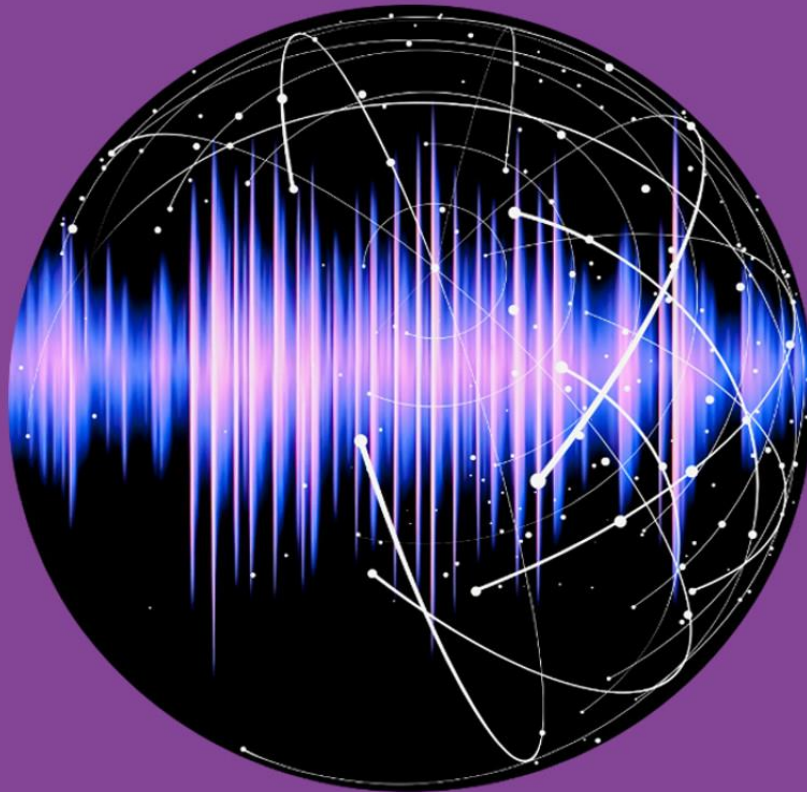


Audio Live User Guide



Version History

| Date | Version | Release by | Reason for changes |
|------------|---------|------------|--|
| 02/12/2016 | 1.0 | J Metcalf | Document originated (1 st proposal) |
| 09/08/2017 | 1.1 | J Metcalf | Updated for V1.0.1.28 |
| 02/01/2018 | 1.2 | J Metcalf | Updated for V1.2.1.8 |
| 05/02/2018 | 1.3 | J Metcalf | Modified running order of document. |
| 14/06/2018 | 1.4 | J Metcalf | Updated for V1.3.0.16 |
| 14/06/2019 | 1.5 | J Metcalf | Updated for V1.5.0.14 |
| 29/07/2019 | 1.6 | J Metcalf | Updated for V1.5.1.0 |
| 12/09/2019 | 1.7 | J Metcalf | Updated for V1.5.2.10 |
| 17/12/2019 | 1.8 | J Metcalf | Updated for V1.5.2.18 |
| 30/01/2020 | 1.9 | J Metcalf | Minor correction |
| 08/03/2021 | 1.10 | J Metcalf | Add note about default gateway. Rebrand |
| 08/10/2021 | 1.11 | J Metcalf | Update for V.1.5.3.42 |

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1. Definitions, Acronyms and Abbreviations

Table 1: Table of Terminology

| Term | Definition |
|------------------------------|--|
| DDS domain | Acronym: Data Distribution service |
| Essence | A general term used to describe a media component; video, audio or metadata. |
| Flow | Sequence of RTP packets of a single essence. The term Flow can be considered synonymous with Stream. |
| IP | The Internet Protocol (IP) is the principal communications protocol in the Internet protocol suite for relaying datagrams across network boundaries. Its routing function enables internetworking, and essentially establishes the Internet. |
| IPMI | The Intelligent Platform Management Interface (IPMI) is a set of computer interface specifications for an autonomous computer subsystem that provides management and monitoring capabilities independently of the host system's CPU, firmware (BIOS or UEFI) and operating system. |
| Multicast enabled IT network | In computer networking, multicast (one-to-many or many-to-many distribution) is group communication where information is addressed to a group of destination computers simultaneously. |
| NTP | Network Time Protocol (NTP) is a networking protocol for clock synchronization between computer systems over packet-switched, variable-latency data networks. |
| Orbit | GV's Automated Control and Monitoring suite. |
| PTP | The Precision Time Protocol (PTP) is a protocol used to synchronize clocks throughout a computer network. On a local area network, it achieves clock accuracy in the sub-microsecond range, making it suitable for measurement and control systems. |
| QFSP | Quad Small Form-factor Pluggable The QSFP is a compact, hot-pluggable transceiver used for data communications applications. It interfaces networking hardware to a fibre optic cable or active or passive electrical copper connection. It is an industry format jointly developed and supported by many network component vendors, allowing data rates from 4x4 Gbit/s. |
| QFSP+ | Enhanced Quad Small Form-factor Pluggable. 4x10Gb which can be combined into a single 40Gb Ethernet link or QSFP28 is a single 100Gb or 4x25Gb |
| RollCall | SAM's propriety control and monitoring protocol common across the product range |
| RTP | Real Time Transport Protocol (RTP) is a network protocol for delivering audio and video over IP networks. RTP is used in communication and entertainment systems that involve streaming media. |
| SFP | Small Form-factor Pluggable - A small transceiver that plugs into the SFP port of a network switch and connects to Fibre Channel and Gigabit Ethernet (GbE) optical fibre cables at the other end. |

| Term | Definition |
|---------------|--|
| SFP+ | Enhanced Small Form-factor Pluggable: The SFP+ is an enhanced version of the SFP that supports data rates up to 16 Gbit/s. |
| Source | Originator of one or more flows i.e. a set of one or more senders |
| Spigot | Generic term for a source or destination |
| Stream | Generic term for a collection of flows. In context of Audio, an AES 67 stream carries audio flows also known as channels |
| GV Live Host | Host software application which manages one or more GV Live Agents. |
| GV Live Agent | Virtual software instantiation which adopts a product type to perform a particular media processing function. |

2. Introduction

This guide is designed to give simple step-by-step instructions explaining how to use the Audio Live IP router.

