subscribers that are attached to a particular host by selecting the host in the Enterprise tree.

To view a list of subscribers:

- 1. Expand the Enterprise tree in Gateway.
- 2. As the tree expands, a list of subscribers is shown in the tree, below the host.



3. As with other areas of Gateway, a single-click launches the basic editor – double-click launches the advanced editor.



Configuring Panels and Subscribers

- To configure an 8-button rack panel, 10-button rack panel, 8-key desktop panel, 16-key rack panel or 32-key rack panel, see the "Hardware Panels - Panel Configuration" chapter in this Configuration Guide.
- To configure an FXO subscriber, see the "Foreign Exchange Office (FXO) - Subscriber Configuration" chapter in this Configuration Guide.
- To configure an FXS subscriber, see the "Foreign Exchange Station (FXS) - Subscriber Configuration" chapter in this Configuration Guide.
- To configure an E&M subscriber, see the "Ear and Mouth (E&M) -Subscriber Configuration" chapter in this Configuration Guide.
- To configure a 4-wire subscriber, see the "4-wire Subscriber Configuration" chapter in this Configuration Guide.
- To configure a telephone subscriber, see the "Telephone Subscriber Configuration" chapter in this Configuration Guide.
- To configure a radio port, see the "Radio Port and Radio Phone Subscriber Configuration" chapter in this Configuration Guide.
- To configure a radio phone, see the "Radio Phone Subscriber Configuration" chapter in this Configuration Guide.
- To configure a Virtual Panel Square, see the "Virtual Panel Configuration" chapter in this Configuration Guide.



Chapter 8 - Hardware Panel – Subscriber Configuration

Overview

A Hardware Panel is added to a host using the Host Editor.

Once you have added the panel, you must define the key targets, key modes and key settings using the Panel Configuration Editor.

- For more information on adding panels to a host, see the "Hosts and the Host Editor" chapter in this Configuration Guide.
- For basic information about opening the Panel Configuration Editor and an explanation of key modes, see the "Panel and Subscriber Configuration" chapter in this Configuration Guide.

As with other areas of the Gateway application, basic and advanced editors are provided. The basic editor provides an intuitive drag and drop interface, making it straightforward to add, delete and edit key targets. The basic editor appears within the advanced editor as the **Sources Layout** tab.



Hardware Panel Editor Tabs

The advanced Hardware Panel editor comprises seven tabs, outlined below.

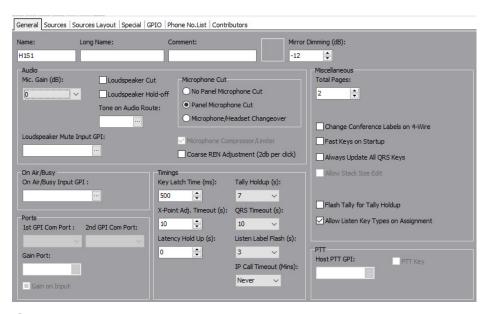
- General
- Sources
- Sources Layout
- Special
- GPIO
- Phone No. List
- Contributors

These are considered in the following sections.



Hardware Panel - General

The Panel Configuration - General tab is used to specify basic details about a Hardware Panel.



Changing Panel Details

You can change the name of the panel and also enter a comment to provide further information about the panel. A picture can be added to identify the panel on a Virtual Panel. Further fields provide control over the operation of the panel cut switch, microphone gain the number of available shift pages and other aspects of behaviour.



NOTE: The name of the Hardware Panel is the target name which is allocated when the Hardware Panel is added to the host using the Host Editor. The default name will usually be edited as appropriate. This is the name which other people will see when they speak to you.

To change panel details:

- Select a Hardware Panel from the list of panels/subscribers attached to a host and open the Panel Configuration Editor. The **Details** tab is selected by default.
- Make the required changes. If you change the name of the panel, the name will be updated if the panel has been added as a target on another subscriber or panel.
- 3. Click the button to post the changes to the database.



MORE INFORMATION: Refer to the "Panel Configuration - Details tab" for a description of the fields on the Panel Configuration - Details tab.



Hardware Panel - Sources

The Panel Configuration - Sources tab is used to assign sources to Hardware Panel. You can assign a full range of sources to the panel by selecting a source type as the target for the particular key on the panel.

Defining Key Targets

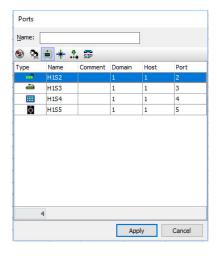
Hardware Panels have from 8 to 32 buttons/keys and a shift button/key which allows further sources to be defined. Each source type that you add will have a default speak/listen mode, which can be changed, if required.



NOTE: You can change the number of shift pages on the General tab to provide access to up to 4 pages, if required.

To add a panel key target:

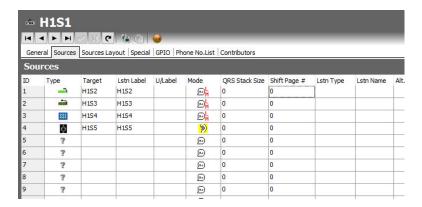
- 1. Select a Hardware Panel from the list of panels/subscribers attached to a host and open the Panel Configuration Editor.
- 2. Click the Sources tab.
- 3. Click in the **Target** field next to the ID of the key that you want to assign.
- 4. Click the button and select the required target from the Panels/Sources dialog box. You can change the target type to a Group, Conference, GPIO/ Audio Route, IFB or SIP connection by clicking the appropriate button at the top of the dialog box.



- 5. Click the **Apply** button to assign the target.
- 6. Add a user-label (U/Label), if required. L/Label defaults to the name of the target and cannot be changed; if a U/Label is entered, then this is used instead of the L/ Label.
- 7. Click in the **Mode** field, click the button and select the operation type for the key. The provided default is usually the correct mode.
- 8. To display a picture on a receiving Virtual Panel when the key is activated, click in the Alt. Picture field, click the button and select the required image.
- Change other settings, if required. You can dim the panel loudspeaker when the target subscriber speaks to the panel user (**Dimming**) and also adjust the incoming (listen) crosspoint gain for the targeted source (**Xpt Gain**).



- 10. If you do not want users to be able to change this key locally to choose a new destination, clear the **Assign** check box.
- 11. Add additional key targets, as required.
- 12. Click the button to post the changes to the database.





MORE INFORMATION: Refer to the "Panel Configuration - Sources tab" for a description of the fields on the Panel Configuration - Sources tab.

Changing a Key Target

Key targets (sources) on a Hardware Panel can be changed, as required. To do this, change the target for the relevant port.

To change a key target:

- 1. Select a Hardware Panel from the list of panels/subscribers attached to a host and open the Panel Configuration Editor.
- 2. Click the Sources tab.
- 3. Click in the **Target** field next to the ID of the key that you want to change.
- Click the button and select the required target.
- 5. Change the other settings, as needed.
- 6. Click the button to post the changes to the database.

Deleting a Key Target

You can delete a key target (source) by clearing the target for a particular port.

To delete a target:

- 1. Select a Hardware Panel from the list of panels/subscribers attached to a host and open the Panel Configuration Editor.
- 2. Click the Sources tab.
- 3. Click in the **Target** field of the source that you want to delete.
- 4. Press the **Delete** or **Backspace** key on your keyboard.
- 5. The target is immediately deleted and the action cannot be undone.



Hardware Panel – Source Layout

The Panel Configuration – Source Layout tab provides an easy to use, intuitive interface to assign sources to Hardware Panel. You can assign a full range of sources to the panel by selecting a source type as the target for the particular key on the panel. The Source Layout tab also serves as the basic hardware panel editor, opened by single-clicking on a hardeware panel from the enterprise tree.

Defining Key Targets

Hardware Panels have from 8 to 32 buttons/keys and a shift button/key which allows further sources to be defined. Each source type that you add will have a default speak/listen mode, which can be changed, if required.

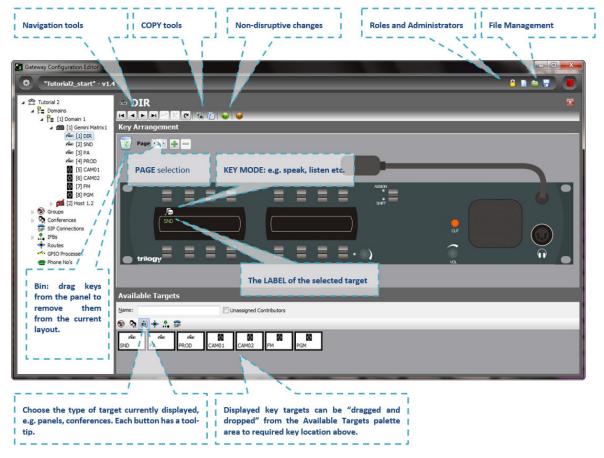


NOTE: You can change the number of shift pages on the General tab to provide access to up to 4 pages, if required.



To add a panel key target:

Ensure the tree view on the left is fully expanded to show the configured subscribers. Single click on the first of your panels to launch the panel editor.



To add keys to a control panel, simply click on one of the available targets from the palette area in the lower part of the screen. Now drag and drop this target on to the required key location area of the panel. As you reach the correct "drop" point, a small rectangle will be highlighted.

- You can also rearrange the layout of keys on the panel using the same drag and drop mechanism.
- To speed up the process, you can select multiple targets from the
 palette using the normal Windows control and shift key functions. As
 you drop these multiple targets onto the panel, any existing
 assignments will be overwritten.
- Many of the Trilogy control panels use lever keys which have both an up and a down action. This allows much greater flexibility since every key can be programmed with a number of *modes*. For example:
 - Press upwards to open a latched speaking route (i.e. hands free). Press downwards for a momentary route (i.e press-to-talk).
 - Press upwards to listen to a 4-wire circuit (e.g. a camera).
 Press downwards to speak to that 4-wire.
- The key mode is preset and will be intelligently set according to the type of target you have selected. To edit the preset mode, click on the settings icon near the top right corner of the panel pane.



 To edit the mode of a key once it has been assigned, left click on the key icon. A pop-up selection will appear. The most widely used modes are described.



- Speak. This key operates as a latching speak key if you briefly press
 the key, otherwise it is a momentary speak, making a speak route
 only for the duration of the keypress.
- Latching /Momentary Speak. This key will always be a momentary speak if the key is pressed downwards, and is always latching in the upwards direction. To turn off the latched speak, press up for a second time.
- **Speak/listen**. Has the latching/momentary characteristics of the **speak** key, but the downwards motion is for speaking and the upwards motion for listening.
- QRS. The QRS is a smart "quick response" key that allows a reply to
 the most recent incoming call to be made without first having to
 locate the specifically configured key for that caller on the panel. The
 label of a QRS key will change from < > to show the identity of the
 caller.

Gateway offers an array of key types, all of which may be selected from the dialogue that pops up. An explanation of all the available key types is given later.

Configure keys as required for the panels in your system.



NOTE: Panel-to-panel intercom almost always uses **speak** keys, with **no need** for corresponding listen keys at panels. Communications with 4-wires almost always need a listen keys, but may need a speak/listen key.



Editor Shift Page Selection



NOTE: additional "shift" pages are also available, increasing the number of key targets that a given panel may have – the available pages can be easily selected within Gateway and by the panel operator. Additional shift pages may be added using the Advanced Editor but try to avoid adding too many or operators may become confused.



Hardware Panel - Special

The Special tab is used to specify special key features including the PTT action of a key on the panel.

Changing Special Panel Settings

Special panel settings can be changed, as required. For example, you can enable signal presence which indicates when a key on the panel has audio, set the maximum number of speaks and listens for conferences and enable ringing for a telephony source on the panel.

To change special panel settings:

- 1. Select a Hardware Panel from the list of panels/subscribers attached to a host and open the Panel Configuration Editor.
- 2. Click the **Special** tab.



NOTE: Details of any targets that have been defined on the Sources tab are displayed. Note you can add additional sources by clicking in the Target field on the Special tab.

- 3. Adjust the special settings for each target, as required.
- 4. Click the <u>under the state of the state o</u>





MORE INFORMATION: Refer to the "Panel Configuration - Special tab" for a description of the fields on the Panel Configuration - Special tab.



Hardware Panel - GPIO

The Panel Configuration - GPIO tab is used to define GP inputs and outputs for a Hardware Panel. GPIO routing statements are explained in the "GPIO and the GPIO Processes Editor" chapter in this Configuration Guide.

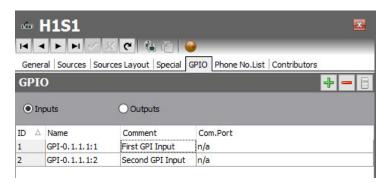
- A maximum of three inputs and outputs can be added on a Hardware Panel. Buttons at the top of the GPIO tab let you:
 - Add a new GPI or GPO.
 - Delete a selected GPI or GPO.
 - Delete all GPIs or GPOs at the same time (after confirmation).

Adding a GP Input or Output to a Hardware Panel

You can add up to three GP inputs and three GP outputs (two may be defined on a desktop panel). These will be automatically named by the system. A comment can be added to provide more information, if required.

To add a GP input or output:

- 1. Select a Hardware Panel from the list of panels/subscribers attached to a host and open the Panel Configuration Editor.
- 2. Click the GPIO tab.
- 3. Click the **Inputs** or **Outputs** radio button, as appropriate.
- 4. Click the green + button to add an input or output.
- 5. Click in the **Comment** field and type a comment, if required.
- 6. Click the button to post any changes you make.





MORE INFORMATION: Refer to the "Panel Configuration - GPIO tab" for a description of the fields on the Panel Configuration - GPIO tab.

Deleting GP Inputs and Outputs

Any GP inputs or outputs that are not required on Hardware Panels can be deleted. You can either delete selected inputs/outputs (using the Delete button) or delete all inputs/outputs at the same time (using the Clear All button).



IMPORTANT: If you delete a GPIO it will be removed from any GPIO processes that have been configured to use the selected GPIO. The process will not be deleted, but will not function as expected.



Hardware Panel - Phone Number List

If you have defined globally-available phone or SIP numbers using the Phone Number Editor, these can be added to the configuration and then recalled on a Hardware Panel.

- Phone numbers are outgoing digits that may be used by panels as speed- dials when accessing FXO, FXS or E&M ports. The digits are dialled as DTMF when the call is placed.
- SIP numbers are outgoing digit sequences that map to either fully formed URIs or are digits that are passed to a Proxy Server where they are fully resolved.

Buttons at the top of the Panel Configuration - Phone No. List tab let you:

- Add a phone/SIP number.
- Delete a selected phone/SIP number.

Adding a Number to a Hardware Panel

To add a number to a Hardware Panel, you must first ensure that you have created the required numbers using the Phone Number Editor. See the "Phone Numbers and the Phone Number Editor" chapter in this Configuration Guide for more information on phone and SIP numbers.



NOTE: Phone and SIP numbers are available globally within the system and the same number can be added to more than one panel. When you add phone or SIP numbers you select from the list of numbers added using the Phone Number Editor and as such you cannot make changes to these numbers on the Panel Configuration - Phone No. List tab.



To add a number:

- 1. Select a Hardware Panel from the list of panels/subscribers attached to a host and open the Panel Configuration Editor.
- 2. Click the Phone No. List tab.
- Click the Add button to open the Phone Numbers list. Both phone numbers and SIP numbers (defined using the Phone Number Editor) are shown.



4. Select the number that you want to add (to select more than one number, hold down the Ctrl key) and click the Apply button to add the number(s) to the panel.





MORE INFORMATION: Refer to the "Panel Configuration - Phone No. List tab" on page 9-16 for a description of the fields on the Panel Configuration - Phone No List tab.

Removing Numbers from a Hardware Panel

Any numbers that are not required on a Hardware Panel can be deleted from the Phone No. List tab, if required. You can either delete selected numbers (using the Delete button) or delete all numbers at the same time (via the right-click context menu).



NOTE: Deleting numbers from a Hardware Panel does not delete the numbers from the Phone Number Editor or from other panels which may be using the numbers.



Hardware Panel - Contributors

The Hardware Panel – Contributors tab shows a list of reverse key target assignments. The list shows other panels which have keys targeting the current panel.

In the example below, panel H1S1 has been selected.



Thus, panels H1S3 and H1S4 have keys which are "pointing at" H1S1.

Most fields are read-only but the following may be adjusted:

- Null mirror dimming
- X-Point level

Click the button to post any changes you make.



Panel Configuration (Hardware Panel) - Field Definitions

Panel Configuration - General tab

Field/Display Item	Description
Name	The name of the Hardware Panel. This is usually defined when the panel is added to the host using the Host Editor. See the "Hosts and the Host Editor" chapter in this Configuration Guide. The name can be changed using the Panel Configuration Editor, if required.
	Maximum length of the name is 20 characters.
Comment	Comments relating to the selected panel. Add your comments in the space provided.
Picture	The picture that has been assigned to the Hardware Panel. To assign a picture, or change an existing one, click the Load button and then browse for and select the required image file.
Mirror Dimming	Mirror dimming helps to reduce acoustic feedback. Range 0 to -60 dB. Default = -8 dB (rack-mounted panels) and -12 dB (desktop panel).
Mic. Gain	The amount of gain that can be applied to the microphone or input associated with this panel. Default (recommended) = 0 dB. Range 12 to -9 dB in 1 dB intervals.
Key latch time	Used to determine the period of time at the panel that differentiates between a momentary and latching action. Default = 500 ms. Range = 200–5000 ms.
Change Conference Labels on 4-Wire	Keys assigned to conferences can be set so that they change their label when a subscriber joins a conference. If the subscriber is a 4-wire it is often a permanent "standing feed" into the conference, so the default conference label will always be overridden to show the 4-wire name. This setting disables such behaviour so the label will not change for 4-wire sources. Default = Not selected.
Loudspeaker Hold-off	Presets the volume at which a panel speaker will operate at its the lowest possible value. Default = Not selected.
Loudspeaker Cut	When selected, cuts the loudspeaker when the Cut key/button is pressed. Default = Not selected.
Microphone Cut	Defines the action of the Cut key/button on Hardware Panels. No panel microphone Cut - Disables microphone cut on a panel. Panel microphone Cut (the default) - Pressing the button cuts the microphone completely. Microphone/Headset Changeover - Pressing the button transfers Audio/Mic to the headset jack of the panel.



Field/Display Item	Description
Shift Pages	Specifies the number of pages available on the
	panel. Default = 2. Maximum = 32.
PTT Host GPI	Defines which Host GPI will be used to provide
	PTT functionality for the Hardware Panel.
Assign-mode Timeout(s)	Specifies when a panel enters assign mode. (This is used to configure panel levels and key types). If no input is received in the timeout period, the panel leaves assign mode and enters normal operation. This normally does not need be changed. Default = 10. Range = 1–60 in 1 second intervals.
Tally Holdup(s)	The timeout of a yellow listen tally if another panel speaks to the user and then hangs up. A listen tally is displayed after hang-up for this timeout value.
	Default = Disable. Range = 1–60 seconds.
QRS time-out(s)	The timeout value for the last calling subscriber to be displayed on the QRS key/button. Default = Never. Range 15–300 seconds.
Always update QRS	When selected, the QRS label changes whenever
	any route to the panel is made regardless of whether there is a button to make the connection or not. Connections can therefore be 'viewed' from any page.
Latency hold up (s)	Applies to 'Fast Keys'. These work by establishing the IP channel needed when the key is first pressed and thereafter simply make/break local audio crosspoints each time the key is pressed/released. If the key is not pressed for the "latency hold up" period the IP route is torn down. Default = 0 seconds. Range = 0–3600 seconds.
Fast keys on startup	If enabled, any fast key IP routes needed are made at TBC startup in anticipation of them being needed straight away. If disabled, they are not made until required. Default = Not selected.
Tone on Audio Route	Configuration of an audio tone that is played out every time a call is received.
Coarse Ren Adjustment	Changes the DB steps used when adjusting the XPT level using a REN.
Loudspeaker Mute Input GPI	Configuration of a GPI to be used to mute the Loudspeaker when switched on.
OnAir/Busy GPI	Allows the configuration of a GPI that will prevent routes from being made to the panel whilst On Air. Essentially an external Do Not Disturb switch.
GPI Com Ports	Configuration of PC Com ports to be used as GPIs
Gain Port	
XPT Adjustment Timeout	The amount of time you can adjust the XPT level using a REN after a call is placed/received.



Field/Display Item	Description
Listen Label Flash	Amount of time to flash the label when using Listen Label functionality.
IP Call Timeout	
Flash Tally for Tally Holdup	Whether the Tally Holdup flashes or is solid after a call is ended.
Allow Listen Keys on Assignment	If unchecked will prevent Listen keys from being assigned using the Assign key on a panel. Prevents snooping.

