nevion

AES-IP-MUX/-SFP
ADA-IP-MUX/-SFP
User Manual

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2 Revision History

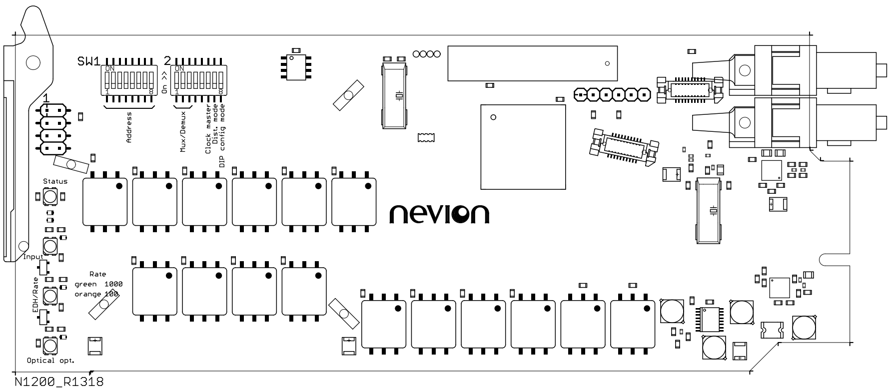
Revision	Date	Comments
A	2016-04-29	First revision
B	2016-08-24	IGMPv2 added
C	2016-11-21	PTP domain control added
D	2017-09-07	New ADA-IP-MUX-C1 backplane with duplex LC connector
E	2018-07-20	Ping updated. Delay request specification changed. Gyda graphics updated
F	2019-06-25	Added ADA-IP-MUX channel tables
G	2019-07-16	Corrected ADA-IP-MUX direction setting for ADDA-AES8
H	2020-06-26	Updated support information

3 Product Features

- IP audio infrastructure scalable to enormous networks
- Digital audio transport and routing on gigabit LAN
- Uses existing CAT5 cabling and SOHO ehthernet switches
- Low latency AES67 linear audio (down to 250us packets)
- Transparent AES3 mode
- Network PTP clock reference
- Unicast or multicast audio transport
- AES-IP-MUX:16 AES3 ports per module, configurable direction
- Digital audio sync output. AES11 or wordclock.
- Optical network connection on SFP option
- ADA-IP-MUX:8 AES3 ports per module, 8 analogue audio channels. Three direction combinations.

4 Introduction

4.1 Top view



4.2 Product Description

The AES-IP-MUX and ADA-IP-MUX are used to transport a large number of digital audio signals over a dedicated IP local area network.
They may be used for:-

- Large scale audio infrastructure (iPath mode)
- Smaller scale matrix type infrastructure up to 128x128 stereo (matrix mode)
- Point to point links over IP (fixed matrix mode)

The **ADA-IP-MUX** product is a combination of an AES-IP-MUX, one of three audio converter modules and a common backplane. Most of this manual refers to the AES-IP-MUX. The audio converter modules are ADC-AES8, DAC-AES8 and the ADDA-AES8.
The -AES8 converter digital audio channels are connected to the IP-MUX AES channels 7-10. This is to allow the use of the ADDA-AES8 converter which has two digital inputs and two digital outputs.

The AES-IP-MUX has 16 AES3 audio ports which may be used as inputs or outputs, sources or destinations in the network.
The AES-IP-MUX uses a central timing reference (PTPv2) and all digital audio signals are locked to this reference.
The AES-IP-MUX encodes AES3 linear PCM audio into AES67 2 channel packets. The packet times supported are 0.25ms, 0.5ms, 1ms and 4ms. The sample frequency supported is 48 kHz. The audio signals are transported either in a standard 16 or 24 bit packet format or completely bit transparently AES3.
The audio transport has a minimum latency of just over the selected packet time plus the network delay. The network delay is less than 0.1 ms in a normal gigabit LAN.

The encoding parameters are set on the AES3 input (source) ports and are *automatically* detected on the output (destination) ports.
The module is intended to be used in a centrally managed system with 'out of band' management. i.e. The Multicon frame controller is used for routing and configuration which may be connected to a separate management LAN. The AES-IP-MUX does not use 'client to client' managed routing or session management.
Larger systems, spanning multiple audio LAN segments may be easily realised with Video iPath.

There are two operational modes for the modules.

1. iPath mode.
2. Multicon Matrix mode.

Large systems may be realised using classic ip routing or Video iPath and modules in iPath mode. This mode uses Multicon GYDA but the ip settings of the modules are controlled directly from iPath.

The Multicon Matrix mode can provide up to 128 2-channel audio connections. Routing control may be dynamically controlled with hardware router control panels or web panels, or may be static for point to point links.

The modules may also be equipped with an SFP providing optical link capability.

4.3 Connector Backplanes

Three backplane type are presently available with a standard network ethernet connector.

- AES-IP-MUX-C1 has two DB-25 connectors (16 AES3). Sync output on DIN 1.2/3.
- ADA-IP-MUX-C1 has two DB-25 connectors (8 AES3, 8 analogue audio)
- ADA-IP-MUX-C2 has 16 Molex KK connectors (Flashcase, 8 AES3, 8 analogue audio)

The SFP option adds an SFP cage and short fibres for connection to the backplane SC connectors. Standard MSA-SFPs with DOM (digital optical monitoring) may be used. The Nevion SFPs range may be found at:-

https://nevion.com/products/gbe-sfps-range

The ADA-IP-MUX-C1 also has a duplex LC optical connector.
Three single width backplanes in the VMUX range may be also used when the SFP option is present and *no* electrical ethernet connection is required.

- AES-VMUX-C1 has a single DB-25 connector (8 AES3)
- AES-VMUX-C2 has 16 Molex KK connectors (Flashcase, 16 AES3)
- ADA-VMUX-C2 has 12 Molex KK connectors (Flashcase, 4 AES3, 8 analogue audio)

The optical input provides an alternative network interface but the actual network connection is chosen automatically. The input switches if a suitable signal is not present, or if the signal disappears. The optical interface has priority if both interfaces are present on power-on.

The following dual width backplane is available for the AES-IPMUX with both optical and electrical ethernet connectors with all 16 AES channels available.

- AES-MUX-C1 has two DB-25 connectors (16 AES3) and dual optical connectors. Sync output on BNC.

4.4 The Nevion IP audio concept

The Nevion AES-IP-MUX produces a standard AES67 audio stream. The packets are *always* 2 channel with a sample rate of 48 kHz. The packet times are from 1/4ms to 4 ms and the audio word lengths may be 16 or 24 bit or transparent AES3 as used in Ravenna networks. The Sample rates and packet times are limited to reflect the typical usage in broadcast infrastructures requiring low latency, high quality and few 'flavours'. An infrastructure format should be one supported by ALL connected devices.

The audio data is packed into the AES67 packets in either a standard RTP L16 or L24 bit packing or an AES3 transparent packing mode using 32 bits per channel. The standard RTP audio packet does not support the V, U and C bit of AES3 audio, so Nevion has used the same AES3 packing format as the Ravenna IP audio network, which is based on a format described in firewire audio.

The AES3 sample clock is derived from the PTP time reference. The AES-IP-MUX is to be used in a synchronous broadcast infrastructure.

