



QE20 Fibre Networking User Manual

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The QE20 is a product of

Johnson Controls

17 Mary Muller Drive

Christchurch

NEW ZEALAND

Phone : +64-3-389-5096

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AMENDMENT LOG

24 May 2023	1.0	Initial Release

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1. Overview

Introduction and Scope

This manual describes the design, set up and operation of a VIGILANT QE20 Fibre Network, and in particular the configuration and operation of the VIF, PIB, ATA and Fibre Switch contained in the QE20 Fibre Network Kit.

For information on the design of QE20 systems, including general networking operation, refer to the QE20 Design Manual (LT0726).

For details on the installation and wiring of the QE20 Fibre Networking Kits refer to the FP2024/2025 QE20 Fibre Network Kit Installation Instructions (LT0714).

Related Manuals

The following manuals should be consulted for more information: –

LT0519 PIB User Manual

LT0714 FP2024/2025 QE20 Fibre Network Kit Installation Instructions

LT0726 QE20 System Design Manual

Cisco ATA 191 and ATA 192 Analog Telephone Adapter User Guide

Cisco ATA 191 and ATA 192 Analog Telephone Adapter Administration

MOXA EDS-405A Ethernet Switch User Manual

Terminology Used In This Manual

1.1.1 VIGILANT Products

EWIS	Emergency Warning and Intercommunication System	As per an EWS but also with an emergency phone system.
EWS	Emergency Warning System	A system that generates and amplifies emergency tones, recorded speech messages, and live speech. The system size may range from a single zone to hundreds of zones.
I-HUB	Intelligent Hub	An intelligent “hub” (actually a switch) for Panel-Link that can route and filter messages, and offers ring networking.
PA	Public Address	Older term for a live speech function or announcement on QE20. Naming of speech audio path on QE20, VIF, etc.
Panel-Link		VIGILANT protocol for networking fire alarm panels, EWIS panels, and other products.
PIB	Panel-Link IP Bridge	VIGILANT module for interfacing Panel-Link to an IP network. It also provides functionality (VoIP proxy) to transmit (stream) audio to many VoIP callers.
QE20		VIGILANT Multi-zone EWIS system.
Speech	Live Speech	Function of QE20 to make PA announcements.
SID	System Identifier	An address in the range 1 - 254, used to identify devices on a Panel-Link network.
T-GEN	Tone Generator	A VIGILANT module that generates and amplifies emergency tones, recorded speech messages, and optionally live speech.
VIF	VoIP Interface	A VIGILANT module that controls and monitors an ATA to support VoIP.

1.1.2 IP Related

ARP	Address Resolution Protocol	A protocol to find the physical address (e.g. MAC address) of a device on a local area network that has been assigned a given IP address.
ATA	Analog Telephone Adapter	A device designed to convert an analogue telephone to VoIP.
DHCP	Dynamic Host Control Protocol	A protocol for an IP connectable device to obtain its IP address automatically from a DHCP Server.
Ethernet		A set of frame-based technologies for local area networks. Common variants are 10 Base-T – 10 Mbps over UTP 100 Base-T – 100 Mbps over UTP 100 Base-FX – 100 Mbps over fibre.
HTTP	Hyper Text Transfer Protocol	The protocol used for fetching and delivering “web pages”.
ICMP	Internet Control Message Protocol	A protocol for checking connectivity and reporting errors on a network.
IP	Internet Protocol	A protocol suite used to provide packet switched communication within and between local area networks. IP runs on many types of media (Twisted pair, fibre, WiFi, etc.) An IP address that is unique in the world (or at least unique in a customer’s wide area network) is used to identify devices. The IP address is different to the LAN address, which need not be unique across a WAN (although paradoxically it is in the case of an Ethernet MAC address).
LAN	Local Area Network	A network covering a small physical area.
NAT	Network Address Translator	A device that is interposed between a private network and the Internet, that translates private IP addresses on the private network, to one or several public IP addresses on the Internet.

RTP	Real-time Transport Protocol	A standardized packet format for delivering audio and video over an IP network.
SAS	Streaming Audio Server	A “server” that is present in some ATA and transmits (one-way) digital audio to multiple clients. Whilst functionally similar, the number of callers supported is limited compared to the VoIP proxy of the PIB.
SIP	Session Initiation Protocol	A protocol used to set up and take down phone calls across IP.
TCP	Transmission Control Protocol	A protocol used for reliable transmission of a stream of bytes between two end-points, used for HTTP and Telnet for example.
Telnet		The protocol for connecting an ASCII terminal to a host across a network.
UDP	Universal Datagram Protocol	A protocol for transmission of packets of data from one point to any other point of an IP network. Delivery is on a best-effort basis, i.e., is not guaranteed.
UTP	Unshielded Twisted Pair	Wire cables with 2 or 4 pairs, used for 10 Mbit, 100 Mbit, and 1 Gbit Ethernet connections. At least 2 pairs are required for 10 Mbit / 100 Mbit Ethernet. 4 pairs are required for Gigabit Ethernet.
VoIP	Voice over IP	The use of IP to transport voice or audio.
VoIP Proxy		A function of the PIB that transmits (streams) audio to many VoIP callers.
WAN	Wide Area Network	A network encompassing a large physical area that joins multiple LANs together.

1.1.3 General

Broadcast	A message that is addressed to multiple recipients.
Hub	A device that provides a centre point for a “Star” network connection. All data received on each port is transmitted on every other port .

DTMF	Dual Tone Multi Frequency: an interference resistant technique for encoding telephone dial key presses into audio frequencies.
Multidrop	A network segment with potentially more than two devices. Data messages need to be addressed, and there needs to be a means to ensure that only one device transmits at a time.
Point to Point	A network segment with only two devices – one at each end. Data messages do not need to be addressed to reach their immediate destination. With a “full duplex” connection both devices can transmit at the same time, while with a “half duplex” connection only one device can transmit at a time.
Router	<p>The same as a hub, but data is transmitted on only the required other ports (based on IP address).</p> <p>Routers typically are capable of working out the best (fastest) route to a final destination if there are several routes, and working out alternative routes if the usual route is “down”.</p>
Switch	The same as a hub, but data is transmitted on only the required other ports (based on LAN address, e.g., Ethernet MAC address).

Description of QE20 Fibre Networking

A fibre networked VIGILANT QE20 system uses an IP (Internet Protocol) network to send data and voice (Speech, WIP, paging console, background music) between QE20 panels.

The VIGILANT Panel-Link IP Bridge (PIB) is used to provide data networking between panels as well as one-to-many VoIP audio streaming. A third-party Analogue Telephone Adaptor (ATA) (Cisco ATA 192) is used to convert voice signals into VoIP and provide one-to-one VoIP calls between nodes as well as act as an audio source for the PIB VoIP proxy functionality. The PIB's Ethernet connection is wired via the ATA (it has two Ethernet ports working in bridged mode), so a communications fault is generated if the ATA fails, is powered down, or loses its configuration.

The VIGILANT VIF0907 module (VIF) is used to interface the QE20's Speech and WIP audio signals into the ATA, as well as set up and monitor the call functions of the ATA.

A third-party Ethernet Switch and fibre optic converter (Moxa EDS-405A) sends the IP traffic over the fibre ring. It monitors the ring for a break, and when one is detected it automatically sends the packets in both directions, so all other nodes receive the information. The ring break signal is output by a fault relay in the switch, wired to the PIB so that the QE20 can generate a fault condition.

Each QE20 panel must include an RS485 Network Module to control networking operation and provide the data and audio interface points for the Fibre Network Kit.

This general arrangement is shown in Figure 1. The highlighted items are included in the QE20 Fibre Networking Kit.

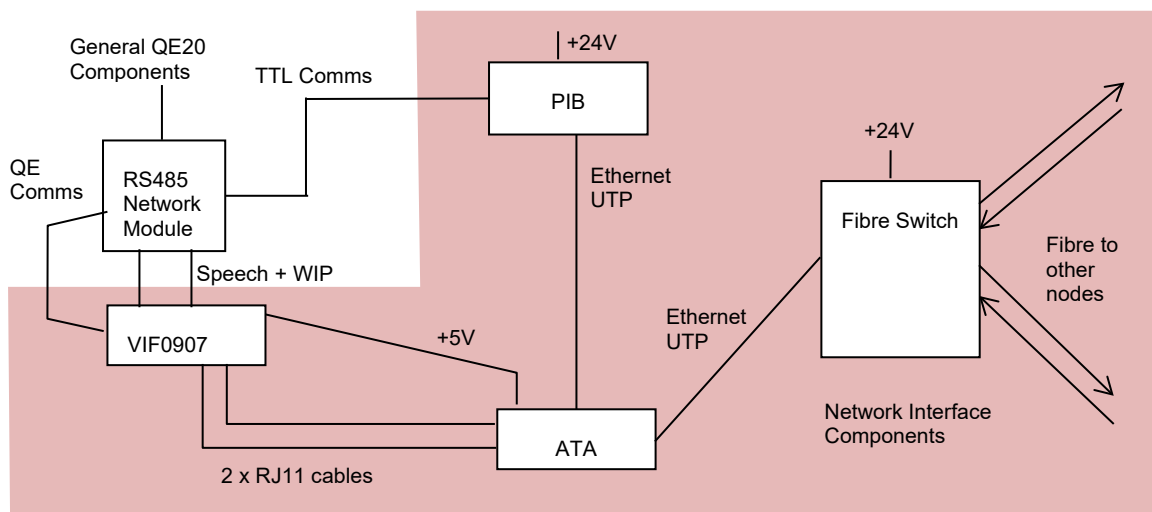


Figure 1 –General Arrangement of QE20 Fibre Networking Kit

The fibre network can be used to extend QE20 networking over long distances or between locations where it is not economic or practical to install the multiple cable pairs needed for a QE20 RS485 copper network (plus extra pairs for background music and/or paging console audio when required).

The fibre network can also be used to distribute paging console audio and/or background music. Additional VIF modules, ATAs, and PIBs may be required.

For the QE20 two different Fibre Network Kits are available – depending on the type of fibre used.

- FP2024 FP, QE20 FIBRE NETWORK KIT, SINGLE MODE
- FP2025 FP, QE20 FIBRE NETWORK KIT, MULTI MODE

Typically, multi-mode fibre will allow up to ~3km between QE20 panels, whereas single mode fibre will extend to ~40km between panels.

Generally, the whole ring will be run in the same fibre type, but it is possible to use different types of fibre. This will require additional equipment at those QE20 panels that need to support both types of fibre – at least an additional fibre switch will be needed.

The fibre network kits are sufficient for voice communication with up to 31 other QE20 panels. If the QE20 needs to simultaneously send Speech and/or WIP All-Call to more than 16 other QE20 panels, then additional PIBs and other equipment will need to be added. Formulae for determining the quantities required are described in this manual and implemented in QE20Cost.

If background music and or paging audio must be transferred around the network, then additional equipment will be required. For each pair of additional audio signals a VIF and ATA is required, plus an additional PIB will be needed for every 31 remote nodes receiving the audio.

The QE20 fibre network cannot be used to increase the existing limits of a QE20 network or a Panel-Link network regarding the maximum number of nodes supported by a QE20. But multiple QE20 networks could be contained on one large fibre network.

Programming of each QE20 site configuration for fibre networking is mostly the same as RS485 networking. However, configuration is required for the ATA dialling plans to enable VoIP calls to be set up around the network. Additionally, the PIB, ATA, Fibre Switch and VIF will require configuration. This is described in this manual.

A Fibre-Copper Bridge QE20 allows the connection of an RS485 copper network onto a fibre ring. This could be used to convert the main fibre network into a local building RS485 copper network for wiring to other QE20 panels within the building. The bridge QE20 includes a second RS485 Network Module to provide the PA and WIP Audio signals for the copper network (the first Network Module handles the main fibre network signals).

The Fibre-Copper Bridge QE20 provides audio bus joining so that it may hold separate Speech and WIP conversations with the QE20s on its copper segment, while other conversations are taking place on the fibre network.

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2. Operation of Fibre Network

Data Communication

When a QE20 panel sends data it may be sent as a packet to the specific QE20 (SID number) that needs to receive it, or as a broadcast to all QE20s. With fibre networking each message is sent by the RS485 Network Module to the PIB, which looks up a table to identify the IP address of the destination QE20 SID and then sends the message to the PIB at that IP address using UDP. The receiving PIB passes the message on to the QE20 Network Module, which forwards it to the Controller to process. For broadcast messages, the PIB will send the message to each PIB that it knows about (unless filtering is programmed).

The PIB can have a pre-programmed table of SID numbers and IP addresses, or it can automatically build this table from messages it receives from the QE20 panels and PIBs on the network.

The Moxa fibre switch takes care of sending the UDP and VoIP packets around the fibre ring, especially when the ring is broken.

Audio Streaming – PIB VoIP Proxy

When Speech and WIP ALL-CALL is sent on the network, it quite often needs to be sent to many places at once. For example, a Speech announcement to the whole network would require the speech audio to be sent to every other node. This presents quite a burden on the sending device, and as the ATAs do not support streaming of audio to more than 4 devices, something else needs to do this.

The PIB can be configured to stream audio to up to 31 other nodes. This function is called PIB VoIP Proxy in the PIB. Each PIB can have up to four streaming channels configured. When it receives a streaming request on a channel it will attempt to make a VoIP call to a pre-programmed IP address to get the source VoIP data stream (usually the ATA in the same panel). Any data the PIB receives is then streamed to all the remote devices that have connected to the channel. This way each PIB can fan out the streaming audio to 31 other nodes. By having multiple PIBs connect to the original source ATA, or daisy-chain from one to another, the single audio source can be fanned out to a virtually infinite number of destinations.

Speech Audio Communication

The following is a simplified description of how Speech audio is distributed in a QE20 fibre network. In the event that something is not working, it may be useful to track the actual operation against the description below.