Panel Configuration - Sources tab

Field/Display Item	Description
ID	A number which corresponds to a key/button on the Hardware Panel. IDs prefixed by P2 (to P32) are keys/buttons on the shifted pages.
Туре	The key's target subscriber type indicated by an icon.
Target	The key's target subscriber name. This cannot be changed here but can be changed by opening the target in the relevant editor.
L/Label	Listen Label. This is the name of the target that will be displayed on the panel. It cannot be changed, but can be overridden if you specify a U/Label.
U/Label	User Label. A customisable user label. If this is specified, it will be used instead of the L/Label. Type the label that you want to use into this field.
Mode	The key's operation type. To change the operation
	type, click in the field and then click the button to choose the required mode.
QRS Stack Size	For QRS keys only, sets the size of the QRS stack. If the stack size is exceeded, the oldest calls are thrown away.
Shift Page #	On panels with multiple shift pages, allows a key to be defined to "jump" straight to a specific shifted page. This can bypass the normal multiple press panel Shift key process.
L/Type	Displays the icon associated with the listen key's target or IFB listener.
L/Target	Allows a panel to "split" a key so that the listen and speak are attached to two different targets. If the key is targetting an IFB and a listener has
	been defined, this field will show the name of the defined listener.
Alt. Picture	Displays a picture when the key is activated. This is useful for showing the operator a graphical representation of the caller on a Virtual Panel.
Dimming	Dims the panel loudspeaker when the target subscriber speaks to the panel user. This loudspeaker dimming dims ALL incoming audio.
	Default = 0 dB. Range is 0 to -28 dB in 2 dB steps, or -100 dB.



Field/Display Item	Description
Xpt. Gain	A value applied to the crosspoint volume between the configured panel and the destination subscriber. This value is used to adjust the incoming (listen) crosspoint gain for the targeted source and can be used to compensate for an inconsistent audio level from that source compared to all other sources.
	Default = 0 dB. Minimum = -60 dB, maximum = 12 dB.
Auto-dial	Specifies a dial sequence that is automatically dialled when a phone key/button is pressed on the panel.
	Applicable to telephony and SIP connections only.
Assign	When selected, (the default), allows the panel user to customise this particular key, choosing a new destination (Subscriber, Conference, Group or Route) and Type (Speak, Listen etc.). If you clear this check box, users will not be able to change this key. You could also consider using Zones and Areas if more security or restrictions are required. Restricting key presses is useful to prevent panels from talking to certain subscribers. Default = Selected.
	Default - Selected.



Panel Configuration - Special tab

Field/Display Item	Description
ID	A number which corresponds to a key/button on the Hardware Panel. IDs prefixed by P2 (to P32) are keys/buttons on the shifted pages.
Туре	The key's target subscriber type.
Target	The key's target subscriber name.
Mode	The key's operation type. This varies depending on the panel type being configured.
Signal Pres.	If selected, allows a panel user to see when a particular key has audio via a slow flashing yellow tally on the key. The user can then elect to listen to it or not. (Conference, 4-wire, telephony and radio keys only). Default = Not selected.
GPO On	When selected, the GPO will be on when the key is pressed.
GPI	The host COM port (1 or 2) to be used as a PTT trigger.
PTT	Defines the PTTaction for the key. Can be set to Mute or Unmute. Mute prevent the user from being heard by the listen user. It does not tear down the route; it simply mutes the volume when the key is pushed.
Key Set	Groups keys together to provide some sort of interaction between the keys in the set. Default = n/a. Range 1–256.
Max. Listens	The maximum listens per given key set group. This is key-driven to prevent more than the defined number of listens. For example, if this is set to 1, then only one key on the set may listen at any one time. Default = n/a. Range 1–256. Can only be set if you have specified a key set.
Max. Speaks	The maximum speaks per given key set group. This is key-driven to prevent more than the defined number of simultaneous speaks. For example, if this is set to 1, then only one key on the set may speak at any one time. Default = n/a. Range 1–256. Can only be set if you have specified a key set.
Ringing	Enables ringing for an E&M, FXS, FXO, SIP connection or telephone. Only one 'ringing source' can be selected.
Tone	If ringing has been enabled, click and select the required ringing tone.



Conference Priority	Only editable if the key is targeting a conference. Gives that key a priority to the specified conference, so that if the conference hits its max speak limit, panels will be blocked from speaking based on their priority. For example, if a conference has max speaks of 3 and two priority 1 and one priority 2 panels are already talking, a priority 3 panel will have to wait for one of the speakers to break their route. However, if a priority 1 panel tried to make a route to the conference then the priority 2 panel would be removed.
Hold/Dialpad	This allows you to place a FXO call on hold and then use the Dialpad to DDI.
Disable Keypad	
Auto Ans (SIP)	Any incoming SIP calls are automatically answered at tyh panel, giving a more intercom like experience for the operator.

Panel Configuration - Phone No. List tab

Field/Display Item	Description
Name	The name of the phone number entry as defined using the Phone Number Editor. Cannot be changed here.
Telephone Number	The phone number entry as defined using the Phone Number Editor. Cannot be changed here.



Panel Configuration - GPIO tab

Field/Display Item	Description
ID	The ID of the GPI in the database.
Name	The name of the GPIO.
	For GPIs this is the subscriber EDHS prefixed with GPI and suffixed with the GP pin value. For example, GPI-0.1.1.3:1
	Names do not have to be unique.
	Maximum length for the name is 20 characters.
Comment	Comments relating to the GPI. You can add comments in the space provided, as required.
	Maximum length for comments is 255 characters.

Panel Configuration - Contributors tab

Field/Display Item	Description
Туре	An icon indicating the type of subscriber contributing, e.g. another panel, a radio or telephone.
Name	The name of the contributing subscriber
Host Name	The name of the host, to which the contributing subscriber is connected.
EDHS	Full EDHS address of the contributor.
Targets	Icon.
Null Mirror Dimming	Check the box to override the application of mirror dimming for this subscriber pair.
X-Point level	The gain level of the contributing "source" to this subscriber.



Chapter 9 - Foreign Exchange Office (FXO) - Subscriber Configuration

Overview

Foreign Exchange Office (FXO) Telephony ports require the correct expansion board to be fitted to a Mercury Interface Unit (MIU). The FXO board supports up to 4 telephone ports which must be connected to either PTT telco analogue provided circuits or to a suitable in-house PABX.

The addition of an FXO board limits the range of supported coding profiles and Gateway will automatically select a suitable coding profile on this host. However, if you are designing a multi-host Mercury system you must take account of this and adjust the coding profile of other hosts. Any incompatibilities will cause warning messages until resolved.

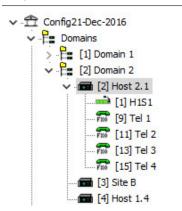
A Foreign Exchange Office (FXO) subscriber is added to a host using the Host Editor. Once you have added an FXO subscriber, you must define the phone controls (if used), GPIs and operational settings using the FXO Subscriber Configuration Editor.

- For more information on adding FXO ports to a host, see the "Hosts and the Host Editor" chapter in this Configuration Guide.
- For basic information about opening the Subscriber Editor and an explanation of key modes, see the "Panel and Subscriber Configuration" chapter in this Configuration Guide.



Viewing a list of FXO Ports

When suitable hardware has been added and FXO Subscribers added using the Host Editor, you can view a list by expanding the Enterprise Tree. The example below shows an MIU with a single FXO board and 4 defined telephone lines.



NOTE: Only odd numbered ports appear in the editor. The even numbered ports provide a power fail loop-through telephone connection and do not require any configuration.

To open the advanced FXO Subscriber editor, double-click on the required telephone symbol.

FXO Subscriber Editor Tabs

The advanced Hardware Panel editor comprises seven tabs, outlined below.

- General
- SIP Options
- GPIO
- Phone Controls
- Contributors

These are considered in the following sections.



FXO Subscriber - General

The FXO - General tab is used to specify basic details about the FXO subscriber.

Changing FXO Details

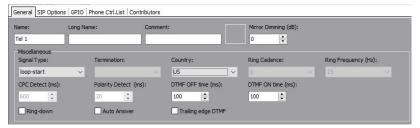
You can change the name of an FXO subscriber and also enter a comment to provide further information about the FXO subscriber. A picture can be added to identify the FXO subscriber on a Virtual Panel.



NOTE: The name of the subscriber is the name allocated when the FXO subscriber is added to the host using the Host Editor.

To change FXO details:

 Select an FXO subscriber from the list of subscribers attached to a host and double-click to open the FXO Subscriber Editor. The General tab is selected by default.



- 2. Make the required changes. If you change the name of the FXO subscriber, the target name will be updated if the FXO subscriber has been added as a source on another subscriber or panel.
- 3. To add a picture to identify the FXO subscriber on a Virtual Panel, click the **Load** button and browse for and select the required image file.
- 4. Some telephony parameters are also presented on this tab and these must be set in accordance with your installation.
- Click the button to post the changes to the database.



FXO Subscriber – SIP Options

The SIP Options tab is used to edit SIP specific details of the FXO subscriber.

To change SIP Options:

 Select an FXO subscriber from the list of subscribers attached to a host and double-click to open the FXO Subscriber Editor. Click on the SIP Options tab.



2. Click the button to post the changes to the database.



FXO Subscriber - GPIO

The GPIO tab is used to define a GPI input for the selected FXO subscriber. GPIO routing statements are explained in the "GPIO and the GPIO Processes Editor" chapter in this Configuration Guide.



The GPI that you add to an E&M subscriber is an input that changes when the phone subscriber that it is connected to goes OFF and ON hook. For example, if a key on a Virtual or Hardware Panel is assigned to the phone target (with a GPI defined) and Signal Presence is selected on the panel setup, then the key on the panel will flash whenever the phone port goes OFF hook. This key will stop flashing when the phone port goes ON hook.

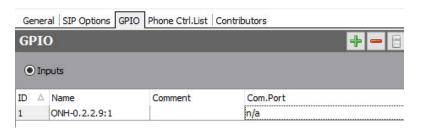
- A maximum of one GPI can be added on an FXO subscriber. Buttons at the top of the GPIO tab let you:
 - Add a GPI.
 - Delete a GPI.

Adding a GPI Input

You can add one GP input. This will be automatically named by the system. A comment can be added to provide more information, if required.

To add a GP input to an FXO subscriber:

- Select an FXO subscriber from the list of subscribers attached to a host and open the Panel Configuration Editor.
- 2. Click the GPIO tab.
- 3. Click the green + button to add the input.



- 4. Change the name and add a comment relating to the GPI, if required.
- 5. Click the button to post any changes to the database.

Deleting GP Inputs

The GPI defined for an FXO subscriber can be deleted, if required. To delete the GPI, click the Delete button.



IMPORTANT: If you delete the GPI it will be removed from any GPIO processes that have been configured to use the selected GPI. The process will not be deleted, but will not function as expected.



FXO Subscriber - Phone Ctrl. List

If you have defined phone controls using the Phone Number Editor, these can be added to the configuration and then used to direct a call to the configured subscriber. SIP numbers, defined using the Phone Number Editor, can also be added to the FXO subscriber, if required.

Buttons at the top of the Phone Ctrl. List tab let you:

- Add a phone control or SIP number.
- Delete a selected phone control or SIP number.

Adding a Phone Control to an FXO Subscriber

Phone controls are incoming DDI digits that are used to direct an answered call directly to a configured subscriber, for example a conference, Virtual Panel or radio phone. See "Phone Numbers and the Phone Number Editor" for more information on adding phone controls.



NOTE: When you add phone controls you select from the list of global phone controls added using the Phone Number Editor and as such you cannot make any changes to the phone control details from here.

Adding a SIP Number to an FXO Subscriber

SIP numbers are numbers that can be used to call a SIP connection. They are outgoing digit sequences that map to either fully formed URIs or are digits that are passed to a Proxy Server where they are fully resolved. See "Phone Numbers and the Phone Number Editor" for more information on adding SIP numbers.

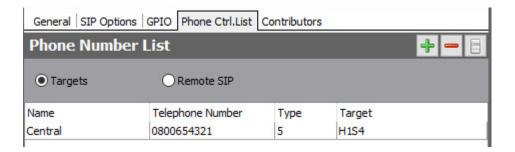


NOTE: When you add SIP numbers you select from the list of global SIP numbers added using the Phone Number Editor and as such you cannot make any changes to the details from here.

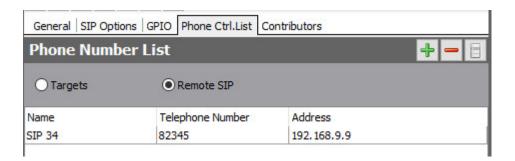
To add a phone control or SIP Number:

- Select an FXO subscriber from the list of panels/subscribers attached to a host and open the FXO Subscriber Editor.
- 2. Click the Phone Ctrl. List tab.
- Press the radio button ("Targets" or "Remote SIP") according to the type of item you wish to add.
- 4. Click the Add button to open the Phone Numbers list.
- 5. Select the number that you want to add and click the **Apply** button.

Examples are shown below.







Removing Phone Controls and SIP Numbers from an FXO Subscriber

Any phone controls or SIP numbers that are no longer required can be removed from the FXO subscriber. You can either remove phone controls or SIP numbers individually (using the Delete button) or remove all phone controls or SIP numbers at the same time (using the Clear All button).

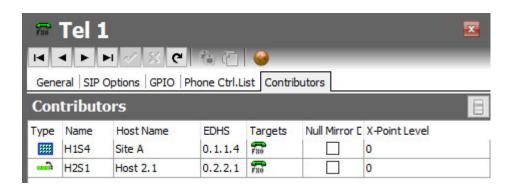


NOTE: Deleting phone controls or SIP numbers from the selected FXO subscriber does not delete these numbers from the global phone control and SIP number lists maintained by the Phone Number Editor or from other subscribers.



FXO Subscriber - Contributors

The Contributors tab shows a list of reverse key target assignments. The list shows panels which have keys targeting the current FXO subscriber. In the example below, FXO Subscriber Tel 1 has been selected.



Thus, panels H1S4 and H2S1 have keys which are "pointing at" TEL 1, allowing those 2 panels to make and receive phone calls.

Most fields are read-only but the following may be adjusted:

- Null mirror dimming. Null mirror dimming is normally automatically applied when 2 subscribers have open bi-directional routes. The audio level in both directions is reduced by a predetermined amount to reduce the risk of acoustic feedback. With certain types of subscriber such as headsets or telephones, due to the physical layout, the risk is minimal and the null mirror dimming can be overridden.
- X-Point level

Click the button to post any changes you make.



FXO Subscriber Editor - Field Definitions

FXO Configuration - General tab

Field/Display Item	Description
Name	The name of the FXO subscriber. This is usually defined when the FXO subscriber is added to the host using the Host Editor. See the "Hosts and the Host Editor" chapter in this Configuration Guide. The name can be changed using the Panel Configuration Editor, if required.
	Maximum length of the name is 20 characters.
Comment	Comments relating to the FXO subscriber. Add your comments in the space provided.
Picture	The picture that has been assigned to the FXO subscriber. To assign a picture, or change an existing one, click the Load button and then browse for and select the required image file.
Mirror Dimming	Mirror dimming helps to reduce acoustic feedback.
	Range 0 to -60 dB. Default = 0 dB.
Signal Type	Specifies the telephony signal type.
Country	Specifies the country of use.
DTMF Off time (mS)	50 – 2000 mS, default 100 mS
DTMF On time (mS)	50 – 2000 mS, default 100 mS
Trailing Edge DTMF	On / Off
Auto-Answer	When selected, forces a FXO port to auto-answer an incoming call and then treat the port as a normal 4-wire. Make sure it is not selected if you wish to use DTMF via phone controls to create routes within the system. Default = Not selected.

FXO Configuration - SIP Options tab

Display Item	Description
Assume SIP on free dial	On / Off

FXO Configuration - GPIO tab

Field/Display Item	Description
ID	The ID of the GPI in the database.
Name	The name of the GPI. By default, this is the subscriber EDHS prefixed with OFH (Off Hook) and suffixed with the GP pin value.
	For example, OFH-0.1.1.3:1
	Names do not have to be unique.
	Maximum length for the name is 20 characters.
Comment	Comments relating to the GPI. You can add comments in the space provided, as required.
	Maximum length for comments is 255 characters.
Com. Port	Not applicable.



FXO Configuration - Phone Ctrl. List tab

Display Item	Description
Targets	Click this radio button to add a phone control.
Remote SIP	Click this radio button to add a SIP number.
Name	The name of the phone control as defined using the Phone Number Editor. This cannot be changed here.
Telephone Number	The phone control number as defined using the Phone Number Editor. This cannot be changed here.
Туре	An icon showing the target for the phone control.
Target	The name of the target. This cannot be changed here.
Address	The IP address or URL corresponding to the SIP number. This cannot be changed here.
Target	The name of the target. This cannot be changed here.

FXO Configuration - Contributors tab

Field/Display Item	Description
Туре	An icon indicating the type of subscriber contributing, e.g. another panel, a radio or telephone.
Name	The name of the contributing subscriber
Host Name	The name of the host, to which the contributing subscriber is connected.
EDHS	Full EDHS address of the contributor.
Targets	Icon.
Null Mirror Dimming	Check the box to override the application of mirror dimming for this subscriber pair.
X-Point level	The gain level of the contributing "source" to this subscriber.



Chapter 10 - Foreign Exchange Station (FXS) - Subscriber Configuration

Overview

Foreign Exchange Station (FXS) Telephony ports require the correct expansion board to be fitted to a Mercury Interface Unit (MIU). The FXS board supports up to 4 telephone ports which must be connected to analogue telephone handsets.

The addition of an FXS board limits the range of supported coding profiles and Gateway will automatically select a suitable coding profile on this host. However, if you are designing a multi-host Mercury system you must take account of this and adjust the coding profile of other hosts. Any incompatibilities will cause warning messages until resolved.

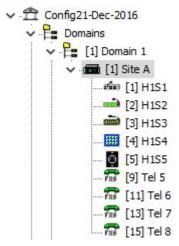
A Foreign Exchange Station (FXS) subscriber is added to a host using the Host Editor. Once you have added an FXS subscriber, you must define the phone controls (if used), GPIs and operational settings using the Panel Configuration Editor.

- For more information on adding FXS ports to a host, see the "Hosts and the Host Editor" chapter in this Configuration Guide.
- For basic information about opening the Panel Configuration Editor and an explanation of key modes, see the "Panel and Subscriber Configuration" chapter in this Configuration Guide.



Viewing a list of FXS Ports

When suitable hardware has been added and FXS Subscribers added using the Host Editor, you can view a list by expanding the Enterprise Tree. The example below shows an MIU with a single FXS board and 4 defined telephone subscribers.



NOTE: Only odd numbered ports appear in the editor.

To open the advanced FXS Subscriber editor, double-click on the required telephone symbol.

FXS Subscriber Editor Tabs

The advanced Hardware Panel editor comprises seven tabs, outlined below.

- General
- SIP Options
- GPIO
- Phone Controls
- Contributors

These are considered in the following sections.



FXS Subscriber - General

The FXS - General tab is used to specify basic details about the FXS subscriber.

Changing FXS Details

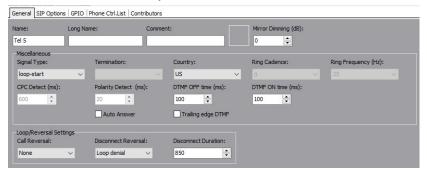
You can change the name of an FXS subscriber and also enter a comment to provide further information about the FXS subscriber. A picture can be added to identify the FXS subscriber on a Virtual Panel.



NOTE: The name of the subscriber is the name allocated when the FXS subscriber is added to the host using the Host Editor.

To change FXS details:

 Select an FXS subscriber from the list of subscribers attached to a host and double-click to open the FXS Subscriber Editor. The General tab is selected by default.



- 2. Make the required changes. If you change the name of the FXS subscriber, the target name will be updated if the FXS subscriber has been added as a source on another subscriber or panel.
- To add a picture to identify the FXS subscriber on a Virtual Panel, click the **Load** button and browse for and select the required image file
- 4. Some telephony parameters are also presented on this tab and these must be set in accordance with your chosen model of handset.
- 5. Click the button to post the changes to the database.



FXS Subscriber – SIP Options

The SIP Options tab is used to edit SIP specific details of the FXS subscriber.

To change SIP Options:

 Select an FXS subscriber from the list of subscribers attached to a host and double-click to open the FXS Subscriber Editor. Click on the SIP Options tab.



2. Click the button to post the changes to the database.



FXS Subscriber – GPIO

The GPIO tab is used to define a GPI input for the selected FXS subscriber. GPIO routing statements are explained in the "GPIO and the GPIO Processes Editor" chapter in this Configuration Guide.



The GPI that you add to an E&M subscriber is an input that changes when the phone subscriber that it is connected to goes OFF and ON hook. For example, if a key on a Virtual or Hardware Panel is assigned to the phone target (with a GPI defined) and Signal Presence is selected on the panel setup, then the key on the panel will flash whenever the phone port goes OFF hook. This key will stop flashing when the phone port goes ON hook.