The AES-IP-MUX is intended for use in a network where gigabit ethernet links are available without unpredictable traffic causing packet loss or congestion. The high network bandwidth means that it does not need any of the bit-reduction codecs or forward error correction schemes, reducing the cost of the modules and keeping the latency of the transport low.

4.5 AES67

The AES-IP-MUX uses AES67 compliant streams. The audio is packed into RTP (Real time protocol) packets. The PTP (precision Time Protocol) is used as a centralized 'house' synchronisation clock. AES67 covers more than the stream format.

AES67 was created primarily as a standard for equipment interconnect. These connections require 'end-to-end client' management to manage audio content and connections. The Nevion IP audio concept is simpler than AES67 in that a central 'out of band' management (control network is not the same network used to convey the audio) is used to control the network. This is closer to a traditional broadcast facility audio infrastructure.

The AES-IP-MUX is simpler to use and more secure than most AES67 interfaces as it does not require the session control protocols or the automatic discovery mechanisms mandated by AES67.

4.6 Packet times and latency

A default packet time of 1 ms has been chosen as it is the mandated packet time for AES67 and results in a packet which is far below the normal Maximum-Transport-Unit (MTU) of 1500 octets. This is the normal maximum packet size on ethernet. 4 ms of 48 kHz AES3-packed bit stereo audio gives a packet payload of 1544 octets. Modern switches and the AES-IP-MUX have no problems handling the larger packets but some PC interfaces have problems with packets larger than 1500 octets. All other combinations of sample rates and packet sizes will give smaller packets but lower network efficiency due to the ethernet and IP wrappers.

The AES-IP-MUX has only one sample rate and four packet times as it is intended for broadcast use. This also reduces the complexity of the module. 1 ms AES3-packed packets only contain 384 octets of payload. All RTP-based audio packets have a packet overhead as shown in the following table.

Overhead type	octets
Interframe gap	12
Preamble	8
Ethernet header	14
IP header	20
UDP header	8
RTP	12
Frame checksum	4
Total	= 78

The network overhead ratio with 1ms AES3 packets at 48kHz is 78 / (384 + 78), about 17% of the traffic.
The total latency of audio over AES-IP-MUX modules is mostly due to the AES67 source buffer. This buffering collects the samples to fill each packet. The transmission of the packet to the remote module takes a very short time as long as the network is not busy. The module at the destination may begin to output the audio samples immediately after reception.

The speed of sound in air is about 340m/s so a 1ms delay is equivalent to the sound travelling 0.34m. The delay of the audio packet traversing a gigabit IP network with 1 ms packets only takes microseconds.
The distance from audio sources to microphones is often larger than 1 meter.
Foldback of audio over the IP network to commentators for example may give comb-filter effects but will not be detectable as echoes. The limit for detectable echoes is about 20ms. 4ms packets will still give acceptable latency for most usage while 1ms packets may be preferable for signals that will be used for foldback.

4.7 Unicast broadcast and multicast

Unicast is the standard method of addressing packets with both 'to' and 'from' ip addresses. The routing information is sent to both modules involved. i.e. Both source and destination modules must be told the other endpoint address of the connection. The source module must also detect the ethernet destination MAC address and uses the ARP protocol to do this.

Multicast is used to send the audio stream to all destinations on the current network segment. It is used when the audio source is to be sent to more than one destination. The modules have a packet filter so that any unneccessary multicast packets may be discarded without blocking the interface. Multicast produces more traffic on the network than unicast but does not require the source to send more than one packet per packet time period which greatly simplifies the module. A gigabit network can support more than 256 source channels *all* set to multicast.

Multicast is also used when the audio is to be sent to several destinations on other *network segments*. The multicast stream will be broadcast on the local network segment but edge routers will be aware of the stream and announce this to the network beyond the source segment. The audio will not be sent to any destination before the destination communicate to its router that it wishes to receive the stream. This negotiation is handled by the IGMP protocol.

Routing may also be acheived with the OpenFlow protocol which results in a centrally managed switched network. This network has a predictable latency, switching and traffic behaviour. This is used in the Nevion Video iPath control system.

4.8 Destination packet filtering

The audio packets have source and destination ip address information. Each destination channel will filter the incoming packets based on either the source or destination address. The value of the remote_ip_address setting controls which filter mode is used.

- Unicast addresses set a source address filter.
- Multicast addresses set a destination filter.
- If the remote and local ip addresses are the *same*, a destination filter will be used.

This done to avoid many ip address fields in the configuration when there is only one active value.

5 Specifications

5.1 Copper ethernet

Standard	1000-Base-T
Auto-negotiation	On
Duplex	Full
MDI	Auto MDI-X

5.2 Optical ethernet

Standard	1000-Base-X

Optical range depends on the output power of the transmitter, the sensitivity of the receiver of the receiver module and the attenuation of the eventual multiplexing filters.
See the Nevion SFP datasheets available from support or the Nevion web site.

5.3 AES3

Inputs and outputs according to	AES3-2003
Physical interface	110 ohm transformer balanced
Sampling frequency	48 kHz
Double sample rates using AES3	Single channel double sampling frequency mode is supported. i.e. 96kHz Mono signal using both sub-frames of the AES3 signal.

AES3 Outputs may be configured to switch off completely if there is no stream received (automute active) or generate a silent AES3 stream.

5.4 Sample buffers

The audio sample buffer is 512 stereo audio samples per channel irrespective of packet word length.

5.5 Audio Latency

Maximum buffer latency will be: (512 - current packet size).
Maximum possible latency with 0.25ms packets is: 512 -12 = 500 samples at 48kHz = 10.4ms.

The AES-IP-MUX will adjust the latency automatically to avoid stream corruption.
The audio latency will always be **either** the configured *target latency or* the sum of the network latency and the the packet time. The latter applies if the configured target value is too short.
The value used for each destination channel is reported on the Multicon info page. AES3 outputs are always time aligned at the AES3 left preamble.

5.6 DARS output

Locked to the PTP signal. Unbalanced AES11 or 48 kHz square wave.

Impedance	75 ohm
Terminated voltage	800mV
Coupling	AC
Maximum DC voltage	16V
Wordclock Polarity	(not)L/R

The wordclock signal may not work with other wordclock reference inputs being AC coupled and only 1.6V unterminated.

5.7 AES67 streams

Only 2 channel packets are produced or received. This corresponds to a direct mapping to an AES3 signal. Bundles of multiple channels must be handled by the control software.

Audio word length and packet times are recovered automatically in the AES67 receiver from received packets as long as the packets contain two channels.

5.7.1 Packet times

0.25 ms | 0.5 ms | 1 ms | 4 ms

0.33 ms is not supported and the 0.5 ms setting is not included in AES67.

5.7.2 Audio word length

16 bit | 24 bit | transparent AES3

Transparent AES3 packing is also used in Ravenna audio networks but is not included in AES67.

5.7.3 RTP timestamp offset

Always zero. All packet timestamps are referred to the TAI epoch.

5.8 PTPv2

5.8.1 Domain

The default PTP domain of 0 is highly recommended but may be changed to any value, 0 to 127 inclusive.

5.8.2 Modes

Delay request/response mode. Also called end to end (e2e) mode.

Multicast sync and delay requests/responses.

One step and two step servers may be used.

Sync packets should be 8 per second but may be as low as 1 per second or up to 16 per second.

Delay requests are initially sent every other sync packet until the transmission delay correction has been applied. The rate is then reduced to every 32 sync packets.