2.1.1 Speech

When a Speech announcement is initiated from a QE20 that involves distribution of the speech to other QE20 panels on the fibre network the following happens in sequence: –

1. The user at the QE20 selects one or more zones for Speech. This panel is hereafter referred to as panel A.

2. Panel A transmits which zones are selected for speech to the other QE20 panels via the PIB.
3. Each other QE20 maps panel A's zones to its own zones, and if any do map to local zones, it waits for the PTT signal to become active, then instructs its VIF module to dial the phone number configured to reach panel A.
4. The ATA uses the dialled phone number and its configured dial plan to determine the IP address and UDP port number for the VoIP proxy (PIB) or ATA at panel A.
5. The ATA at each such panel makes the call to the VoIP proxy or ATA at panel A.
6. On receiving the first incoming call the VoIP proxy then calls the ATA at panel A.
7. The ATA at panel A signals the VIF it has an incoming call by ringing, and the VIF answers the VoIP call.
8. The ATA at panel A transmits digitised audio to (1) the VoIP proxy which will then distribute this to each of the ATAs at the other panels that have joined the same call or (2) the ATA at the other panel.
9. The operator makes the speech announcements to the occupants of the zones selected.
10. The operator at Panel A releases the PTT button or deselects zones for speech.
11. This is transmitted across the data network to the other QE20 panels.
12. The other panels see that speech is no longer required for their zones and so terminate their call to the ATA or VoIP proxy at panel A, which will then terminate the call it made to the ATA at panel A once all remote nodes have disconnected.

WIP Audio Communication

2.1.2 WIP Call initiated by User Interface

When a WIP call to a field WIP at a remote panel is initiated on a QE20, the following happens in sequence: –

1. The operator at the QE20 initiates the call by pressing the WIP call button. This panel is hereafter referred to as panel A.
2. Panel A transmits the pressed WIP button number to the other QE20 panels (via the PIB and fibre networking).
3. The other QE20s determine if the button maps to one of its WIP buttons and, if so, sets the button to the ring state. If that button maps to a field WIP then the WIP module rings that WIP circuit.
4. When the field WIP is answered (off-hook), the remote panel informs Panel A of the fact via the data network.
5. Panel A sees the called WIP is off-hook and instructs its VIF module to dial the phone number configured to reach the remote panel.
6. The ATA at Panel A uses its configured dial plans (and most likely speed-dial) settings to find the IP address of the ATA at the remote panel.
7. The ATA at Panel A makes a VoIP call to the ATA at the remote panel.
8. The ATA at the remote panel signals the incoming call by ringing the VIF, which answers the call.
9. The ATAs at panel A and the remote panel transmit digitised audio between each other (both ways) using RTP.

10. The two parties talk to each other.
11. The operator at Panel A disconnects the call by pressing the WIP call button again.
12. Panel A transmits the call-on-hold condition to the remote panel via the PIB and networking.
13. Panel A terminates the VoIP call.

Note: if the call is terminated by the remote party hanging up their WIP phone, refer to the final steps of the next sequence.

2.1.3 WIP Call Initiated by Field WIP

When a field WIP is picked up, the following happens in sequence (assuming the call is answered at a remote QE20): –

1. A field WIP is picked up. The QE20 (panel or amprack) this WIP connects to is hereafter referred to as panel A.
2. Panel A maps the field WIP to its local button and sets this flashing.
3. Panel A transmits the WIP button to the other QE20 panels (via the PIB and IP networking).
4. The other QE20s map panel A's WIP button to their WIP buttons, and if the calling WIP button maps to a local WIP button rings the master WIP and flashes the appropriate WIP LED.
5. The operator at the QE20 panel answers the WIP call by pressing the flashing call button. This panel is hereafter referred to as panel B.
6. Panel B sets this WIP button to the connected condition.
7. Panel B then transmits the "connected" condition for the WIP button to panel A via the PIB and fibre networking.
8. Panel A maps this WIP button to its local WIP button and sets it to the connected condition, along with the field WIP.
9. Panel B instructs its VIF module to dial the phone number configured to reach panel A.
10. The ATA at panel B uses its configured dial plans (and most likely speed-dial) settings to find the IP address of the ATA at panel A.
11. The ATA at panel B makes a VoIP call to the ATA at panel A.
12. The VIF at panel A sees the incoming ringing and answers the VoIP call.
13. The ATAs at panel A and panel B transmit digitised audio between each other (both ways) so the two parties can talk to each other.
14. The WIP connected to Panel A is hung up.
15. Panel A transmits the WIP button idle condition to panel B (via the PIB and IP networking).
16. Panel B receives the idle condition for the WIP button, extinguishes the WIP LED, and terminates the VoIP call.
17. The ATAs hang up.

Note: if the call is terminated by a user at the QE20, refer to the final steps of the previous sequence.

2.1.4 WIP All-Call

This works in a similar way to Speech – refer above. When a remote QE20 receives an All-Call from another QE20 and it is configured to accept this All Call, it will instruct its VIF module to ring the phone number configured for All Call at the calling

SID. It does not wait for any WIPs to be answered. The ATA maps the dialled number to the IP address and port of the VoIP Proxy or ATA at the calling panel and makes the call.

2.1.5 Inter-QE20 WIP Call

This works in a similar way to the User Interface-initiated call described above. There is no on-hold condition with inter-QE20 ECP calls. Note it is not possible to hold simultaneous calls with the ECP WIP and field WIPS at the same QE20.

Additional Audio Signal Communication

2.1.6 Background Music

Panels that receive background music over the fibre network will make a (usually permanent) VoIP call to the panel that distributes the music. If the source of the music can change (e.g., it is controlled by a paging console), then the VoIP calls made will depend on the current music source.

2.1.7 Paging Console Audio

Panels that receive paging announcements over the fibre network will make a call to the panel that sources the paging console audio, i.e., the panel which connects to the paging console(s).

If there is more than one panel with a paging console, then the VoIP calls will be made to the panel with the active paging console. Otherwise, VoIP calls can be made always to a fixed panel. In both cases, the QE20 script should be programmed to make the VoIP calls when required, i.e., when paging is in use.

Network Data Requirements

Please refer to the information in the PIB User Manual (LT0519).

In addition to the Ethernet traffic requirements listed in LT0519, you must allow for the VoIP traffic. This is approximately 80 kbps for each possible simultaneous connection. For a standard call this data is transmitted both ways, while for a streamed connection (e.g., Speech or WIP All-Call), data is one-way from the source to each destination. For example, if a panel can be sending speech to 6 other nodes and a (two-way) WIP call could be in progress at the same time, the total traffic would be $6 \times 80 \text{ kbps} + 2 \times 80 \text{ kbps}$ or approximately 640 kbps.

If additional audio signals are transported over the network then the data requirements of these will need to be added in.

Ports / Protocols Used by the ATA and PIB

2.1.8 ARP

ARP is required to map IP addresses to MAC addresses.

2.1.9 ICMP

ICMP is supported to the extent that replies are sent to PING requests.

2.1.10 TCP

TCP Port 80 provides a web server that is used for configuration and diagnostic monitoring of the ATA, PIB and fibre switch. Access to these web servers is highly recommended for service personnel, as many diagnostic details are provided by only these web servers.

2.1.11 SIP

SIP over UDP is used by the ATA and PIB VoIP proxy to set up and take down calls. By default, the ATA uses UDP ports 5060 and 5061. The PIB VoIP proxy uses UDP ports 5060 to 5063 by default.

An ATA that wants to initiate a call to another ATA or PIB VoIP proxy will send a SIP INVITE message. The remote VoIP device may then send a TRYING / RINGING response, and if the call is answered by the remote VIF, will then send an "OK" response. The two VoIP devices will then send audio data each way using RTP. At the end of the call, one of the VoIP devices will send a BYE message.

2.1.12 RTP

RTP over UDP is used by the VoIP devices to transfer digitised voice data between QE20s. The ATA uses UDP ports 16384 to 16482 by default. The PIB VoIP proxy uses UDP ports in the range 16000 to 19998 inclusive.

Router / Firewall / VLAN Configuration

Dedicated fibre networking should be used for the QE20 network.

If this is not the case, then the evacuation network must be configured into a separate VLAN so its traffic is isolated from that of other users of the network. Refer to PIB User Manual (LT0519) for further information on this.

Routers/Firewalls within the IP network must be configured not to block any addressed (i.e. non-broadcast) TCP, UDP, SIP, and RTP packets to / from the PIBs and ATAs on the network. Refer to section 2.7 for the protocols / ports used by the ATA and PIB for VoIP communication, and to the PIB User Manual (LT0519) for the protocols / ports used by the PIB for Panel-Link networking.

Note, SIP sometimes does not work well with systems employing Network Address Translation (NAT) because the ATA embeds its IP address in the message it sends to the other connecting ATA. If NAT is employed, special programming of the SIP ATAs, PIB, and the router providing NAT will most likely be needed.

3. Design & Configuration

This chapter describes the design and configuration normally done when a QE20 system with Fibre Networking is ordered. It is not intended that the installer does this. The factory will prepare the information, configure any equipment being ordered, and include a documentation pack with each fibre-networked QE20 that will contain the information necessary to configure any modules that might need to be replaced or upgraded in the field.

When upgrading or adding to existing fibre networks, it is possible that the installer will need to alter the configuration of already installed equipment. Details of what needs to be changed will generally be provided as part of the upgrade package from the factory.

CALCULATION OF PIB, VIF & ATA QUANTITIES

QE20Cost includes a number of questions that determine the quantities of equipment required for fibre networking at a QE20 panel.

The screenshot shows a configuration window for 'Networking'. At the top, there are two dropdown menus: 'Networking:' set to 'None' and 'Number of Nodes:' set to 'Range [1, ..., 5]'. To the right of the second dropdown is a checkbox labeled 'In NonStock'. Below these are four rows of configuration options, each with a checkbox and a quantity box (a yellow box with up/down arrows):

- Checkbox: Receives PA from other fibre connected panels to amplifiers at this panel. Quantity box: 0. Label: Number of fibre connected panels to which this panel transmits PA.
- Checkbox: Supports WIP Calls between this panel and other fibre connected panels. Quantity box: 0. Label: Number of fibre connected panels to which this panel transmits WIP All Call.
- Checkbox: Receives Paging Console audio from other fibre connected panels to amplifiers at this panel. Quantity box: 0. Label: Number of fibre connected panels to which this panel transmits Paging Console audio.
- Checkbox: Receives Music audio from other fibre connected panels to amplifiers at this panel. Quantity box: 0. Label: Number of fibre connected panels to which this panel transmits music audio.

At the bottom, there are two more checkboxes:

- This panel is a bridge between the fibre network and the RS485 network.
- Dedicated network - factory to assign IP addresses.

In particular the four “Receives / Supports” tick boxes and the four “Number of fibre-connected” quantity boxes are critical.

These are described as follows.

If:

“Receives PA from other fibre connected panels to amplifiers at this panel” is ticked, or

“Number of fibre connected panels to which this panel transmits PA” is non-zero;

Then a portion of a VIF and ATA is needed for Speech (PA) and the number of panels to which speech is transmitted counts towards the number of PIBs needed.

If:

“Supports WIP Calls between this panel and other fibre connected panels” is ticked, or

“Number of fibre connected panels to which this panel transmits WIP All-Call” is non-zero;

Then a portion of a VIF and ATA is needed for WIP calls and the number of panels to which All-Call is transmitted counts towards the number of PIBs needed.

If:

“Receives Paging Console audio from other fibre connected panels to amplifiers at this panel” is ticked, or

“Number of fibre connected panels to which this panel transmits Paging Console audio” is non-zero;

Then a portion of VIF and ATA is needed for paging console audio and the number of panels to which paging console audio is transmitted counts towards the number of PIBs needed.

If:

“Receives Music from other fibre connected panels to amplifiers at this panel” is ticked, or

“Number of fibre connected panels to which this panel transmits Music audio” is non-zero;

Then a portion of a VIF and ATA is needed for music and the number of panels to which music is transmitted counts towards the number of PIBs needed.

The Fibre Networking Kit provides the first VIF, ATA, and PIB. If the number of portions of VIF & ATA is greater than 1, then a second VIF and ATA will be needed.

For every 31 nodes above the first 31 that can have Speech, All-Call, Music and Paging audio sent to, add an extra PIB. If the number of PIBs exceeds 3, a 5-port Ethernet switch will need to be added.

VoIP Design Procedure

This section outlines the general procedure for designing the VoIP system.

Allocate a 2-digit SID to each QE20 panel (20 – 89 is suggested to leave some SID numbers for fire panels on the same network). Allocate IP addresses to each of the fibre networking components (refer section 3.3).

For each QE20, determine how many other panels will need to make Speech and WIP VoIP calls to this panel at the same time. Refer to section 3.4.4.

For each panel, if it needs to receive or send BGM or paging audio over the fibre network, determine how many other panels will need to receive background music and paging audio from this panel at the same time. Refer to sections 3.4.3 and 3.4.4.

This will determine the number of Speech (PA) lines and WIP lines required and from that the number of VIF modules, ATAs, and PIB VoIP Proxy modules required.

Then, design the dialling schemes so that the panel being configured can call each of the other panels it needs to call. The recommended principle is that the QE20 should transmit a 2-digit ‘phone number’ to the ATA, and the ATA convert this phone number to a URI that will address the required line of the ATA or PIB VoIP proxy at each remote QE20.

See the “Recommended Dial Plans” in section 3.5.2. For receiving All-Call use dial numbers 90-99 and enter specific mapping to the panels that need to send All-Call audio here.

Assignment of IP Addresses

Each PIB, ATA and fibre switch must be configured with a unique static IP address. Generally, on a standalone network, it is convenient to use private addresses in the 10.0.0.0 - 10.255.255.255 address range.

The suggested addressing scheme (where xx is the SID of the QE20) is:

10.0.0.xx for the Moxa.

10.0.1.xx for the first PIB.

10.0.2.xx for the second PIB.

10.0.3.xx for the third PIB.

etc

10.0.8.xx for the first ATA at that location.