- A maximum of one GPI can be added on an FXS subscriber. Buttons at the top of the GPIO tab let you:
 - Add a GPI.
 - Delete a GPI.

Adding a GPI Input

You can add one GP input. This will be automatically named by the system. A comment can be added to provide more information, if required.

To add a GP input to an FXS subscriber:

- Select an FXS subscriber from the list of subscribers attached to a host and open the Panel Configuration Editor.
- 2. Click the GPIO tab.
- 3. Click the green + button to add the input.



- 4. Change the name and add a comment relating to the GPI, if required.
- Click the button to post any changes to the database.

Deleting GP Inputs

The GPI defined for an FXS subscriber can be deleted, if required. To delete the GPI, click the Delete button.



IMPORTANT: If you delete the GPI it will be removed from any GPIO processes that have been configured to use the selected GPI. The process will not be deleted, but will not function as expected.



FXS Subscriber - Phone Ctrl. List

If you have defined phone controls using the Phone Number Editor, these can be added to the configuration and then used to direct a call to the configured subscriber. SIP numbers, defined using the Phone Number Editor, can also be added to the FXS subscriber, if required.

Buttons at the top of the Phone Ctrl. List tab let you:

- Add a phone control or SIP number.
- Delete a selected phone control or SIP number.

Adding a Phone Control to an FXS Subscriber

Phone controls are incoming DDI digits that are used to direct an answered call directly to a configured subscriber, for example a conference, Virtual Panel or radio phone. See "Phone Numbers and the Phone Number Editor" for more information on adding phone controls.



NOTE: When you add phone controls you select from the list of global phone controls added using the Phone Number Editor and as such you cannot make any changes to the phone control details from here.

Adding a SIP Number to an FXS Subscriber

SIP numbers are numbers that can be used to call a SIP connection. They are outgoing digit sequences that map to either fully formed URIs or are digits that are passed to a Proxy Server where they are fully resolved. See "Phone Numbers and the Phone Number Editor" for more information on adding SIP numbers.

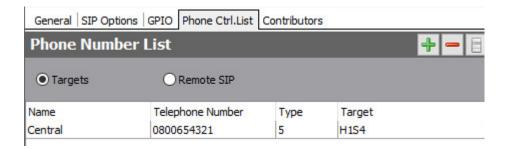


NOTE: When you add SIP numbers you select from the list of global SIP numbers added using the Phone Number Editor and as such you cannot make any changes to the details from here.

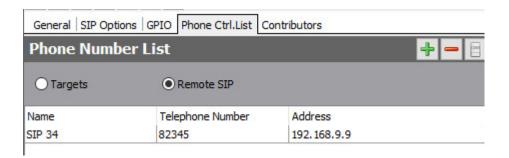
To add a phone control or SIP Number:

- 1. Select an FXS subscriber from the list of panels/subscribers attached to a host and open the FXS Subscriber Editor.
- 2. Click the Phone Ctrl. List tab.
- 3. Press the radio button ("Targets" or "Remote SIP") according to the type of item you wish to add.
- 4. Click the Add button to open the Phone Numbers list.
- 5. Select the number that you want to add and click the Apply button.

Examples are shown below.







Removing Phone Controls and SIP Numbers from an FXS Subscriber

Any phone controls or SIP numbers that are no longer required can be removed from the FXS subscriber. You can either remove phone controls or SIP numbers individually (using the Delete button) or remove all phone controls or SIP numbers at the same time (using the Clear All button).



NOTE: Deleting phone controls or SIP numbers from the selected FXS subscriber does not delete these numbers from the global phone control and SIP number lists maintained by the Phone Number Editor or from other subscribers.



FXS Subscriber - Contributors

The Contributors tab shows a list of reverse key target assignments. The list shows panels which have keys targeting the current FXS subscriber. In the example below, FXS Subscriber Tel 5 has been selected.



Thus, panels H1S1 and H1S2 have keys which are "pointing at" TEL 5, allowing those 2 panels to make and receive phone calls with handset Tel 5.

Most fields are read-only but the following may be adjusted:

- Null mirror dimming. Mirror dimming is normally applied automatically when 2 subscribers are in duplex conversation, to reduce the risk of acoustic feedback. With some types of subscribers such as telephones and headsets the risk is minimal and mirror dimming may safely be disabled.
- X-Point level

Click the button to post any changes you make.



FXS Subscriber Editor - Field Definitions

FXS Configuration - General tab

Field/Display Item	Description
Name	The name of the FXS subscriber. This is usually defined when the FXS subscriber is added to the host using the Host Editor. See the "Hosts and the Host Editor" chapter in this Configuration Guide. The name can be changed using the Panel Configuration Editor, if required.
	Maximum length of the name is 20 characters.
Comment	Comments relating to the FXS subscriber. Add your comments in the space provided.
Picture	The picture that has been assigned to the FXS subscriber. To assign a picture, or change an existing one, click the Load button and then browse for and select the required image file.
Mirror Dimming	Mirror dimming helps to reduce acoustic feedback.
	Range 0 to -60 dB. Default = 0 dB.
Signal Type	Specifies the telephony signal type.
Country	Specifies the country of use.
DTMF Off time (mS)	50 – 2000 mS, default 100 mS
DTMF On time (mS)	50 – 2000 mS, default 100 mS
Trailing Edge DTMF	On / Off
Auto-Answer	When selected, forces a FXS port to auto-answer an incoming call and then treat the port as a normal 4-wire. Make sure it is not selected if you wish to use DTMF via phone controls to create routes within the system. Default = Not selected.
Loop Reversal: Call	Select:
Reversal	 None Local Remote Local & Remote Default = None
Loop Reversal:	Select:
Disconnect Reversal	Loop DenialPolarity ReversalDefault = Loop Denial
Loop Reversal: Disconnect Duration	Range 0 – 65535, default = 850



FXS Configuration - SIP Options tab

Display Item	Description
Assume SIP on free dial	On / Off

FXS Configuration - GPIO tab

Field/Display Item	Description
ID	The ID of the GPI in the database.
Name	The name of the GPI. By default, this is the subscriber EDHS prefixed with OFH (Off Hook) and suffixed with the GP pin value. For example, OFH-0.1.1.3:1 Names do not have to be unique. Maximum length for the name is 20 characters.
Comment	Comments relating to the GPI. You can add comments in the space provided, as required.
	Maximum length for comments is 255 characters.
Com. Port	Not applicable.

FXS Configuration - Phone Ctrl. List tab

Display Item	Description
Targets	Click this radio button to add a phone control.
Remote SIP	Click this radio button to add a SIP number.
Name	The name of the phone control as defined using the Phone Number Editor. This cannot be changed here.
Telephone Number	The phone control number as defined using the Phone Number Editor. This cannot be changed here.
Туре	An icon showing the target for the phone control.
Target	The name of the target. This cannot be changed here.
Address	The IP address or URL corresponding to the SIP number. This cannot be changed here.
Target	The name of the target. This cannot be changed here.



FXS Configuration - Contributors tab

Field/Display Item	Description
Туре	An icon indicating the type of subscriber contributing, e.g. another panel, a radio or telephone.
Name	The name of the contributing subscriber
Host Name	The name of the host, to which the contributing subscriber is connected.
EDHS	Full EDHS address of the contributor.
Targets	Icon.
Null Mirror Dimming	Check the box to override the application of mirror dimming for this subscriber pair.
X-Point level	The gain level of the contributing "source" to this subscriber.



Chapter 11 - 4-wire - Subscriber Configuration

Overview

A 4-wire subscriber is added to a host using the Host Editor. Once you have added the subscriber, you must configure it using the subscriber editor.

Unlike other subscriber types, there are no targets, key modes and key settings required to define for a 4-wire subscriber.

- For more information on adding a 4-wire to a host, see the "Hosts and the Host Editor" chapter in this Configuration Guide.
- For basic information about opening the Subscriber Configuration Editor and an explanation of key modes, see the "Panel and Subscriber Configuration" chapter in this Configuration Guide.



A 4-wire is two pairs of balanced analogue audio input and output signals, generally associated with a single item of external equipment. This is connected using a total of four wires: audio in +, audio in -, audio out +, audio out -.

Since the 4-wire connects to an external device there are very few editable parameters.



4-wire - Subscriber - General

The 4-wire - Subscriber Configuration - General tab is used to specify basic details about the 4- wire subscriber.

Changing 4-Wire Details

You can change the name of the 4-wire and also enter a comment to provide further information about the subscriber. A picture can be added to identify the 4-wire subscriber on a Virtual Panel.



NOTE: The name of the subscriber is the target name which is allocated when the 4-wire is added to the host using the Host Editor.

To change 4-wire details:

- Select a 4-wire port from the list of panels/subscribers attached to a host and open the Panel Configuration Editor. The **Details** tab is selected by default.
- 4. Make the required changes. If you change the name of the 4-wire, the target name will be updated if the 4-wire has been added as a source on another subscriber or panel.
- 5. Click the button to post the changes to the database.





4-wire - Subscriber - Contributors

The Contributors tab shows a list of reverse key target assignments. The list shows panels which have keys targeting the current 4-wire subscriber. In the example below, 4-wire Subscriber H1S5 has been selected.



Thus, panels H1S1 and H2S1 have keys which are "pointing at" the 4-wire allowing those 2 panels to send audio (speak) to the 4-wire. Note that if the panels have keys just to listen to the 4-wire, they are no longer contributing, so will not appear in this list.

Most fields are read-only but the following may be adjusted:

- Null mirror dimming. Null mirror dimming is normally automatically applied when 2 subscribers have open bi-directional routes. The audio level in both directions is reduced by a predetermined amount to reduce the risk of acoustic feedback. With certain types of subscriber such as headsets or telephones, due to the physical layout, the risk is minimal and the null mirror dimming can be overridden.
- X-Point level

Click the button to post any changes you make.



4-wire - Subscriber - Field Definitions

General tab

Field/Display Item	Description
Name	The name of the 4-wire. This is usually defined when the 4-wire is added to the host using the Host Editor. See the "Hosts and the Host Editor" chapter in this Configuration Guide. The name can be changed using the Panel Configuration Editor, if required.
	Maximum length of the name is 20 characters.
Comment	Comments relating to the selected 4-wire subscriber. Add your comments in the space provided.
Picture	The picture that has been assigned to identify the 4- wire. To assign a picture, or change an existing one, click the Load button and then browse for and select the required image file.
Mirror Dimming	Mirror dimming helps to reduce acoustic feedback.
	Range 0 to -60 dB. Default = 0 dB.

Contributors tab

Field/Display Item	Description
Туре	An icon indicating the type of subscriber contributing, e.g. another panel, a radio or telephone.
Name	The name of the contributing subscriber
Host Name	The name of the host, to which the contributing subscriber is connected.
EDHS	Full EDHS address of the contributor.
Targets	Icon.
Null Mirror Dimming	Check the box to override the application of mirror dimming for this subscriber pair.
X-Point level	The gain level of the contributing "source" to this subscriber.



Chapter 12 - Radio Port and Radio Phone- Subscriber Configuration

Overview

A radio port or radio phone is added to a host using the Host Editor. Once you have added the port or phone, you must configure it using the Subscriber - Radio Configuration Editor.

- For more information on adding radio ports and radio phones to a host, see the "Hosts and the Host Editor" chapter in this Configuration Guide.
- For basic information about opening the Panel Configuration Editor and an explanation of key modes, see the "Panel and Subscriber Configuration" chapter in this Configuration Guide.

The radio phone port type is designed for users equipped with DTMF enabled handsets. With extra programming, these handsets can be used to make direct contact with other Mercury subscribers by keying a pre-arranged DDI style short code. Users of these handsets can contact other radio channels and phone subscribers (FXS directly) (FXO by reaching the port, then dialling more digits), call a Virtual or Hardware Panel directly or join a conference. The radio base (connected to the MIU) typically does not require any modification—it just passes the DTMF straight through to the RIB port.

When a Radio Port is added, the following tabs appear in the editor:

- General
- RIB
- VOX
- Contributors

When a Radio Phone is added, the following tabs appear in the editor:

- General
- RIB
- SIP Options
- Sources
- Phone No. List
- Contributors



Radio Port - General

The General tab is used to specify basic details about the radio port.

Changing Radio Port Details

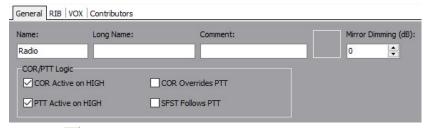
You can change the name of the radio port and also enter a comment to provide further information about the port. A picture can be added to identify the radio port on a Virtual Panel.



NOTE: The name of the radio port is the target name which is allocated when the radio port is added to the host using the Host Editor.

To change radio port details:

- Select a radio port from the list of subscribers attached to a host and open the Radio Configuration Editor. The **General** tab is selected by default.
- 2. Make the required changes. If you change the name of the radio port, the target name will be updated if the radio port has been added as a source on another subscriber or panel.
- 3. To add a picture to identify the radio port on a Virtual Panel, click the **Load** button and browse for and select the required image file.



4. Click the button to post the changes to the database.

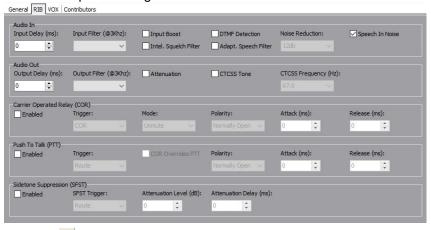


Radio Port - RIB

The RIB tab is used to specify more advanced details about the radio port.

To change radio port details:

- 1. Select a radio port from the list of panels/subscribers attached to a host and open the Radio Configuration Editor. Select the RIB tab.
- 2. Make the required changes.



3. Click the button to post the changes to the database.

Radio Port - VOX

The VOX tab is used to specify the VOX settings for the radio port.

To change radio port settings:

- 1. Select a radio port from the list of panels/subscribers attached to a host and open the Radio Configuration Editor.
- 2. Click the VOX tab.
- 3. Make the required changes to the settings.



4. Click the button to post any changes to the settings.



Radio Port - Contributors

The Contributors tab shows a list of reverse key target assignments. The list shows panels which have keys targeting the current 4-wire subscriber. In the example below, 4-wire Subscriber H1S5 has been selected.



Thus, panels H1S1 and H2S1 have keys which are "pointing at" the 4-wire allowing those 2 panels to send audio (speak) to the 4-wire. Note that if the panels have keys just to listen to the 4-wire, they are no longer contributing, so will not appear in this list.

Most fields are read-only but the following may be adjusted:

- Null mirror dimming. Null mirror dimming is normally automatically applied when 2 subscribers have open bi-directional routes. The audio level in both directions is reduced by a predetermined amount to reduce the risk of acoustic feedback. With certain types of subscriber such as headsets or telephones, due to the physical layout, the risk is minimal and the null mirror dimming can be overridden.
- X-Point level

Click the button to post any changes you make.



Radio Phone General

The General tab is used to specify basic details about the radio phone.

Changing Radio Phone Details

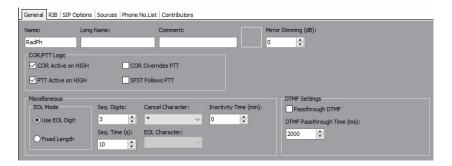
You can change the name of the radio phone and also enter a comment to provide further information about the phone. A picture can be added to identify the radio phone on a Virtual Panel.



NOTE: The name of the radio phone is the target name which is allocated when the radio phone is added to the host using the Host Editor.

To change radio phone details:

- Select a radio phone from the list of panels/subscribers attached to a host and open the Radio Configuration Editor. The **General** tab is selected by default.
- 2. Make the required changes. If you change the name of the radio phone, the target name will be updated if the radio phone has been added as a source on another subscriber or panel.
- 3. To add a picture to identify the radio phone on a Virtual Panel, click the **Load** button and browse for and select the required image file.



4. Click the button to post the changes to the database.

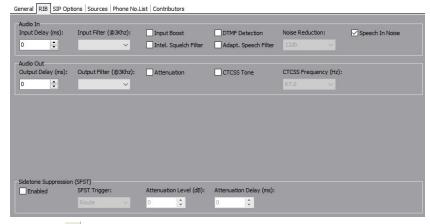


Radio Phone - RIB

The RIB tab is used to specify more advanced details about the radio phone.

To change radio phone details:

- 1. Select a radio phone from the list of subscribers attached to a host and open the Radio Configuration Editor. Select the RIB tab.
- 2. Make the required changes.



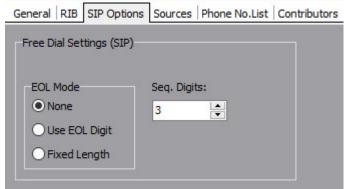
3. Click the button to post the changes to the database.

Radio Phone - SIP Options

The SIP Options tab is used to specify SIP related options.

To change radio phone details:

- Select a radio port from the list of panels/subscribers attached to a host and open the Panel Configuration Editor. The **Details** tab is selected by default.
- 2. Make the required changes.



3. Click the button to post the changes to the database.



Radio Phone - Sources

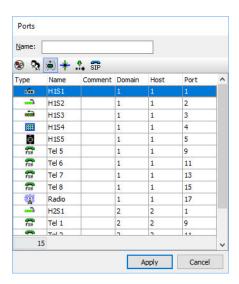
The Sources tab is used to assign sources to a radio phone. Each source is automatically allocated a DDI. Phone controls can also be added to a radio phone. For more information on phone controls, see the "Phone Numbers and the Phone Number Editor" chapter in this Configuration Guide.

Adding a Source

Up to 256 sources to be defined for each radio phone. Each source type that you add will have a default speak/listen mode and DDI. If you add phone controls, the configured phone control number will be used.

To add a source:

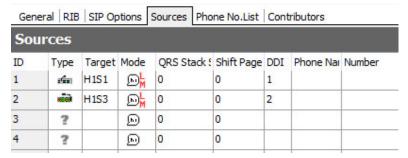
- 1. Select a radio phone from the list of panels/subscribers attached to a host and open the Radio Configuration Editor.
- 2. Click the **Sources** tab.
- 3. Click in the **Target** field and click the button to open the Panels/Sources dialog box.
- 4. To change the target type, click the Groups, Conferences, GPIO/Audio Routes, IFBs or SIP Connections button and make your selection.



- Select the required panel/source (to select more than one panel/source, hold down the **Ctrl** key) and click the **Apply** button.
- 6. To change the mode, click in the **Mode** field, click and select the required mode. The default mode is usually correct.
- 7. Change the DDI, if required. The DDI automatically increments when sources are added.
- 8. To add a phone control, click in the **Ph. Book Name** field and click to open the Panels/Sources dialog box.
- 9. Select the required phone control from the list and click the **Apply** button. Note that the Ph. Book Number field is automatically populated with the phone control number. The DDI field is filled in automatically with the phone control number if you are using Fixed Length as the EOL Mode.
- 10. Add additional sources, as required.



11. Click the button to post the changes to the database.



Changing a Source

Source types on radio phone can be changed, as required. To do this, change the target for the relevant port.

To change a source:

- 1. Select a radio phone from the list of subscribers attached to a host and open the Radio Configuration Editor.
- 2. Click the **Sources** tab.
- 3. Click in the **Target** field next to the ID of the key that you want to reassign.
- 4. Click the button and select the required target.
- 5. Change the other settings, as needed.
- 6. Click the button to post the changes to the database.

Deleting a Source

Any sources that are no longer required can be deleted.

To delete a source:

- 1. Select a radio phone from the list of panels/subscribers attached to a host and open the Panel Configuration Editor.
- 2. Click the Sources tab.
- 3. Click in the **Target** field and press the **Delete or Backspace** key on your keyboard.



Radio Phone: Phone Number List

If you have defined SIP numbers using the Phone Number Editor, these can be added to the configuration and then dialled on the radio phone.

Buttons at the top of the Phone No. List tab let you:

- Add a SIP number.
- Delete a selected SIP number.
- Delete all SIP numbers added to the radio phone.

Adding a Number to a Radio Phone

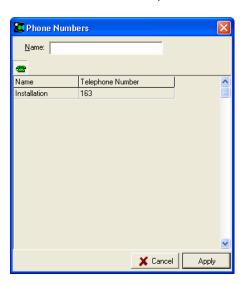
SIP numbers are outgoing digit sequences that map to either fully formed SIP URIs or are digits that are passed to a Proxy Server to be fully resolved.



NOTE: You cannot make any changes to the SIP number details from the Panel Configuration Editor - Phone No. List tab.

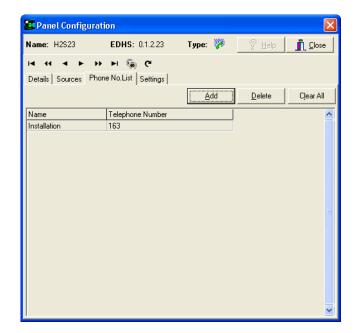
To add a number to a radio phone:

- 1. Select a radio phone from the list of panels/subscribers attached to a host and open the Panel Configuration Editor.
- 2. Click the **Phone No. List** tab. Any numbers that have already been added are shown.
- 3. Click the Add button to open the Phone Numbers list.