All delay requests have randomised launch times.

5.9 Other network protocols

ARP

- Probes are used to prevent the use of duplicate IP addresses.
- Requests are used by sources for MAC acquisition with unicast transmission.
- Gratuitous announcements are made from destinations to keep ARP caches from flushing in switches.

IGMPv2: report suppression supported.

ICMP Echo (FPGA firmware > 1.16) Responds to all host addresses.

5.10 Power

All AES set to inputs	
+5V	0.29A 1.5W
All AES set to outputs	
+5V	0.67A 3.4W
with SFP	
+5V	add 1.0 W

5.11 Multicon

The AES-IP-MUX is compatible with Multicon GYDA firmware 5.2 or later.

6 Configuration

The AES-IP-MUX **must** be conencted to an ethernet segment where a PTPv2 server with a domain matching the setting of the modules is connected. The default domain for AES67 is 0.

The module has two operational modes.

1. iPath mode.
2. Multicon matrix mode.

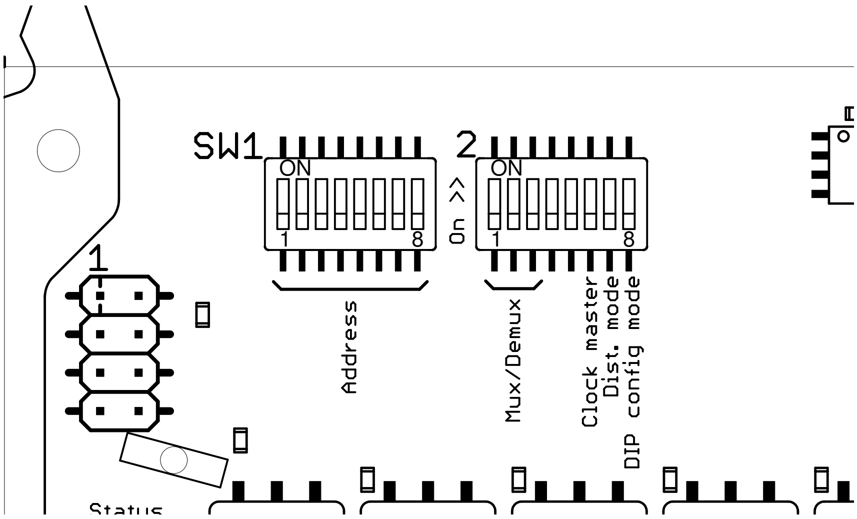


Figure 6.1 Module DIP switches.

The operational mode DIP switch SW2.7 is *always* read.
The DIP switch are only used to configure the module if SW2.8 is on.
The DIP switches are only read at power-on.

Table 6.1 DIP switch usage

Switch bank	Switches	Usage	Label
SW1	1-4	Local IP base address	Address
SW1	5-6	Remote IP base address (iPath only)	Address
SW2	1+2	Port direction	Mux/Demux
SW2	3+4	Audio packet time	Mux/Demux,-
SW2	5+6	Audio word length	-/Clock Master
SW2	7	Operational mode	Dist. mode
SW2	8	DIP config mode	DIP config mode

6.1 iPath mode

SW2.7 (labelled Dist Mode) is set on. This is the normal IP stream based mode. The GYDA web GUI may be used to monitor and control the IP stream parameters. DIP switches are used to set the parameters if SW2.8 is set on. Remove the module and switch off SW2.8, then re-insert the module to allow Multicon control of the configuration.
A Video iPath controller will control some of the parameters of the module through its own interface in Multicon GYDA. The module may be controlled with the local Muticon web GUI but once Video iPath is operational, some of the settings may be overwritten.

6.2 Multicon Matrix mode

Multicon Matrix mode is used for a system with up to 128 AES3 outputs with conventional router controls.
The DIP switches are used to set up the module with SW2.8 on, but SW2.7 must be set off.
All IP address settings are handled by the AES-IP-MUX modules and the system is viewed as a conventional matrix.
Up to 16 modules may be used in a system.IGMP snooping may be used but an IGMP querier must be present. IGMP snooping may also cause switching delays.
If dynamic routing is to be used, remove the module and switch off SW2.8 then reinsert the module.
The Nevion Configurator must be used to combine the modules to become part of the 128 x 128 stereo matrix.
The routing of the matrix can *only* be controlled with Vikinx web or hardware panels or in the Multicon virtual router web control GUI.

Take the following steps to configure the modules with DIP switches.

- Set the port directions on the modules to be all input, all output or 8 input and 8 output.
- Assign base address so that each *input* card/channel has a corresponding *output* card/channel with the SAME base address.
- Set the required audio encoding parameters.
- Connect all the AES-IP-MUX modules to a common ethernet segment.

If the sytem is to be used as a matrix:-

- Plug in all the modules with SW2.8 on.
- Remove the modules and switch SW2.8 off.
- Connect the Multicons. The multicons on different frames must be on a single network segment but it does not need to be the same segment as the audio.
- Use the Nevion Configurator connected to the Multicon network.
- All the Multicons with participating AES-IP-MUXes must be added to a new system.
- Create a 128x128 virtual matrix and add all the levels of the participating modules.

- Use the configurator to fill in default vaules for all the levels in the virtual matrix.
- Assign panels and web panels to control the virtual matrix.
- Upload the configuration to the Multicons and Panels.

The virtual router display in Multicon may show that some of the outputs report a breakaway. This is when a module is not synchronised with the virtual matrix. It will be synchronised the next time a command is sent controlling that output. A quick(er) way to synchronise many breakaways is to reload a stored configuration in Multicon(-> Configuration-> Flashlink-> Load).

6.3 Fixed Matrix mode

Used for simple point to point operation. Take the following steps to configure the modules with DIP switches.

- Configure the modules with DIP switches as described above.
- Leave SW2.8 on and SW2.7 set off.
- Connect all the AES-IP-MUX modules to a common ethernet segment.

The 'matrix routing' is fixed as a 'diagonal'.
The module with AES3 inputs with one base address will be connected to the module with AES3 outputs with the same base address.
All streams are unicast and point to point.
The Configurator steps are not necessary.

6.4 DIP switch descriptions

6.4.1 Base Addresses SW1 1-8

The module assigns a local IP address to each of the 16 channels. Audio routing involves two IP addresses, the local, and the remote.

- The *local* address is the IP address of the audio channel on that module.
- The *remote* IP address is where the audio should be sent to, or where it comes from; depending on the direction of the interface.
- AES3 inputs and outputs get different local IP addresses from the *same* base address setting. This is done to make it simple to make fixed configuration, point to point links. AES-input-card-1 will connect to AES-output-card-1 etc.

6.4.1.1 Matrix mode

There are eight possible base addresses in a 128x128 matrix so only the first 8 combinations (shown with * in the following table) are used for the base address. The routing matrix is set to a diagonal when SW2.8 is on.

6.4.1.2 iPath mode

The base addresses for both local and remote ports are set as shown in the following table.

Table 6.2 Base address DIP switch settings.