10.0.9.xx for the second ATA at that location, etc.

To make network configuration and fault finding easier, the following table should be copied and filled in with the IP addresses for the various SIDs and devices on the network.

SID #	1				
Location	Example				
Moxa IP Address	10.0.0.1				
PIB #1 IP Address	10.0.1.1				
PIB #2 IP Address	10.0.2.1				
PIB #3 Address	10.0.3.1				
ATA #1 IP Address	10.0.8.1				
ATA #2 IP Address	10.0.9.1				
ATA #3 IP Address	10.0.10.1				

SID #					
Location					
Moxa IP Address					
PIB #1 IP Address					
PIB #2 IP Address					
PIB #3 Address					
ATA #1 IP Address					
ATA #2 IP Address					
ATA #3 IP Address					

SID #					
Location					
Moxa IP Address					
PIB #1 IP Address					
PIB #2 IP Address					
PIB #3 Address					
ATA #1 IP Address					
ATA #2 IP Address					
ATA #3 IP Address					

Configuration of the QE20

3.1.1 Speech Call Origination

For Speech, each remote QE20 containing amplifiers for the selected zones will dial the QE20 making the speech announcement.

If there is more than 1 remote location that can be receiving speech at the same time, at least one PIB VoIP proxy must be configured at this QE20. A PIB VoIP proxy will be required for every 31 simultaneous callers.

3.1.2 WIP Call Origination

For WIP calls to/from a field WIP, the VoIP call is made from the QE20 to the location where the field WIP is terminated.

For inter-ECP WIP calls, the VoIP call is made from the location that connects the call. VoIP calls are not made until the WIP call is connected.

For WIP All-Call (if it is configured), VoIP calls are made from the location(s) where field WIPs are terminated to the QE20 that initiates All-Call (usually to a VoIP Proxy function on the PIB). These VoIP calls are made as soon as All-Call is received (if accepted), regardless of whether any phones have been picked up.

3.1.3 Paging Console Audio / Background Music Call Origination

Calls for paging console audio and background music are controlled by the QE20 scripting. Calls are made from the panels with amplifiers to the panel which generates the audio. Note when there are multiple panels that could generate the paging audio, a call will need to be made to the relevant panel when that source is active.

Refer to Section 3.6 for an example.

3.1.4 Use of Multiple VIF, ATA, and PIBs

If there are multiple remote nodes requiring Speech from a given panel, or WIP All-Call is implemented, or the panel connects to a paging console or background music source, or the panel receives paging console audio or background music from a remote panel, then multiple VIF modules, ATAs, and PIBs may be required.

To calculate the number of VIF modules, ATAs, and PIBs required, the easiest way is to use the QE20 pricing program QE20COST. QE20COST will: –

- Calculate the quantities of VIF modules, ATAs and PIBs required.
- Check that there is enough room in the chosen cabinet.
- Calculate the battery and power supply requirements.
- Allow for additional modules that may be required for distributing Paging Console audio and/or background music.
- Calculate the price of a complete panel with all fibre networking equipment pre-installed and tested.

Note that QE20COST uses the following rules to determine the number of VIFs, ATAs, and PIBs, and to design their interconnections: –

- An ATA line can be used for two-way connections between two panels, receiving audio broadcast by another panel, and to act as an audio source to

one PIB VoIP proxy that re-broadcasts the audio to up to 31 other panels. The use of the line as unidirectional or bi-directional is changed “on the fly” through software negotiation when calls are set up.

- When the QE20 needs to dial out for a Speech call it uses the PA channel of the first VIF, and when it needs to dial out for a WIP call it uses the WIP channel of the first VIF.
- Paging console audio and background music may use any channels (PA / WIP) on additional VIFs.
- A single PIB VoIP proxy can distribute audio to 31 other panels, but this capacity is shared across all the active audio channels (up to 4) the proxy connects to. For large systems with multiple audio sources being broadcast multiple PIBs may be required. The PIB used for Panel-Link networking can also do the VoIP proxy function.
- A VIF can power at most two ATAs, and it will be arranged that a VIF does not power any ATAs that it is not otherwise associated with.

If you do not have access to QE20COST, it should be assumed half a VIF will be required for each of the WIP, Speech, PABX, or background music functions. Similarly, half an ATA will be required for each of these functions. One PIB is required for 31 simultaneous callers. If multiple PIBs are needed, it should be calculated on the basis that audio sources aren’t split across PIBs.

3.1.5 Example Arrangements

See Figure 2, Figure 3, and Figure 4 for some example arrangements, and drawing 699-295 for generic wiring. Note the PIB for Panel-Link networking is not shown.

Figure 2 shows an example where one line is required for Speech (PA) and one line for WIP. The Speech line can be used for incoming Speech to amplifiers at this node and/or sending Speech to only one remote node. The WIP line can be used for 1 to 1 calls, receiving All-Call, or making All-Call to only one remote node. Speech and All-Call cannot be made to multiple nodes.

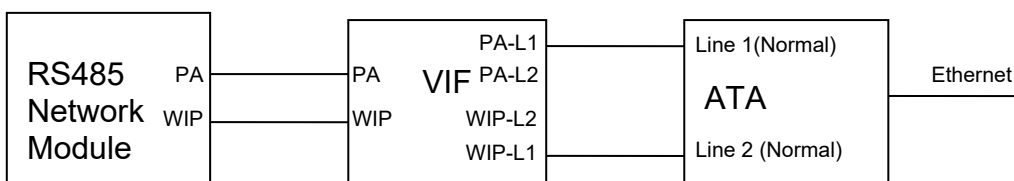


Figure 2 – 1 Line PA + 1 Line WIP

Figure 3 adds the PIB for the VoIP Proxy function so this QE20 can also send speech and / or WIP All-Call to multiple nodes.

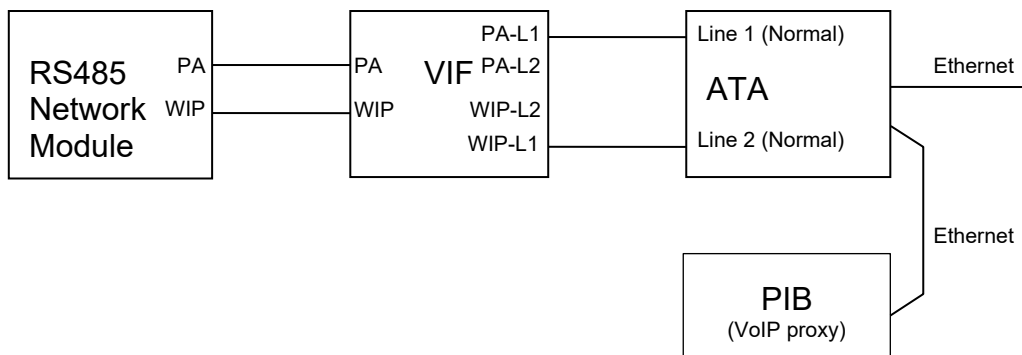


Figure 3 – 1 Line PA + 1 Line WIP + PIB VoIP Proxy

Figure 4 shows an expansion to Figure 3 so that BGM (music) audio and / or paging console audio can be sent to or received from other QE20s.

A second VIF and ATA is added to each QE20, and the PIB configured for additional channels of VoIP Proxy functions. Each VIF + ATA pair provides two channels of audio that can be sent or received over the network.

The QE20 panel is configured with Script commands to control VIF #1 (and so the ATA #2) so that calls can be made or received.

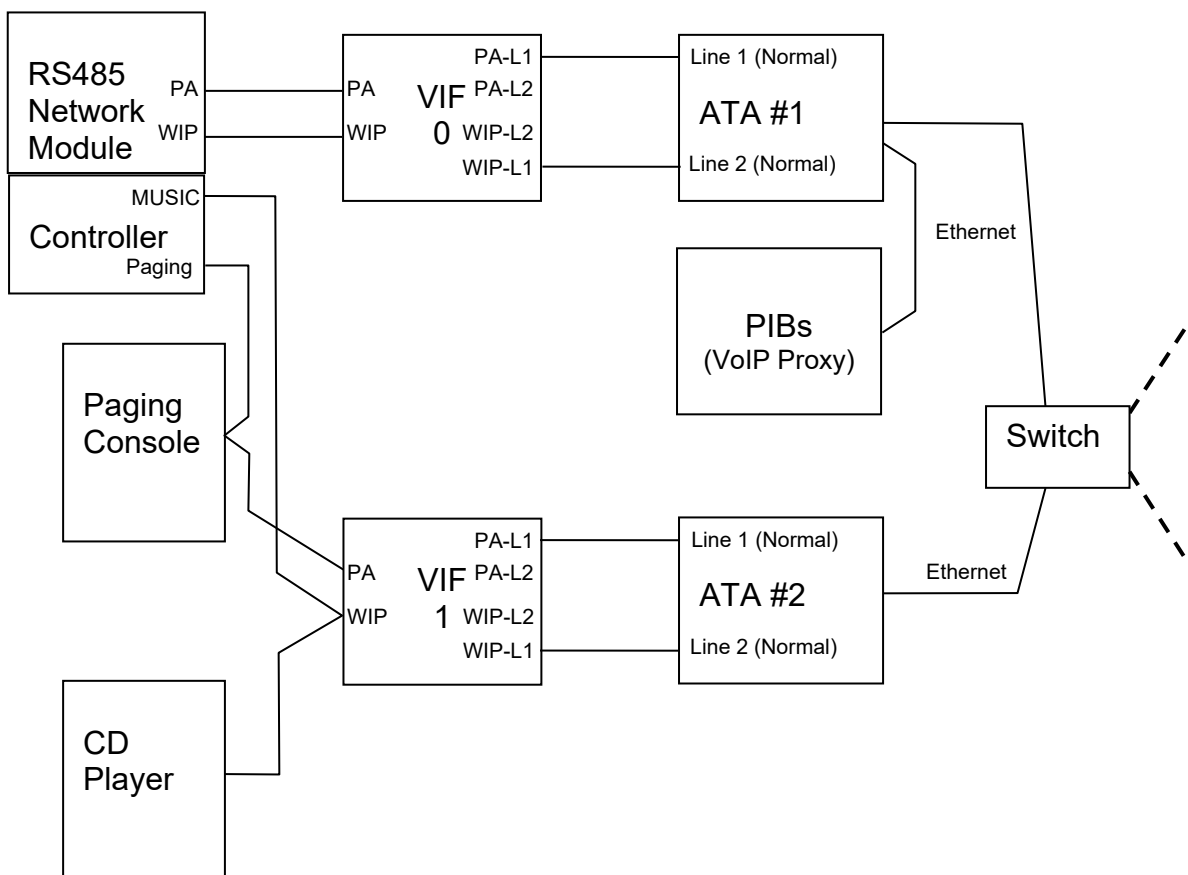


Figure 4 – PA In / Out, WIP In / Out including All-Call, Paging Console out, and Music out (using PIB VoIP Proxy)

3.1.6 Configuration of VoIP Dial Numbers

In QE20Config the QE20 must be configured with the phone numbers for each remote node that must be reached by VoIP. This 2-digit number is “dialed” by the VIF module whenever it is required to set up a VoIP call to the remote node. Separate numbers can be entered for Speech, WIP 1:1 calls, and WIP All-Call. The **same** number may be used for Speech (PA) and WIP - however, in the ATA configuration, PA lines can be set up to call only PA lines and WIP lines are set up to call only WIP lines, by using for example line 1 at port 5060 for PA and line 2 at port 5061 for WIP.

When WIP All-Call is to be received from another QE20, a different number - that of the WIP VoIP PIB proxy – needs to be dialed. Use the “VoIP WIP All-Call Dial Number” for this purpose.

The screenshot in Figure 5 shows an example set up of “VoIP Dial Numbers”. This QE20 needs to receive WIP All-Call from SIDs 21 and 23. Specific dial numbers (91 and 92) have been entered to reach the PIB VoIP Proxy at the remote QE20s for this.

Refer to Section 3.5.1 for how to design the dial plan – the conversion of the dialled number to IP address and port number.

File Edit Select Profile View Check Window Licence Tools Help

SID Number	QE90	Panel ID	Panel Description	Audio Segments to reach	Accept All Call	Accept Silence Buzzer	Accept Reset	Accept Silence Warning	Accept Remote Faults	VoIP Dial Number	VoIP WIP Dial Number	VoIP WIP All-Call Dial Number
21	<input type="checkbox"/>			1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	21	21	91
22	<input type="checkbox"/>			1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	22	22	
23	<input type="checkbox"/>			1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	23	23	92

Figure 5 – Example QE20 Remote SIDs Configuration

3.1.7 Configuration for “Idle Lines”

The “Idle Line” is the line on which the VIF listens for an incoming call when it is idle. The VIF will always call out on Line 1 when it needs to originate a call.

In general, both the PA and WIP Idle Lines should be sent to “Line 1” in QE20Config.

3.1.8 Configuration for Paging Console Audio and Music Distribution

It is not possible to change the idle state of the 2nd lines of the VIF, as this forces the line relay to be switched to the 2nd line. This is not a problem as systems utilising the VoIP PIB Proxy function shouldn’t require this 2nd line. The VIF, VoIP PIB proxy, and ATA lines can be used to distribute paging console audio and music over fibre, in a very similar way to emergency PA and WIP audio. This is controlled by scripting in the QE20 configuration. Refer to Section 3.6 for an example.

3.1.9 Configuration for a Fibre-Copper Bridge

In a mixed copper / Fibre network there needs to be one QE20 with both a Fibre interface and an RS485 Copper interface for each copper segment. This QE20 is called a Fibre-Copper Bridge and will have 2 Network Modules fitted.

- Panels on the copper segment that don’t have the fibre interface will send a message to the Fibre-Copper Bridge QE20 when a call is required on the fibre

network on their behalf. These panels need to have the SID number of the Fibre-Copper Bridge programmed into their “VoIP Proxy SID” field on the Network Page (note: VoIP Proxy in this context does not refer to the functionality present within the PIB). Additionally, the “Number of VIF Cards” must be set to 0, and “PA Idle Line” and “WIP Idle Line” must be set to “VoIP not used”.