4. Select the number that you want to add (to select more than one number, hold down the **Ctrl** key) and click the **Apply** button to add the number(s) to the radio phone.







MORE INFORMATION: Refer to the "Panel Configuration (Radio Phone) - Phone No. List tab" on page 16-12 for a description of the fields on the Panel Configuration - Phone No List tab.

Removing Numbers from a Radio Phone

Any SIP numbers that are not required on the radio phone can be deleted, if required. You can either delete selected numbers (using the Delete button) or delete all numbers at the same time (using the Clear All button).



Radio Phone - Contributors

The Contributors tab shows a list of reverse key target assignments. The list shows panels which have keys targeting the current 4-wire subscriber. In the example below, 4-wire Subscriber H1S5 has been selected.



Thus, panels H1S1 and H2S1 have keys which are "pointing at" the 4-wire allowing those 2 panels to send audio (speak) to the 4-wire. Note that if the panels have keys just to listen to the 4-wire, they are no longer contributing, so will not appear in this list.

Most fields are read-only but the following may be adjusted:

- Null mirror dimming. Null mirror dimming is normally automatically applied when 2 subscribers have open bi-directional routes. The audio level in both directions is reduced by a predetermined amount to reduce the risk of acoustic feedback. With certain types of subscriber such as headsets or telephones, due to the physical layout, the risk is minimal and the null mirror dimming can be overridden.
- X-Point level

Click the button to post any changes you make.



Subscriber - Radio Port - Field Definitions

Configuration (Radio Port) - General tab

Field/Display Item	Description
Name	The name of the radio port. This is usually defined when the radio port is added to the host using the Host Editor. See the "Hosts and the Host Editor" chapter in this Configuration Guide. The name can be changed using the Panel Configuration Editor, if required. Maximum length of the name is 20 characters.
Comment	Comments relating to the selected radio port. Add
	your comments in the space provided.
Picture	The picture that has been assigned to the radio port. To assign a picture, or change an existing one, click the Load button and then browse for and select the required image file.
Mirror Dimming	Mirror dimming helps to reduce acoustic feedback.
	Range 0 to -60 dB. Default = 0 dB.
COR active on HIGH	Select this check box to allow the radio port to override COR activity.
	Default = Selected.
PTT active on HIGH	Allows the active state of the PTT to be configured as either logical 1 (high level signal) or logical 0 (low level signal). Default = Selected.
COR overrides PTT	Select this check box to keep transmitting and enter more digits when dialling an FXO port.
	If this check box is not selected, you will not be able to enter more digits when dialling an FXO port after a dial tone is received.
	Default = Not selected.
SFST follows PTT	SFST = Safety of Flight Side Tone.
	With this setting turned on any changes to output applied by SFST will be driven by the PTT Output state. It has the same effect as turning on SFST and select PTT on the RIB tab. Used when PTT is VOX rather than Route driven.



Configuration (Radio Port) - RIB tab

Field/Display Item	Description
	-
Audio In: Audio Delay	Audio delay in milliseconds applied to incoming audio on the selected port. Not frequently used but can be applied to ensure any routes being driven by a COR input are established before audio from radio is delivered.
Audio In: Input Filter	Applies the described filter to the audio incoming to the port.
Audio In: Input Boost	Can be switched on for a gain boost to the incoming audio.
Audio In: Intelligent Squelch Filter	Attempts to remove the static blips that can occur from radio units.
Audio In: DTMF Detection	Turn this on if you want the radio to be able to drive an FXO port, should the appropriate route be made for it.
Audio In: Adaptive Speech Filter	Applies processing to the incoming audio to reduce noise – db reduction can be configured by adjusting the Noise Reduction field.
Audio In: Noise Reduction	
Audio In: Speech in Noise	
Audio Out: Output Delay	Delay applied to outgoing audio. Much more commonly used so that various signals can be transmitted and the radio base unit in transmit before audio arrives so that the destination doesn't miss vital audio.
Audio Out: Output Filter	Applies the described filter to the audio outgoing from the port.
Audio Out: Attenuation	Negative gain applied to outgoing audio.
Audio Out: CTCSS Tone	
Audio Out: CTCSS Frequency (Hz)	
COR: Enabled	When this setting is enabled Mercury will use the configuration of the following settings to enable or disable audio from the radio port to other subscribers. For example, a panel permanently listening to the radio port will see a flashing tally until the COR signal is received, then the tally will become solid and audio will be heard – this reflects processing going on the background where the actual route is being mute or inhibited.
COR: Trigger	Can select between COR and VOX to control when Mercury will apply the configured action. COR – electrical input coming into the RIB port



Field/Display Item	Description
COR: Mode	Controls whether all listen routes to the radio port are muted or inhibited when the COR trigger is off.
COR: Polarity	Inverts the sense of COR trigger. Not recommended for use with VOX but would be used if radio base unit provides the appropriate electrical signal.
COR: Attack (mS)	Configures how long the trigger condition has to be true before the action will be implemented.
COR: Release (mS)	Configures how long the action continues to be applied once the trigger condition is false.
PTT: Enabled	When this setting is enable Mercury will turn on the associated output signal for the selected radio port when a configured trigger is true. The following fields configure how the action is triggered and performed.
PTT: Trigger	Two options used to trigger the output to be switched on. Route – Whenever a speak route is made to the port, the output signal is switched on. VOX – Whenever audio is outgoing from the point, the output signal is switched on.
PTT: COR Overrides PTT	If this is enable then if COR is true then PTT cannot be triggered, this can be useful with certain radio base stations.
PTT: Polarity	Inverts the sense of the PTT output.
PTT: Attack (mS)	Configures how long the trigger condition has to be true before the action will be implemented.
PTT: Release (mS)	Configures how long the action continues to be applied once the trigger condition is false.
Sidetone Suppression: Enabled	When enabled the audio from the radio will be dimmed or delayed when the trigger condition is true.
Sidetone Suppression: SFST Trigger	Two options to control when SFST is applied. Route – When any speak route is made to the radio port, SFST is applied. PTT – When PTT is turned on, SFST is applied. Usually used when PTT has been set to be triggered by VOX.
Sidetone Suppression: Attenuation Level (dB)	The amount of negative db applied when SFST is turned on.
Sidetone Suppression: Attenuation Delay (mS)	The amount of delay used when SFST is turned on.



Configuration (Radio Port) - VOX tab

Field/Display Item	Description
Input VOX: Attack (mS)	Amount of time the supplied Audio is above the Level before the VOX trigger is turned on. Recommend approx. 500ms, so that little blips of background noise don't register but not so long that the trigger misses the start of any audio.
Input VOX: Release (mS)	How long the trigger stays on after audio has dropped below the specified Level. Used to avoid the trigger turning off when there are short pauses or breaths when speaking.
Input VOX: Level (dB)	The db Level that audio has to be above in order for the VOX trigger to turn on.
Output VOX: Attack (mS)	As above, but for the RIB port audio output
Output VOX: Release (mS)	As above, but for the RIB port audio output
Output VOX: Level (dB)	As above, but for the RIB port audio output

Configuration (Radio Port) - Contributors tab

Field/Display Item	Description
Туре	An icon indicating the type of subscriber contributing, e.g. another panel, a radio or telephone.
Name	The name of the contributing subscriber
Host Name	The name of the host, to which the contributing subscriber is connected.
EDHS	Full EDHS address of the contributor.
Targets	Icon.
Null Mirror Dimming	Check the box to override the application of mirror dimming for this subscriber pair.
X-Point level	The gain level of the contributing "source" to this subscriber.



Configuration (Radio Phone) - Field Definitions

Configuration (Radio Phone) - General tab

Field/Display Item	Description
Name	The name of the radio port. This is usually defined when the radio port is added to the host using the Host Editor. See the "Hosts and the Host Editor" chapter in this Configuration Guide. The name can be changed using the Panel Configuration Editor, if required. Maximum length of the name is 20 characters.
Comment	Comments relating to the selected radio port. Add your comments in the space provided.
Picture	The picture that has been assigned to the radio port. To assign a picture, or change an existing one, click the Load button and then browse for and select the required image file.
Mirror Dimming	Mirror dimming helps to reduce acoustic feedback.
	Range 0 to -60 dB. Default = 0 dB.
COR active on HIGH	Select this check box to allow the radio port to override COR activity.
PTT active on HIGH	Default = Selected.
PTT active on HIGH	Allows the active state of the PTT to be configured as either logical 1 (high level signal) or logical 0 (low level signal).
	Default = Selected.
COR overrides PTT	Select this check box to keep transmitting and enter more digits when dialling an FXO port. If this check box is not selected, you will not be able to enter more digits when dialling an FXO port after a dial tone is received. Default = Not selected.
SFST follows PTT	SFST = Safety of Flight Side Tone.
	With this setting turned on any changes to output applied by SFST will be driven by the PTT Output state. It has the same effect as turning on SFST and select PTT on the RIB tab. Used when PTT is VOX rather than Route driven.
EOL Mode	The mode used to identify the end of a line. Two modes are available:
	Use EOL Digit The end of line is identified with a preselected character (EOL Character).
	Fixed Length The end of line is identified after a preset number of digits has been entered (Seq. Digits). Default = Use EOL Digit.
Seq. Digits	The number of digits used to identify an end of line. Used if Fixed Length is selected as the EOL Mode.
Seq Time (s)	The length of time the RIB will wait for a digit sequence to be entered before timing out. Default = 10 seconds.



Description
The end of line character. This character will be used if Use EOL Digit is selected as the EOL Mode.
The character used to cancel an input.
Default = *.
If a call is not cancelled, the radio will still be active and cannot be called.
A timeout can be specified which will automatically cancel a call if no PTT or COR is detected within the specified time period.
Default = 0 min.
Select this check box to pass DTMF digits to an outside line when connecting to an FXO port. Default = Not selected.



Configuration (Radio Phone) - RIB tab

Field/Display Item	Description
Audio In: Audio Delay	Audio delay in milliseconds applied to incoming audio on the selected port. Not frequently used but can be applied to ensure any routes being driven by a COR input are established before audio from radio is delivered.
Audio In: Input Filter	Applies the described filter to the audio incoming to the port.
Audio In: Input Boost	Can be switched on for a gain boost to the incoming audio.
Audio In: Intelligent Squelch Filter	Attempts to remove the static blips that can occur from radio units.
Audio In: DTMF Detection	Turn this on if you want the radio to be able to drive an FXO port, should the appropriate route be made for it.
Audio In: Adaptive Speech Filter	Applies processing to the incoming audio to reduce noise – db reduction can be configured by adjusting the Noise Reduction field.
Audio In: Noise Reduction	
Audio In: Speech in Noise	
Audio Out: Output Delay	Delay applied to outgoing audio. Much more commonly used so that various signals can be transmitted and the radio base unit in transmit before audio arrives so that the destination doesn't miss vital audio.
Audio Out: Output Filter	Applies the described filter to the audio outgoing from the port.
Audio Out: Attenuation	Negative gain applied to outgoing audio.
Audio Out: CTCSS Tone	
Audio Out: CTCSS Frequency (Hz)	
COR: Enabled	When this setting is enabled Mercury will use the configuration of the following settings to enable or disable audio from the radio port to other subscribers. For example, a panel permanently listening to the radio port will see a flashing tally until the COR signal is received, then the tally will become solid and audio will be heard – this reflects processing going on the background where the actual route is being mute or inhibited.
COR: Trigger	Can select between COR and VOX to control when Mercury will apply the configured action. COR – electrical input coming into the RIB port



Field/Display Item	Description
COR: Mode	Controls whether all listen routes to the radio port are muted or inhibited when the COR trigger is off.
COR: Polarity	Inverts the sense of COR trigger. Not recommended for use with VOX but would be used if radio base unit provides the appropriate electrical signal.
COR: Attack (mS)	Configures how long the trigger condition has to be true before the action will be implemented.
COR: Release (mS)	Configures how long the action continues to be applied once the trigger condition is false.
PTT: Enabled	When this setting is enable Mercury will turn on the associated output signal for the selected radio port when a configured trigger is true. The following fields configure how the action is triggered and performed.
PTT: Trigger	Two options used to trigger the output to be switched on. Route – Whenever a speak route is made to the port, the output signal is switched on. VOX – Whenever audio is outgoing from the point, the output signal is switched on.
PTT: COR Overrides PTT	If this is enable then if COR is true then PTT cannot be triggered, this can be useful with certain radio base stations.
PTT: Polarity	Inverts the sense of the PTT output.
PTT: Attack (mS)	Configures how long the trigger condition has to be true before the action will be implemented.
PTT: Release (mS)	Configures how long the action continues to be applied once the trigger condition is false.
Sidetone Suppression: Enabled	When enabled the audio from the radio will be dimmed or delayed when the trigger condition is true.
Sidetone Suppression: SFST Trigger	Two options to control when SFST is applied. Route – When any speak route is made to the radio port, SFST is applied. PTT – When PTT is turned on, SFST is applied. Usually used when PTT has been set to be triggered by VOX.
Sidetone Suppression: Attenuation Level (dB)	The amount of negative db applied when SFST is turned on.
Sidetone Suppression: Attenuation Delay (mS)	The amount of delay used when SFST is turned on.



Configuration (Radio Phone) – SIP Options tab

Field/Display Item	Description
Free Dial Settings (SIP)	The mode used to identify the end of a line for free dial SIP settings. When enabled, the digits dialled will be sent to the configured Proxy Server rather than matched against the phone number list. For example, dialling digits 1234 will cause a call to 1234@proxy_server_ip_address to be attempted, rather than a lookup of 1234 in the phone number list. This can be used if the SIP Server allows Mercury to use its dial plan obviating the need to replicate the dial plan in Mercury. Default = None. Three modes are available: None Free dial settings are not enabled. Use EOL Digit The end of line is identified with a preselected character (EOL Character).
	Fixed Length The end of line is identified after a preset number of digits has been entered (Seq. Digits). Seq Digits The number of digits used to identify an end of
	line. Used if Fixed Length is selected as the EOL Mode. Default = 3.

Configuration (Radio Phone) - Sources tab

Field/Display Item	Description
ID	The ID of the source in the database.
Туре	Shows an icon to identify the source type, for example a telephone icon, hardware panel icon or conference icon,
Target	The target that is associated with the DDI.
Mode	The key's operation type. This varies depending on the panel type being configured.
	The mode for FXS, FXO, telephone and radio phone defaults to Phone Action Key . This means that the radio phone will target a phone device with phone-action attributes, for example it will ring a target.
DDI	The DDI code which will be used to direct a call to the configured subscriber. This is automatically populated by the system. If you are adding a phone control, the DDI field is populated with the phone control number if you are using Fixed Length as the EOL Mode. Note that if this number is longer or shorter than the EOL setting (Seq. Digits) for the radio, then the DDI will be lengthened (with '0s' or truncated.
Ph. Book Name	Click the <u>under the list.</u> button and select a phone control from the list.
Ph. Book Number	Shows the phone control number if a phone control has been selected.



Configuration (Radio Phone) - Phone No. List tab

Display Item	Description
Name	The name of the SIP number as defined using the Phone Number Editor. Cannot be changed here.
Telephone Number	The SIP number as defined using the Phone Number Editor. Cannot be changed here.

Configuration (Radio Phone) - Contributors tab

Field/Display Item	Description
Туре	An icon indicating the type of subscriber contributing, e.g. another panel, a radio or telephone.
Name	The name of the contributing subscriber
Host Name	The name of the host, to which the contributing subscriber is connected.
EDHS	Full EDHS address of the contributor.
Targets	Icon.
Null Mirror Dimming	Check the box to override the application of mirror dimming for this subscriber pair.
X-Point level	The gain level of the contributing "source" to this subscriber.



Chapter 13 - Virtual Panel – Subscriber Configuration

Overview

Virtual Panels are PC applications that provide a Graphical User Interface (GUI) for Mercury users. They allow users to communicate with each other, or provide a console for operators to manage talk groups with external devices.

A Virtual Panel is added to a host using the Host Editor. Once you have added the panel, you must define the sources, key modes and key settings using the Panel Configuration Editor.

- For more information on adding panels to a host, see the "Hosts and the Host Editor" chapter in this Configuration Guide.
- For basic information about opening the Panel Configuration Editor and an explanation of key modes, see the "Panel and Subscriber Configuration" chapter in this Configuration Guide
- For information about the local configuration and settings available on the Virtual Panel, see the Mercury V3 Virtual Panel Configuration Guide", document 700-90-610.



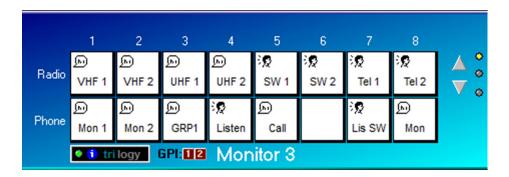
Virtual Panel Editor Tabs

The Subscriber - Virtual Panel Editor has six tabs:

- General
- Sources
- Special
- GPIO
- Phone No. List
- Contributors

With the following exceptions, these are the same as for the Subscriber – Hardware Panel Editor explained in an earlier chapter.

- There is no Source Layout tab. Key targets may only be assigned on the spreadsheet style "Sources" tab. This is because the shape and size of the Virtual Panel is configured at the panel itself, not by the editor. Hence Gateway does not provide a graphical view of the panel.
- The Special tab has additional fields to allow the configuration of keysets. See the following page for an explanation.
- The Virtual Panel does not provide any GPI Outputs.



See the Hardware Panel Configuration chapter of this guide for more information.



Keysets

Using keysets

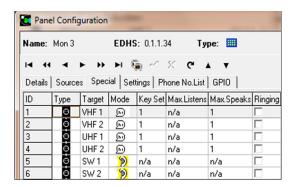
Keysets allow the definition of a linked group of keys on a virtual panel. Normally, keys are not part of any keyset and operate individually, with no relation to other keys on the panel. If a number of keys are added to the same keyset, they form an interlocked set. A common application is to allow only a single speak button to be active on a panel at a time, such that the selection of a second speak button deselects the first.

Adding keys to a keyset

As explained above, by default, keys are not part of any keyset. The keyset value is part of the Special tab and will normally be set to none or n/a. To add a key to a keyset, highlight the key and select a value in the keyset column. To produce an interlocking keyset, a number of keys are set to the same value.

An additional setting ensures that only one speak route is available at any time. The value in the Max, Speak column must be set to 1.

In the example below, the first four keys are all part of keyset 1.





NOTE: To complete the application, one further setting is required. This is on the Virtual Panel itself, under the Preferences menu. On the general tab, a checkbox entitled "Automatically Break Routes in keysets" must be checked. This ensures that when a second button is pressed, the first one is automatically deselected.



MORE INFORMATION: Refer to the "Panel Configuration Editor - Special tab" on page 17-13 for a description of the fields on the Panel Configuration - Special tab. Also for information on the Virtual Panel, see Mercury V3 Virtual Panel Guide, document # 700-90-610.



Panel Configuration (Virtual Panel) - Field Definitions

Panel Configuration - General tab

Field/Display Item	Description
Name	The name of the Hardware Panel. This is usually defined when the panel is added to the host using the Host Editor. See the "Hosts and the Host Editor" chapter in this Configuration Guide. The name can be changed using the Panel Configuration Editor, if required.
	Maximum length of the name is 20 characters.
Comment	Comments relating to the selected panel. Add your comments in the space provided.
Picture	The picture that has been assigned to the Hardware Panel. To assign a picture, or change an existing one, click the Load button and then browse for and select the required image file.
Mirror Dimming	Mirror dimming helps to reduce acoustic feedback. Range 0 to -60 dB. Default = -8 dB (rack-mounted panels) and -12 dB (desktop panel).
Mic. Gain	The amount of gain that can be applied to the microphone or input associated with this panel. Default (recommended) = 0 dB. Range 12 to -9 dB in 1 dB intervals.
Key latch time	Used to determine the period of time at the panel that differentiates between a momentary and latching action. Default = 500 ms. Range = 200–5000 ms.
Change Conference Labels on 4-Wire	Keys assigned to conferences can be set so that they change their label when a subscriber joins a conference. If the subscriber is a 4-wire it is often a permanent "standing feed" into the conference, so the default conference label will always be overridden to show the 4-wire name. This setting disables such behaviour so the label will not change for 4-wire sources. Default = Not selected.
Loudspeaker Hold-off	Presets the volume at which a panel speaker will operate at its the lowest possible value. Default = Not selected.
Loudspeaker Cut	When selected, cuts the loudspeaker when the Cut key/button is pressed. Default = Not selected.
Microphone Cut	Defines the action of the Cut key/button on Hardware Panels. No panel microphone Cut - Disables microphone cut on a panel. Panel microphone Cut (the default) - Pressing the button cuts the microphone completely. Microphone/Headset Changeover - Pressing the button transfers Audio/Mic to the headset jack of the panel.