SW1.1	.2	.3	.4	Local base address	Matrix mode
SW1.5	**.6**	**.7**	**.8**	**Remote base address**	
off	off	off	off	0	*
off	off	off	on	16	*
off	off	on	off	32	*
off	off	on	on	48	*
off	on	off	off	64	*
off	on	off	on	80	*
off	on	on	off	96	*
off	on	on	on	112	*
on	off	off	off	128	
on	off	off	on	144	
on	off	on	off	160	
on	off	on	on	176	
on	on	off	off	192	
on	on	off	on	208	
on	on	on	off	224	
on	on	on	on	240	Do not use

Two ranges of IP addresses are used for the default addressing scheme. The sources have the lower range, e.g. 172.16.0.xx while the destinations have the range where the third IP octet is one larger,172.16.1.xx. These address ranges have been chosen as they are private IP address ranges like 10.x.x.x or 192.168.x.x but are seldom used.

IP addresses are written as 4 decimal numbers separated by dots. Each number is called an octet and can be from 0 to 255. Each channel's local IP address is derived is the following way:-

- The upper two octets are fixed to 172.16.
- The third octet in the IP address is given by the AES3 channel direction. 0 is an AES3 input and an AES67 source, 1 is an AES3 output and an AES67 destination.
- The last number is derived from the base address plus the channel number.

The default gateway address is set to 172.16.0.254.
The default network mask is set to \23 i.e. 255.255.254.0.

6.4.2 Audio channel direction SW2 1+2

The 16 audio channels are controlled in blocks of eight. There are only 4 valid combinations and these are set with DIP switches SW2 1+2.

Table 6.3 AES3 direction DIP settings

SW2.1 SW2.2	AES9-16	AES1-8	
off off	inputs	inputs	(ADA-IP-MUX + ADC-AES8 compatible setting)
off on	inputs	outputs	(ADA-IP-MUX + ADDA-AES8 compatible setting)
on off	outputs	inputs	
on on	outputs	outputs	(ADA-IP-MUX + DAC-AES8 compatible setting)

The module has two 8 in/ 8 out settings. Bi-directional connections may be realised with two IP-MUX cards with the same base addresses, but they must each have different settings of SW2 1+2. Unfortunately, only one of the settings is compatible with the ADA-IP-MUX backplanes when used with an ADDA-AES8. A point to point connection between two such module sets requires a manual configuration of the IP addresses on one of the modules with Multicon GYDA.
The Multicon GYDA debug terminal command to set ADDA-AES8 compatible direction mode is:- 'mtx 6 1 0'.

6.4.3 Packet time SW2 3+4

The packet time is set globally for the AES67 sources (AES3 inputs) with SW2.3+4 as shown in the following table.

SW2.3 SW2.4	Packet time (ms)
off off	0.25
off on	0.5
on off	1
on on	4

6.4.4 Audio word length SW2 5+6

The packet time is set globally for the AES67 sources with SW2.5+6 as shown in the following table.

SW2.5 SW2.6	Audio word length (bits)
off off	16
off on	24
on off	AES3
on on	AES3

6.4.5 Operational mode SW2.7

Labelled DIST.MODE. Matrix mode if the switch is off, iPath mode if the switch is on.

> **Warning:** This switch must be set correctly. The hot-swap configuration in GYDA will be lost if a module is inserted into a live system with this switch set incorrectly.

6.4.6 DIP config mode SW2.8

DIP SW2.8 is on: All settings will be overwritten with values set by the switches and Multicon will not be able to change the settings. If off, only SW2.7 is read.

6.4.7 Example 1:Point to point

Source module of point to point link, base address 0. 16 bit, 250us packets.

16 AES inputs/sources. Local adresses 172.16.0.1 to 172.16.0.16. Fixed Multicon matrix mode.

DIP SW1 setting off,off,off,off, off,off,off,off

DIP SW2 setting off,off,off,off, off,off,off,on (DIP config mode)

AES input 1 with local address 172.16.0.1 will send audio to remote address 172.16.1.1.

AES input 2 with local address 172.16.0.2 will send audio to remote address 172.16.1.2. etc.

The other end of the link will be set up the same except that SW2.1+2 will be on.

6.4.8 Example 2: Multicon matrix module Outputs 25 to 32

One end of a bi-directional link, base address 16.
8 AES inputs/sources. Adresses 172.16.0.17 to 172.16.0.24.
8 AES outputs/destinations. Adresses 172.16.1.25 to 172.16.1.32.
1ms packets, 24 bit.

DIP SW1 setting off,off,off,on, off,off,off,off

DIP SW2 setting off,on,on,off, off,on,off,on

The initial diagonal setting of the matrix will set the following routing for the source channels:-
AES input 1 with local address 172.16.0.17 will send audio to remote address 172.16.1.17.

AES input 2 with local address 172.16.0.18 will send audio to remote address 172.16.1.18.
etc. up to..

AES input 8 with local address 172.16.0.24 will send audio to remote address 172.16.1.24.

Then for the destinations:-
AES output 9 with local address 172.16.1.25 will receive audio from remote address 172.16.0.25.

AES output 10 with local address 172.16.1.26 will receive audio from remote address 172.16.0.26.
etc.

This module will have the AES3 outputs controlled by the corresponding outputs in the matrix 25 to 32.

The modules must be powered up in the frames and be found by the Multicon GYDA controller. They must then be removed and the DIP switch SW2.8 must be switched off to allow matrix control with Multicon. The module will have a 128x128 matrix available for control in the Multicon. This must be combined with the similar matrices in the other modules with the Nevion configurator as described earlier.

Hint: To set up two 8 input + 8 output cards to route to each other:-

- Set both cards to the same base address with DIP SW1.
- Set one card SW2.1+2 to 'off,on' and the other to 'on,off'

6.4.9 Example 3: iPath control Unicast

Source 172.16.0.4 to stream 24 bit 4ms packets to destination 172.16.1.36

6.4.9.1 Source

DIP SW1 setting off,off,off,off, off,off,on,off

DIP SW2 setting off,off,on,on, off,on,on,on (then replug and change to off)

The card is set to 16 inputs.

The local base address is 0 so channel AES4 has a local IP address of 172.16.0.4.

The remote IP address is that of the *destination* AES67 channel. The remote base address (lower than 36) is 32 so the destination will be channel AES4 on that module.

6.4.9.2 Destination

DIP SW1 setting off,off,on,off, off,off,off,off

DIP SW2 setting on,on,off,off, off,off,on,on (then replug and change to off)

The card is set to 16 outputs.

The local base address is 32 so channel AES4 has a local IP address of 172.16.1.36.

The remote IP address is that of the *source* AES67 channel. The remote base address is 0 so the source will be channel AES4 on that module.

Packets will be sent from the source channel to the destination as soon as there is an AES3 signal present on the source input. These packets will only flow between the two modules and the intermediate switches and will not be visible elsewhere on the switch ports unless a managed switch is used with port mirroring.

Unicast routing normally needs configuration changes on both source and destination channels.

6.4.10 Example 4: iPath control Multicast within an ethernet segment

Multicast routing is not possible to set with DIP switches but DIP switch SW2.7 must be on. The Channel directions must be set with the DIP switches and the module inserted with SW2.8 on. Multicon must 'see' the module before it is re-inserted with SW2.8 switched off.

The audio packet parameters may be set on the source module.

6.4.10.1 Source channel

The local IP address must be unique in the local ethernet segment.

The remote IP address must be that of a Multicast address. This is an IP address where the first octet is between 224 and 239 inclusive. The address *should* preferably be unique but the transmission will still function if it is not.

The local IP address must be within the network mask range.

6.4.10.2 Destination channel

The local IP address must be unique in the local ethernet segment.