- At all panels, the “dial numbers” entered in the Remote SIDs table for SIDs on copper segments must be the dial numbers required to reach the Fibre-Copper Bridge for that copper segment.
- Each copper segment needs to be given a unique Audio Bus Segment (number) so that the audio segments needed to reach QE20 panels on each copper segment can be identified. For example, SID 80 is the Fibre Copper Bridge, the main fibre ring is Audio Bus Segment 1, and it provides a RS485 + audio copper network – Audio Segment 3. SID 81 is connected to SID 80 by copper cables, so is assigned the matching Audio Bus Segment number.

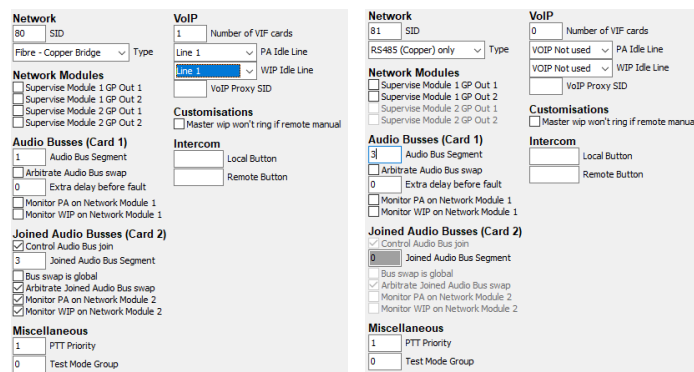


Figure 6 – Network Configs for QE20 Fibre-Copper Bridge and Copper-Connected QE20

Dial Plans

3.1.10 ATA Dial Plans

The ATA must be configured to translate the phone numbers dialled by the QE20 and VIF into IP addresses and destination ports. This could be done using a “Dial Plan” that:

- allows a single digit to be dialled and maps this to a “speed dial” for each required digit (not recommended as it is slow), or
- generates an IP address with part of the IP address incorporating the dialled number (recommended).

Some examples are given here in sufficient detail to allow them to be amended for particular networks, but a full description of ATA dial plans is beyond the scope of this manual. For more information refer to the specific ATA Administration Guide.

3.1.11 Using a Dial Plan to Generate an IP Address from a Phone Number

For information on creating dial plans refer to the specific ATA Administration Guide.

The methods in these examples use <input:output> blocks in the dial plan, where input represents the number dialled, and output represents the resulting digits forming the IP address:Port number.

Simple Example

This simple dial plan will convert the dialled digit (2 – 4) to a URL.

Dialled Digit	URL
2	2@10.1.1.20:5061
3	2@10.1.3.21:5061
4	2@10.1.99.22:5061

(<2:2@10.1.1.20:5061>|<3:2@10.1.3.21:5061>|<4:2@10.1.99.22:5061>)

In each < > part, the digit(s) before the first : are the dialled digit(s) and the characters after the first : are the resulting URL. The URL has a format of line_number@ip_address_of_destination_ATA:UDP_port_number of the line number at that ATA or PIB. Multiple options are separated by the | character, and all are included in the (). Note the line number needs to match the User ID assigned into the line in the ATA, or the channel number of the PIB VoIP proxy.

Dial By SID Example

A dial plan of:

```
(<:2@10.0.1.>xx<:5061>)
```

Will produce an “output” of 2@10.0.1., then the two digits dialled xx, then :5061.

So, the plan will generate, for example, a destination address of 2@10.0.1.89:5061 when the 2-digit number 89 is dialled. This can be used for “dial by SID” (where the last part of the IP address is the QE20 SID number) and is useful when a QE20 must be able to connect (one at a time) to many other QE20s.

In the QE20 configuration, the VoIP dial number for each remote panel should be set to the SID number of the remote panel. Note: all dial numbers need to be two digits, so 01 - 09 needs to be entered for SIDs 1-9.

Random IP address dial plan example

The dial plan:

```
(<90:2@10.0.1.21:5061> | <91:2@10.0.1.50:5061>)
```

Can be used to convert specific two-digit numbers to a URL.

Number Dialled	Number generated (URL)
90	2@10.0.1.21:5061
91	2@10.0.1.50:5061

This example can be used as the basis for creating an arbitrary table of URLs based on the number dialled (which may not be the target panel SID number). This will typically be used for All-Call VoIP Proxy addresses.

Mixed Dial by SID and Random IP addresses dial plan example

A dial plan of:

```
(<:2@10.0.1.>[0-8]x<:5061>|<90:2@10.0.0.21:5061>|<91:2@10.0.0.50:5061>)
```

Will produce a destination address of 2@10.0.1.xx:5061 when the 2-digit number 00 to 89 (xx) is dialed, and specific URLs when 90 and 91 are dialed. Other two-digit numbers are not accepted.

Recommend Dial Plans

The recommended dial plan for Line 1 (Speech) is

```
(<:1@10.0.1.>[0-8]x<::5060>)
```

So that the PIB VoIP Proxy host on the first PIB is called for speech from SIDs 00-89, assuming the suggested IP addressing scheme in Section 3.3 is used.

The general recommended dial plan for Line 2 (WIP) is

```
(<:2@10.0.8.>[0-8]x<::5061> | <90:2@10.0.1.21:5061> | <91:2@10.0.1.50:5061>)
```

This will dial the remote QE20 first ATA WIP line directly for numbers 00 to 89 for 1:1 WIP calls. Specific URLs need to be added for each SID that can send WIP All Call to this QE20 (change the 90 and 91 combinations to the required dial numbers and PIB IP address:Port combinations and add additional options if other QE20s can send WIP All Call here).

Design Example

Figure 7 shows an example system requiring many of these features.

SID 50 is the MECP and a Fibre-Copper Bridge, effectively joining the two remote “amp racks” on the fibre ring (SIDs 21 and 22) to the copper connected SECP at SID 20.

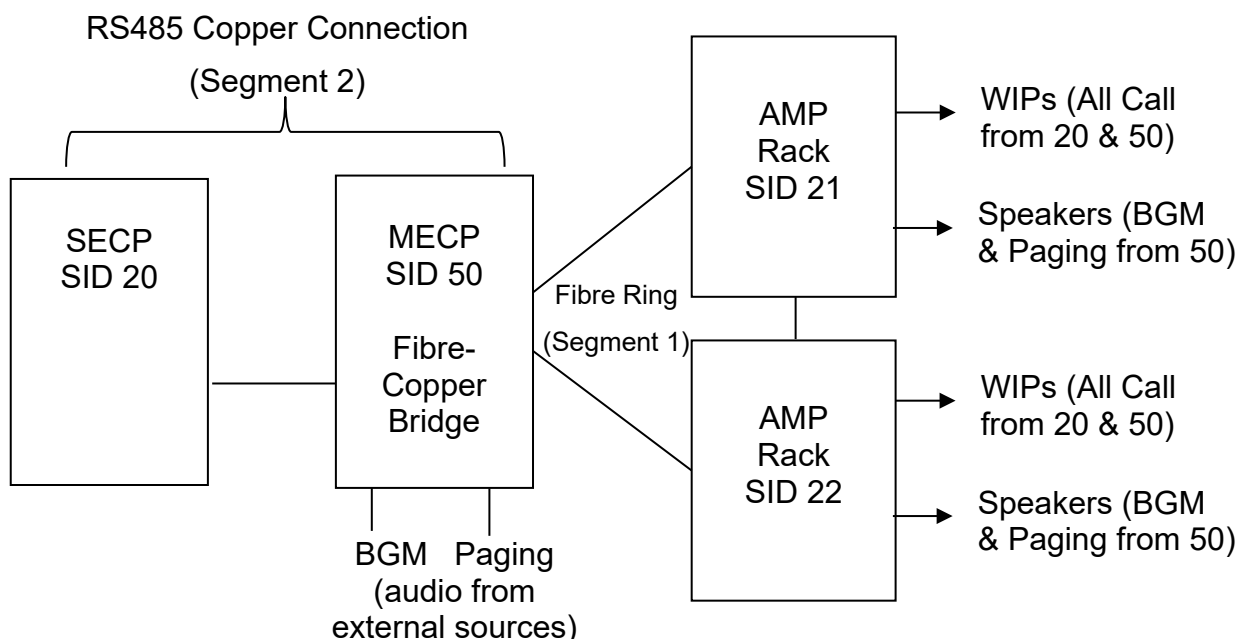


Figure 7 – Example Network

Speech (PA) needs to be sent from the MECP and SECP, SIDs 50 and 20, to the amp racks, SIDs 21 and 22.

WIP All Call functions are enabled at SID 20 and 50, which need to be received on WIPs at SIDs 20, 50, 21 and 22.

The MECF, SID 50, also receives BGM and Paging audio that needs to be sent over the fibre network to the amp racks, SIDs 21 and 22. A 2nd VIF and ATA combination will be required at each of these QE20s to send and receive this audio.

The fibre ring is allocated an Audio Bus Segment number 1 and the copper connection is allocated segment 2.

This table shows the questions and answers required by QE20Cost so that the correct panel designs can be done and the quantity of VIFs, ATAs and PIBs calculated.

QE20 Cost Question / SID Number	20	50	21	22
Network Type	RS485	Fibre	Fibre	Fibre
Number of Nodes	3	3	3	3
This panel is a fibre-copper bridge		Y		
Receives PA from other fibre-connected			Y	Y
Number of fibre panels sends PA to		2		
Supports WIP Calls		Y	Y	Y
Number of fibre panels sends All-Call to		2		
Receives Paging Console Audio			Y	Y
Number of fibre panels sends paging to		2		
Receives Music Audio			Y	Y
Number of fibre panels sends music to		2		

The resulting quantities are.

Module Quantities / SID	20	50	21	22
Network Type	RS485	Fibre (and RS485)	Fibre	Fibre
Network Modules	1	2	1	1
PIB		1	1	1
VIF		2	2	2
ATA		2	2	2

IP addresses can then be assigned to the Moxa, PIB, and ATAs on the fibre-connected QE20s.

SID #	20	50	21	22
Location				
Moxa IP Address		10.0.0.50	10.0.0.21	10.0.0.22
PIB #1 IP Address		10.0.1.50	10.0.1.21	10.0.1.22
PIB #2 IP Address				

PIB #3 Address				
ATA #1 IP Address		10.0.8.50	10.0.8.21	10.0.8.22
ATA #2 IP Address		10.0.9.50	10.0.9.21	10.0.9.22
ATA #3 IP Address				

Dial plans need to be prepared for the first ATAs (the BGM/Paging ones will be covered later).

For Line 1 (Speech) all ATAs can use the default dial plan, as long as it is mapped to the IP address of the PIB at each SID and the PIB is set for VoIP Proxy on channel 1. VoIP calls will be made at only SIDs 21 and 22 (and both call only SID 50), as SID 50 does not receive Speech from these panels.

<:1@10.0.1.>[0-8]x<::5060>

For Line 2 (WIP) the dial plan can use the default (mapped to the IP address of the ATA), with the addition of a special number (90) at SIDs 21 and 22 to receive All-Call from the PIB VoIP Proxy at SID 50. It doesn't matter if this dial plan is also used at SID 50, as the number 90 will never be dialled there.

<:2@10.0.8.>[0-8]x<::5061> | <90:2@10.0.1.50:5061>

The PIB at SID 50 needs to be configured for the VoIP Proxy function on channels 1 (PA / Speech) and 2 (WIP All Call) (channels 3 & 4 will be covered later). The URIs need to map to the 1st ATA ports 1 and 2 at SID 50, and a codec of G711u is selected to match the ATA default.

VoIP Proxy			
Enter configuration for each VoIP proxy channel. The SIP port is that on which inc address><:sip port>. An empty SIP URI means that the channel is disabled. The de applications.			
Channel	SIP Port	Audio Source SIP URI	Codec
1	5060	1@10.0.8.50:5060	G.711 μ-Law ▼
2	5061	2@10.0.8.50:5061	G.711 μ-Law ▼
3			G.711 μ-Law ▼
4			G.711 μ-Law ▼

In QE20Config the Network and Remote SID programming for the MECP SID 50 is:

SID Number	QE90	Panel ID	Panel Description	Audio Segments to reach	Accept All Call	Accept Silence Buzzer	Accept Reset	Accept Silence Warning	Accept Remote Faults	VoIP Dial Number	VoIP WIP Dial Number	VoIP WIP All-Call Dial Number
20	<input type="checkbox"/>	SID 20	SECP	2	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>			
21	<input type="checkbox"/>	SID 21	Amp Rack	1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	21	21	
22	<input type="checkbox"/>	SID 22	Amp Rack	1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	22	22	

At the SECP, SID 20, the Network and Remote SID setups are as follows (note the assignment of SID 50 as the VoIP Proxy SID, the Audio Bus Segment as 2, and the ordering of the Audio Segments to reach SIDs 21 and 22).