Field/Display Item	Description
Shift Pages	Specifies the number of pages available on the
	panel. Default = 2. Maximum = 32.
PTT Host GPI	Defines which Host GPI will be used to provide
i i i ilost oi i	PTT functionality for the Hardware Panel.
Assign-mode Timeout(s)	Specifies when a panel enters assign mode. (This is used to configure panel levels and key types). If no input is received in the timeout period, the panel leaves assign mode and enters normal operation. This normally does not need be changed.
	Default = 10. Range = 1–60 in 1 second intervals.
Tally Holdup(s)	The timeout of a yellow listen tally if another panel speaks to the user and then hangs up. A listen tally is displayed after hang-up for this timeout value. Default = Disable. Range = 1–60 seconds.
QRS time-out(s)	The timeout value for the last calling subscriber to
	be displayed on the QRS key/button.
	Default = Never. Range 15–300 seconds.
Always update QRS	When selected, the QRS label changes whenever any route to the panel is made regardless of whether there is a button to make the connection or not. Connections can therefore be 'viewed' from any page.
Latency hold up (s)	Applies to 'Fast Keys'. These work by establishing the IP channel needed when the key is first pressed and thereafter simply make/break local audio crosspoints each time the key is pressed/released. If the key is not pressed for the "latency hold up" period the IP route is torn down.
	Default = 0 seconds. Range = 0–3600 seconds.
Fast keys on startup	If enabled, any fast key IP routes needed are made at TBC startup in anticipation of them being needed straight away. If disabled, they are not made until required. Default = Not selected.
1 st GPI Com Port	The COM port used to connect the physical switch
	that will act as GPI 1 for the Virtual Panel Square.
	Click : to select the required GPI.
2 nd GPI Com Port	The COM port used to connect the physical switch that will act as GPI 2 for the Virtual Panel.
	Click : to select the required GPI.
PTT Host GPI	Defines which Host GPI will be used to provide PTT functionality for the Virtual Panel.



Panel Configuration - Sources tab

Field/Display Item	Description
ID	A number which corresponds to a key/button on the Hardware Panel. IDs prefixed by P2 (to P32) are keys/buttons on the shifted pages.
Туре	The key's target subscriber type indicated by an icon.
Target	The key's target subscriber name. This cannot be changed here but can be changed by opening the target in the relevant editor.
L/Label	Listen Label. This is the name of the target that will be displayed on the panel. It cannot be changed, but can be overridden if you specify a U/Label.
U/Label	User Label. A customisable user label. If this is specified, it will be used instead of the L/Label. Type the label that you want to use into this field.
Mode	The key's operation type. To change the operation
	type, click in the field and then click the button to choose the required mode.
QRS Stack Size	For QRS keys only, sets the size of the QRS stack. If the stack size is exceeded, the oldest calls are thrown away.
Shift Page #	On panels with multiple shift pages, allows a key to be defined to "jump" straight to a specific shifted page. This can bypass the normal multiple press panel Shift key process.
L/Type	Displays the icon associated with the listen key's target or IFN listener.
L/Target	Allows a panel to "split" a key so that the listen and speak are attached to two different targets.
	If the key is targetting an IFB and a listener has been defined, this field will show the name of the defined listener.
Alt. Picture	Displays a picture when the key is activated. This is useful for showing the operator a graphical representation of the caller on a Virtual Panel.
Dimming	Dims the panel loudspeaker when the target subscriber speaks to the panel user. This loudspeaker dimming dims ALL incoming audio.
	Default = 0 dB. Range is 0 to -28 dB in 2 dB steps, or -100 dB.
LS Dim	Loudspeaker Dim
Auto-dial	Specifies a dial sequence that is automatically dialled when a phone key/button is pressed on the panel.
	Applicable to telephony and SIP connections only.



Field/Display Item	Description
Assign	When selected, (the default), allows the panel user to customise this particular key, choosing a new destination (Subscriber, Conference, Group or Route) and Type (Speak, Listen etc.). If you clear this check box, users will not be able to change this key. You could also consider using Zones and Areas if more security or restrictions are required. Restricting key presses is useful to prevent panels from talking to certain subscribers. Default = Selected.
Admin. TEB	



Panel Configuration Editor - Special tab

Field/Display Item	Description
ID	A number which corresponds to a key on the
	Virtual Panel.
Туре	The key's target subscriber type indicated by an icon.
Target	The key's target subscriber name. This cannot be changed here but can be changed by opening the target in the relevant editor.
Mode	The key's operation type. To change the operation
	type, click in the field and then click the button to choose the required mode.
Signal Pres.	If selected, allows a Virtual Panel user to see when a particular key has audio via a slow flashing yellow tally on the key. The user can then elect to listen to it or not. Applies to conference, 4-wire, telephony and radio keys only. Default = Not selected.
Assoc. GPO	Click in this field to select the GPO that you want
	to associate with the key.
GPO On	When selected, the GPO will be on when the key is pressed.
GPI	The host COM port (1 or 2) to be used as a PTT trigger.
PTT	Defines the PTTaction for the key. Can be set to Mute or Unmute. Mute prevent the user from being heard by the listen user. It does not tear down the route; it simply mutes the volume when the key is pushed.
Key Set	Groups keys together to provide interaction between the keys in the set.
	Default = n/a. Range 1–256.
Max. Listens	The maximum listens per given key set group. This is key-driven to prevent more than the defined number of listens. For example, if this is set to 1, then only one key on the set may listen at any one time.
	Default = n/a. Range 1–256.
	Can only be set if you have specified a key set.
Max. Speaks	The maximum speaks per given key set group. This is key-driven to prevent more than the defined number of simultaneous speaks. For example, if this is set to 1, then only one key on the set may speak at any one time.
	Default = n/a. Range 1–256.
	Can only be set if you have specified a key set.
Ringing	Enables ringing for an E&M, FXS, FXO, SIP connection or telephone. Only one 'ringing source' can be selected.
Tone	If ringing has been enabled, click and select the required ringing tone.
Hold Dialpad	



Field/Display Item	Description
Disable Keypad	
Auto Answer (SIP)	

Panel Configuration - Phone No. List tab

Field/Display Item	Description
Name	The name of the phone number entry as defined using the Phone Number Editor. Cannot be changed here.
Telephone Number	The phone number entry as defined using the Phone Number Editor. Cannot be changed here.

Panel Configuration - GPIO tab

Field/Display Item	Description
ID	The ID of the GPI in the database.
Name	The name of the GPIO.
	For GPIs this is the subscriber EDHS prefixed with GPI and suffixed with the GP pin value. For example, GPI-0.1.1.3:1
	Names do not have to be unique.
	Maximum length for the name is 20 characters.
Comment	Comments relating to the GPI. You can add comments in the space provided, as required.
	Maximum length for comments is 255 characters.

Panel Configuration - Contributors tab

Field/Display Item	Description
Туре	An icon indicating the type of subscriber contributing, e.g. another panel, a radio or telephone.
Name	The name of the contributing subscriber
Host Name	The name of the host, to which the contributing subscriber is connected.
EDHS	Full EDHS address of the contributor.
Targets	Icon.
Null Mirror Dimming	Check the box to override the application of mirror dimming for this subscriber pair.
X-Point level	The gain level of the contributing "source" to this subscriber.



Chapter 14 - Groups and the Groups Editor

Overview

A group is a user-defined collection of subscribers. Pressing a single key on a panel enables communication with all members (targets) of the group. A group call makes multiple unicast routes and can be used as an alternative to a conference in operating environments where multicast is not feasible.

Groups are similar to conferences, as a single panel key or button can be used to talk to multiple targets and both can be used within GPIO logic statements. However, whereas a user opts to listen to a conference in order to receive audio, groups are treated as direct calls; when a user speaks to a group; all members in the group automatically listen.

Groups use multiple unicast VoIP to remote members in a Mercury intercom system. As many channels as receiving hosts are used by the originating host.

Groups are added to an Enterprise using the Enterprise Editor. (See the "Enterprises and the Enterprise Editor" chapter in this Configuration Guide for further details.) Groups may also be added using the Groups Overview form. Once groups have been added, the Group Editor is used to specify the members of each group.

A typical application is where multiple control panels are deployed within a large operations control room. An outsider, calling into that control room, cannot see which positions are currently manned. The solution is to create a Group containing all panels in the control room and provide the "outsider" with a key to that group. After the initial group call, the reply will come from an appropriately manned position and the conversation can continue on a 1:1 basis.

Another way to consider the group is as a "macro". On a panel, instead of pressing six separate keys to call six people simultaneously, create a group and add the six "members". The panel can then be configured with one key instead of six.



NOTE: A target can be added to a group provided that the target has been specified as a subscriber using the Host Editor. If a group has been specified as a key target for a panel that has been added to a group, details of that source will be shown in the Groups Editor.



Using the Group Editor

Viewing a List of Groups

You can view a list of all groups that have been defined in the Enterprise by selecting the Groups node in the Enterprise tree. The same form allows additional groups to be added, or any not required to be deleted.

To view a list of groups:

- 1. Expand the Enterprise tree in Gateway.
- 2. Single-click on Groups.

The Details panel lists the groups that have been defined.

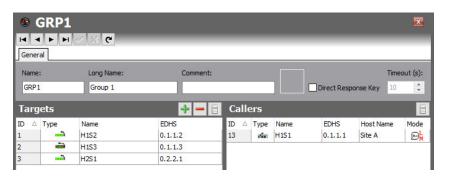


Opening the Groups Editor

The Group Editor is used to specify basic details about a group and to maintain the group. As usual there are Basic and Advanced editors.

To open the Group Editor:

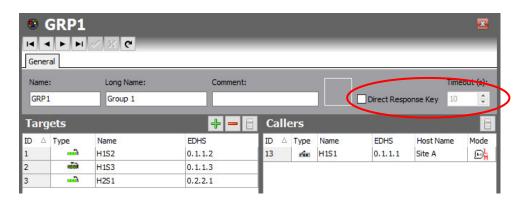
- 1. Expand the Enterprise tree in Gateway.
- 2. Click the "right-arrow" next to **Groups** in the Enterprise tree to show a list of groups.
- 3. Double-click on the required group to open it in the Group Editor.





Group Editor

The Group Editor has a single tab and the layout is almost identical for both the basic and advanced editors. There is just one additional setting on the advanced editor, which is highlighted below.



The left pane of the editor, headed "Targets" comprises the members of the Group. As is evident from the + and – buttons, members must be added to (or removed from) the Group.

The right pane of the editor, headed "Callers" is read-only. This lists those panels who are outside the group but who have a key to speak to the group.

The name, long name and comment may be edited as usual.

Click the button to post the changes to the database.



Group Editor - Field Definitions

Group Editor - General tab

Field/Display Item	Description
Name	The name of the group. This is usually defined using the Enterprise Editor when the group is added to the Enterprise. The name can be changed using the Groups Editor, if required. Names do not have to be unique. Maximum length for the group name is 20 characters.
Comment	Any comments relating to the selected group. To add comments, select the group (click once on the line), click in this field and enter your comments.
Picture	Maximum length for comments is 255 characters. A picture that has been assigned to the group. To assign a picture or change an existing picture, click the Load button and then browse for and select the required image file.
	Only visible on Virtual Panels.
Direct Response Key	On / off to enable a direct response key for this group
Direct Response Key timeout (s)	
Targets: Type	The type of panel or source (identified by an icon).
Targets: Name	The name of the panel or source (as specified in the Panel Configuration Editor). This cannot be changed here. See "Panel and Subscriber Configuration" for more information.
Targets: EDHS	The EDHS address of the panel or source. This provides a quick reference to the location of the group members. This cannot be changed here.
Callers: Type	An icon representing the type of panel. (Read-only).
Callers: Name	The key number on the panel to which the group has been allocated. (Read-only).
Callers: EDHS	The EDHS address of the group member. (Read-only).
Callers: Host Name	The name of the host to which the group member is connected. (Read-only).
Callers: Mode	An icon representing the type of key configured on the caller panel. (Read-only).



Chapter 15 - Conferences and the Conference Editor

Overview

Conferences provide a many-to-many communication facility where each participant hears a mix of everyone currently active in the conference, minus their own audio contribution (mix-minus).

Typically, a conference will involve multiple panel subscribers and 4-wire audio ports (including, for example, radio and/or telephone channels).

A 4-wire source is a balanced pair of analog audio input and output signals, generally associated with a single item of external equipment. This is connected using a total of four wires: audio in +, audio in -, audio out +, audio out -.

Panels can also take part in conferences, provided they have been specified as sources using the Panel Configuration Editor (see the "Panel and Subscriber Configuration" chapter in this Configuration Guide.)

Panel users will be given keys to the conference, allowing them to select whether they speak and/or listen to the conference.

4-wire members will default to being permanent speakers and listeners: a 4-wire port will always be heard by any panel member listening to the conference, and the 4-wire port will always receive audio from any other speaking member of the conference. GPI routing can be used to provide real-time control of the speak/ listen status of 4-wire ports.

Where subscribers are local to a given Mercury host, all audio connections are made and broken by the host's audio mixing engine, but when a subscriber is on a different host, IP audio channels must be used to transport audio between the hosts. In a multi-host, multi-subscriber system, it is possible to rapidly use up all available IP channels.

Multicast

Mercury usually allocates a unicast IP channel (i.e. a point-to-point connection that conveys audio uniquely between two hosts) for inter-host communications. Conferences, however, use multicast IP audio. This allows a single IP channel to place audio onto the network without requiring a specific destination IP address (i.e. host) to be specified. Once notified of the presence of the multicast audio, multiple hosts may then listen to the same audio data: this provides a dramatic reduction in the number of IP channels required.

Multicast requires the use of multiple, special IP addresses. A single, unique, "multicast address" is used for each individual multicasting source in the system, and these are configurable within Gateway.

Multicast places special demands on the network infrastructure: specifically, multicast will not (by default) pass through IP routers and is certainly not possible on the public Internet. Care must therefore be taken when



considering the use of conferences and consultation with the network Administrator is recommended.

Multicast allows a subscriber to use just a single IP channel to send audio to all other hosts involved in the conference. However, multiple IP channels are still required to receive audio from other hosts, and at least one channel will be needed for every active, speaking host in a conference.

As an example, a conference involving five hosts, with subscribers of each host concurrently speaking and listening will require:

- Each host to allocate a single speak channel, to multicast audio to the other four hosts
- Each host to allocate four IP channels to receive audio from the other four hosts.
- Each host therefore requires one transmit IP channel, but four receive IP channels.

Simplistically, multicast can dramatically reduce the number of transmit IP channels required, but does not reduce the number of receive IP channels required. A conference can only have up to 48 members defined, although they do not have to be all active at the same time.

If a large conference is configured, it is possible to rapidly run out of IP channels at a host should too many people speak at once. To provide control of this, each conference has a facility to restrict the maximum number of speaking members. Note that this does not restrict the number of listen channels between subscribers on the same host: such communications is free and unblocked unless a key for that conference is not on a user panel.

Conferences may be given a name, used as the default label on panel keys. A selectable option allows this label to dynamically change on panel displays to show the name of the most recent conference caller. There is a perconference setting to enable/disable this function completely, and a per-panel setting to disable it for 4-wire sources (only applicable to Hardware Panels). In addition, Virtual Panels display the current number of speakers and listeners for the conference in the volume control/picture pop-up.

Conferences are similar to groups, as a single panel key or button can be used to talk to multiple targets and both can be used within GPIO logic statements. However, conferences use multicast to transmit VoIP to remote members in a Mercury Intercom system, thus saving DSP channels.

A host must also be pre-configured (using the Hosts Editor) in order to take part in conference audio, particularly if users are using definable keys to conferences. A conference must have at least one speaker and one listener in order to be deemed as active and passing audio. A group is active as soon as the first listen or speak is made.

Conferences are added and deleted at Enterprise level using the Enterprise Editor. The Conference Editor is then used to specify basic details about a conference and to maintain conference 4-wire sources, telephone ports and panel members.



NOTE: It is possible to create a route using the Routes Editor that potentially creates an audio route to or from a conference from a given subscriber (e.g. a panel, radio etc.) that is not defined as a normal member of the conference. Care must be taken that the host on which the subscriber is based has another subscriber already in that conference, or is defined on the Additional Hosts page for that conference in the Conference Editor. For this route to be established it must either be defined as fixed, or have a GPIO logic process defined in the GPIO Processes Editor.



Using the Conference Editor

Viewing a List of Conferences

All of the conferences that are added using the Enterprise Editor are grouped together under 'Conferences' in the Enterprise tree.

To view a list of conferences:

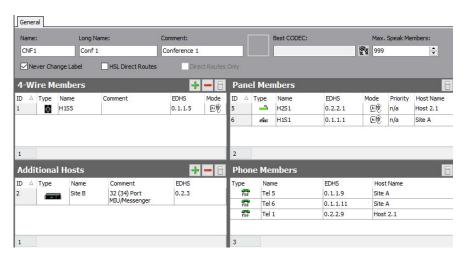
- 1. Expand the Enterprise tree in Gateway.
- 2. Click on Conferences.

The Details panel lists all of the conferences that have been defined and allows additional conferences to be added or existing ones deleted. Basic details of each conference can be edited. The same options are provided at Enterprise level.



To open the Conference Editor:

- 1. Expand **Conferences** in the Enterprise tree as shown above.
- 2. Double-click on the required conference.





Areas of the Conference Editor

The Conference Editor has five areas which are used to configure a conference. The form allows a conference to be viewed in a single form. Differences between the basic and advanced editors are highlighted below and information about each area is in later sections.

- Details. Available in both editor versions with the exception of the following which are only present in the Advanced editor.
 - Max Speaking Members
 - Never Change Label
 - HSL Direct Routes
 - o Only Direct Routes
- 4-Wire members
- Additional Hosts (only in Advanced)
- Panel Members (read-only)
- Phone Members (read-only)



Conference Details

The name and description (comment) of a conference can be changed, as required. The name and description are entered when the conference is created. A picture can be added to identify the conference on a Virtual Panel. The long name appears on hardware panels with displays supporting >5 characters.

To change conference details:

- 1. Open the Conference Editor.
- 2. Make the required changes.
- 3. Click the button to post the changes to the database.



4-Wire Sources

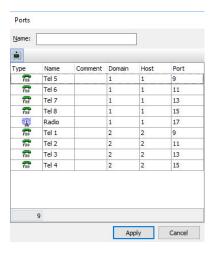
The Conference Editor - 4-Wire Sources area is used to maintain the list of 4-wire sources (4-wire, radio port, telephone, virtual port) for the conference. Sources can be added or deleted, as required. You can define how the 4-wire is used in the conference as a speaking member, a listening member or both.

Buttons at the top of the Conference Editor - 4-Wire Sources tab let you:

- Add a 4-wire source to the conference.
- Delete a selected 4-wire source.

To add a 4-wire source to a conference:

- 1. Open the Conference Editor and click the **4-Wire Sources** tab.
- 2. Click the green + button to open the Panels/Sources dialog box.



- Select the required source from the Panels/Sources list and click the Apply button. To add more than one source at the same time, hold down the Ctrl key when selecting.
- 4. To add or change comments, click in the **Comments** field and type your comments.
- 5. Change the mode, if required. To do this, click in the Mode field and then click it to display a pop-up window listing the available modes. Select the required mode.
- 6. Click the button to post the changes to the database.

Deleting a 4-wire Source

4-wire sources for the conference can be deleted, as required. You can either delete selected sources (using the Delete button) or delete all sources for the conference at the same time (using the right click context menu).



NOTE: Deleting 4-wire sources from a conference does not delete the 4-wire sources from the database.



Additional Hosts

The Conference Editor - Additional Hosts tab lists all of the hosts that may want to join a conference at some time. This enables the TBCs for each host to be able to communicate with each other and participate in the conference.

If panels attached to a host have no keys to conferences, then the TBC on the host makes no provision for conferences and does not organise any host-to-host communication and ignores multicast packets of conferences that the host is not interested in. To allow for local key assignments targeting a conference, additional hosts must be specified to inform the host TBC that conferences may be used at some point in the future.



NOTE: This is a powerful feature and may result in panels speaking/listening to a conference to which they are not invited so should be used with care.

Buttons at the top of the Additional Hosts tab let you:

- Add additional hosts.
- Delete selected additional hosts.

Adding an Additional Host

Any other hosts in any of the domains in your configuration can be added as additional hosts for the selected conference.

To add an additional host:

- Open the Conference Editor.
- Click the green + button to open the Hosts dialog box.



- 3. Select the required host(s) and click the Apply button to add the hosts.
- 4. To add or change comments, click in the Comments field and type your comments.
- 5. Click the button to post the changes to the database.

Deleting an Additional Host

Additional hosts can be deleted, if required. When deleting additional hosts, you can delete selected hosts (using the Delete button) or delete all additional hosts at the same time (using the right-click context menu).

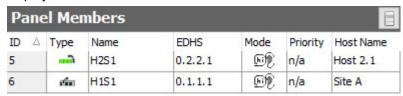


NOTE: Deleting additional hosts from a conference does not delete the host records from the database.