Audio Live is a multi-channel audio routing and processing solution for the IP domain. It supports AES67 / RFC3190 (L24) / SMPTE2110-30 streams and offers a powerful 2048x2048 channel routing matrix.

Routing of audio streams requires that input audio streams be synchronous, PTP timestamps must be present in the extended headers and the RTP timestamps must be present and locked to the PTP timestamps for successful operation. Where inputs are encountered that are asynchronous, Audio Live offers the ability to synchronize up to 16 asynchronous audio stream inputs. Audio Live offers the unique ability to be configured as an asymmetric router. The input and output can be independently configured from the following configurations:

Table 2: Asymmetric Routing Configurations

| Routing Configurations |
|-----------------------------------|
| 32 Streams x 64 Channels |
| 64 Streams x 64 Channels |
| 64 Streams x 32 Channels |
| 128 Streams x 16 Channels |
| 192 Streams x 8 Channels |
| 8 x 64, 32 x 16, 128 x 8 Channels |
| 8 x 64, 64 x 16, 64 x 8 Channels |
| 16 x 64, 16 x 16, 96 x 8 Channels |
| 16 x 64, 32 x 16, 64 x 8 Channels |
| 16 x 64, 56 x 16, 32 x 4 Channels |
| 8x64, 96x16 Channels |
| 8x64, 160x8 Channels |
| 16x64, 64x16 Channels |

Audio Live offers an input realignment buffer, enabling various upstream transmission delays present on each input stream to be compensated for and aligned with the designated reference stream. The maximum size of the input tolerance buffer ranges between 20ms and 1 seconds depending on which router input configuration is selected. The Timestamp tolerance is affected by the System Input and Output channel configuration - see Table 4 in Appendix D.

The reference stream can be manually forced or an automatic mode can be used. Timestamps are then used to synchronize all the flows and provide flow control at the output. From a time stamp perspective Audio Live is completely transparent.

Additional channel delay can be added to the input and output channels if required up which ranges between 2 seconds and 5 seconds before the audio router again depending on the System Input and Output channel configuration – see Table 5 in Appendix E.

The control application for Audio Live and the GV Live software, is RollCall Control Panel (RollCall Lite may be used and is available as a free download). As this is not a RollCall/RollCall Lite tutorial, it is assumed that RollCall Lite is already installed and operational. For larger systems we recommend the use of GV's IP Routing Control Server (V9.8.14 or above) to configure spigots and the Audio Live internal cross point.

3. System Requirements / Prerequisites

1. **IT Network:** In order to successfully use Audio Live, a multi-cast enabled IP network is required. To send/receive 'AES 67' Audio Stream, Audio Live must have the same DDS Domain ID as the source/destination devices.

2. **RCP (RollCall Control Panel):**

The control surface for Audio Live can be either:

- Orbit.
- RollCall RCP (RollCall Control Panel).

Orbit: The Orbit Lite control application is available as a free download from the GV Website here:

[GV Orbit Lite Download](#)

Orbit should be installed on a remote PC either running Windows 7 or Windows 10.

RollCall Control Panel is available within the Orbit application.

At the time of writing, the latest RollCall Suite version is: V5.3.0

RollCall RCP: RCP (RollCall Control Panel) is a component part of the RollCall application. RollCall should be installed on a remote Windows computer.

At the time of writing, the latest RollCall Suite version is: V4.19.8

RollCall Lite, of which RCP is a component part, is available as a free download from the GV Website here:

[RollCall Lite - Download](#)

Note that there must be an IP network connection between the PC running RCP and the Alchemist Live Host machine.

The default IP address on NIC 1 is 192.168.0.1 (see Appendix A for more details).

Note that this manual demonstrates control via RollCall RCP. However, Orbit RCP templates are virtually identical to those of RollCall RCP.

3. **IP Routing Controller:** In order to successfully and easily create specific routes we recommend the use of GV's IP Routing Controller.

4. **Audio Live License:** For successful operation of Audio Live a valid license is required.

5. **Input Streams:** For synchronous input streams, PTP timestamps must be present in the extended headers and the RTP timestamps must be present and locked to the PTP timestamps. Up to 16 asynchronous input streams can be received utilizing the 'Synchronize' feature.

4. GV Live Architecture

The **GV Live** software consists of a **Host**, **Agent** and **License Server**.

The **GV Live Host** manages the **GV Live Agents**. Typically there would be one host per physical compute instance. From a RollCall perspective, the **GV Live Host** can be considered as a virtual IQ frame.

GV Live Agents are spawned by the Host and adopt a product type based on the nominated product license. Each agent has its own unique RollCall address just like a module would in a frame and can therefore be considered as a virtual slot.

The **License Server** is the local daemon that manages the product license/licenses. The daemon must be present on start for the product to successful use of the product. If for any reason the license disappears there is a 7 day grace period before the product will stop working.

5. The Console

5.1 Introduction

Audio Live application runs on Linux Ubuntu OS. A Console is provided to offer simplified configuration of key elements of the system.