SID Number	QE90	Panel ID	Panel Description	Audio Segments to reach	Accept All Call	Accept Silence Buzzer	Accept Reset	Accept Silence Warning	Accept Remote Faults	VoIP Dial Number	VoIP WIP Dial Number	VoIP WIP All-Call Dial Number
21	<input type="checkbox"/>	SID 21	Amp Rack	2, 1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	21	21	
22	<input type="checkbox"/>	SID 22	Amp Rack	2, 1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	22	22	
50	<input type="checkbox"/>	SID 50	MECP	2	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>			

At the amp rack, SID 21, the Network and Remote SID setups are as follows – note that to reach SID 20 for Speech and WIP the VoIP and WIP dial numbers for the Fibre-Copper Bridge, SID 50, are used:

Network

21 SID

Fibre only Type

Network Modules

Supervise Module 1 GP Out 1

Supervise Module 1 GP Out 2

Supervise Module 2 GP Out 1

Supervise Module 2 GP Out 2

Audio Busses (Card 1)

1 Audio Bus Segment

Arbitrate Audio Bus swap

0 Extra delay before fault

Monitor PA on Network Module 1

Monitor WIP on Network Module 1

Joined Audio Busses (Card 2)

Control Audio Bus join

0 Joined Audio Bus Segment

Bus swap is global

Arbitrate Joined Audio Bus swap

Monitor PA on Network Module 2

Monitor WIP on Network Module 2

Miscellaneous

1 PTT Priority

0 Test Mode Group

VoIP

2 Number of VIF cards

Line 1 PA Idle Line

Line 1 WIP Idle Line

VoIP Proxy SID

Customisations

Master wip won't ring if remote manual

Intercom

Local Button

Remote Button

SID Number	QE90	Panel ID	Panel Description	Audio Segments to reach	Accept All Call	Accept Silence Buzzer	Accept Reset	Accept Silence Warning	Accept Remote Faults	VoIP Dial Number	VoIP WIP Dial Number	VoIP WIP All-Call Dial Number
20	<input type="checkbox"/>	SID 20		1, 2	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	50	50	90
22	<input type="checkbox"/>	SID 22		1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	22	22	
50	<input type="checkbox"/>	SID 50		1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	50	50	90

The configuration for the other amplifier rack, SID 22, is similar.

BGM and Paging Configuration.

The PIB at the MECP, SID 50, needs to have VoIP Proxy enabled on channels 3 and 4, with the URI mapped to lines 1 and 2 of the second ATA at the MECP, using a G711u codec, so that BGM and paging audio can be streamed to the amp racks SIDs 21 and 22.

VoIP Proxy			
Enter configuration for each VoIP proxy channel. The SIP port is that on which incoming calls are received. The SIP address is that of the audio source. An empty SIP URI means that the channel is disabled. The default codec is G.711 μ-Law.			
Channel	SIP Port	Audio Source SIP URI	Codec
1	5060	1@10.0.8.50:5060	G.711 μ-Law ▼
2	5061	2@10.0.8.50:5061	G.711 μ-Law ▼
3	5062	1@10.0.9.50:5060	G.711 μ-Law ▼
4	5063	2@10.0.9.50:5061	G.711 μ-Law ▼

The 2nd ATAs at the amp racks, SIDs 21 and 22, need a dial plan to reach the PIB at the MECP, SID 50, on channels 3 and 4 and using the correct UDP port for BGM and Paging audio respectively.

For Line 1:

```
(<:3@10.0.1.>[0-8]x<::5062>)
```

For Line 2:

```
(<:4@10.0.1.>[0-8]x<::5063>)
```

The MECP SID 50 needs scripting to enable VIF #1 to answer calls on both the PA and WIP lines when they are received for BGM and paging. The following logic can be included in the Init Script table, as nothing needs to change once it is set up.

```
VC1PA1=1
VC1WA1=1
```

The amp racks, SIDs 21 and 22, need scripts to activate calls on their VIF #1 PA and WIP lines to the relevant PIB VoIP Proxy ports at SID 50. Again, these can be placed in the Init Script table, as they don't need to change once set up. If there were changing sources of BGM or paging, then the scripts would need to be dynamic and be included in the Main Script table.

```
VC1P50=1
VC1W50=1
```

PC Setup for Device Configuration

In the following sections it is assumed the PC Ethernet port is set for acquiring its IP address from a DHCP server.

For configuring the Moxa switch from factory default, a specific IP address will need to be assigned to the PC Ethernet port and the PC's Internet firewall may need to be turned off. Refer to the Windows Network and Internet Settings and Security Settings for these. Admin access will be required on the PC.

For reliable operation of the networking on the PC (e.g., to use PIB_Finder and the Moxa Edscfgui utility) it may be necessary to temporarily turn off WiFi operation (e.g., select flight mode) and disable other "Ethernet" network devices in the PC if they are configured.

Configuration of ATAs

3.1.12 Steps of Setting up an ATA

The two main stages to setting up an ATA are:

- 1) Reset the ATA to factory defaults and program the assigned IP address and other networking parameters. This needs to be done only once.
- 2) Configure the ATA over the network using its built-in web server.

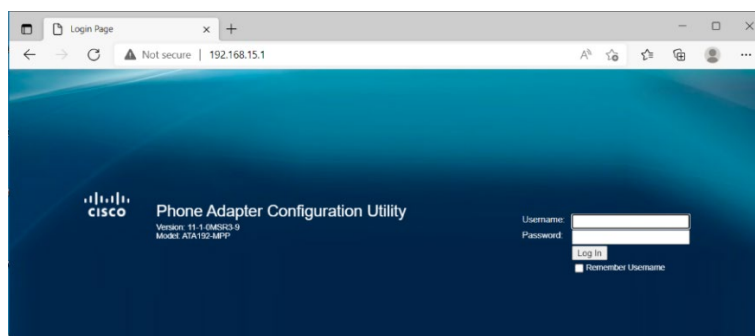
These are covered in the following sections.

3.1.13 Reset ATA to Factory Defaults & Assign IP Address

One method of programming the ATA is to connect a pushbutton telephone to the ATA and use the in-built IVR to program the settings. Refer to the ATA Admin Guide for the instructions.

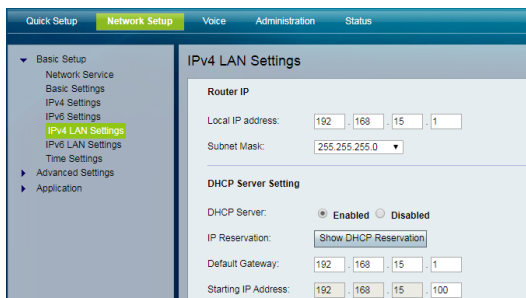
Otherwise, to reset the ATA to factory default and assign its network settings.

- Power up the ATA.
- Push and hold in the RESET pin on the ATA until the Power LED starts flashing, then release. Wait until the POWER LED is on steady (~90s).
- Connect the host PC's Ethernet port to the ATA's ETHERNET port with a RJ45 patch cable.
- On the host PC, open a web browser and browse to the ATA's web server. I.e, type 'http://192.168.15.1' into the address box of the browser and press Enter. The ATA web page should open. [The ATA has an IP Address of 192.168.15.1 and enables a DHCP server on the ETHERNET port by default.]



Screen Sample – ATA Log in page

- Log in with the default username **admin** and password **admin**.
- Click on **Network Setup** in the heading line and then on these pages (page names on LHS) enter the settings shown in the order of the table below.



Screen Sample – ATA Network Setup page

Page	Entry/Option
Basic Setup IPv4 Settings	Connection Type : Static IP Internet IP Address : enter the assigned IP address (e.g. 10.0.8.1) Subnet Mask : 255.0.0.0 Default Gateway : 10.0.10.1 Click Submit .
Basic Setup Network Service	Network Service : Bridge Click Submit .

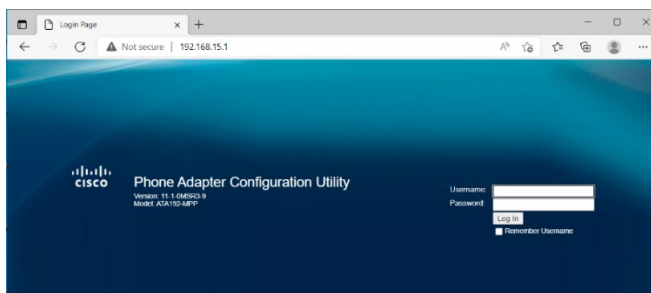
The host PC will lose connectivity to the ATA as the ATA’s IP address will change. Wait until the **POWER** LED on the ATA is on steady.

Further configuration of the ATA can be done using the built-in Web Server.

3.1.14 Configure ATA Using Built-In Web Server

On a working network:

- Enable the DHCP Server function on the PIB by fitting J13 DHCP Server/TAMPER. At most one device (PIB, Moxa switch, ATA) on the QE20 network should have the DHCP Server function enabled.
- Using an RJ45 cable connect the PC’s Ethernet port to a spare port on a Moxa Switch on the network, with the ATA connected to the Moxa switch or network ring with the PIB.
- On the host PC, open a web browser and browse to the ATA’s web server. E.g., type ‘http://10.0.8.1’ (use the correct IP address for the ATA) into the address box of the browser and press Enter. The ATA log in web page should open.



Screen Sample – ATA Log in page

- Log in with the configured username and password (or the default if no specific username and password is assigned).
- Program all the other settings as per Section 3.8.4.

3.1.15 ATA Programming

The following fields are required for QE20 VoIP after the ATA factory default is applied.

Page	Entry/Option
Administration	User List=>admin: Assign a suitable password. User List=>cisco: Assign a suitable password. Click Submit .
Voice Regional	Call Progress Tones=>Holding Tone : (Blank) Control Timer Values=>Hook Flash Timer Min : 0.4 Miscellaneous=>FXS Port Input Gain : 1 Miscellaneous=>FXS Port Output Gain : 6 Click Submit .
Voice Line 1	Call Feature Settings=>Enable IP Dialing : Yes Proxy and Registration=>Register : No Proxy and Registration=>Make Call Without Reg : Yes Proxy and Registration=>Ans Call Without Reg : Yes Subscriber Information=>User ID: 1 Dial Plan=>Dial Plan : as per section 3.5 FXS Port Polarity Configuration=>Idle Polarity : Reverse Click Submit .
Voice Line 2	Call Feature Settings=>Enable IP Dialing : Yes Proxy and Registration=>Register : No Proxy and Registration=>Make Call Without Reg : Yes Proxy and Registration=>Ans Call Without Reg : Yes Subscriber Information=>User ID: 2 Dial Plan=>Dial Plan : as per section 3.5 FXS Port Polarity Configuration=>Idle Polarity : Reverse Click Submit .

PIB Configuration

3.1.16 General

For details refer to the PIB User Manual (LT0519). The PIB must have V4.0 or higher firmware.

Do not select **Use DHCP for IP Settings** as a fixed IP address is required for QE20 networking and the VoIP proxy functionality.

When the VoIP proxy functionality is used without a Panel-Link networking device attached (i.e., all other PIBs than the first) the UDP Port should be set to a different port number to the Panel-Link PIBs to effectively disable Panel-Link networking.

In the VoIP Proxy configuration section, for each channel that is to be enabled, enter the SIP URI of the audio source. The default codec choice of G.729 should be changed to match the ATA – e.g., G.711 μ -Law.

3.1.17 Steps for Setting Up PIB

There are two main stages to setting up a PIB.

- 1) Reset the PIB to factory defaults and program the assigned IP address and other networking parameters. This needs to be done only once.
- 2) Configure the PIB over the network using its built-in web server.

These are covered in the following sections.

3.1.18 Reset PIB to Factory Defaults & Assign IP Address

The steps to reset a PIB to its factory defaults and program its required IP address and network settings are as follows. These should be used for a new PIB or a replacement PIB.

- Connect a PC to the PIB's RS232 port J22 with a null modem cable (LM0076) and using a serial port terminal program (QE20Config, SmartConfig, Wincomms) open the serial port with a setting of 38400, 8 data bits, No parity, 1 Stop Bit.
- Fit a mini jumper to "SERIAL DIAG/DOOR" J10 and leave "DHCP SERVER/TAMPER" J13 open.
- Hold down the RSSI button, then power up the PIB. Release the RSSI button after the Run LED LD3 on the PIB starts flashing. The PC terminal should show PIB power up messages.

```

>>>>>> ASE WK2 Monitor Program Version 2.00 <<<<<<<<
COPYRIGHT TSP CHCH  Nov 17 2009

ASE DTE Number:1048575 (000fffff)

Active ASE Firmware Header ID OK
No ASE Firmware Header ID in Inactive Space
Active ASE firmware header CRC OK
Active ASE firmware CRC OK
Run The Active ASE Firmware
0.095 PIB Powering up. Firmware version 3.00
1.109 Configured to get IP address via DHCP
1.110 Will Try for previous address 10.122.207.45
1.716 CP2200 Oscillator and Self Initialization Complete
3.286 Retrying negotiation
5.388 Auto-Negotiation Passed
5.389 Ethernet Full Duplex
5.390 Ethernet MAC Address 00:0b:3c:68:a7:be
5.405 Panel-Link IP Adapter Running
stat
Firmware compile date Feb 10 2016 16:33:29
PIB Board PA1091
Power Supply Fault Level: 115
Power Supply Actual Value: 237
IP Address=0.0.0.0; Netmask=0.0.0.0; Gateway=0.0.0.0
DHCP State=SELECTING; T1=2; T2=16; DHCP Server 0.0.0.0
61.866 DHCP failure
  
```

Screen Sample – PIB messages on Power Up

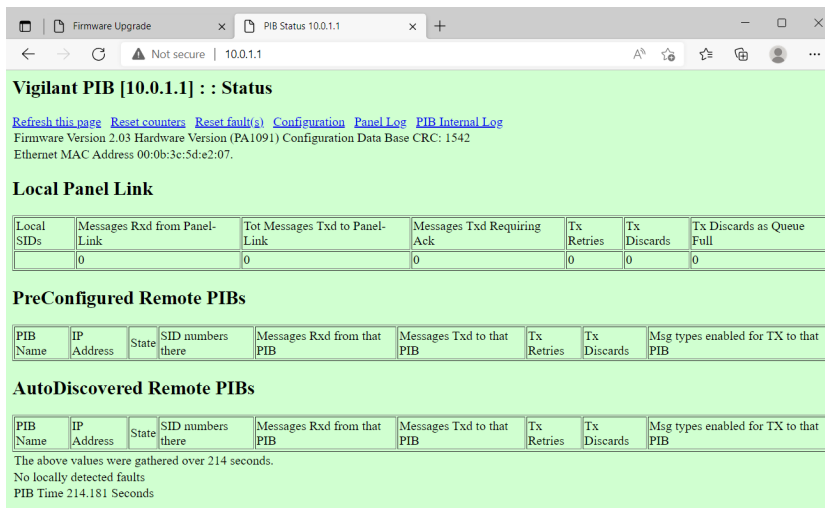
- On the PC terminal type **config** followed by the Enter key, and then enter the values in the table below to assign the required IP settings to the PIB.