Panel Members

This area of the editor is read-only. If a conference has been specified as a key target on a Hardware or Virtual Panel then details of the panel are displayed here.



The two panels listed (H2S1 and H1S1) are both able to listen and / or speak to the conference, as indicated by the mode icon. No details may be edited here: the Subscriber – Panel editor should be used to remove or edit these keys.



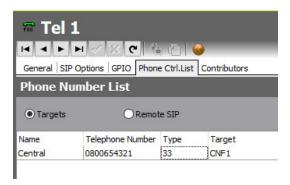
Phone Members

If a phone control has been allocated to a conference *and* the phone control has then been added to the phone control list for an FXO, or FXS port then details of the subscriber are displayed on the Conference Editor - Phone Members tab.

First, here is the original phone control defined in the global Phone Control list:

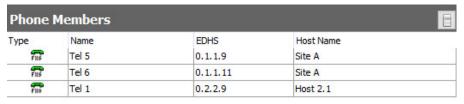


Now, this number has been allocated to a phone port, Tel 1. This is shown in the Subscriber – FXO editor, Phone Control List tab.



Viewing Phone Members

Phone controls targeting conferences that have been added to FXS or FXO ports are shown on the Conference Editor - Phone Members tab. None of the information shown here can be changed.





Conference Editor - Field Definitions

Conference Editor - Details area

Field/Display Item	Description
Name	The name of the conference. This is usually defined using the Enterprise Editor when the conference is added to the Enterprise. The name can be changed using the Conference Editor, if required. The name does not have to be unique.
	Maximum length for the conference name is 20 characters.
Long Name	Alternative to the Name field (above) which will appear on hardware panels with displays >5 characters.
Comment	Comments relating to a selected conference. To add comments, click in this field and enter your comments.
	Maximum length for comments is 255 characters.
Picture	The picture that has been assigned to the conference. To assign a picture or change an existing picture, click the Load button and then browse for and select the required image file.
	Only visible on Virtual Panels.
Never change label	Toggles the display of membership messages on panels (last speaking member) in a Virtual Panel.
HSL Direct Routes	Only available if using Gemini units configured with HSL.
Direct Routes only	Only available if using Gemini units configured with HSL.
Best CODEC	Surveys the configured codecs across all hosts and attempts to picks the best Codec for the conference that is compatible with those Hosts.
Max Speak Members	Sets the maximum number of active speakers permitted at any time. Range 1 to 1632: default 999

Conferences Editor - 4-Wire Sources area

Field/Display Item	Description
Туре	The type of source (identified by an icon).
Name	The name of the source (as specified on the Host Editor).
Comment	Any comments that have been added regarding this source. Click in this field to add or change comments.
	Maximum length for comments is 255 characters.
EDHS	The EDHS address of the source. This cannot be changed.



Field/Display Item	Description
Mode	The mode for the source. This can be changed, if required. To change the mode, select the source and click to the left of the icon to display the button. Click this and select the required mode:
	Speak The 4-wire is always speaking to the conference. As long as there is at least one other active conference member (permanent or otherwise) the audio will always flow from this port. This is the default setting.
	Listen The 4-wire is always listening to the conference.
	Speak/Listen The 4-wire is always speaking and listening to the conference.

Conferences Editor - Additional Hosts area

Field/Display Item	Description
Туре	The type of host (identified by an icon).
Name	The name of the host (as specified on the Host Editor)
Comment	Any comments that have been added regarding this host. Add your own comments here if required.
EDHS	The EDHS address of the host. This cannot be changed.

Conference Editor - Panel Members area

This area is read-only.

Display Item	Description
Name	The name of the source.
Туре	An icon representing the type of source.
Key	The key number on the panel to which the conference has been allocated.
EDHS	The EDHS address of the source.

Conference Editor - Phone Members area

This area is read-only.

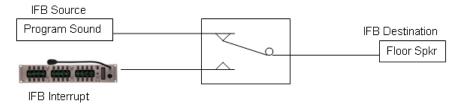
Display Item	Description
Name	The name of the subscriber on the panel.
Туре	The subscriber type (FXO, FXS or E&M).
EDHS	The EDHS address of the subscriber.
Host	The host name.



Chapter 16 - IFBs and the IFB Editor

Overview

Interruptible Foldbacks (IFBs) are audio routes which can be broken into, or interrupted, by a third party replacing the original audio with the third party panel microphone. A typical example of an IFB is shown in Figure 22.1



A typical IFB

An IFB comprises three components:

- IFB Destination usually a 4-wire subscriber.
- IFB Source the "normal" feed sent to the IFB destination.
- **IFB Interrupt** generates the control signal and interrupting audio.

In the example shown in Figure 22.1, the presenter on the studio floor is normally listening to the program sound. When the sound operator presses the speak key on the panel to the presenter, the audio from program sound is switched away and replaced with the operator's sound microphone.

More complex IFBs can be configured using the IFB Editor with the following features available:

- Multiple return audio sources.
- Multiple IFB interrupt sources.
- Cutting or dimming of IFB audio by a pre-selected level.
- Prioritisation of IFB triggers to allow individual triggers to override other triggers within the IFB group.

In common with other parts of the application, there are both basic and advanced IFB Editors. The advanced editor provides a tabular style view whereas the basic editor takes a pictorial approach and will be described in this chapter.



Viewing a List of IFBs

All of the IFBs that are added using the Enterprise Editor are grouped together under 'IFBs' in the Enterprise tree. As seen in the image below, IFBs may also be added or deleted on the IFBs Overview form.

To view a list of IFBs:

- 6. Expand the Enterprise tree in Gateway.
- 7. Click on IFBs.

The Details panel lists all of the IFBs that have been defined.





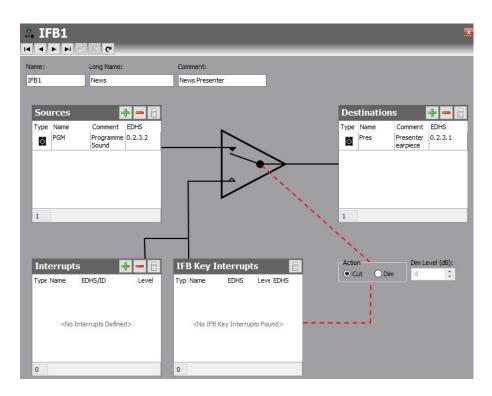
Using the IFB Editor

Opening the IFB Editor

IFBs are maintained using the IFB Editor. In common with other parts of the application, there are both basic and advanced IFB Editors. The advanced editor provides a tabular style view whereas the basic editor takes a pictorial approach and will be described in this chapter.

To open the Basic IFB Editor:

- 1. Expand IFBs in the Enterprise tree.
- 2. Single-click on the required IFB IFB1.



The IFB Editor Form

The IFB Editor form (above) provides a visual interpretation of the IFB action and the form is arranged accordingly. Signal flow is from left (source) to right (destination) with the interrupt occurring in the centre switch. The form has five areas which are used to maintain basic details about the IFB, to add and delete IFB sources, destinations and triggers, to add and delete GPI and GPO triggers, to view panels/sources that have the selected IFB assigned and to allocate a listener to the trigger.



IFB Details

The name and long name of the selected IFB and any comments that have been entered about that IFB may be edited here.



NOTE: IFBs are added at Enterprise level using the Enterprise Editor. See the "Enterprises and the Enterprise Editor" chapter in this Configuration Guide for more information. IFBs may also be added or deleted on the IFBs Overview form.

Changing IFB Details

The name and description (comment) of an IFB can be changed, as required. The name and description are entered when the IFB is created. You can also specify the action (Cut or Dim) of the IFB interrupt.

To change IFB details:

- 1. Open the IFB Editor.
- 2. Make the required changes.
- 3. Specify the interrupt action by selecting either **Cut** or **Dim**. If **Dim** is selected, specify the dim amount (0 to -61 dB), as required.
- 4. Click the button to post the changes to the database.



IFB Sources

The IFB Editor - Sources dialogue shows any sources that have been defined. New sources can be added and existing ones deleted, as required. In most applications only a single source will be present.

Buttons at the top of the IFB Editor - Route Sources tab let you:

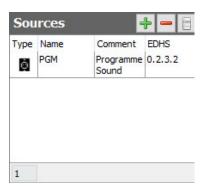
- Add a panel/source as an IFB source.
- Delete a selected IFB source.

Adding Sources

IFB sources are added by selecting from a list of subscribers.

To add an IFB source:

- 1. Open the IFB Editor.
- 2. In the **Sources** dialogue, click the **green +** button to open the pop-up dialogue box.
- 3. Select the required source and click the **Apply** button.
- 4. Click the button to post the route sources to the database.



Deleting IFB Sources

IFB sources that are no longer required for the IFB can be deleted. You can either delete selected route sources (using the Delete button) or delete all route sources at the same time (using the right-click context menu).



IFB Destinations

The IFB Editor - Destination dialogue shows any destinations that have been defined. New route destinations can be added and existing ones deleted, as required. Only a single destination is permitted.

Buttons at the top of the IFB Editor - Route Destination tab let you:

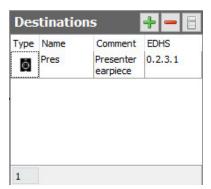
- Add a Subscriber as an IFB destination.
- Remove the IFB destination.

Adding an IFB Destinations

IFB destinations are added by selecting from a list of Subscribers.

To add a destination:

- 1. Open the IFB Editor.
- 2. In the Destinations dialogue, click the **green +** button to open the pop-up dialogue box.
- 3. Select the required destination and click the **Apply** button.
- 4. Click the button to post the route destinations to the database.



Deleting the IFB Destination

The current IFB Destination must be deleted using the ${\bf red}$ – button, prior to adding an alternative destination.



IFB Interrupts

The IFB Interrupts dialogue shows any interrupts that have been defined. New interrupts can be added and existing ones deleted, as required.

Buttons at the top of the IFB Editor - Interrupts dialogue let you:

- Add an IFB interrupt. Three different types of interrupt are presented:
 - Subscriber (panel, radio, telephone etc.)
 - GPI Input
 - o GPI Output
- Delete a selected panel/source IFB trigger.
- Delete all panel/source IFB triggers (after confirmation, from rightclick context menu).

Adding IFB Interrupts

Interrupts are added by selecting from a pop-up dialogue.

To add an IFB Interrupt:

- 1. Open the IFB Editor.
- 2. In the Interrupts dialogue, click the **green +** button and a sub-menu will appear.



 Choose the required type of interrupt – the pop-up dialogue displayed next will be populated accordingly. The three different options are shown below and the selection procedure for each is the same.



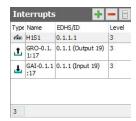




- 4. Select the required interrupt or trigger and click the **Apply** button. To add multiple items, hold down the **Ctrl** key when selecting.
- 5. To change the level, click in the **Level** field and select the priority (1-3) from the drop-down list. The lower the number the higher the priority. If more than one panel wishes to talk to the IFB destination, the panel with the highest priority (lower number) will always take precedence. The default priority is '3.
- 6. Click the button to post the external trigger to the database.



The example below shows interrupts of each type added to IFB1.



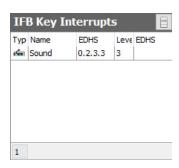
Deleting External Triggers

External triggers that are no longer required for the IFB can be deleted. You can either delete selected external triggers (using the Delete button) or delete all external triggers at the same time (using the Clear All from the context menu).



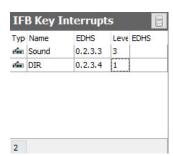
IFB Key Interrupts

The IFB Key Interrupts dialogue shows any panels that have been configured directly with an IFB as a key target. The information here is primarily read-only: IFB Key Interrupts cannot be added or removed here, only as part of panel configuration. See the Subscriber – Hardware Panel editor chapter for more information.



The image above shows the Sound panel with EDHS 0.2.3.3 that has a key targeting IFB1. The only parameter which can be adjusted here is the IFB Trigger level.

When a panel is used to provide the interrupt, this is frequently the fastest and easiest way to set up an IFB. The next image shows the same IFB with 2 panels having key interrupts – Sound and Director. Director has been set with a higher priority, trigger level 1, which means that Director will override Sound should they speak simultaneously.





IFB Editor - Field Definitions

IFB Editor - Details

Field/Display Item	Description
Name	The name of the IFB. This is defined using the Enterprise Editor when the IFB is added to the Enterprise. The name can be changed using the IFB Editor, if required. The name does not have to be unique.
	Maximum length for the name is 20 characters.
Long name	Used on panels with displays supporting 8 characters.
Comment	Comments relating to a selected IFB. To add comments, click in this field and enter your comments.
	Maximum length for comments is 255 characters.
Cut	Specifies the IFB interrupt action – cuts the source audio.
Dim	Specifies that the IFB interrupt action – dims the source by the specified amount. Default = 0 dB. Range 0 to -61 dB.
	Doladit - 0 db. Mange 0 to -01 db.

IFB Editor - Sources

Field/Display Item	Description
Туре	An icon that represents the selected panel/source.
Source	The name of the selected panel/source. This cannot be changed.
EDHS	The EDHS address of the selected panel/source. This cannot be changed.
Comment	Shows comments entered using the Panel Configuration Editor.

IFB Editor - Destination

Field/Display Item	Description
Туре	An icon that represents the selected panel/source.
Source	The name of the selected panel/source. This cannot be changed.
EDHS	The EDHS address of the selected panel/source. This cannot be changed.
Comment	Shows comments entered using the Panel Configuration Editor.



IFB Editor - Interrupts

Field/Display Item	Description
Туре	An icon that represents the selected interrupt. This may be a panel, or a GPI input / output.
Source	The name of the selected interrupt.
EDHS/ID	The EDHS address or ID of the selected interrupt.
Level	Defines the priority of the IFB; the lower the number the higher the priority. If more than one source wishes to trigger the IFB, the source with the highest priority (lower number) will always take precedence. Click in this field and select the required level (1-3).
Comment	Shows comments entered using the Panel Configuration Editor.

IFB Editor – Key Interrupts

Field/Display Item	Description
Туре	Shows an icon that represents the selected panel/ source.
Source	The name of the selected panel/source that has a key targeting the IFB.
EDHS	The EDHS address of the selected panel/source.
Level	Defines the priority of the trigger, the lower the number the higher the priority. Click in this field and select the required level (1-3). Default = 3.



Chapter 17 - SIP and the SIP Connection Editor

Overview

Session Initiation Protocol (SIP) is an open, standards based protocol for negotiating voice and multimedia calls over a network. The addition of SIP provides Mercury with the capability to make and receive SIP VoIP calls to any configured, basic SIP devices that may be present on the network.

A SIP connection is an object used to identify a SIP entity that exists outside of Mercury. SIP connections are added at Enterprise level using the Enterprise Editor. If no SIP connections have been added, the SIP Connections item will not be displayed in the Enterprise tree.

SIP connections are associated with a URL (or alias) using the Host Editor. When a caller attempts to connect to that URL (or alias) on a host, the SIP connection corresponding to the alias is 'activated'. Once activated, a SIP connection key on any panel connected to the host can be targeted to answer the call.

A SIP connection may have a list of phone controls associated with it. These phone controls determine how an incoming call is connected when an alias targeting the SIP connection is called from a remote SIP/radio phone. The types of control objects that can be connected to within a SIP call include conferences, routes and simple intercom connections to and from the SIP/radio phone. A DTMF dial sequence (phone number) is required to be entered at the remote SIP/radio phone to control which of the given subscribers are targeted. Only those control subscribers listed can be targeted.

The SIP Connection Editor is used to change the name of a SIP connection and to associate a picture with the connection. Phone controls can be added and deleted, as required. A list of panel members in the SIP Connection Editor lists all panels that have keys targeting the SIP connection.



Using the SIP Connection Editor

Viewing a List of SIP Connections

You can view a list of all SIP connections that have been defined by selecting and expanding SIP Connections in the Enterprise tree.

The basic and advanced editors carry exactly the same information.

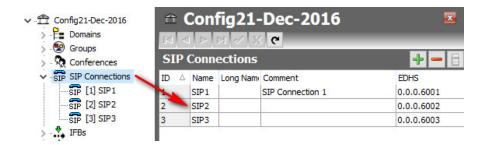


NOTE: If the SIP Connections node does not show the right-arrow in the Enterprise tree, then no SIP connections have been defined.

To view a list of SIP connections:

- 1. Expand Gateway Enterprise tree.
- 2. Click on SIP Connections.

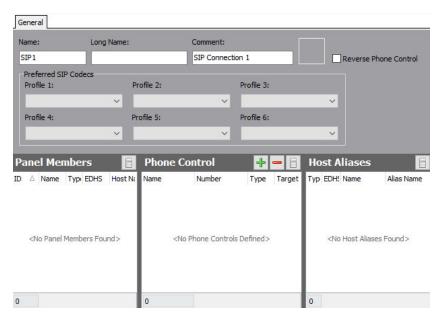
The Details panel lists all of the SIP connections that have been defined. Some basic details can be edited and SIP Connections can be added or removed using the toolbar buttons. The same details are provided at Enterprise level. SIP Connections can also be added or deleted in both places.



To open the SIP Connection Editor:

- 3. Expand the Enterprise tree in Gateway.
- Click the right-arrow next to SIP Connections in the Enterprise tree to show a list of connections.
- 5. Double-click on the required connection to open it in the SIP Connection Editor. The basic and advanced editors carry exactly the same information.





The SIP Connection Editor

The SIP Connection Editor is divided into four areas which are used to change SIP connection details, view panel members, add and delete phone controls and view host aliases.

Details area

The Details tab is used to specify basic SIP connection details.

Panel Members area

The Panel Members area shows a list of panels that have the SIP connection assigned to a panel key.

Phone Control area

The Phone Control area is used to add and delete phone controls.

Host Aliases area

The Host Aliases area shows details of any aliases that have been defined for the SIP connection using the Host Editor.



SIP Details Area

The SIP Connection Editor - Details area shows the name of the selected SIP connection and any comments that have been entered relating to it.



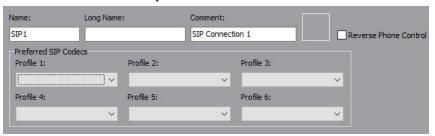
NOTE: SIP connections are added at Enterprise level using the Enterprise Editor or on the SIP Connections overview, as outlined earlier.

Changing SIP Connection Details

You can change the name and description (comment) for the selected SIP connection, as required. A picture can be specified for the connection and the zone specified or changed.

To change SIP connection details:

- Open the SIP Connection Editor. The **Details** area is at the top of the form.
- 2. Make the required changes to the name or comments.
- To add a picture to identify the SIP connection on a Virtual Panel, click the **Load** button and browse for and select the required image file.
- 4. Preferred SIP Codecs may also be defined here.



5. Click the button to post the changes to the database.



SIP Panel Members Area

If a SIP connection has been specified as a key on a panel (see "Panel and Subscriber Configuration"), then details of the source are displayed in the SIP Connection Editor - Panel Members area. This is at the left side of the SIP Connections Editor form.



In the screenshot above, SIP1 has been allocated to a key on panel H1S2. This is shown on the Panel Members tab for SIP1 in the SIP Connection Editor. None of the information shown here can be changed.



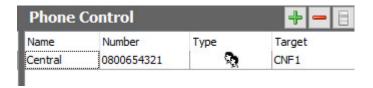
SIP Phone Control Area

The SIP Connection Editor - Phone Control area shows any phone controls that have been associated with the selected SIP connection. New phone controls can be associated and existing ones can be removed, as required.

Phone controls are incoming digits that are used to direct a call, once answered, to the configured subscriber.

Buttons at the top of the Phone Control tab let you:

- Add a new phone control.
- Delete a phone control.



Associating Phone Controls

Phone controls can be associated with a SIP connection, as required. When you add a phone control you select from the list of global phone controls created and maintained using the Phone Number Editor. See the "Phone Numbers and the Phone Number Editor" chapter in this Configuration Guide for details. Phone controls targeting TEB ports or phonesets cannot be selected.

To add a phone control:

- Open the SIP Connection Editor: the Phone Control area is centre screen.
- 2. Click the **green +** button to open the dialogue box.
- 3. Select the required phone controls and click the **Apply** button.



NOTE: You cannot edit any details of phone controls on the SIP Connection Editor - Phone Control tab.

Removing Phone Controls

Phone controls that are not required for a SIP connection can be removed. You can delete individual controls (using the Delete button) or delete all controls associated with the selected SIP connection at the same time (using the right click context menu).



NOTE: When you delete phone controls from a selected SIP connection they are not deleted from other connections or from the global phone control list maintained by the Phone Number Editor.



SIP Host Aliases Area

If an alias has been defined for a SIP connection using the Host Editor (see "SIP Aliases" in the Host Editor chapter of this Configuration Guide), then details of the alias are shown in the SIP Connection Editor - Host Aliases area.