The remote IP address may be that of the *source AES67 channel* or the Multicast address.

Packets will be sent from the source channel to the multicast pseudo-destination as soon as there is an AES3 signal present on the source input. These packets will be broadcast and will be visible elsewhere on the switch ports.

Multicast routing only requires configuration changes on the destination.

6.4.11 Ethernet media control

The default mode of operation is auto-switch between the optical and electrical media when the optical SFP is present. Only one input is in use at any time. If both inputs are connected at boot time, the optical connection will be selected first.

6.5 Multicon controls

6.5.1 IP addresses

In iPath mode, the local and remote IP addresses may be set for each channel. The Gateway and network mask may also be set. The mask is the number of 1s in the binary IPv4 number.

6.5.2 Target system latency

This may be set for destinations on the config page. The delay is set in samples but has a resolution of 8 samples and a maximum useful value of: (512 samples - packet sample size). 1ms packets have 48 samples etc.

The destination channel delay will be set to the target latency whenever the audio stream is interrupted.

The delay used will initially be the target latency but will automatically be adjusted to be just enough to avoid a buffer underrun if the initial setting is too small. A small value of target latency may be used to measure the latency in a network. The latency used for each channel is shown in iPath mode on the module 'info' page on Multicon. The latency will *always* include the number of samples in the AES67 packet.

6.5.3 DARS output format

The AES-IP-MUX module has an unbalanced output that may be used as a digital audio reference signal (DARS). The signal may be an 48kHz audio wordclock or an AES11 signal.

6.5.4 Automute

The AES3 Outputs may be configured to switch off completely if there is no stream received (automute active) or generate a silent AES3 stream.

6.5.5 Audio packet parameters

In iPath mode, the audio packet parameters may be set for each audio source stream, i.e. each AES3 input.

In multicon matrix mode, the packet time is set globally for all channels. The Audio word length may only be set with the DIP switches

6.5.6 PTP domain

The PTP domain may only be changed in Multicon in iPath mode.
The PTP domain will be reset to 0 when the DIP switches are used to configure the module.

6.6 Multicon GYDA

Multicon GYDA is the system controller for the Flashlink frames. The controller itself is inserted into one of the Flashlink frame slots where it communicates over RS422 with the cards in the system.
Up to 8 frames may be connected together with standard CAT-5 cables. Each frame must then be assigned an address with the DIP switch on the power connector backplane.
The controller card has an ethernet connector which must be connected to a TCP/IP network. The controller presents the Flashlink system as an SNMP server and an HTML webserver.

6.6.1 Hot-swap

GYDA stores the configuration of all the modules it controls in non-volatile memory whenever a command is issued. This allows a complex configuration to be stored and restored after a power loss. GYDA uses this to restore the configuration of a module if it is hot-swapped. This hot-swap reconfiguration only occurs if the module is inserted into a running system with an active GYDA. Modules that are hot-swapped must also be of the same type as only one configuration is stored for each slot. A new type of module will overwrite the configuration of the previous module if the type is different.
The DIP switch settings of the AES-IP-MUX *must* be the same for the modules involved in a hot-swap because some of the configuration is only set with the DIP switches.

6.6.2 Operational modes

The setting for operational mode (DIP SW2.7) will completely change the GUI (and type) of the module. The two 'personalities' are shown below.

6.6.2.1 Multicon matrix mode

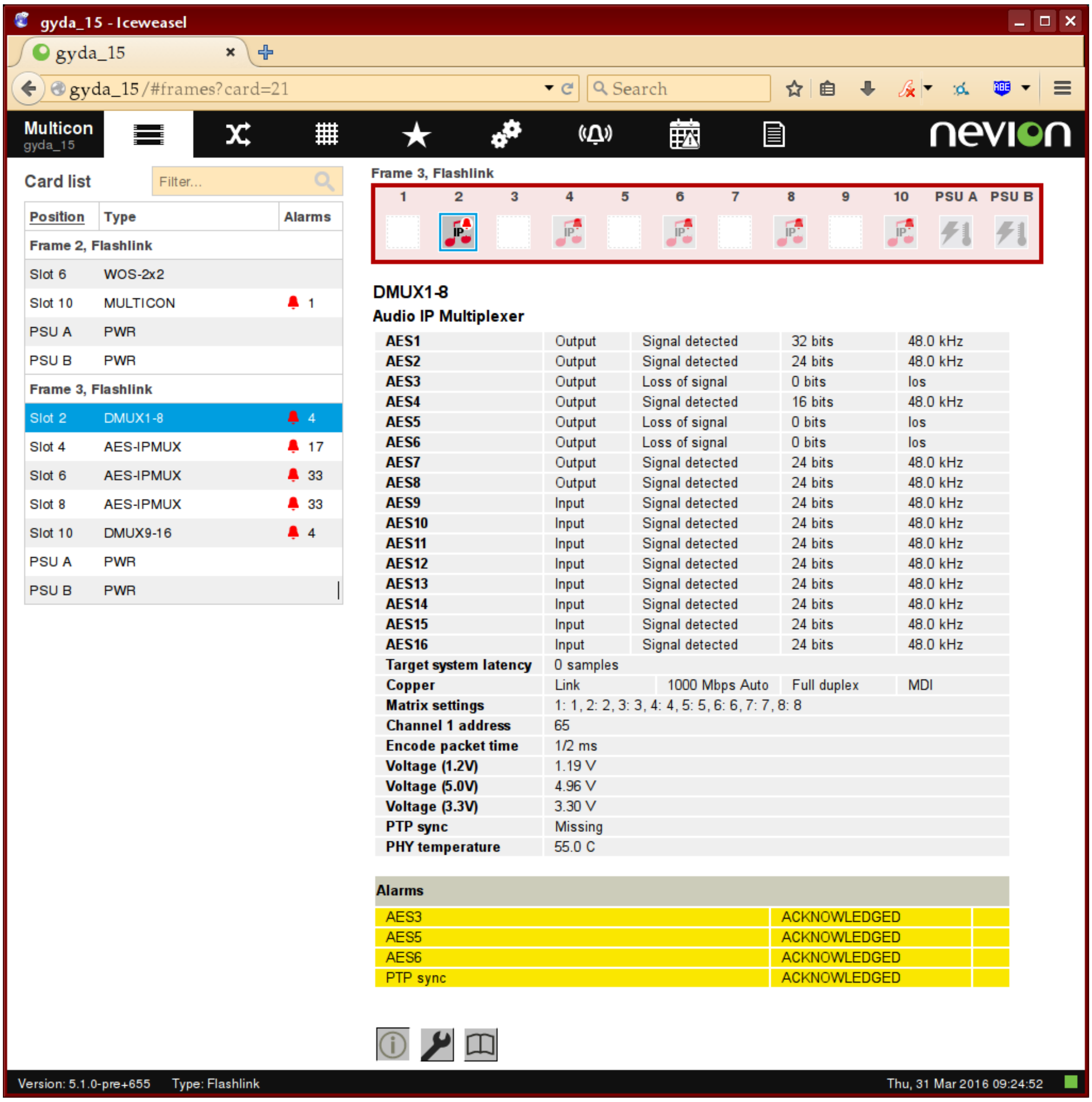


Figure 6.2 Multicon matrix mode info page

The GYDA web configuration GUI for each module cannot be used to control routing. The info page of the module shows the current matrix setting and signal activity.