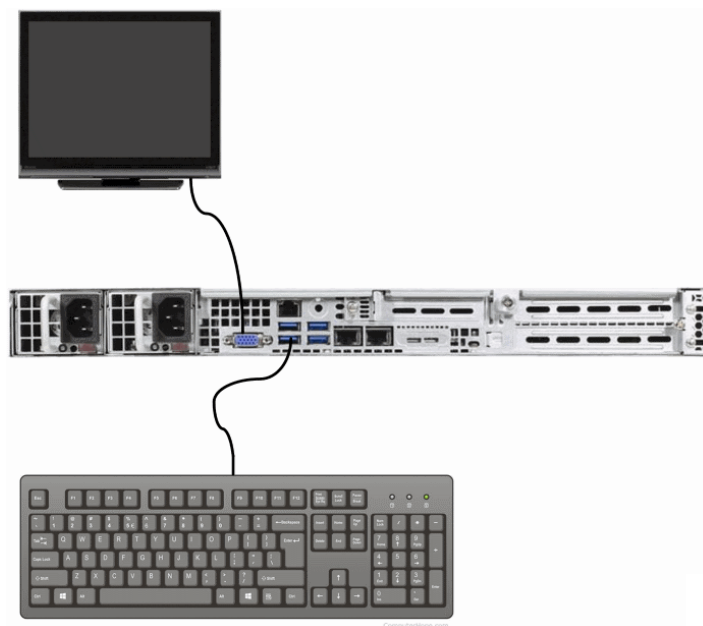
Using the Console the following items can be configured:

- Configure Network – NIC 1 is defined to have a Static IP and cannot be set to DHCP. NIC's 2, 3 & 4 may be configured to be either Static IP or DHCP.
- Advanced Network configuration - allows the User to create static routes on top of the existing default gateway.
- Network Diagnostics
- The time and set may be set.
- An upgrade process may be initiated.
- A new license may be installed.
- The Status of the system may be displayed.
- Operational Logs may be retrieved.
- The Server may be power-cycled.
- The Server may be shut down.

5.2 Accessing the Console

To access the Console connect a monitor to the on-board VGA and connect a keyboard.

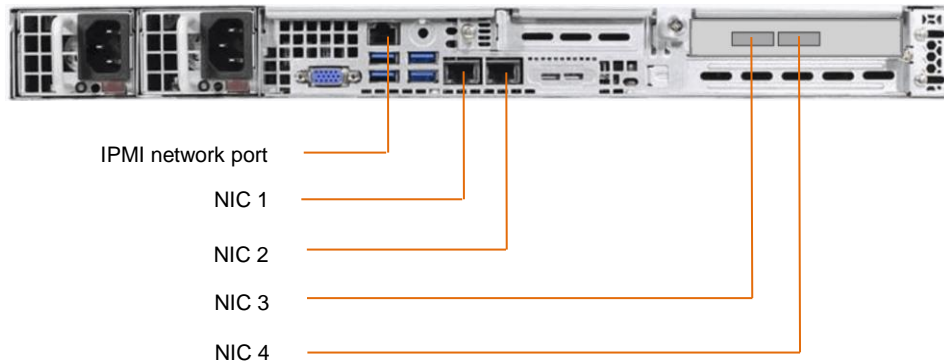
1. Connect a Monitor & Keyboard to the Audio Live Host machine:



Note: a mouse is not required

2. Connect a Network cable to the Server:

Note that there are 5 network ports. Connect the LAN to either NIC1 or NIC2.



- NIC 1 (eno1) is factory configured with the default Static IP Address and /30 subnet mask: **192.168.0.1/30**
NOTE - this port maybe set to another static IP address, mask and gateway but cannot be set to DHCP.
- NIC 2 (eno2) is factory configured as DHCP. This port maybe configured as static or DHCP.
- NIC 3 (ens1f1) - second Mellanox ConnectX4 port and is factory configured with a static address and subnet mask
- NIC 4 (ens1f0) - first Mellanox port ConnectX4 port and is factory configured with a static address and subnet mask
- IPMI port may be ignored at this point (see Appendix A, below).

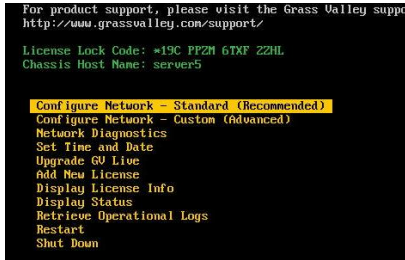
3. Boot up the Audio Live host machine. Once the boot sequence has completed, the Console will be displayed:



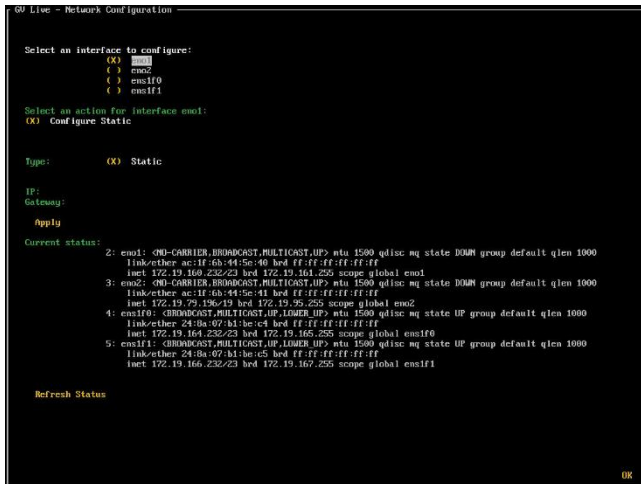
5.2.1 Configure Network – Standard (Recommended)

Using the up/down arrows on the keyboard, highlight the item to be configured.

In this case: **Configure Network**



Press return and the Network Configuration menu will now be displayed:



Note that in the Console:

- **NIC 1** equates to **eno1** *
- **NIC 2** equates to **eno2**
- **NIC 3** equates to **ens1f0 ConnectX4 - Port 1**
- **NIC 4** equates to **ens1f1 ConnectX4 - Port 2**

* This port maybe set to another static IP address, mask and gateway, but **cannot** be set to DHCP.

Using the up/down arrows on the keyboard, highlight the NIC to be configured. In this example NIC 2 is selected:

Now press the *space bar* on the keyboard to select the NIC.