Viewing Host Aliases

You can view details of a Host alias on the SIP Connection Editor - Host Aliases area.

None of the information shown here can be changed.

To view Host Aliases:

Open the SIP Connection Editor: the Host Aliases area is to the right side of the form.





SIP Connection Editor - Field Definitions

SIP Connection Editor - Details area

Field/Display Item	Description
Name	The name of the SIP connection. This is usually defined using the Enterprise Editor when the connection is added to the Enterprise. The name can be changed using the SIP Connection Editor, if required. The name does not have to be unique. Maximum length for the name is 20 characters.
Long Name	The Long Name will be displayed on panels with 8 character displays.
Comment	Comments relating to a selected connection. To add comments, click in this field and enter your comments.
	Maximum length for comments is 255 characters.
Picture	Shows whether a picture that has been assigned to the SIP connection. To assign a picture or change an existing picture, click the Load button and then browse for and select the required image file. Only available on Virtual Panels.
Reverse Phone Control	
Reverse Prione Control	This option can be enabled, and the SIP Connection associated with a Conference and a Phone number, so that when the Conference is activate a SIP call will be automatically dialled to the configured number. This connects the Conference with an external SIP third party. Contact Trilogy for more information.
Preferred CODEC 1–6	Not currently used.

SIP Connection Editor - Panel Members area

This is a read-only area.

Display Item	Description
Name	The name of the source.
Туре	An image of source type.
EDHS	The EDHS address of the source.
Host Name	The name of the host within the specified domain.



SIP Connection Editor - Phone Control area

Display Item	Description
Name	The name of the phone control as defined using the Phone Number Editor.
Number	The number associated with the phone control.
Туре	An icon showing the type of panel or subscriber that the phone control is targeting.
Target	The name of the panel or subscriber that the phone control is targeting.

SIP Connection Editor - Host Aliases area

This is a read-only area.

Display Item	Description
Туре	An icon to identify the host type.
EDHS	The EDHS address of the host on which the alias was defined.
Name	The name of the host on which the alias was defined.
Alias Name	The name of the SIP connection alias as defined using the Host Editor.



Chapter 18 - Routes and the Routes Editor

Overview

A route is a defined audio path created by declaring a source and a destination pair. Routes can be created, broken or inhibited by a GPI, another route, or by a panel key directly.

Routes are created and maintained using the Routes Editor.

To create a route you must specify a source (which can be a panel/subscriber or conference) and a destination (which can also be a panel/subscriber or conference). You cannot route one conference to another

Routes can be fixed and made automatically when the TBC starts, or made as required. Routes that are idle can also be set to break automatically, if required.

Once created, routes can then be used elsewhere in the system, for example when creating GPIO processes.

If you delete a route using the Routes Editor, any GPIO processes that have been defined using the specified route will also be deleted automatically.

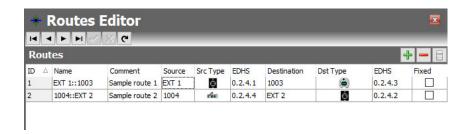


Using the Routes Editor

Opening the Routes Editor

Routes are created and maintained using the Routes Editor.

- 1. Expand the Enterprise tree in Gateway.
- 2. Click on Routes in the Enterprise tree to open the Routes Editor. The basic Routes Editor will open: in this instance the basic and advanced editors are identical.



The Routes Editor shows all routes that have been defined. In the example above, 2 routes have been defined. The routes can be ordered natively (the order in which they were added) or by name. Click the appropriate column heading, ID or Name, to order the routes by your preferred method.

Buttons at the top of the Sources tab let you:

- Add a new route.
- Delete a route.

Other options are available from the right click context menu.

Adding a Route

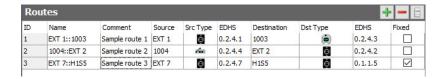
New routes can be added, as required. You can also swap source and destination for existing routes as well as change sources and destinations.

To add a route:

- Open the Routes Editor.
- 2. Click the **green +** button. A single blank entry is created and the **Source** field is highlighted.
- 3. Click the button within the **Source** field to display a list of 'source' panels/sources, or click to show a list of 'source' conferences. Select the required panel/source or conference and then click the **Apply** button.
- 4. Click in the **Destination** field and click the button to display a list of 'destination' panels/sources, or click to show a list of 'destination' conferences.
- 5. Select the required destination panel/source or conference and then click the **Apply** button. Note that you cannot route a conference to a conference.
- To specify that the route is fixed (and made automatically when the TBC starts), select the **Fixed** check box. Note that if you check this box, the route cannot be controlled by any other process such as a GPI Input.



- 7. Click the button to post the route to the database.
- 8. Note that the name field is completed automatically in the format Source : Destination but may be edited.



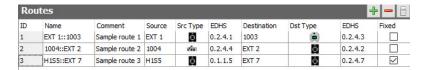
Sample route 3, shown above, is a fixed route between two 4-wire ports.

Swapping a Route

A route can be swapped (i.e. reversed) if the sources and destinations have been defined the wrong way around.

To swap a route:

- 1. Open the Routes Editor.
- 2. Select the route that you want to swap (Sample route 3).
- 3. Click **Swap** from the context menu to swap source and destination.



Deleting a Route

Routes that are no longer required can be deleted. You can delete individual routes (using the Delete button) or delete all routes at the same time (using the context menu).



IMPORTANT: When you delete a route, any GPIO processes that use the deleted route will also be deleted.



Routes Editor - Field Definitions

Routes Editor

Field/Display Item	Description
ID	Non-editable field. Increments in line with route creation.
Name	The name of the route (Source::Destination). This can be changed, if required. The route name does not have to be unique.
	Maximum length for the name is 64 characters.
Comment	Any comment relating to the route. You can add a comment if required.
	Maximum length for comments is 255 characters.
Source	The name of the route source. Click in this field and
	click the button to specify the source.
Source Type	Shows an icon that represents the route source type (e.g. panel, 4-wire etc.).
EDHS	The EDHS address of the route source. This cannot be changed.
Destination	The name of the route destination. Click in this field and click the button to specify the destination.
Dest Type	Shows an icon that represents the route destination type (e.g. panel, 4-wire etc.).
EDHS	The EDHS address of the route destination. This cannot be changed.
Fixed	Fixed routes are routes that are made immediately when the TBC starts. They remain made, unless inhibited, for as long as the TBC managing them is running. They are used when a route with no other mechanism to make the route is required.



Chapter 19 - GPIO Processes and the GPIO Process Editor

Overview

The GP Inputs/Output Processes Editor is used to define the rules by which routes are made and broken within a Mercury system.

Before you can create GPIO statements, you must create routes within the system using the Routes Editor (see the "Routes and the Routes Editor" chapter in this Configuration Guide). You can define more routes that you ultimately use, but you cannot create GPIO statements without first having created one or more routes.

Normally a route is merely a declaration of interest in a specific signal path. It comprises a source and destination pair: the declaration of a route does not necessarily imply that any audio signal is present.

GPI inputs and outputs which are used in conjunction with routes are declared at host level using the Host Editor (see the "Hosts and the Host Editor" chapter in this Configuration Guide).



NOTE: Subject to support by appropriate hardware, you can also define GPIs at panel and subscriber level and these will also be available globally for use in GPIO route statements.



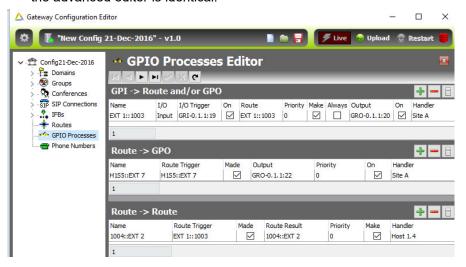
Using the GPIO Processes Editor

Opening the GPIO Processes Editor

GPIO processes are maintained using the GPIO Processes Editor. Routes must be created before GPIO statements can be created or maintained.

To open the GPIO Processes Editor:

- Expand the Enterprise tree in Gateway.
- 2. Click on the **GPIO Processes** branch in the Enterprise tree to open the GP Inputs/Output Processes Editor. The basic editor will open: the advanced editor is identical.



Understanding the GPIO Processes Editor

The best way to read entries in the editor is as a series of logical statements.

IF [trigger action] [is true / false]

THEN [result]

AND [optional extra result]

GPIO Processes Editor

The GPIO Processes Editor has three areas which are described below.

GPIO > Route and/or GPO

Select a GPI input or output trigger for a specified route and also inhibit/uninhibit or make/break the route. A further optional action may be selected to fire a GP output.

Route > GPO

The Route > GPO statement lets you select a route which when made, triggers a selected GPO.

Route > Route

The Route > Route statement lets you select a route trigger and specify a route result.



GPIO > Route Statements

The GPIO > Route section is used to specify a GPI input or output trigger for a particular route.



NOTE: To define route triggers, ensure that you have defined the required GPIOs on the correct host using the Host Editor and the required routes using the Routes Editor.

Buttons at the top of the GPIO > Route tab let you:

- Add a GPIO trigger for a route.
- Delete a selected GPIO route trigger.

Adding a GPIO > Route Statement

A GPI or GPO can be defined as a trigger for a route. The logic is as follows:

IF 'I/O trigger' is [On/Off]

THEN [Make/Inhibit] 'Route' [Always]

AND [Make/Break] 'Optional GP Output'

ON TBC 'Handler'

Where:

- [On/Off] = whether the trigger is on or off. [Make] = make or break the route. [Inhibit] = inhibit or uninhibit the route.
- [Always] = creates a fixed crosspoint that is always on (if selected).

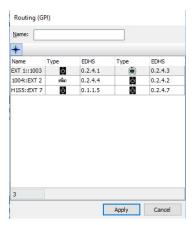
To add a GPIO > route statement:

- 1. Open the GPIO Processes Editor. The **GPIO > Route** statements appear at the top of the editor.
- 2. Click the Add button to add a single new record.
- Click in the I/O Trigger field and click the button to open the GPIO dialogue box. You can toggle between inputs and outputs if required. Inputs are selected by default.



- 4. Select the required trigger and click the **Apply** button.
- 5. Click in the **Route** field and click the button to open the Routes dialogue box.





- 6. Select the required route and click the **Apply** button.
- 7. Tick the required check boxes to specify the logic for the process.
- 8. Optionally, to also trigger an output, click in the **Output** field click the button to open the GP Outputs dialog box. Select the required output and click the **Apply** button.
- 9. Click the button to post the changes to the database.
- 10. You can name the statement either before or after posting to the database. If you leave the name field blank a name will automatically be created.
- 11. The result is shown below.



Deleting a GPIO > Route Statement

GPIO > Route statements can be deleted, as required. You can either delete selected statements (using the Delete button) or delete all GPIO > route statements that have been defined at the same time (using the context menu).



NOTE: Deleting the logic statements does not delete the selected GPIOs or routes from the database.



Route > GPO Statements

The Route > GPO section is used to select a route that will trigger a selected GP Output. Buttons at the top of the Route > GPO tab let you:

- Add a route that will trigger a GP Output.
- Delete a selected statement.

Adding a Route > GPO Statement

A route can be defined as a trigger for a specified GPO. The logic is as follows:

IF 'Route' is [On/Off]

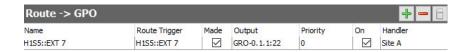
THEN [Make/Inhibit] 'GPO' ON TBC 'Handler'

Where:

- [On/Off] = whether the route is made or not.
- [Make] = make or break the GPO.
- [Inhibit] = inhibit or uninhibit the GPO.

To add a route > GPO statement:

- Open the GPIO Processes Editor. The Route > GPO section is centre screen.
- 2. Click the Add button to add a single new record.
- 3. Click in the **Route Trigger** field and click the button to open the Routes dialog box. Select the required route trigger and click the **Apply** button.
- 4. Click in the **Output** field and click the button to open the GP Outputs dialog box. Select the required output and click the **Apply** button.
- 5. Select the required check boxes to specify the logic for the process.
- 6. Click the button to post the changes to the database.
- 7. You can name the statement either before or after posting to the database. If you leave the name field blank a name will automatically be created.
- 8. The result is shown below.



Deleting a Route > GPO Statement

Route > GPO statements can be deleted, as required. You can either delete selected statements (using the **Delete** button) or delete all Route > GPO statements that have been defined at the same time (using the context menu).



NOTE: Deleting the logic statements does not delete the selected GPOs or routes from the database.



Route > Route Statements

The Route > Route section is used to specify a route trigger and route result. Buttons at the top of the Route > Route tab let you:

- Add a route that will trigger a route result.
- Delete a selected statement.

Adding a Route > Route Statement

A route can be defined as a trigger for another route to be made or broken. The logic is as follows:

IF 'Route Trigger' is [On/Off]

THEN [Make/Inhibit] 'Route Result' ON TBC 'Handler'

Where:

- [On/Off] = whether the route trigger is on or off. [Make] = make or break the route.
- [Inhibit] = inhibit or uninhibit the route.

To add a route > route statement:

- 1. Open the GPIO Processes Editor. The **Route > Route** statements are at the bottom of the screen.
- 2. Click the **Add** button to add a single new statement record.
- 3. Click in the **Route Trigger** field and click the button to open the Routes dialog box. Select the required route trigger and click the **Apply** button.
- 4. Click in the **Route Result** field and click the button to open the Routes dialog box. Select the required route result and click the **Apply** button.
- 5. Tick the required check boxes to specify the logic for the process.
- 6. Click the button to post the changes to the database.
- You can name the statement either before or after posting to the database. If you leave the name field blank a name will automatically be created.
- 8. The result is shown below.



Deleting a Route > Route Statement

Route > Route statements can be deleted, as required. You can either delete selected statements (using the Delete button) or delete all Route > Route statements that have been defined at the same time (using the context menu).



NOTE: Deleting the logic statements does not delete the selected routes from the database.



GPIO Processes Editor - Field Definitions

GPIO Processes Editor - GPIO > Route section

Field/Display Item	Description
Name	A system-generated name based on the route result. This can be changed to allow easy identification of the process. The name does not have to be unique.
	Maximum length for the name is 20 characters.
1/0	Shows the type of input/output selected in the I/O Trigger field.
I/O Trigger	The GPIO or GPO used to trigger the route defined in the Route field.
On	If selected, the GPIO trigger is on and the routing function is performed.
	If not selected, the GPIO trigger is off and the routing function is not performed.
Route	Specifies the route to be made.
Make	If selected, defines the route as being made.
	If de-selected, defines the route as being inhibited.
Always	If selected, creates the route as a fixed crosspoint that is always on.
Output	Allows a GPO to be triggered as a result of a GPI.
On	Flips the logic of the GPO configured in the Output field.
Handler	Shows the host that will be handling the GPIO. This is the host on which the selected route was



GPIO Processes Editor - Route > GPO section

Field/Display Item	Description
Name	A system-generated name based on the route trigger. This can be changed to allow easy identification of the process. The name does not have to be unique.
	Maximum length for the name is 20 characters.
Route Trigger	The route used to trigger the GPO defined in the Output field.
	If more than one GPO needs to be controlled from a single trigger condition, multiple GPIO statements are required, with the same trigger.
Made	If selected, the route is made.
	If de-selected, the route is not made.
Output	Allows a GPO to be triggered as a result of a route being made or not made.
Priority	Specifies the priority of the route.
On	If selected, makes the selected GPO.
	If de-selected, does not make the selected GPO
Handler	Shows the host that will be handling the GPIO. This is the host on which the route trigger was

GPIO Processes Editor - Route > Route section

Field/Display Item	Description
Name	A system-generated name based on the route result. This can be changed to allow easy identification of the process. The name does not have to be unique.
	Maximum length for the name is 20 characters.
Route Trigger	The route used to trigger the second route defined in the Route Result field.
Made	If selected, the route is made.
	If not selected, the route is not made.
Route Result	The resultant route.
Priority	Specifies the priority of the route.
Make	If selected, this defines the route as being made.
	If not selected, this defines the route as being inhibited.
Handler	Shows the host that will be handling the GPIO. This is the host on which the route trigger was



Chapter 20 - Phone Numbers and the Phone Number Editor

Overview

The Phone Number Editor is used to add, change or delete frequently used, globally-available phone and SIP numbers.

Numbers added to the phone and SIP number lists can be added to panels (see the "Panel and Subscriber Configuration" chapter in this Configuration Guide) and used locally.

The Phone Number Editor is also used to define phone sets and phone controls.

- Phone sets are logical groupings of panels that display a ringing tally when an incoming call is detected on the phone line to which the panels have a key.
- Phone controls are incoming DDI digits that are used to direct an answered call to the appropriate panel, conference, FXS, FXO or E&M port, route or phone set.

In common with the other areas of the Gateway application, both basic and advanced editors are provided. In this instance they are identical and the basic version will be described.



Using the Phone Number Editor

Opening the Phone Number Editor

Phone numbers, SIP numbers, phone controls and phone sets are maintained using the Phone Number Editor.

To open the Phone Number Editor:

- 1. Expand Gateway Enterprise tree.
- Click on **Phone Numbers** in the Enterprise tree to open the Phone Number Editor.
- The Details panel is divided into 4 on-screen areas, as shown in the image below.



Details of items 1, 2 and 3 are contained fully on this screen and are described in following sections. Item 4, Phone Sets, are added on this screen and basic details may be edited. Membership of Phone Sets is managed with the Phone Set editor and Phone Sets are listed separately under the Enterprise Tree.

The Phone Number Editor Areas

The Phone Number Editor has four areas which are used to maintain phone numbers, SIP numbers, phone controls and phone sets in the global directory.

1. Phone Numbers

The Phone Numbers tab is used to add and delete phone numbers.

2. Phone Control

The Phone Control tab shows a list of all phone numbers that have been added to call conferences, FXO, FXS or E&M ports and routes.

3. SIP Numbers

The SIP Numbers tab is used to add and delete SIP numbers.

4. Phone Sets

The Phone Sets tab is used to define phone sets.



Phone Numbers

The Phone Number Editor - Phone Numbers area shows a list of all phone numbers that have been added. Phone numbers are outgoing digits that may be used by panels as speed dials when accessing FXO, FXS or E&M ports. Digits are dialled as DTMF when the call is placed.

Buttons at the top let you:

- Add a new phone number.
- Delete a phone number.

Adding a Phone Number

Phone numbers can be added, as required. To add a phone number you must provide a name and number. A picture can be specified for each number, if required.

To add a phone number:

- Open the Phone Number Editor. The **Phone Numbers** area is located at top left.
- 2. Click the **Add** button. A new row is added to the phone number list.
- 3. Click in Name and enter the name of the contact.
- 4. Click in **Number** and enter the number field for that contact.
- To add a picture for the contact, click in **Picture** then click the button and browse for and select the required image file.



6. Click the button to post the number to the database.

Deleting a Phone Number

Phone numbers that are not required can be deleted. You can either delete individual numbers (using the Delete button) or delete all numbers at the same time (using the right-click context menu).



IMPORTANT: Deleting either individual numbers or all numbers from the global directory will also delete them from local panel listings.



SIP Numbers

The Phone Number Editor - SIP Numbers area shows a list of all SIP numbers that have been added. Each SIP number record includes a reference name, quick dial number, subscriber URL or SIP address and an authentication user ID and password. SIP numbers are outgoing digit sequences that map to either fully formed URIs or are digits that are passed to a Proxy Server to be resolved fully.

Buttons at the top let you:

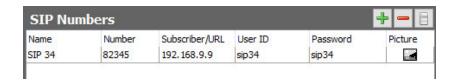
- Add a new SIP number.
- Delete a SIP number.

Adding a SIP Number

SIP numbers can be added, as required. To add a SIP number you must provide a name and number. Authentication details can also be added if these are needed by the connection. A picture can be specified for each number, if required.

To add a SIP number:

- Open the Phone Number Editor. The SIP Numbers area is located lower left.
- 2. Click the Add button. A new row is added to the SIP number list.
- 3. Click in Name and enter the reference name for the number.
- 4. Click in **Number** and enter the quick dial number for that contact.
- 5. Click in **Subscriber/URL** and enter the SIP address. This can be an IP address or URL. If you leave this field blank, it will default to the reference name that you specify.
- 6. Click in **User ID** and enter the required user ID, if required.
- Click in **Password** and enter the authentication password, if required.
- 8. To add a picture for the SIP number, click in **Picture** then click the button and browse to and choose the required image file.



9. Click the button to post the record to the database.

Deleting a SIP Number

SIP numbers that are not required can be deleted. You can either delete individual numbers (using the Delete button) or delete all numbers at the same time (using the right-click context menu).



IMPORTANT: Deleting either individual numbers or all numbers from the global list will also delete them from local panel listings.



Phone Controls

The Phone Number Editor - Phone Control area shows a list of all phone numbers that have been added to call panels, conferences, FXO, and FXS ports, routes and phone sets. Phone controls are incoming DDI digits that are used to direct an answered call to the appropriate panel, conference, FXS, FXO or E&M port, route or phone set.

Buttons at the top of the Phone Number Editor - Phone Control area let you:

- Add a new phone control.
- Delete a phone control.