Routing of the system may be controlled with the Multicon virtual matrix page.

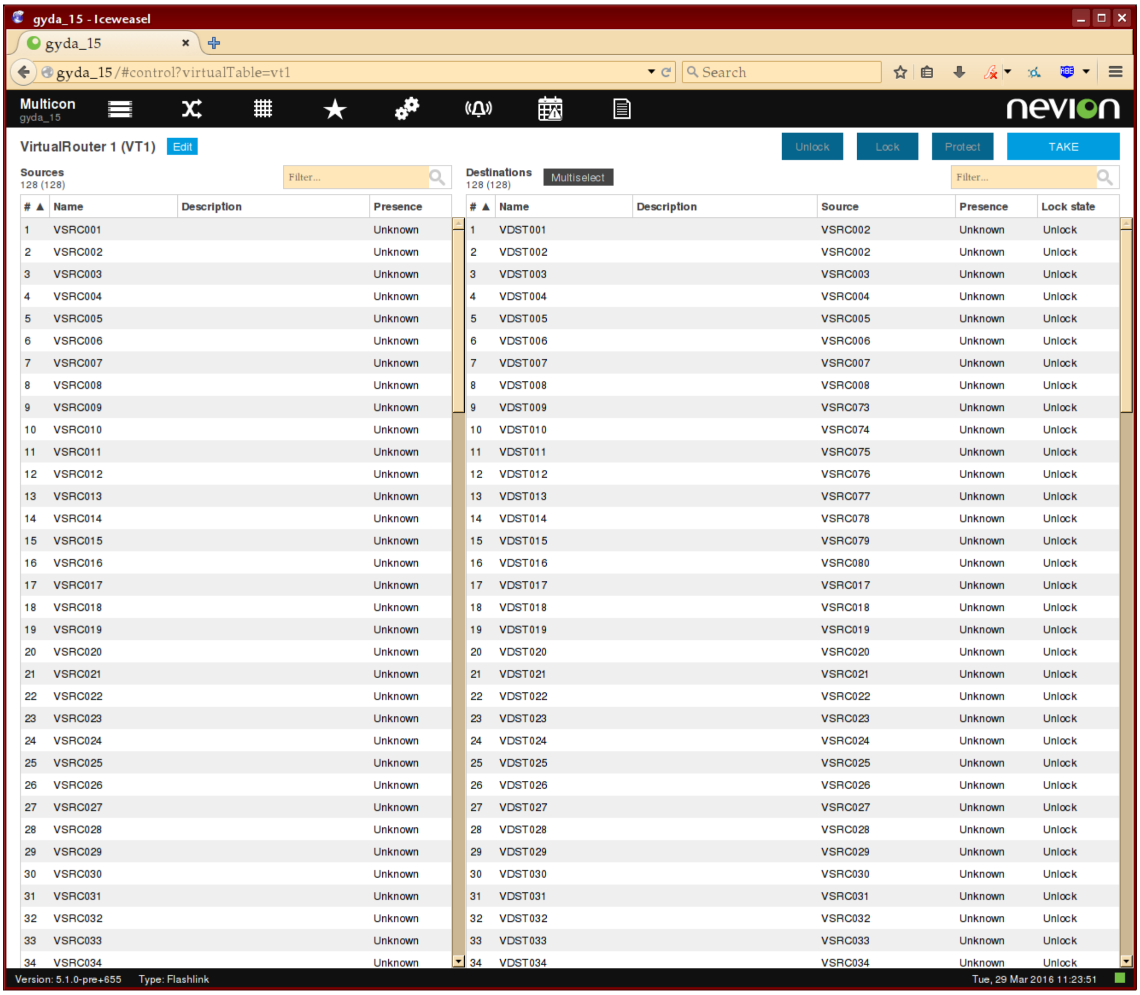


Figure 6.3 Multicon virtual matrix control GUI

6.6.2.2 iPath mode

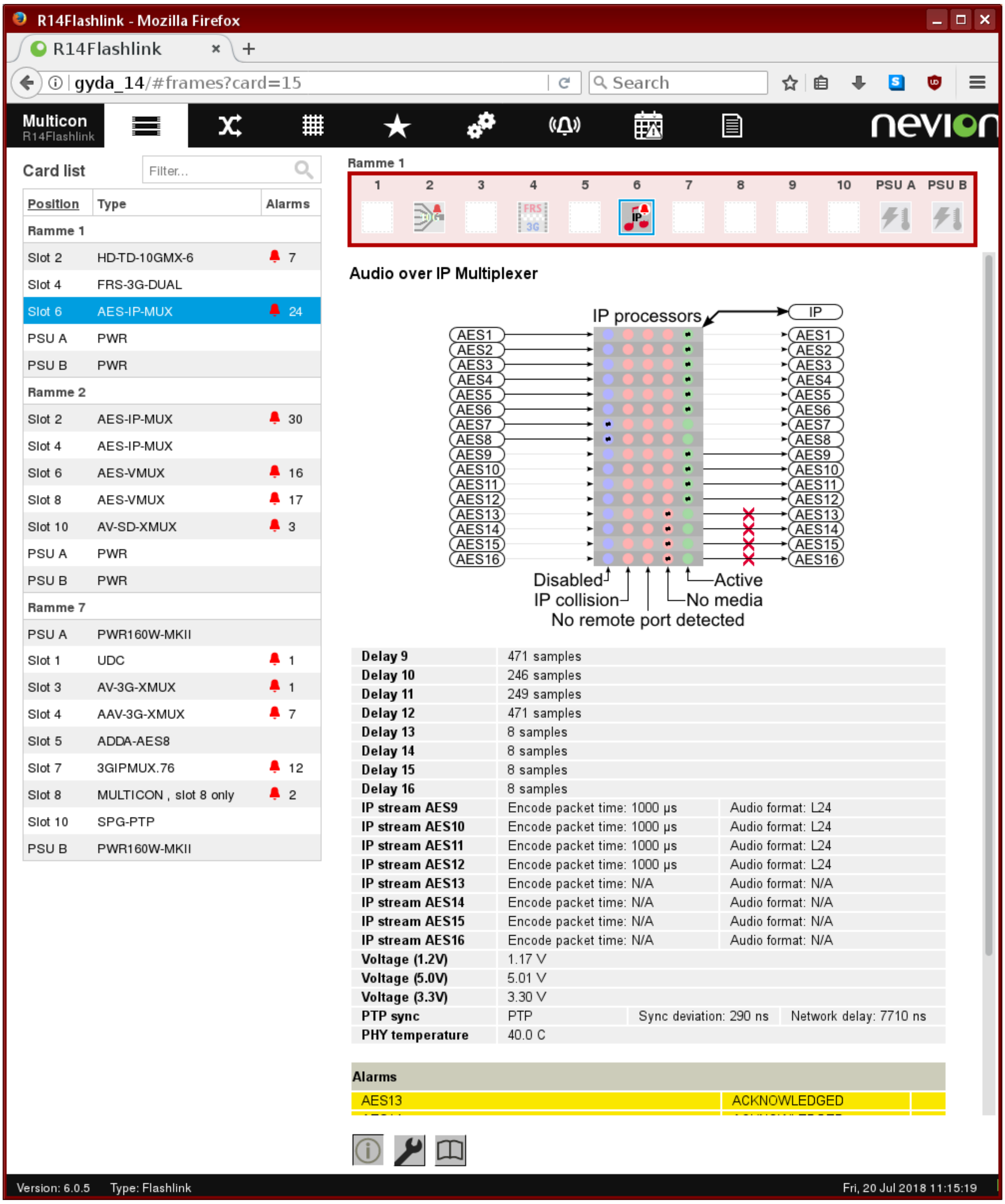


Figure 6.4 iPath mode info page

Values shown on the info page are the values reported from the destination channels.