Using the up/down arrows on the keyboard, move down to the **Configure** section. Note that the menu options will update to:

```
GV Live - Network Configuration

Select an interface to configure:
  ( ) eno1
  (X) eno2
  ( ) ens1f0
  ( ) ens1f1

Select an action for interface eno2:
(X) Configure Static/DHCP
  ( ) Enable
  ( ) Disable

Type:          (X) Static
              ( ) DHCP

IP:
Gateway:
```

Select the desired configuration.

If the selection is **Static**, press the **Tab** button again and the display will update again, this time with fields to enter an **IP Address** and **Gateway** visible:

```
GV Live - Network Configuration

Select an interface to configure:
  ( ) eno1
  (X) eno2
  ( ) ens1f0
  ( ) ens1f1

Select an action for interface eno2:
(X) Configure Static/DHCP
  ( ) Enable
  ( ) Disable

Type:          (X) Static
              ( ) DHCP

IP:
Gateway:
```

Using the up/down arrows on the keyboard, highlight **IP** (IP will turn white):

```
Select an action for interface eno2:
(X) Configure
  ( ) Enable
  ( ) Disable

Type:          ( ) DHCP
              (X) Static

IP:
Gateway:
```

Type in the required address:

```
SAM Live - Network Configuration

Select an interface to configure:
  ( ) eno1
  (X) eno2
  ( ) ens1f0
  ( ) ens1f1

Select an action for interface eno2:
(X) Configure Static/DHCP
  ( ) Enable
  ( ) Disable

Type:          (X) Static
              ( ) DHCP

IP:            192.168.200.2/24
Gateway:      192.168.200.1
Apply
```

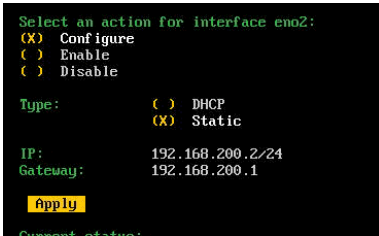
Note:

In Linux the IP Address **must** be appended with: **/xx**

For example: **192.168.200.2/24** with a Gateway **192.168.200.1**

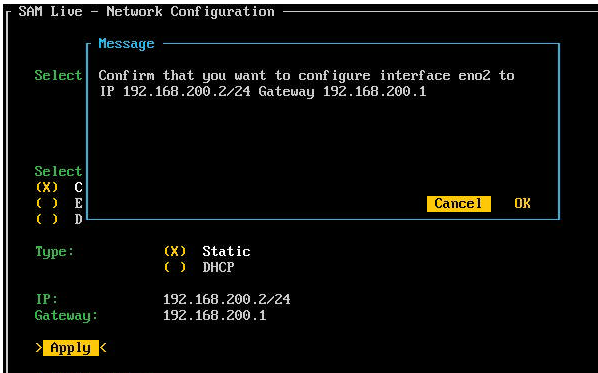
If it is desired to communicate over an associated/trunked/bridged network, then a Gateway Address will be required:

Note – only one NIC should have a Gateway defined. In circumstances where more complex routing is required, then Static routes will need to be defined using the **Configured Network – Custom (Advanced)** menu (see part 5.2.2 below).



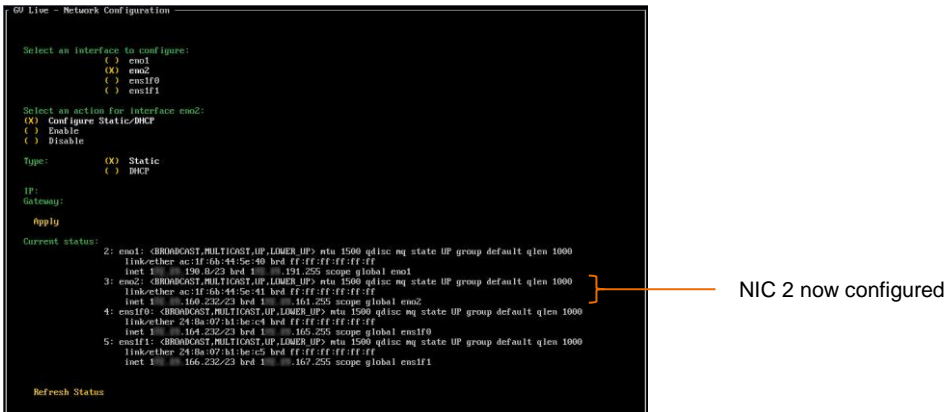
Once **IP Address** and **Gateway** have been configured, using the up/down arrows on the keyboard, navigate to and highlight: **Apply**

Press return and a confirmation of the configuration change will be asked for:



Select **OK**.

Note that confirmation of the new Static IP Address is displayed in the **Current status** section:



If the NIC has been configured to be DHCP, the IP Address will be displayed in the Current Status section.

Next configure the static addresses for **NIC3 (ens1f1)** and **NIC4 (ens1f0)**

```

GV Live - Network Configuration

Select an interface to configure:
  ( ) em0
  ( ) em0c
  (X) ens1f1
  ( ) ens1f11

Select an action for interface ens02:
  (X) Configure Static/DHCP
  ( ) Enable
  ( ) Disable

Type:
  (X) Static
  ( ) DHCP

IP:
Gateway:

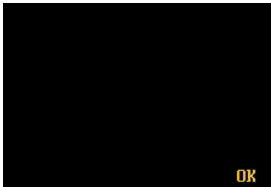
Apply

Current status:
2: em0: BROADCAST,MULTICAST,UP,LOWER_UP  mtu 1500 qlistc mq state UP group default qlen 1000
   link-ether ac:1f:68:44:5e:40 brd ff:ff:ff:ff:ff:ff
   inet 10.190.8/23 brd 10.190.155 scope global em0
3: em0c: BROADCAST,MULTICAST,UP,LOWER_UP  mtu 1500 qlistc mq state UP group default qlen 1000
   link-ether ac:1f:68:44:5e:41 brd ff:ff:ff:ff:ff:ff
   inet 10.160.232/23 brd 10.160.255 scope global em0c
4: ens1f0: BROADCAST,MULTICAST,UP,LOWER_UP  mtu 1500 qlistc mq state UP group default qlen 1000
   link-ether 24:8a:07:b1:be:c4 brd ff:ff:ff:ff:ff:ff
   inet 10.164.232/23 brd 10.164.255 scope global ens1f0
5: ens1f1: BROADCAST,MULTICAST,UP,LOWER_UP  mtu 1500 qlistc mq state UP group default qlen 1000
   link-ether 24:8a:07:b1:be:c5 brd ff:ff:ff:ff:ff:ff
   inet 10.166.232/23 brd 10.166.255 scope global ens1f1

Refresh Status
  
```

NIC 3, 4 configured

To exit the **Configure Network** menu, navigate down to the **OK** symbol (bottom right corner of display):



Select **OK**. This will return the display to the top menu of the Console.