Prompt	Entry
Password	Enter
Prevent PIB from acting DHCP	N Enter
Use DHCP to get IP	N Enter
Enter IP	Assigned IP address (e.g., 10.0.1.1 for SID 1) Enter
Enter Netmask	255.0.0.0 Enter
Gateway	Enter

All other configuration of the PIB can make use of the internal Web server – as described in the following section.

3.1.19 Configure PIB Using Built-In Web Server

- Enable the DHCP Server function on the PIB by fitting J13 “DHCP SERVER/TAMPER”. Note at most one device (PIB, Moxa switch, ATA) on the QE20 network should have the DHCP Server function enabled.
- Using an RJ45 cable connect the PC’s Ethernet port to the PIB’s Ethernet port or to a spare port on a Moxa Switch on the network, if the PIB is connected to the Moxa switch or ring.
- On the host PC, open a web browser and browse to the PIB’s web server. E.g., type ‘http://10.0.1.1’ (using the assigned IP address for the PIB) into the address box of the browser and press Enter. The PIB web page should open. Enter the programmed password if requested.



PIB Status Web Page

- Click on the ‘Configuration’ link at the top of the page to go to the configuration page and then enter the settings in the table below.

Field	Entry
--------------	--------------

Optional name for the PIB	PIB identifier e.g., PIB #1 at SID 1
Default UDP Port	Leave as 2220 unless PIB is used for VoIP only then change to 2221
Panel-Link baud rate	38400
FAS1 is fault contact input (NC)	Tick
VoIP Proxy Audio Source SIP URI	Enter the URI for the ATA in this QE20 for each used channel, refer to section 3.5, if VoIP Proxy is needed
Codec	G711u (to match ATA)
Password	Enter the password if assigned
New Password	Enter a password
Re-type password	Enter a password

- Click **Submit Changes** button to save the new settings.
- Click on **Click this to save the existing (submitted) configuration as a file on your PC**. Copy the file to a suitable storage place on the PC to save the file and rename it for the specific PIB, if required.
- Disconnect the PC.

Once the PC is disconnected remove the J13 “DHCP SERVER/TAMPER” link on the PIB to disable DHCP Server operation.

Fibre Switch Configuration

The QE20 Fibre IP Network Kit consists of a Moxa fibre switch and at least one PIB, ATA and VIF module. On the Moxa Switch the fibre network Ports 4 and 5 are usually used for the network ring, however Ethernet ports could be used for a mixed fibre - ethernet ring (e.g., Ethernet could be used for connecting the ring between two adjacent QE20 panels).

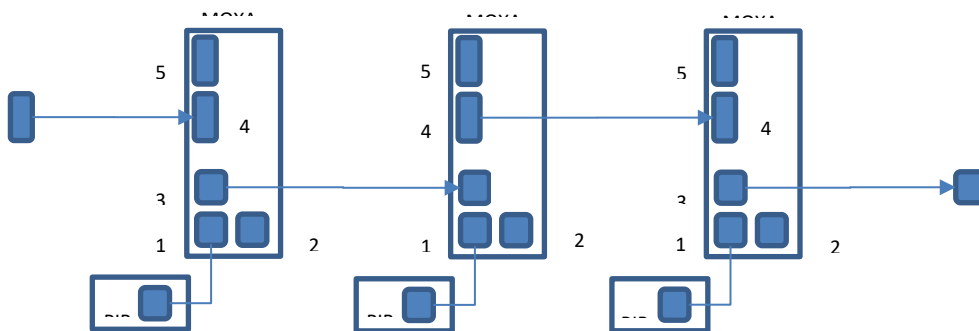


Figure 8 – Example Connection of MOXA Fibre/Ethernet Ports in Ring Network

Figure 8 shows the left-hand MOXA port arrangement with fibre port 4 and ethernet port 3 as the “ring ports”. Port 2 is available for the host PC configuration. Port 1 is initially used for PIB configuration, and when it’s done, for connection to the ATA’s WAN port and the ATA’s ETHERNET port will then be used for connection to the PIB.

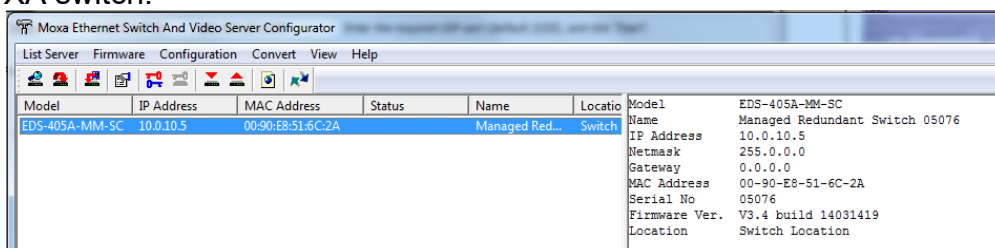
3.1.20 Reset Moxa to Factory Defaults & Assign IP Address

To reset the MOXA to factory default settings:

- The Moxa can be set to factory defaults using a special serial cable and a PC. This is described in the Moxa Managed Ethernet Switch/Extender User’s Manual. This method may be needed if the IP Address of the switch cannot be determined.

Or

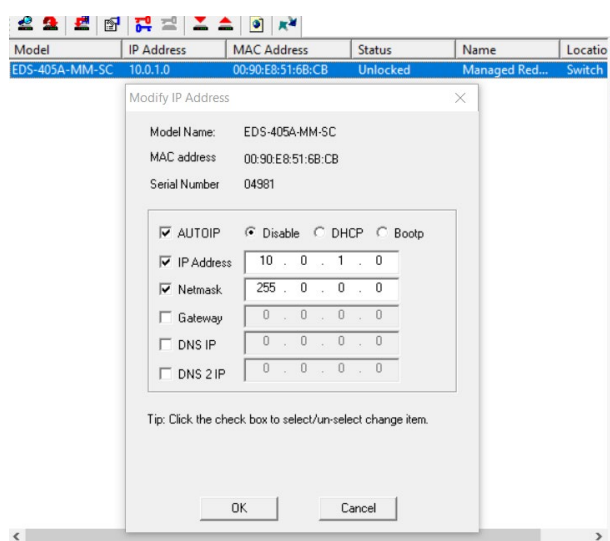
- Connect the PC’s Ethernet port to one of the MOXA’s Ethernet ports (1/2/3) with an RJ45 patch cable.
- Assign an IP address of 192.168.127.1 and a net mask of 255.255.255.0 to the Ethernet port on the PC. It may also be necessary to disable the Firewall in the PC.
- Run the MOXA PC application ‘edscfgui.exe’ on the PC and click the Broadcast Search icon (left most icon of the icon list bar) to scan/find the MOXA switch.



By default, the Moxa switch has an IP address of 192.168.127.253. If the reported IP address for the Moxa switch is not 192.168.127.253 then the Moxa has another IP address assigned. It may be necessary to assign a different IP address and net mask to the PC Ethernet port so the PC and Moxa are on the same sub-net and can communicate over Ethernet.

If the Status column shows Locked right click the entry found, and on the menu select **Unlock**. Enter the assigned username and password – by default these are **admin** and **moxa**. Click **OK** and make sure the status changes to Unlocked.

Right click the entry found, and on the menu select **Modify Configuration**. A Modify IP Address dialog box should be shown.



Tick **AUTOIP** and select **Disable**.

Tick **IP Address** and enter the assigned IP address (e.g., 10.0.1.0).

Tick **Netmask** and enter the subnet mask (e.g., 255.0.0.0).

Click **OK**.

At this point the PC will lose connectivity with the Moxa switch.

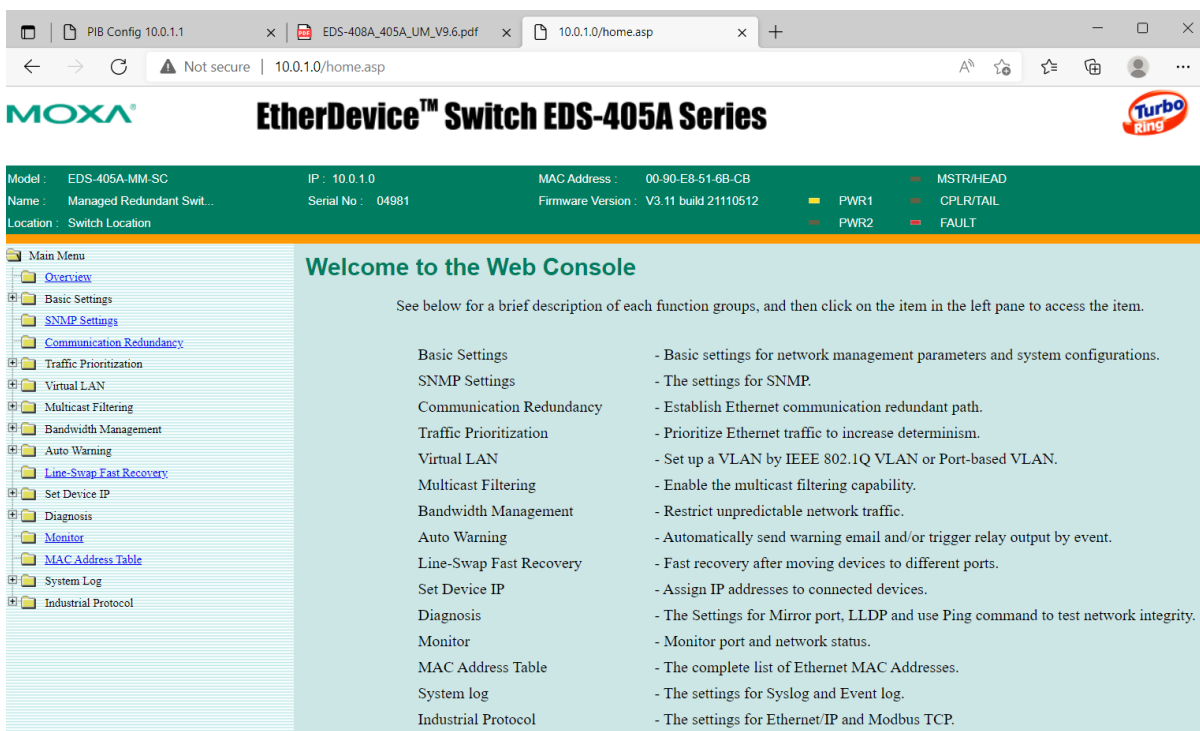
- Remove the IP Address assignment on the PC Ethernet port and select DHCP. Further configuration of the Moxa switch can be done using the built-in Web Server.

3.1.21 Configure Moxa Switch Using Built-In Web Server

To configure the MOXA switch using its assigned IP address:

- Enable the DHCP Server function on the PIB by fitting J13 “DHCP SERVER/TAMPER”. Note at most one device (PIB, Moxa switch, ATA) on the QE20 network should have the DHCP Server function enabled.

- Using an RJ45 cable connect the PC’s Ethernet port to a spare port on the Moxa Switch, or another Moxa Switch if the Moxa Switches are connected to a working network ring along with the PIB.
- On the host PC, open a web browser and browse to the Moxa web server. E.g., type ‘http://10.0.1.0’ (using the assigned IP address for the Moxa switch) into the address field and press Enter.
- Enter the assigned username and password.
- This should display the Moxa web server home page.



- Enter the settings shown in the table below. Click on the + symbol of the Page name on the left hand side to open some menus.

Page=>Field	Entry/Option
Communication Redundancy	Redundancy Protocol : Turbo Ring V2
	Enable Ring 1: Tick Set as Master: Tick for one QE20, untick for all others. Redundant Ports 1 st Port: Select the ring port (usually 4). Redundant Ports 2nd Port: Select the ring port (usually 5). Click Activate .
Auto Warning=>Relay Warning=>Event Setup	Power Input 1 failure: Enable Power Input 2 failure: Disable Turbo Ring Break: Enable

	<p>Select Off for the ring ports and Ignore for the other ports. Generally:</p> <p>PortEvents->Port1: Ignore PortEvents->Port2: Ignore PortEvents->Port3: Ignore PortEvents->Port4: Off PortEvents->Port5: Off</p> <p>Click Activate</p>
Basic Settings=>Turbo Ring DIP Switch	<p>Disable the Turbo Ring DIP Switch: Tick</p> <p>Click Activate.</p>

4. VIF0907 Module

VIF Function

The VIGILANT VIF module is designed as an interface between the QE20 and a 3rd party VoIP ATA. The ATA is used to provide QE20 “backbone” (inter-panel) audio communications by converting the QE20’s audio circuits into digital data and transmitting this between the appropriate panels (one-to-one calls), and the PIB VoIP proxy (one-to-many calls).

The ATA usually provides a means to connect an analogue telephone onto an IP network. It decodes DTMF tones to obtain the dialled digits, then communicates with a SIP server to obtain the IP address of the remote VoIP interface to connect to. However, for the QE20, no SIP server is used, and the ATA is instead configured to internally convert dialled digits to an IP address:port and then call that IP address:port directly (which will be that of another ATA or a PIB VoIP proxy on the QE20 network).

4.1.1 VIF to ATA Interface

The connection point for the VIF to the ATA is the same as for connecting an analog phone directly to the ATA.

Therefore, the VIF has been designed to emulate an analogue phone, i.e., it makes calls by going off-hook (drawing current) then dialling using DTMF tones. It answers incoming calls by recognising a ringing voltage on the line and going off-hook (drawing current).

Additionally, the VIF provides 2 x 5V outputs suitable to power up to 2 ATAs.