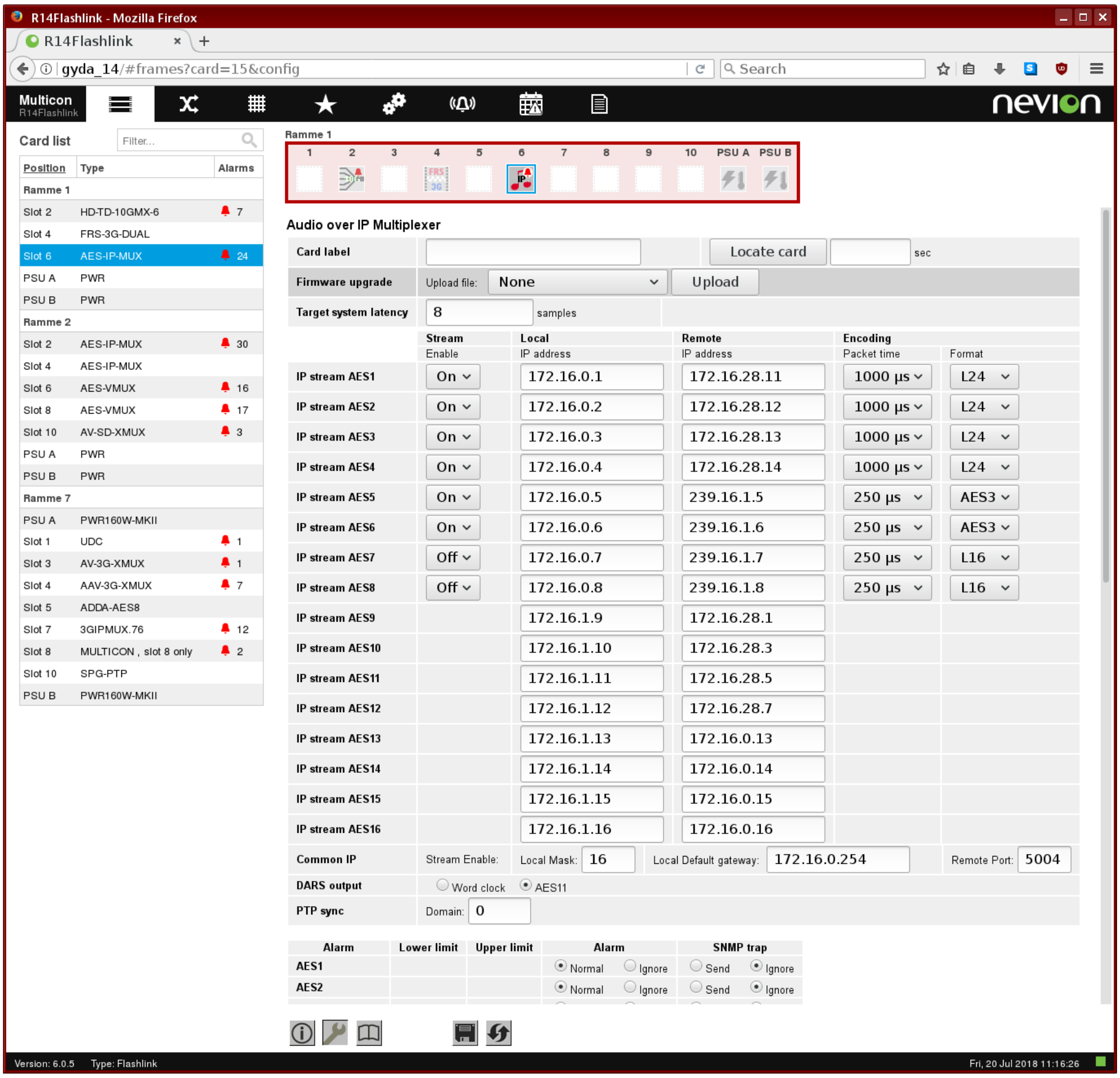


Figure 6.5 iPath mode configuration page

The encoding parameters of the sources are set on the configuration page.

7 Connections

7.1 Standard backplanes

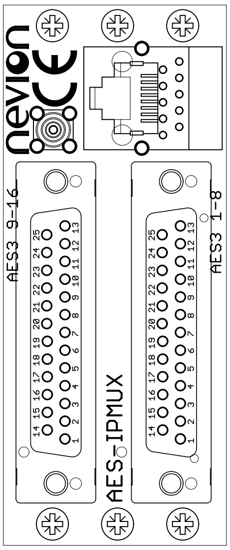


Figure 7.1 AES-IP-MUX-C1: DB-25 AES3 ports 1-16 and DIN sync

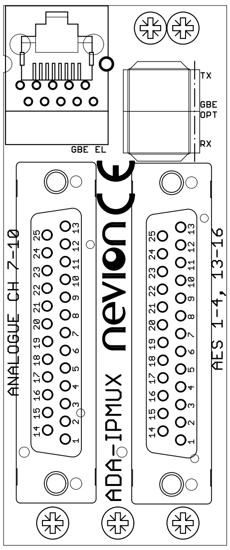


Figure 7.2 ADA-IP-MUX-C1: DB-25 AES3 ports 1-4 and 13-16, analogue audio, duplex LC connector

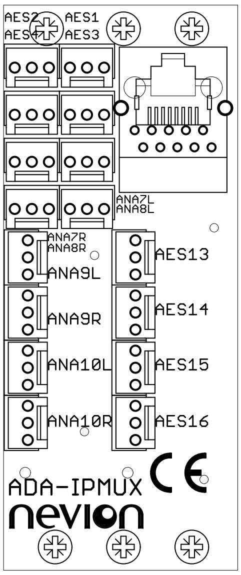


Figure 7.3 ADA-IP-MUX-C2: Flash-case, Molex KK AES3 ports 1-4 and 13-16, analogue audio

7.2 SFP SC option backplanes

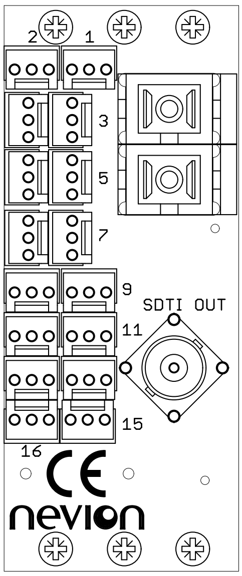


Figure 7.4 AES-VMUX-C2: Flash-case, Molex KK AES3 ports, BNC sync, dual optical connectors

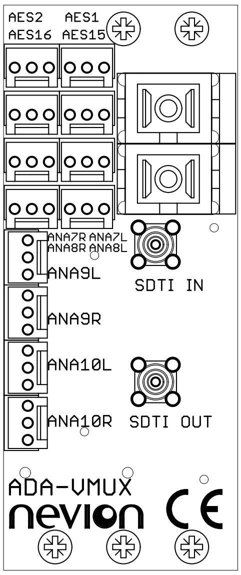


Figure 7.5 ADA-VMUX-C2: Flash-case, Molex KK AES3 ports1,2,15 and 16; DIN sync, dual optical connectors