5.2.2 Configure Networks – Custom (Advanced)

Using the up/down arrows on the keyboard, highlight **Configure Networks – Custom (Advanced)**

```

For product support, please visit the Grass Valley sup
http://www.grassvalley.com/support/

License Lock Code: *19C PP2M 6T8F 22HL
Chassis Host Name: server5

Configure Network - Standard (Recommended)
Configure Network - Custom (Advanced)
Network Diagnostics
Set Time and Date
Upgrade GV Live
Add New License
Display License Info
Display Status
Retrieve Operational Logs
Restart
Shut Down

```

Press return and the **Custom Network Configuration** menu will now be displayed:

```

GV Live - Advanced Network Configuration

Custom Debian style network interfaces script editor:
(Please ensure em01 is given a static IP address to maintain the license lock code)

auto em02
iface em02 inet static
address 172.19.79.196
netmask 255.255.254.0
gateway 172.19.71.20

auto em01
iface em01 inet static
address 172.19.160.232
netmask 255.255.254.0

auto ens1f1
iface ens1f1 inet static
address 172.19.166.232
netmask 255.255.254.0

auto ens1f0
iface ens1f0 inet static
address 172.19.164.232
netmask 255.255.254.0

```

This menu allows the User to create static routes on top of the existing default gateway.

Note that if the **Custom Network Configuration** is used, the ability to configure the network adaptors via RollCall will no longer be possible.

5.2.3 Network Diagnostics

Using the up/down arrows on the keyboard, highlight: **Network Diagnostics**

```

For product support, please visit the Grass Valley support
http://www.grassvalley.com/support/

License Lock Code: *19C PYZH 6TYP Z2HL
Chassis Host Name: server5

Configure Network - Standard (Recommended)
Configure Network - Custom (Advanced)
Network Diagnostics
Set Time and Date
Upgrade GV Live
Add New License
Display License Info
Display Status
Retrieve Operational Logs
Restart
Shut Down

```

Select return to enter the **Network Diagnostics** menu:

```

GV Live - Network Diagnostics

Routing Table: Kernel IP routing table
Destination  Gateway      Genmask      Flags Metric Ref  Use: iface
default      172.19.71.20  0.0.0.0      UH    0     0    0 ems0
172.19.64.0  *            255.255.224.0 U    0     0    0 ems0
172.19.166.0 *           255.255.254.0 U    0     0    0 ens1
172.19.164.0 *           255.255.254.0 U    0     0    0 ens1f0
172.19.166.0 *           255.255.254.0 U    0     0    0 ens1f1

Ping -
Note: To request a response via the high-speed media interfaces, it is necessary to stop the GV Live services
Stop GV Live

IP Name/IP:
Destination:

Ping
Response:

```

The Network Diagnostic menu allows the User to 'ping test' the configured network adaptors.

Note: the GV Live Services must be stopped, prior to a ping test being performed on the high speed networks (ens1f0 & ens1f1).

5.2.4 Set Time and Date

Using the up/down arrows on the keyboard, highlight **Set Time and Date**:

```
For product support, please visit the Grass Valley support
http://www.grassvalley.com/support/

License Lock Code: *19C PP2M 6TKF 2ZHL
Chassis Host Name: server5

Configure Network - Standard (Recommended)
Configure Network - Custom (Advanced)
Network Diagnostics
Set Time and Date
Upgrade GV Live
Add New License
Display License Info
Display Status
Retrieve Operational Logs
Restart
Shut Down
```

Select return to enter the **Set Time and Date** menu:

```
GV Live - Time and Date

Current:
    Local time: Thu 2021-07-29 10:44:45 BST
    Universal time: Thu 2021-07-29 09:44:45 UTC
    RTC time: Thu 2021-07-29 09:44:45
    Time zone: Europe/London (BST, +0100)
    Network time on: yes
    NTP synchronized: yes

New:

Time:      10:44:45
Date:      29 July, 2021
Set time and date

Timezone:  (X) Europe/London
           ( ) Europe/Luxembourg
           ( ) Europe/Madrid
           ( ) Europe/Malta
           ( ) Europe/Mariehamn
           - more -

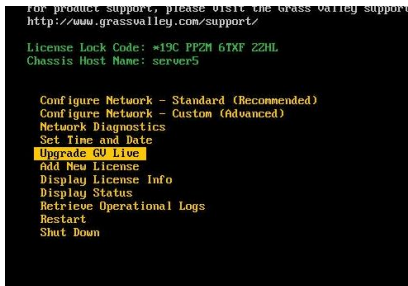
Set timezone
```

Select **OK** to complete the process, and return to the top menu.

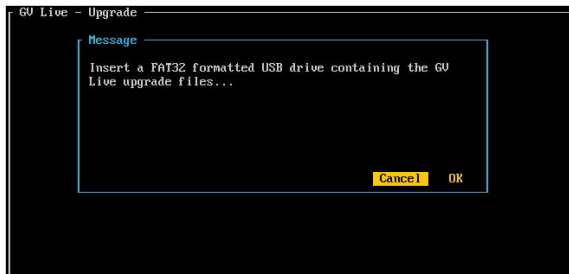
5.2.5 Upgrade GV Live

The Software Update will need to be uploaded to a USB Drive.