Adding a Phone Control

To add a phone control you must specify a name, number and source name. The mode defaults to Speak/Listen but can be changed to either Speak or Listen, as required.

To add a phone control:

- 1. Open the Phone Number Editor. The **Phone Control** list is located lower left of screen.
- 2. Click the **Add** button. A new row is added to the phone control list.
- 3. Click in **Name** and enter the name of the control. If you leave this blank, the name will default to the name of the selected Source.
- 4. Click in **Number** and enter the number corresponding to the control.
- 5. Click in **Source** then click the button and browse to and choose the required conference, panel, GPIO/audio route or phone set.
- 6. Click in **Mode** and then click the button to change the mode, if required.
- 7. Click the button to post the phone control to the database.



Deleting a Phone Control

Phone controls that are not required can be deleted. You can either delete individual phone controls (using the Delete button) or delete all phone controls at the same time (using the right-click context menu).



IMPORTANT: Deleting phone controls from the configuration will affect local panel listings.



Phone Sets

Phone sets are a logical grouping of panels that display a ringing tally when an incoming call is detected on the phone line to which the panels have a key. Any of the panels in the phone set can answer the incoming call.

Buttons at the top of the Phone Number Editor - Phone Sets tab let you:

- Add a new phone set.
- Delete a phone set.

Adding a Phone Set

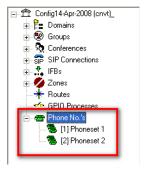
Phone sets can be added to the configuration, as required. Each phone set is given a default name, which can be changed. An optional description may be added.

To add a phone set:

- 1. Open the Phone Number Editor. The **Phone Sets** area is located top right of the Details Panel.
- 2. Click the Add button.
- 3. Specify how many sets you wish to add and click the **OK** button.
- 4. If you want to change the default name, select a phone set and click in **Name** and enter the name of the phone set.
- 5. Click in **Description** and enter a description (optional).
- 6. Click the button to post the phone set to the database. The phone set you added is listed under **Phone Numbers** in the Enterprise tree.



7. Click the right arrow next to **Phone Numbers** in the Enterprise tree to list any phone sets that have been defined. Currently, the phone set has no members. Click on a phone set to open it in the **Phone Set Editor**. Details of the Phone Set Editor are in the next chapter.



Deleting a Phone Set

Any phone sets that are not required can be deleted. You can either delete individual phone sets (using the Delete button) or delete all phone sets at the same time (using the Clear All button).



Phone Number Editor - Field Definitions

Phone Number Editor - Phone Numbers

Field/Display Item	Description
Name	The name of the 'contact'. The name can be changed, as required. The name does not have to be unique.
Number	The phone number. The number can be changed, as required.
	The number string can include the digits 0-9, *, # and ,
	Maximum length is 20 digits/special characters.
Picture	Shows the picture that has been assigned to the number. To assign a picture, or change an existing picture, click the ellipsis button and then browse for and select the required image file. Visible on Virtual Panels only.

Phone Number Editor - Phone Control tab

Field/Display Item	Description
Name	The name of the control.
	The name can be changed, as required.
Number	The phone control number.
	The number can be changed, as required.
Mode	The mode defaults to Speak/Listen but can be changed by clicking on the icon and then clicking the button.
Туре	Shows an icon that represents the source.
Source	Click in this field and then click the button to select or change the source (conferences, panel sources, GPIO audio routes or phone sets).



SIP Number Editor - SIP Numbers tab

Field/Display Item	Description
Name	The reference name of the SIP number. The name can be changed, as required. The name does not have to be unique.
Number	The quick dial number. The number can be changed, as required.
	The number string can include the digits 0-9, *, # and ,
	Maximum length is 20 digits/special characters.
Subscriber/URL	The SIP address. You can enter an IP address or URL. If you leave this field blank, it will default to the Name when you save the record.
User ID	The authentication user ID for the SIP number.
Password	The authentication password for the SIP number (and user ID).
Picture	Shows the picture that has been assigned to the number. To assign a picture, or change an existing picture, click the ellipsis button and then browse for and select the required image file. Visible on Virtual Panels only.

Phone Number Editor - Phone Sets tab

Field/Display Item	Description
ID	The ID of the phone set.
	This cannot be changed.
Name	The name of the phone set.
	The name can be changed, as required.
Description	A description of the phone set.
	You can enter a description into the space provided.



Chapter 21 - Phone Sets and the Phone Sets Editor

Overview

Phone sets are logical groupings of panels that will display a ringing tally when an incoming call is detected on the phone line to which the panels have a key. Any of the panels in the phone set can answer the incoming call.

Phone sets are created using the Phone Number Editor. See the "Phone Numbers and the Phone Number Editor" chapter in this Configuration Guide for details on how to create a phone set.

The Phone Set Editor is then used to add, change and delete IP phone sources from phone sets.

In common with the other areas of the Gateway application, both basic and advanced editors are provided. In this instance they are identical and the basic version will be described.



Using the Phone Sets Editor

Opening the Phone Sets Editor

Phone sets are maintained using the Phone Sets Editor.

To open the Phone Sets Editor:

- 1. Expand the Enterprise tree in Gateway.
- 2. Click the right arrow next to Phone Numbers in the Enterprise tree to show a list of phone sets.
- 3. Click on the required phone set to open it in the Phone Sets Editor.



All relevant information is now shown in the details panel.

Details

The Details fields are used to set the name of a phone set, or to add or change the comment about a phone set.

IP Phone Sources

The IP Phone Sources fields are used to add and delete IP phone sources that define which panels ring when a specific number is called.

Phone Set Details

To change phone set details:

- 4. Open the Phone Sets Editor.
- 5. Change the name of the phone set and enter a comment, if required.
- 6. Click the button to post the changes to the database.



Phone Set IP Phone Sources

Adding IP Phone Sources to a Phone Set

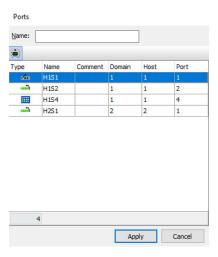
An IP phone source is a panel that will ring if a particular phone is dialled. This allows you to define which panels will display an incoming call and subsequently which users will see that call. Once answered by an IP phone source panel, all other panels in the phone set will stop ringing.



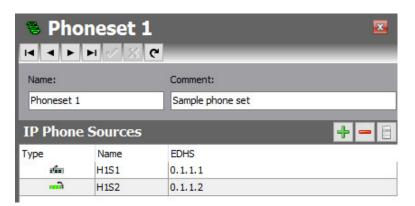
NOTE: A panel/source will only be available for inclusion in a phone set if it has a key targeting a TEB or SIP connection.

To add an IP phone source:

- 1. Open the Phone Sets Editor.
- 2. Click the **green +** button to open the Panels/Sources dialog box.



3. Select the panels/sources that you want to add and click the **Apply** button.



Deleting IP Phone Sources

IP phones sources that are not required in the phone set can be deleted. You can delete individual sources (using the Delete button) or delete all sources at the same time (using the right click context menu).



Phone Set Editor - Field Definitions

Phone Sets Editor - Details

Field/Display Item	Description
Name	The name of the phone set. The name can be changed, as required. The name does not have to be unique.
	The maximum length for a phone set name is 64 characters.
Comment	Any comments entered when the phone set was created using the Phone Number Editor. Comments may be added or changed.
	Maximum length for comments is 255 characters.

Phone Sets Editor - IP Phone Sources

Field/Display Item	Description
Туре	An icon showing the type of source selected, e.g. Virtual Panel.
EDHS	The EDHS address of the selected source. Cannot be changed.
Name	The name of the selected source. Cannot be changed.



Chapter 22 - Mercury Terminology

4-wire	A balanced pair of analog audio input and output signals, generally associated with a single item of external equipment. This is connected using a total of four wires, audio in +, audio out +, audio out
access level	A number, in the range 0 to 32, which is assigned to a zone to specify its access level. 0-no access; 1-highest; 32-lowest.
acoustic feedback	Feedback which occurs when a sound loop exists between an audio input and an audio output.
active client	A host that is connected to the Mercury database.
additional host	A host that needs to be interested in a given conference. Assigning additional hosts enables the TBCs for each host to communicate with each other and participate in the conference.
audio crosspoint	A connection between a specific source and destination "pair" made in the analog domain within an MIU.
Audio Expansion Board (AEB)	An expansion board that may be installed into an MIU. Provides eight RJ45 connectors for eight channels of 4-wire audio or intercom panels.
audio loopback	Loops ports when no connection is present. It enables the routing of services to address the sources in the configuration that might not have physical addresses.
audio trunking	See trunking.
bandwidth	A measure of how much data can be transferred in a fixed amount of time.
bandwidth profile	Settings that allow an Administrator to select a specific IP payload over a range of 10 kbps to 100 kbps.
bit	A single binary digit. It is the standard for quantifying data in computing.
byte	A unit of measurement that is equivalent to 8- bits.
cache	A collection of data duplicating original values stored elsewhere.
Carrier Operated Relay (COR)	An on/off signal sent from a radio to indicate that it is in receive mode.
client	A device, such as an IP phone, that makes or answers a call.
codec	A device or program capable of encoding and decoding on a digital data stream or signal.



coding profile	The name by which bandwidth profile was previously known.
command line files	Files that can be used to send commands directly to the firmware when starting the TBC.
Commander	A highly-scalable digital matrix with a minimum capacity of 6 ports and a maximum of 576 in a single matrix.
conference	A one-to-many or many-to-one communication facility provided by Mercury where each participant hears a mix of everyone currently active in the conference, minus their own audio contribution (mix- minus).
Conference Editor	Part of Gateway that is used to configure conferences.
configuration	A set of files that defines the overall structure of a Mercury system.
Configuration Editor	An application that is used to manage the configuration details in the database. It provides a GUI to allow users to visualise and interact with the database.
cut key	A key on a panel that can be configured to cut sound when pressed.
database	A shared collection of logically related data that specifies the configuration of a Mercury system.
dataport	The IP port that the DB master or slave uses for data connections.
Database Supervisor	See Mercury DB Supervisor.
dial plan	In Mercury, a dial plan determines the SIP URIs that a Mercury panel is allowed to dial and the incoming DDI digits that a Mercury host will recognise, and the Mercury subscribers that are associated with these DDI numbers.
domain	A logical grouping of hosts. Used to identify a system within an organisation.
Domain Editor	Part of Gateway that is used to configure domains.
driver	A piece of software used to link the program and its associated hardware.
endpoint	A device, such as an IP phone, that makes or answers a call.
Enterprise	Used to identify an organisation. The Enterprise is the "top level" of a configuration.
Enterprise Editor	Part of Gateway that is used to define the overall structure of your Enterprise.
fast inhibits	An enterprise level setting in Gateway that forces all hosts to break DSP audio routes without contacting the remote host.
firmware	Software that is embedded in a hardware device.
fixed route	A route that is made immediately when the TBC starts. A fixed route remains made, unless inhibited, for as long as the TBC managing the route is running.
flash memory	A type of non-volatile computer memory that can be electrically erased and reprogrammed.



full-duplex	The transmission of data in two directions simultaneously. For example, a telephone is a full-duplex device because both parties can talk at once. In contrast, a walkie-talkie is a half-duplex device because only one party can transmit at a time.
FXO	An FXO (Foreign Exchange Office) is a device that, from the point of view of a telephone exchange, seems to be a regular telephone. As such, it should be able to accept ring signals, go on-hook and off-hook, and send and receive voice signals. Analog telephone handsets, fax machines and (analog) modems are FXO devices. FXO interfaces are also available for computers and networking equipment to allow these to interact directly with a Plain Old Telephone Service (POTS). These are commonly found in devices acting as gateways between VoIP systems and the PSTN. The FXO must be connected to the FXS interface. The FXS interface delivers the familiar dial tone and ring tone to the FXO, and supplies the power for the FXO device to work.
FXS board	An expansion board that provides four channels of FXO for telephone lines plus four fail-safe telephone handsets.
FXS	An FXS (Foreign Exchange Station) is any device that, from the point of view of a telephone, seems to be a telephone exchange. As such, it should be able to supply power to the connected telephone, provide ring signals and dial tone, understand when the phone goes on-hook or off-hook, and send and receive voice signals.
FXS board	An expansion board that provides four channels of FXS for telephone handsets.
gateway	A node (a router) on a computer network that serves as an access point to another network.
General Purpose Input (GPI)	Allows the Mercury system to receive a signal from an external device using an on/off command.
General Purpose Output (GPO)	Allows the Mercury system to send a signal to an external device using an on/off command.
General Purpose Input Output (GPIO)	Used to refer collectively to General Purpose Input and General Purpose Output. See above.
GPIO Processes Editor	Part of Gateway that is used to configure GP Inputs and Outputs.
group	A group is a collection of user-defined subscribers.
Group Editor	Part of Gateway that is used to configure groups.
Hardware Panel	A Mercury control panel that does not require a PC GUI to operate. Hardware panels are dedicated devices available in a range of offerings, with choice of form factor and size.
heartbeat	Network messages sent from all hosts in the enterprise to each other to determine whether hosts are available.
host	A host is a constituent part of a domain. It is the focal point for all user input. A host can be a dedicated piece of hardware, such as an MIU, or a PC with hardware added either internally (PCI card) or externally (USB adapter).
Host Editor	Part of Gateway that is used to configure Mercury hosts.



Interruptible Foldback (IFB)	An audio route which can be broken into, or interrupted, by a third party, replacing the original audio with the third party panel microphone.
IFB Editor	Part of Gateway that is used to configure IFBs.
initiator	The party initiating a call.
instate	Replace cached configuration files with new working files following a configuration update.
interest	A count of how many different TBCs are monitoring a route for the purpose of GPI logic functionality. Most routes will have an interest of 0 in a simple configuration.
invited member	A panel or source that may be invited to participate in a conference.
IP address	A unique number that devices use in order to identify and communicate with each other on a computer network using the Internet Protocol standard.
IP phone	A panel that will ring if a particular phone is dialed.
logging	ASCII text files created on an MIU, recording communication activity.
master database	A database that holds master configuration for all domains and hosts on the Mercury system.
Mercury	A system providing bi-directional audio communication in an IP environment.
Mercury DB Server	A service installed within Windows that runs the Mercury DB Supervisor.
Mercury DB Supervisor	A middleware application that provides an interface between the Mercury database and the Mercury software suite running on the hosts.
Mercury host	See host.
Mercury Installation Tool	An application that allows the location of the database and other host settings to be specified.
Mercury Interface Unit (MIU)	A Mercury host. It can host many user interfaces and external devices. Up to four interface boards may be fitted, to serve 32 audio interface channels. It is suitable in system setups that support multiple users. It is fitted with a Mercury PCI card that provides two channels of audio input and output, and an optional GPIO card providing 16 GP Inputs and 16 GP Outputs.
minimise time	The time taken for an inactive panel to minimise.
mirror dimming	Used to help reduce acoustic feedback when full-duplex communication is taking place.
mix-minus	A feature of conferences where each participant hears a mix of everyone currently active in the conference, minus their own audio contribution.
multicast	The delivery of audio to a group of destinations simultaneously using the most efficient strategy to deliver the messages over each link of the Mercury system only once and only create copies when the links to the destinations split. Multicasting is used to minimise bandwidth usage and is transmitted as a single stream that can be listened to by multiple receivers.



Network Address Translation /Port	NAT/PAT translates TCP or UDP connections made to a host and port on an outside network to a host and port on an inside network.
Address Translation (NAT/PAT)	
network stack	A particular software implementation of a computer networking protocol suite.
noise gate	An electronic device or software logic that is used to control the volume of an audio signal.
non-volatile memory	Computer memory that can retain stored information even when the computer is not powered-up.
operator	An individual that can login to Gateway with a username and password.
Opto	Optical coupling of signals used to isolate any power components.
panel	An interface that enables Mercury users to place and receive calls. Panels include Virtual Panels for PC users and Hardware Panels for non-PC users.
Panel Configuration Editor	Part of Gateway that is used to configure Mercury panels and subscribers.
panel member	A panel that has been specified as a source for a conference.
peer-to-peer operation	A network that relies primarily on the computing power and bandwidth of the participants in the network rather than concentrating it in a relatively low number of servers.
Peripheral Component Interconnect (PCI)	A connection socket for a peripheral designed for the local bus on a computer motherboard.
phone control	Incoming DDI digits that are used to direct an answered call directly to the configured subscriber.
phone number	Outgoing digits that may be used by panels as speed-dials when accessing FXO, FXS or E&M ports. The digits are dialled as DTMF when the call is placed.
Phone Number Editor	Part of Gateway that is used to store frequently used phone numbers. These numbers can be allocated to panels.
phone set	A logical grouping of panels that displays a ringing tally when an incoming call is detected on the phone line to which the panels have a key.
Phone Sets Editor	Part of Gateway that is used to configure phone sets.
pinout	A term used in electronics to describe how an electrical connector is wired. Each pin of the connector has a purpose which is briefly described in the pinout.
Push-to-Talk (PTT)	An input on a radio that allows external equipment to actuate the transmit mode.
Radio Interoperability Board (RIB)	An expansion board that provides eight RJ45 connectors for direct connection to eight radios.
recipient	The party receiving a call.
route	A defined audio path created by defining a source and a destination. Routes can be created, broken or inhibited by a GPI, another route, or by a panel key directly.
route destination	The end point of a particular route.



Routes Editor Part of Gateway that is used to define and maintain routes. serial trunking See trunking. An application that starts when Windows is booted and runs in the background as long as Windows is running. Session Initiation Protocol (SIP) An application-layer control (signalling) protocol for creating, modifying, and terminating sessions with one or more participants. These sessions include Internet telephone calls, multimedia distribution, and multimedia conferences. Sidetone Sound that is picked up by a telephone's mouthpiece and reproduced by the earpiece of the same handset, acting as feedback that the phone is really working. SIP address The Uniform Resource Identifier (or Locator) of an endpoint. Of the form: sip:trilogy@trilogycomms.com or sip:0123456789@87.98.65.43 SIP alias A URL (or alias) assigned to a SIP connection using the Host Editor. Whe a caller attempts to connect to that URL (or alias) on a host, the SIP connection corresponding to the alias is 'activated' Once activated, a SIP connection An object used to identify a SIP entity that exists outside of Mercury. SIP connections are added at Enterprise level using the Enterprise Editor. SIP Connection Part of Gateway that is used to maintain SIP connections and assign phone controls to connections. SIP number Outgoing digit sequences that map to either fully formed URIs or are digits that are passed to a Proxy Server to be fully resolved. SIP Proxy Server Receives "call" requests from a SIP User Agent and forwards these requests to one or more addresses to locate the destination and returns a "redirect" response to the User Agent with the destination and returns a "redirect" response to the User Agent with the destination and returns a "redirect" response to the User Agent with the destination and resurns a "redirect" response to the User Agent sannouncing their presence in the network. SIP URI See SIP address. SIP User Agent A SIP User Agent (or Endpoint) generally contains both User Agent Client and User Agent Serve		
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network to be broken down into what appears (logically) to be several	supervisor port	The IP port on which the database master or slave listens for control data.
	subnet	network to be broken down into what appears (logically) to be several



subnet mask	A local bit mask (set of flags) that specifies which bits of the IP address specify a particular IP network or host within a sub network. An IP address of 128.66.12.1 with a subnet mask of 255.255.255.0 specifies a host 1 on subnet 128.66.12.0. The subnet mask determines the maximum number of hosts on a sub network.
Talkback Controller (TBC)	The host application that is the heart of the Mercury system. It manages the peer-to-peer socket connections between Mercury hosts, manages IP audio routes and local audio crosspoints, and acts as the interface to the host Mercury hardware.
tally	A universal term for the speak and listen status lamps used to inform the user about open speak and listen routes to a target. Tallies are conveniently displayed near a key on hardware panels to show the user clearly the status of any audio routes.
Transistor-Transistor Logic (TTL)	The electrical characteristics of the interface.
trunking	A facility provided on the MIU that enables audio (and, if required, RS232 data) to be routed from one unit to another.
unicast	The delivery of audio to a single destination. A specific and unique connection is set up between the hosts. If the same data must be sent to multiple hosts, either multiple unicast connections are required, or multicast may be used.
Universal Serial Bus (USB)	An interface on the computer that enables you to connect a USB device. USB 1.0 is an external bus standard that enables data transfer rates of 12 Mbps (12 million bits per second). USB 2.0 has speeds equivalent to Firewire 1394 of 400 Mbps.
User Agent Client	Makes "calls" to other SIP devices.
User Agent Server	Receives "calls" from other SIP devices.
User Datagram Protocol (UDP)	One of the protocols for data transfer that is part of the TCP/IP suite of protocols. UDP is a "stateless" protocol, in that UDP makes no provision for acknowledgement of packets received.
Virtual Panel	PC applications that provide a GUI for Mercury users.
Voice Operated Transmission (VOX)	An on/off command that indicates the presence of an audio signal.