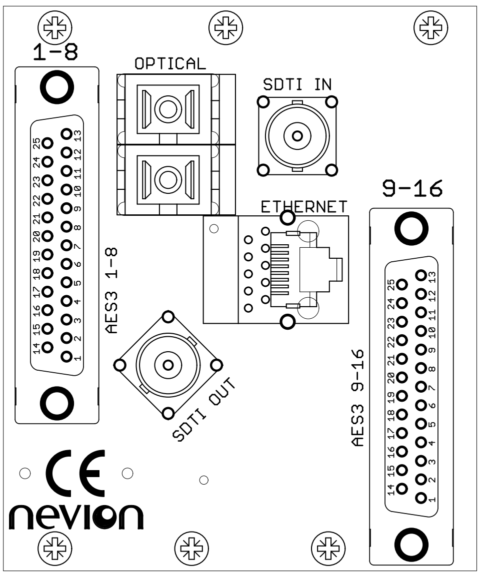


Figure 7.6 AES-MUX-C1: DB-25 AES3 ports 1-16, BNC sync and dual optical connectors

7.3 Pin connections

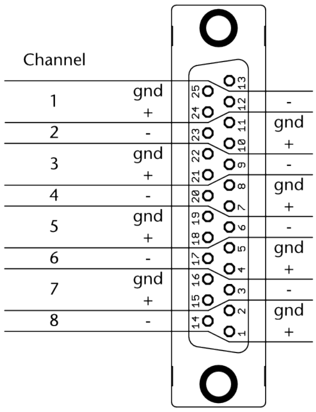


Figure 7.7 The backplanes use the TASCAM standard pin assignment for 8 balanced audio channels on the female DB-25.

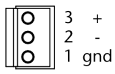


Figure 7.8 Molex KK connections

7.3.1 ADA-IP-MUX channel connections

7.3.1.1 ADDA-AES8

ADDA	Analogue Connector	IP-MUX channel	IP-MUX direction
Output 1L	1	7 Left	output
Output 1R	2	7 Right	"
Output 2L	3	8 Left	output
Output 2R	4	8 Right	"
Input 1L	5	9 Left	input
Input 1R	6	9 Right	"
Input 2L	7	10 Left	input
Input 2R	8	10 Right	"

- C1 Digital connector

Connector channel	IP-MUX channel	IP-MUX direction
1	1	output
2	2	output
3	3	output
4	4	output
5	13	input
6	14	input
7	15	input
8	16	input

7.3.1.2 ADC-AES8

ADC	Analogue Connector	IP-MUX channel	IP-MUX direction
Input 1L	1	7 Left	input
Input 1R	2	7 Right	"
Input 2L	3	8 Left	input
Input 2R	4	8 Right	"
Input 3L	5	9 Left	input
Input 3R	6	9 Right	"
Input 4L	7	10 Left	input
Input 4R	8	10 Right	"

- C1 Digital connector

Connector channel	IP-MUX channel	IP-MUX direction
1	1	input
2	2	input
3	3	input
4	4	input
5	13	input
6	14	input
7	15	input
8	16	input

7.3.1.3 DAC-AES8

DAC	Analogue Connector	IP-MUX channel	IP-MUX direction
Output 1L	1	7 Left	output
Output 1R	2	7 Right	"
Output 2L	3	8 Left	output
Output 2R	4	8 Right	"
Output 1L	5	9 Left	output
Output 1R	6	9 Right	"
Output 2L	7	10 Left	output
Output 2R	8	10 Right	"

- C1 Digital connector

Connector channel	IP-MUX channel	IP-MUX direction
1	1	output
2	2	output
3	3	output
4	4	output
5	13	output
6	14	output
7	15	output
8	16	output

8 LEDs

The module has four LEDs.

8.1 Status LED

This turns red for 1 second when power is applied and then turns green when the FPGA code has been loaded correctly.

The LED is green

- the module is programmed and functioning normally.

The LED is orange

- the module is not programmed or is in the process of being programmed

The LED is red

- there is something wrong with the module power supply levels OR
- the FPGA code has not loaded correctly

8.2 LAN link LED

The LED is green

- The active ethernet port has link established

The LED is orange

- the active ethernet port is negotiating link

The LED is red

- there is no link

8.3 PTP LED (marked EDH)

The LED is green

The module has full lock to PTP and delay request responses are being received.

The LED is orange

The module has received PTP sync packets and is locking.

The LED is red

The module has not received PTP sync packets or they have been absent for more than the announce timeout, typically 3 times the announce period..

8.4 Optical option LED

The LED is orange

- an optical SFP option is not present OR
- the SFP is not a recognised SFP.

The LED is red:

- the input signal is too low or high power OR
- the laser has failed

The LED is green:

- the optical input signal is present and correct AND
- the laser is operating with the correct power

8.5 On-site re-programming.

The module may be re-programmed on site with a GYDA Multicon system controller. Firmware and instructions will be provided by Nevion support when necessary.

9 General environmental requirements

The equipment will meet the guaranteed performance specification under the following environmental conditions:

Operating room temperature range	0°C to 45°C
Operating relative humidity range	<90% (non-condensing)

The equipment will operate without damage under the following environmental conditions:

Temperature range	10°C to 55°C
Relative humidity range	<90% (non-condensing)

10 Product Warranty

The warranty terms and conditions for the product(s) covered by this manual follow the General Sales Conditions by Nevion, which are available on the company web site:
http://www.nevion.com

Appendix A Materials declaration and recycling information

A.1 Materials declaration

For product sold into China after 1st March 2007, we comply with the "Administrative Measure on the Control of Pollution by Electronic Information Products". In the first stage of this legislation, content of six hazardous materials has to be declared. The table below shows the required information.

組成名稱 Part Name	Toxic or hazardous substances and elements					
	鉛 Lead (Pb)	汞 Mercury (Hg)	镉 Cadmium (Cd)	六价铬 Hexavalent Chromium (Cr(VI))	多溴联苯 Polybrominated biphenyls (PBB)	多溴二苯醚 Polybrominated diphenyl ethers (PBDE)
AES-IPMUX	O	O	O	O	O	O
AES-IPMUX-SFP	O	O	O	O	O	O
ADA-IPMUX	O	O	O	O	O	O
ADA-IPMUX-SFP	O	O	O	O	O	O

O: Indicates that this toxic or hazardous substance contained in all of the homogeneous materials for this part is below the limit requirement in SJ/T11363-2006.

X: Indicates that this toxic or hazardous substance contained in at least one of the homogeneous materials used for this part is above the limit requirement in SJ/T11363-2006.

This is indicated by the product marking:

A.2 Recycling information

Nevion provides assistance to customers and recyclers through our web site:
http://www.nevion.com/. Please contact Nevion's Customer Support for assistance with recycling if this site does not show the information you require.
Where it is not possible to return the product to Nevion or its agents for recycling, the following general information may be of assistance:
Before attempting disassembly, ensure the product is completely disconnected from power and signal connections.
All major parts are marked or labeled to show their material content.
Depending on the date of manufacture, this product may contain lead in solder.
Some circuit boards may contain battery-backed memory devices.