Using the up/down arrows on the keyboard, highlight **Upgrade GV Live**:



Press *return* and the system will prompt will appear to insert the USB Drive.



Follow the on screen instruction to complete the process.

5.2.6 Add New License

The License File will need to be uploaded to a USB Drive.

Using the up/down arrows on the keyboard, highlight **Add New License**:

```
For product support, please visit the Grass Valley support
http://www.grassvalley.com/support/

License Lock Code: *19C PP2M 6TXF 2ZHL
Chassis Host Name: server5

Configure Network - Standard (Recommended)
Configure Network - Custom (Advanced)
Network Diagnostics
Set Time and Date
Upgrade GV Live
Add New License
Display License Info
Display Status
Retrieve Operational Logs
Restart
Shut Down
```

Press **return** and a prompt will appear to insert the USB Drive.

```
GV Live - Install License
Message
Insert a FAT32 formatted USB drive containing the
license file you want to apply...
Cancel OK
```

Follow the on screen instruction to complete the process.

5.2.7 Display License Info

```

For product support, please visit the Grass Valley support
http://www.grassvalley.com/support/

License Lock Code: *19C PP2M 6TXF ZZHL
Chassis Host Name: server5

Configure Network - Standard (Recommended)
Configure Network - Custom (Advanced)
Network Diagnostics
Set Time and Date
Upgrade GV Live
Add New License
Display License Info
Display Status
Retrieve Operational Logs
Restart
Shut Down
    
```

When selected, the details of the installed License will be displayed.

```

GV Live - License Information

License Lock Code: *19C PP2M 6TXF ZZHL

Mellanox:      (PN) Part number: MXX4160-BCA1
               (EC) Engineering changes: AB
               (SN) Serial number: HT1715R00915

Licenses:      Audio Live Base
               License Version: 1.0
               Quantity: 1
    
```

5.2.8 Display Status

Using the up/down arrows on the keyboard, highlight: **Display Status**

```

For product support, please visit the Grass Valley support
http://www.grassvalley.com/support/

License Lock Code: *1LY 2W3Y TJ69 6BEB

Configure Network - Standard (Recommended)
Configure Network - Custom (Advanced)
Network Diagnostics
Set Time and Date
Upgrade GV Live
Add New License
Display License Info
Display Status
Retrieve Operational Logs
Restart
Shut Down
    
```

Press **return** and the **Status** page will be displayed:

```

GV Live - Status

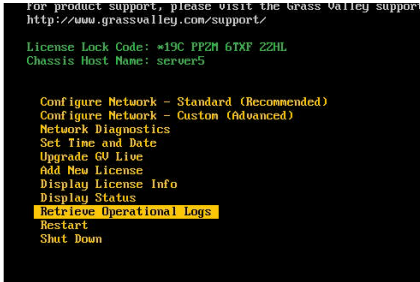
GV Live Host      Active: active (running) since Mon 2021-07-26 16:14:14 BST; 2 days ago
                 Tasks: 66
                 Memory: 93.9M
BTP Handling      Active: active (running) since Tue 2021-07-27 08:39:06 BST; 2 days ago
                 Tasks: 20
                 Memory: 25.8M
Licensing         Active: active (running) since Mon 2021-07-26 16:14:14 BST; 2 days ago
                 Tasks: 1
                 Memory: 6.3M

RAID Status
  Raid Level : raid1
  Array Size : 51200000 (48.03 GiB 52.43 GB)
  State : clean
  Active Devices : 2
  Working Devices : 2
  Failed Devices : 0
  Spare Devices : 0
    
```

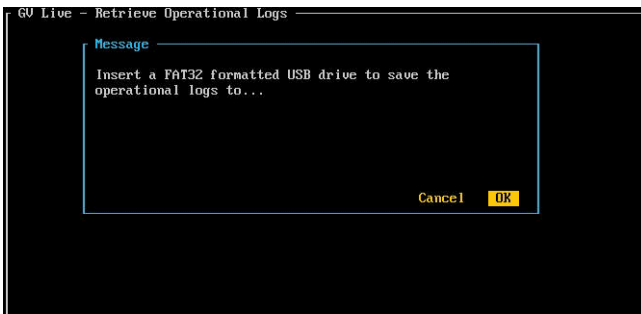
5.2.9 Retrieve Operational Logs

The Operational logs will be downloaded onto a USB Drive.

Using the up/down arrows on the keyboard, highlight: **Retrieve Operational Logs**

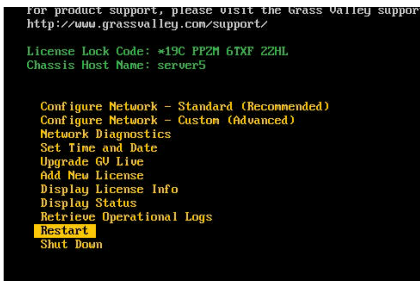


Press **return** and a message will appear prompting the Operator to insert a USB drive:



5.2.10 Restart

Using the up/down arrows on the keyboard, highlight: **Restart**



Press **Return** to power-cycle the Audio Live Host machine.

5.2.11 Shut Down

Using the up/down arrows on the keyboard, highlight: **Shut Down**

```
For product support, please visit the Grass Valley support
http://www.grassvalley.com/support/

License Lock Code: *19C PP2M 6TKF 2ZHL
Chassis Host Name: server5

Configure Network - Standard (Recommended)
Configure Network - Custom (Advanced)
Network Diagnostics
Set Time and Date
Upgrade GO Live
Add New License
Display License Info
Display Status
Retrieve Operational Logs
Restart
Shut Down
```

Press **Return** to power-down the Audio Live Host machine.

6. Installing RollCall Lite

The Audio Live product does not have a dedicated control interface, but instead utilizes RCP.

Note – this guide is not a RollCall tutorial.