4.1.2 VIF to QE20 Interface

The VIF connects to the Speech (PA) and WIP terminals of the QE20 RS485 Network Module. The VIF also connects to the QEComms terminals on the RS485 Network Module, so that it can be controlled by, and send status to, the QE20 Controller. In particular the QE20 commands the VIF to ‘dial’ remote nodes as required (i.e. when a Speech (PA) and/or WIP connection is required between nodes).

If other audio signals – paging console, background music, etc, are being carried over the fibre network they are connected to the WIP and PA terminals on the additional VIF module(s) fitted.

4.1.3 Block Diagram

Figure shows the major functional blocks of the VIF0907. It has a microprocessor that communicates with and is controlled by the QE20 ECM. The remaining blocks make up two sections that are virtually identical; one for PA and the other for WIP. Each of these sections has a transformer to provide isolation to the QE20, a DTMF generator controlled by the microprocessor, a ‘current sink’ enabled by the microprocessor, a ring detector that feeds into the microprocessor, and a relay controlled by the microprocessor that enables it to connect to one of two ATAs.

The ability to switch between two ATAs with relay control is not typically used in newer systems that utilise the PIB VoIP proxy. It is needed in older systems where an ATA line is hard configured to be either bi-directional or uni-directional (SAS mode). This meant that one ATA line was needed typically for incoming PA audio, or a one-to-one bi-directional WIP call, and the other (in SAS mode) used for unidirectional outgoing audio to multiple places i.e. PA or WIP All-Call originating here.

With the PIB VoIP proxy functionality the ATA lines are always configured to allow bi-directional operation. This allows a line to be used for both one-to-one and one way outgoing audio – what is required for a given call is negotiated by the software within the VoIP devices (ATAs or PIB) at the call endpoints when the call is setup.

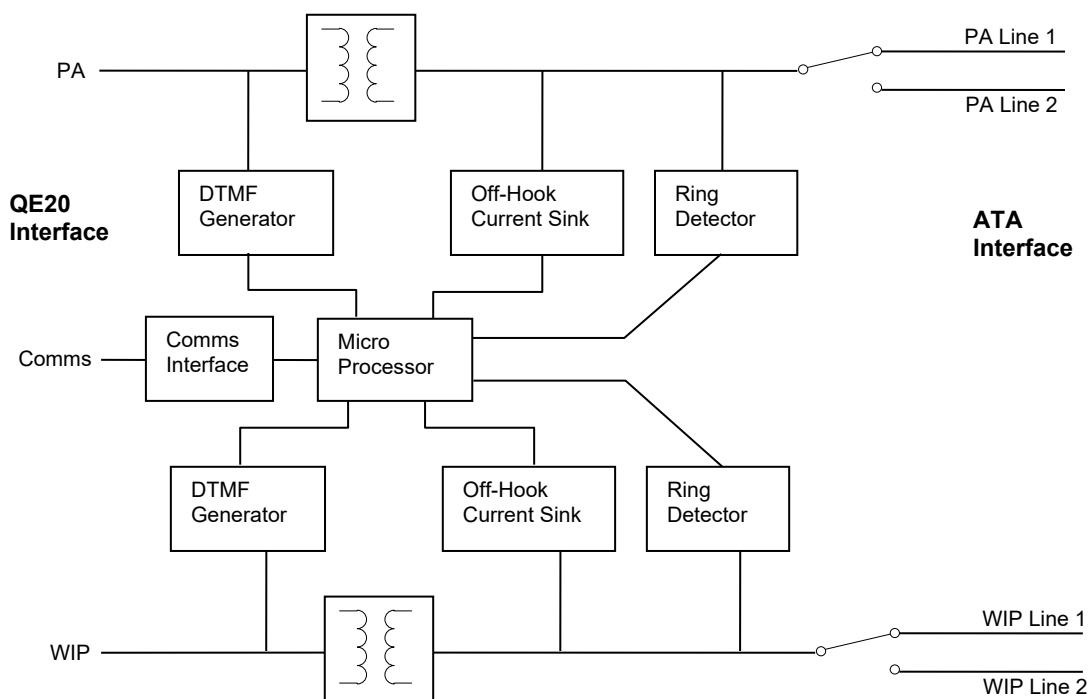


Figure 9 – VIF0907 Functional Block Diagram

Physical

Size	Circuit board assembly 180 x 93 x 30mm.
Mounting	4 holes 3.0mm diameter in the same pattern as a QE20 FIB Module / WIP termination card / SPIF card. Supplied on DIN rail mounting bracket.

Environmental

Operating Temperature	-5 to +45°C
Humidity	Up to 95% non-condensing

Power Requirements

Voltage	15 - 28V DC
---------	-------------

Current	0.16A quiescent / 0.33A in-use* (@24V) when one 5V output is used to power one ATA.
	0.29A quiescent / 0.60A in-use* (@24V) when both 5V outputs are used to power two ATA.
	0.03A + 0.01A per operated relay when the 5V power outputs are not used.
	* in-use means each line has a call in progress.

Electrical Inputs / Outputs

24V Power (J12)	In	Screw Terminals (2)
5V Power (J13)	Out	2-pin header
5V Power (J14)	Out	2-pin header
PA Line 1 (J1)	In / Out	RJ11 to ATA
WIP Line 1 (J4)	In / Out	RJ11 to ATA
PA Line 2 (J2)	In / Out	RJ11 to ATA
WIP Line 2 (J5)	In / Out	RJ11 to ATA
QE20 Comms Bus	In / Out	Screw Terminals (4)
RS232 (J8)	In / Out	DB9 Socket, DCE pinout. Used for Flash programming and diagnostics.
QE20 PA	In / Out	Screw Terminals (2) to RS485 Network or audio
QE20 WIP	In / Out	Screw Terminals (2) to RS485 Network or audio
Call Server	In	Screw Terminals (2), currently not used.

Links

LK1	Enable Flash Program	Insert when updating the firmware from a PC/laptop. Otherwise, this link must be removed, or parked on one pin only.
LK2	Flash Program / Call Server	Place in the Flash Program position when updating the VIF's firmware. Otherwise, the position doesn't matter (currently).

DIP Switches

The address of the VIF must be set using the DIP switches. The first VIF must be set to address 90 hex (all switches off). Additional VIFs (if any) must be set to sequential addresses starting from 91 hex.

ADDR1 - 8	QE20 Comms Address	ADDR 8	ADDR 4	ADDR 2	ADDR 1	Address (hex)
		Off	Off	Off	Off	90
		Off	Off	Off	On	91
		Off	Off	On	Off	92
		Off	Off	On	On	93
		Off	On	Off	Off	94
		Off	On	Off	On	95 etc.
FUNC A / FUNC B	Function	Both Off (currently unused)				

LEDs

LD1-LD6 mimic the operational state of the corresponding relays RL1-RL6.

LD1	PA Off Hook	On indicates that the VIF PA line is off-hook – i.e. is making or has answered a PA call.
LD2	PA Line 2	On indicates that the VIF is using line 2 for PA. Otherwise it is using line 1.
LD3	PA Connect	On indicates that the VIF has dialled an outgoing call or has answered an incoming call for PA. The audio signal to/from the ATA is switched through to the QE20.
LD4	WIP Off Hook	On indicates that the VIF WIP line is off-hook, i.e., is making or has connected a WIP call.
LD5	WIP Line 2	On indicates that the VIF is using line 2 for WIP. Otherwise it is using line 1.
LD6	WIP Connect	On indicates that the VIF has dialled an outgoing call or has answered an incoming call for WIP. The audio signal to/from the ATA is switched through to the QE20.
LD7	RUN	Flashes approx 2 times per second normally. Flashes slowly if there is a Flash CRC fault. Off if no power. Anything else indicates a serious software issue.
LD8	QE20 Comms	Flashes in response to messages received from the QE20.

Refer to sections 3.4.1 and 3.4.2 for details about which end of a required connection makes the 'call'.

Refer to section 7.1 for details of fault finding using the VIF LEDs.

Fault Conditions

The following conditions generate faults which will be indicated on the QE20.

- A VIF module configured in the site config is not responding on the QE20 Comms bus. This fault is generated by the RS485 Network Module and shown on the QE20 as QECOMMS BUS FAULT.
- When trying to make a speech announcement or a WIP call to / from a remote QE20, if there is no voltage on the local line to the ATA, or the VoIP call is not answered by the remote ATA / VIF / PIB VoIP proxy / QE20. This fault is generated only when the VoIP call is required, i.e., there is a WIP call or Speech broadcast across the fibre network. This is shown as "PA Segment Fault" for a failed speech connection and "WIP Segment Fault" for a failed WIP connection at the QE20 that attempted to make the call.

Serial Port Diagnostics

A PC running a terminal emulator program can be connected to the RS232 port (DIAGS/PROGRAMMING J8) on the VIF module, using a straight-through 9 pin male to 9 pin female RS232 cable (such as LM0138). In the terminal emulator select the required comms port, 9600 baud, no parity, 1 stop bit.

The VIF will log attempted calls, incoming ringing, etc. Note the first 4 characters of each line are the time in milliseconds displayed in hexadecimal. This free-running timer wraps from FFFF to 0000 every 65.536 seconds.

The VIF logs the following events.

Message	Meaning
VIF Software Vx.xx	Displayed on power-up. The VIF firmware version is x.xx.
DIP SWITCH ADR xx	The board address set on the DIP switches has changed to xx.
PA Ring	A ringing signal has been detected on the PA line.
WIP Ring	A ringing signal has been detected on the WIP line.
PA Fwd On	A voltage of 'forward' polarity has been detected on the PA line.
PA Fwd Off	A previously detected 'forward' polarity on the PA line has ended.
PA Rev On	A voltage of 'reverse' polarity has been detected on the PA line.
PA Rev Off	A previously detected 'reverse' polarity on the PA line has ended.
WIP Fwd On	A voltage of 'forward' polarity has been detected on the WIP line.
WIP Fwd Off	A previously detected 'forward' polarity on the WIP line has ended.
WIP Rev On	A voltage of 'reverse' polarity has been detected on the WIP line.
WIP Rev Off	A previously detected 'reverse' polarity on the WIP line has ended.
PA Answer	The VIF has answered a call on the PA line.
WIP Answer	The VIF has answered a call on the WIP line.
PA disc	The VIF has disconnected a call on the PA line.
WIP disc	The VIF has disconnected a call on the WIP line.
*PA Calling xx	The VIF is calling phone number xx on the PA line.
*WIP Calling xx	The VIF is calling phone number xx on the WIP line.
**PA Dialling xx	The VIF is dialling phone number xx on the PA line.
**WIP Dialling xx	The VIF is dialling phone number xx on the WIP line.
**PA Dial Complete	The VIF has completed dialling the phone number on the PA line.

**WIP Dial Complete	The VIF has completed dialling the phone number on the WIP line.
PA call ended	The VIF has “hung up” the phone on the PA line
WIP call ended	The VIF has “hung up” the phone on the WIP line

Some examples of what you should expect to see on this log are given below.

Example power-up sequence:

```
0001 VIF Software V1.04
0001 DIP SWITCH ADR 01
```

Example call out on PA line:

```
3602 PA FWD On
36D0 PA Dialling 8
3735 PA Dial Complete
38CA PA FWD Off
38ED PA REV On
```

Note: FWD On indicates voltage on the line in an idle condition. REV On indicates a connected call.

Example call answered on WIP line:

```
381B WIP Ring
382A WIP Answer
383A WIP FWD On
384B WIP FWD Off
385E WIP REV On
386F WIP FWD On
3881 WIP FWD Off
3894 WIP FWD On
38A5 WIP REV Off
38B8 WIP FWD Off
38DC WIP REV On
```

Note: FWD / REV alternate rapidly until the adapter discovers the line has been answered. The final connected state should be REV On, indicating the call connected state.

Flash Programming/Firmware Update

New versions of firmware for the VIF microprocessor may become available from time to time. Table 4-11 below lists the versions currently released. These versions will be supplied with a firmware file such as **SF0446_V1.05.mot**, a utility **vig_avrsp.exe**, and some more files. Extract these files to a folder on your PC.

Connect an RS232 serial port of your computer to the DIAGS / PROGRAMMING (J8) port on the VIF module using a straight-through 9 pin male to 9 pin female RS232 cable (such as LM0138). Insert both LK1 and LK2 on the VIF module in the ENABLE FLASH PGM position and power up the VIF.

Open a command prompt, and “change directory” to the folder you extracted the files to, and run the following command on your PC (for example): -

```
vig_avrsp.exe -pv1 -d20 SF0446_V1.05.mot
```

–pv1 is the PC comms port number. Set it to some other number if you use a port other than COM1. –d20 is a delay parameter. Try increasing the number if you have a very fast PC and get an error. SF0446_V1.05.mot is the new firmware file to download.

Check that the PC displays (this process takes up to ~1 minute).

```
Detected device is ATmega164P.
```

```
Flash: Erasing...Writing...Verifying...Passed
```

Then run a terminal emulator on the PC, as described in section 4.10. After you have started the terminal emulator, remove LK1 or place it on one pin only, and the VIF should restart with the new firmware. The position of LK2 does not matter now. Check that the version displayed on power up matches the new version you just programmed.

Note: this procedure works best with a real serial port. With a USB to serial port emulator/convertor it should work eventually, but will take many minutes.

Table 4-11 – VIF Firmware Versions

Version	Features
1.01 – 1.03	Earlier versions. These are not compatible with QE20. Either the firmware must be updated to at least V1.06, or the VIF replaced with a newer one.
1.04	Added delay before sending reply on QEComms. Revised some timeout values for use with ATA 192.
1.05	Removed delay for use on QE90.
1.06	Changed delay so removed click on audio. Required for use with QE20.

5. Wiring

Fibre Network Kit to QE20 Connections

Installation and wiring of the Fibre Networking Kit is detailed in FP2024/2025 QE20 Fibre Network Kit Installation Instructions (LT0714).

The QE20 Fibre Network Kit requires:

- 24V DC power to be supplied from a PSE.
- A 4-way loom carrying PA & WIP audio to the VIF to be plugged into J11 VIF on the RS485 Network Module.
- The 2-way loom to be plugged into J5 QECOMMS+- on the RS485 Network Module.
- The 10-way header loom to be plugged into J3 TTL SERIAL on the RS485 Network Module.

Drawing 1922-197 Sheet 2 (included in this manual) details the wiring connections between the Fibre Network Kit and the RS485 Network Module, and the internal wiring of the Fibre Network Kit.

Internal Wiring of Fibre Network Kit

5.1.1 Power

24V DC power is wired to the 24V terminals of the PIB, “PWR1” terminals of the Moxa fibre switch, and 24V J12 on the VIF.

+5V DC power for each ATA is wired from the +5V outputs of the VIF (J13, J14) to each ATA “DC 5V Power” socket using a 2-way loom with suitable connectors.

5.1.2 Ethernet

The ATAs are programmed to operate in Bridge mode, with their Ethernet port connected to the Internet port. So, a number of ATAs can be connected in series.

The PIB that is used for Panel-Link networking should be connected to the last ATA so that it monitors the ATA units for network connectivity and functionality. Disconnection or failure of any ATA will then cause the PIB to stop receiving messages and thus generate faults for the QE20 network.

The “Ethernet” RJ45 connector J2 on the PIB is wired to the “Ethernet” RJ45 connector on the ATA using a short RJ45 cable. Connect another short RJ45 Ethernet cable from the ATA’s “Network” socket to the Ethernet port of the next ATA or to one of the RJ45 connectors “1”, “2”, or “3” on the fibre switch.

Additional ATAs or PIBs can be connected to the other Ethernet RJ45 ports of the fibre switch.

5.1.3 Fibre Switch Relay Output

The “Relay” output of the Moxa Fibre Switch must be wired to the FAS1 & 0V input of the PIB so that a fault on the fibre ring is signalled to the PIB and onto the QE20 for annunciation.

The “Relay” contacts are normally energised by the switch so are closed, opening for a ring fault, switch failure, or power loss.

Any additional Fibre Switches should have their “Relay” contacts wired in series with the first, so that any fibre switch opening its “Relay” contacts will create an open circuit and cause a fault on the PIB.

5.1.4 ATA Phone Lines

The PA Line 1 J1 and WIP Line 1 J4 RJ11 connectors of the VIF Module are connected to the Phone 1 and Phone 2 ports of the corresponding ATA using RJ11 phone cables (straight through connections).

Fault Wiring : VoIP Proxy Only PIBs

For a PIB that is performing Panel-Link networking for the QE20 it is not necessary to connect its fault output directly to the QE20. This is because the RS485 Network Module will request the PIB status and indicate a fault if this not received or the PIB is indicating a fault condition is present.

However, PIBs that are being used for only VoIP proxy functionality (the Panel-Link networking is not utilised) must have their fault outputs monitored. OC2 on the PIB is normally turned on, going open on fault or power fail.

The fault output OC2 of each such PIB must be connected to the FAS1 Fault input of a preceding PIB. The OC2 Fault output of this PIB should be wired at the end of the Moxa “Relay” chain, so that it provides a connection to 0V when normal for the Moxa switches and the first (Panel-Link Networking) PIB FAS1 input.

The additional PIBs will also require:

- 24V DC power as per Section 5.2.1.
- Ethernet connection to an ATA or Moxa Fibre Switch as per Section 5.2.2.

Additional VIF & ATA Wiring

If an additional ATA and VIF are required for extra network audio functions (e.g., background music and paging), then connect the Music and Paging audio signals to the PA and WIP inputs of the VIF.

Connect 24V to the VIF, and 5V DC Power, and the phone lines from the VIF to the ATA as described in Section 5.2.1 and shown in general on drawing 1922-197 Sheet 2.

6. Fitting Fibre Parts in a QE20 Cabinet

The QE20 Fibre Networking Kit is a pre-assembled module containing a PIB, VIF, ATA, and a Moxa switch. It occupies 3 x 60mm module positions of the QE20 mounting frame, usually fitted within 1 frame distance of the RS485 Network Module so the supplied cables can be used.

Additional PIBs, or VIF & ATA pairs can be mounted on FP2029 General Purpose brackets. This bracket occupies 1 x 60mm position of the mounting frame and can fit one PIB, or 1 VIF and 1 ATA. It should be mounted in the very right-hand position of a frame so it can overhang into the spare space.

Note that the Fibre Network Kit cannot be mounted in a shallow cabinet.

Details for fitting and wiring the Fibre Kit are contained in its installation guide LT0714.

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7. Fault Finding

Fault Finding Using VIF, ATA, and PIB LEDs

If there is a problem with the networked Speech or WIP VoIP connections, the first check is to make sure there are no faults at the affected QE20s, with the fibre ring, or associated equipment.

If the problem is the calls are established and no fault is generated indicating the VoIP calls failed, but no audio is transferred then check the cable connecting the VIF PA/WIP signals on the RS485 Network Module to the VIF Module in the Fibre Network kit at each end of the call.

If the calls are not being established (results in a PA Segment or WIP Segment fault being generated soon after making the call) the LEDs on the VIF modules and ATA modules at each end of the audio connection should be checked.

If the VIF "Off Hook" LED is repeatedly turning on for ~0.2 seconds then off for about 1.0 seconds, this indicates that the VIF is trying to make a call, but there is no voltage on the line from the ATA. Check the cable to the ATA and that the ATA is powered up and not in fault. If the RJ11 cable has just been replaced (the polarity may be different), it may be necessary to alter the ATA settings as described in section 3.8.4. Check that the ATA has an Ethernet connection – its Power LED should be ON steady, and its Network LED should be ON, with some flickering.

If the VIF "Off hook" LED is repeatedly: turning on, then after ~0.2 seconds the Connect LED is turning on for ~2.0 seconds then both LEDs turn off for about 1 second, this indicates that the VIF and ATA are trying to make the call, but it is not being answered. At the QE20 that should be receiving the call check the ATA that is being called is powered up, has an Ethernet connection – its Power LED should be ON steady, its Network LED should be ON with some flickering, and the RJ-11 cable connects to the VIF. Check that the receiving ATA is visible on the network - try pinging or browsing to its IP address with a laptop plugged into a spare Ethernet port at the same location as the ATA that is unsuccessfully trying to connect. Check the configuration of the QE20s and ATAs.

Note: if an ATA Power LED is flashing, this may indicate no Ethernet connection, the ATA is determining its IP address, it is rebooting, or it could not boot up correctly. Consult the ATA User Guide for an explanation of the flash cadence.

If a PIB VoIP proxy is utilised for the call, the power LED of the PIB should be lit, the Ethernet Link LED should be flickering if a call is active, and if no call is active it should at least be lit if there is an Ethernet cable attached. The PIB system fault LEDs may indicate if a fault is present as well. To get specific diagnostic information for the VoIP proxy functionality the PIB status web page should be visited – see Section 7.2.

Fault Finding Using ATA and PIB Status Pages

Additionally, it is useful to check the voice status page of the web interfaces of each of the ATAs. This will show if there is a call being attempted, in-progress, and when there is a connected call, it will show the IP address of the remote ATA or PIB VoIP proxy.

When the PIB VoIP proxy is utilised, the PIB status page will also show useful information. The channel status can be seen – if it is not “streaming” then the call hasn’t connected properly. The IP addresses of callers to a channel are displayed here as well. Check to see the calling ATA is listed. If the PIB knows of a problem that caused a call to fail, then this will be logged. Log entries are shown on the PIB Internal Log web page.

If the required call is “connecting”, but audio is not being received, it is likely that the problem is in the PA and WIP audio connections from the QE20 Network Module to the VIF (at each end), a faulty VIF or ATA, or (less likely) something in the IP network is blocking RTP traffic. The latter condition could be confirmed by the number of “RTP packets received” not incrementing on the ATA status page.

If the required call is not “connecting”, this should be confirmed by the LEDs on the VIF, and by a fault being generated on one or both QE20s. Check the status page of the ATA that should be making the call and see if (every 1 - 2 seconds) **Call 1 state** changes from **Idle** to **Dialling**. If not the problem may be with the VIF card or ATA at that end. If it does change to **Dialling** but never to **Connected**, the problem could be a configuration issue, an issue with the IP network, a faulty VIF, faulty or misconfigured PIB, or a faulty ATA at the receiving end.

Note: The Problem Reporting Tool button (PRT) on the ATA cannot be used as there is no server on the QE20 network for the ATA to send the report to.

Fault Finding Using a Phone

A standard “touch tone” phone with an RJ11 cable can be used for some diagnosis.

- Plug it (instead of the VIF) into the ATA Phone port and dial the number that the VIF should be dialling to reach a particular remote panel. Check that the call is answered. If applicable check that you can hear audio from the far end, e.g., when someone at the panel is making a Speech (PA) announcement that should be getting to the panel you are at.
- Plug the phone (instead of the VIF) into the ATA Phone port and check that it rings when other panels or the PIB VoIP proxy attempts to make a connection to it, e.g., by connecting a WIP call.

Be sure that you take account which end initiates the VoIP calls – refer to sections 3.4.1, 3.4.2, 3.4.3.

Fault Finding a Ring Network

Note: if the ring is broken somewhere the master Moxa switch will turn its red fault LED On and this will be annunciated at that QE20 as PIB Input Fault.

Note: if a QE20 panel is turned off the fibre links on each side of the panel will appear “down”. However, all other panels will keep working as long as only one panel is turned off. If you need to power down a panel for an extended time, you could use a fibre connector to bypass the fibre switch.

To locate where a break is, use a web-browser to browse to each of the Moxa switches in turn until one that is adjacent to a break is found. If you know the order the switches are wired in, you could check only every second switch.

On the screen that comes up check the details along the heading. MSTR/HEAD will be on/flashing for the Master switch. FAULT will be on if there is a fault at that node.

IP : 10.0.1.0	MAC Address : 00-90-E8-51-6B-CB	— MSTR/HEAD
Serial No : 04981	Firmware Version : V3.11 build 21110512	■ PWR1 — CPLR/TAIL
		— PWR2 ■ FAULT

The Communications Redundancy page gives further details.

Healthy indicates the ring is complete. **Break** indicates the ring is broken somewhere.

Master indicates this switch is the ring master, and **Slave** indicates this switch is not the master. This information is not relevant in determining the location of a break.

Check the assigned ring ports on the Moxa switch.

Link Down indicates the ring segment connected to this port is physically broken. Check the status of the switches on either side to find the other end of the broken segment.

Forwarding indicates the segment connected to the port is good and data is being forwarded over it.

Blocked indicates the segment connected to the port is good but data is not being forwarded over it, to deliberately break the ring under normal conditions so packets don't circulate forever. This won't be shown if the ring is physically broken somewhere.

Advanced Fault Finding in a Large Network

Some other techniques that could be used for locating ring breaks in a large network are listed below. Some of these will require a degree of IP experience to set up.

- Ping each device from a PC on the network. If devices are not accessible it could be due to ring breaks, devices powered off or disconnected from the network.
- Create a second break on the ring and determine which panels then go off-line. Use the wiring diagram to determine where the two breaks are.
- Set up a syslog server on a PC attached to the network, and set up the Moxa switches to send a fault log to this server.
- Use some network management software from the switch supplier - http://www.moxa.com/product/network_management_software.htm.
- Use an SNMP (Simple Network Management Protocol) product.

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8. Module Replacement

Part Numbers for Ordering

FP2024 FP, QE20 FIBRE NETWORK KIT, SINGLE MODE

FP2025 FP, QE20 FIBRE NETWORK KIT, MULTI MODE

The complete Fibre Networking kit with either single mode or multi-mode fibre switch.

FP2029 FP, QE20 GP MOUNTING BRACKET, 1 MODULE WIDE

Bracket for mounting PIB, ATA & VIF, or other modules

PA0644 PCB ASSY VIF0907 VOIP INTERFACE BOARD (DIN)

Includes 1 x LM0448 and 2 x LM0552, and DIN rail mounting hardware.

SU0321 SUNDRY CISCO VOIP ADAPTER (ATA 192)

SU0319 SUNDRY MOXA 5 PORT E/NET SW (2 MULTI MODE FIBRE)

SU0320 SUNDRY MOXA 5 PORT E/NET SW (2 SINGLE MODE FIBRE)

SU0325 SUNDRY MOXA 5 PORT E/NET SW

SU0326 SUNDRY MOXA 8 PORT E/NET SW

FP0986 FP PIB PANEL-LINK IP BRIDGE

LM0138 LOOM DB9M-DB9F ALL PINS STRAIGHT THRU 1.8M

Suitable for Re-Flashing VIF module firmware or connecting to its diagnostic port.

LM0448 LOOM, QE20 VoIP ADAPTER POWER

LM0552 LOOM RJ11 TO RJ11 1.5M ALL PINS STRAIGHT THROUGH

Re-Configuration

8.1.1 Configuring a Replacement VIF

The VIF does not require any configuration, apart from setting its address on the DIP switches and the two links the same as on the module being replaced.

Don't use an old VIF in QE20, as the firmware version may not be compatible. At least firmware V1.06 is needed.

8.1.2 Configuring a Replacement ATA

QE20 is designed to use the ATA 192. Do not replace the ATA 192 with an earlier or other manufacturer's model. The firmware in the ATA 192 must be at least version 11-2-3.

The replacement ATA **must** be configured as documented in each site's documentation pack.

Refer to Section 3.8 for configuring the ATA and the ATA Administration Guide for further information on the settings.

8.1.3 Configuring a Replacement PIB.

Refer to the PIB User Manual (LT0519) for full instructions.

The PIB needs to have V4.0 firmware or later installed for compatibility with QE20. It should be configured as per the documentation pack. Instructions are contained in Section 3.9.

8.1.4 Configuring a Replacement Fibre Switch.

Refer to the PIB User Manual (LT0519) for instructions.

The Fibre Switch should be configured correctly as per the documentation pack, or the saved configuration restored. Refer to Section 3.10.