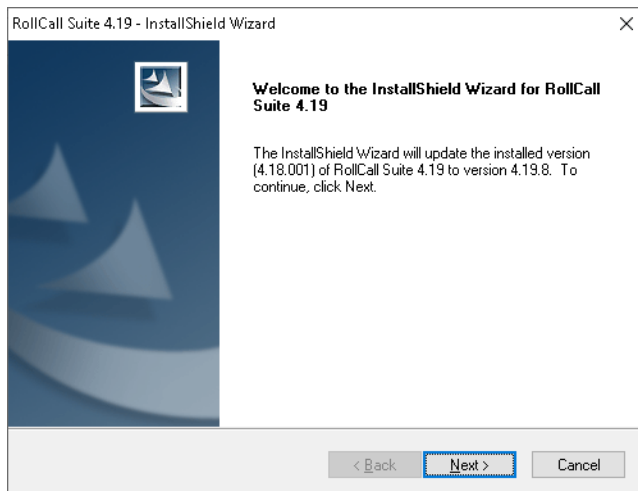
RCP is the control application for the GV product range and is available to GV customers free of charge. RollCall Lite will most likely be installed on a remote PC with a network connection to the host machine.

The following procedure describes how to install RCP as part of a RollCall installation. RCP can also be run from within an Orbit installation (see section 8).

Installation is very easy. Just launch the RollCall Suite Installer and follow the Install Wizard. Note that part way through the install process, the question will be asked; which RollCall components are to be installed.

All components can be installed but from an Audio Live perspective, the important components are:

- Control Panel
- IP Control Proxy Service



Follow the RollCall Installer wizard to complete the process.

Once installed, control of GV Live products is possible using Control Panel. It is recommended to use Control Panel via IP Proxy service

Routes can be made to and from Audio Live using GV's IP Routing Controller.

Note – a 'free of charge' version of RollCall is available, called RollCall Lite. This can be used to control Audio Live. The full RollCall Suite can also be used, but this is a chargeable option.

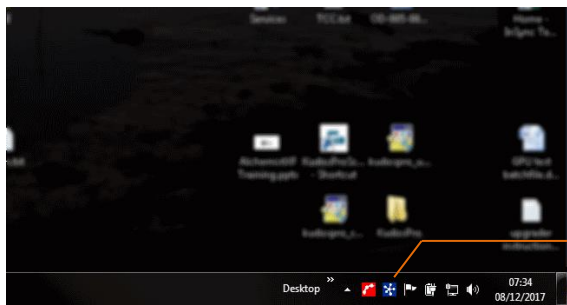
7. Using RCP via IP Proxy Service

7.1 Configuring RollCall Proxy

The following procedure will describe how to set up RollCall IP Control Proxy Service, on a remote PC on the same network as the Audio Live host machine.

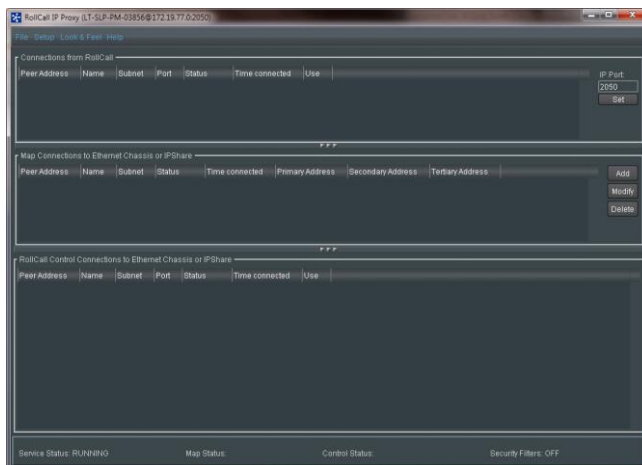
Note that the advantage of using the IP Control Proxy Service is that control of multiple GV products can be managed from a single RollCall control panel session, without the need to keep changing the connection IP Address. By using IP Control Proxy Service, Control Panel can be set to Localhost and all devices configured in the IP Control Proxy Service will be visible in the connection pane of Control Panel.

1. On the remote PC, where RollCall has been installed, note the RollCall launch icon, present on the task bar
(Note – if it's not visible, go to: **Start/All Programs/Snell** and the RollCall Proxy application can be picked from the listed RollCall components)

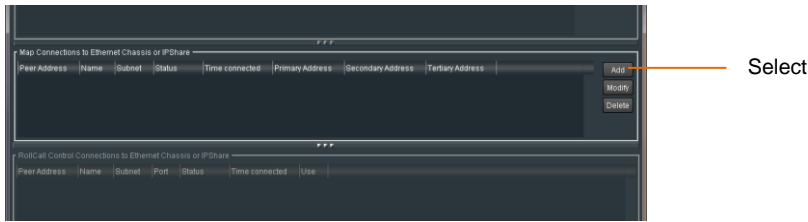


RollCall Proxy
application
launch

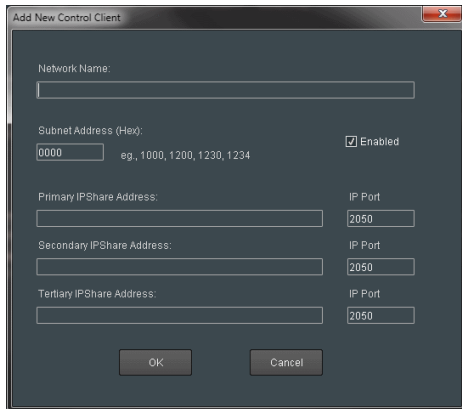
Launch RollCall Proxy. The **RollCall IP Proxy** window will open:



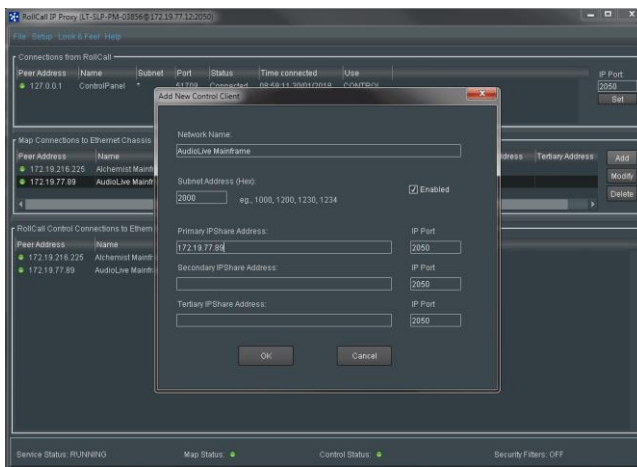
2. In the Map Connections to Ethernet Chassis or IP Share pane, select **Add**:



The **Add a new Client** window will appear:



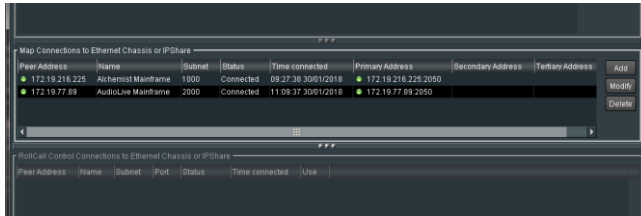
3. Populate the **Add a new Client**:



- The **Network Name** is usually an intuitive name that will allow easy identification of the device within the **Map Connections to Ethernet Chassis or IP Share** pane.
- The Subnet address needs to be set. The actual hex value set isn't important, as long as it unique. If the Audio Live is the first item configure into RollCall Proxy, use the hex address **1000** (as in this example). If it's the second, use hex address **2000** (assuming 1000 has already been used).
- Enter the **Primary IPShare Address**. This is the IP Address of the Audio Live host machine. If this is not known at this point in time, see Appendix 1 below.

Click **OK**.

The configured Client will now appear in the **Map Connections to Ethernet Chassis or IPShare** pane.



The screenshot shows a software interface with a table titled "Map Connections to Ethernet Chassis or IPShare". The table has columns for Peer Address, Name, Subnet, Status, Time connected, Primary Address, Secondary Address, Tertiary Address, and buttons for Add, Modify, and Delete. Two rows are visible, both with a green status indicator and "Connected" status.

| Peer Address | Name | Subnet | Status | Time connected | Primary Address | Secondary Address | Tertiary Address | Add | Modify | Delete |
|----------------|---------------------|--------|-----------|---------------------|---------------------|-------------------|------------------|-----|--------|--------|
| 172.19.216.225 | Alchemist Mainframe | 1000 | Connected | 09:27:38 30/01/2018 | 172.19.216.225.2050 | | | | | |
| 172.19.77.89 | AudioLive Mainframe | 2000 | Connected | 11:08:37 30/01/2018 | 172.19.77.89.2050 | | | | | |

Below this table is another section titled "Local Control Connections to Ethernet Chassis or IPShare" with a similar table structure, but it is currently empty.

Note: this example also includes a configured connection for an Alchemist Live, as well as an Audio Live, hence both appear in the:

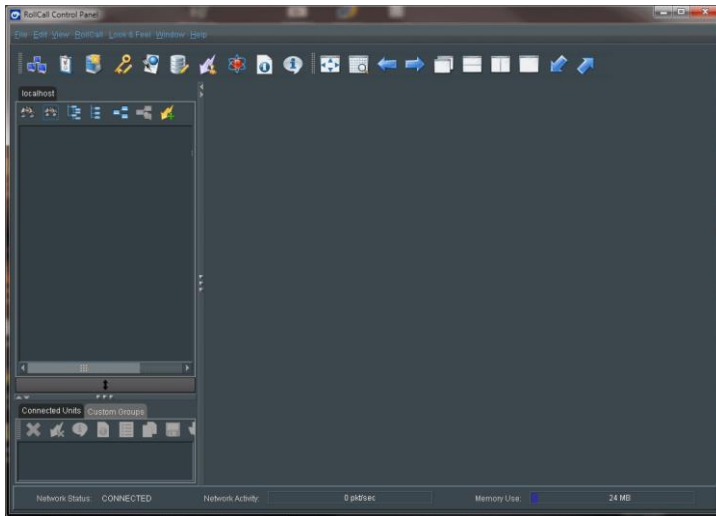
Map Connections to Ethernet Chassis or IPShare pane.

7.2 Connecting to Audio Live Using RollCall Lite

1. On the remote PC, open **RollCall Control Panel**, via the desktop icon:



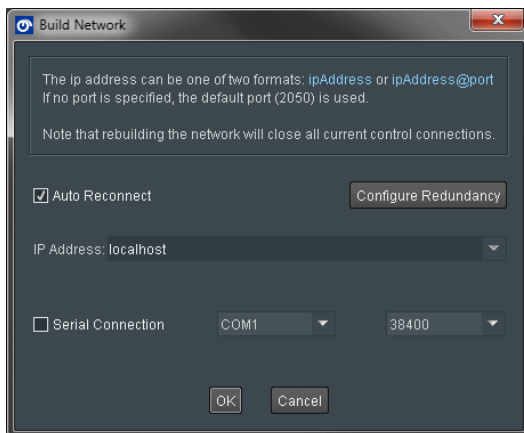
The RollCall Control Panel will open:



2. Click on the **Connection** icon:



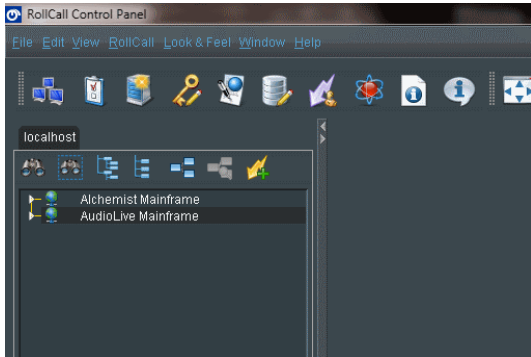
The **Build Network** Window will now appear:



The IP Address will default to: **localhost**. This is the required setting. If the IP Address is set to anything else, set it to **localhost**.

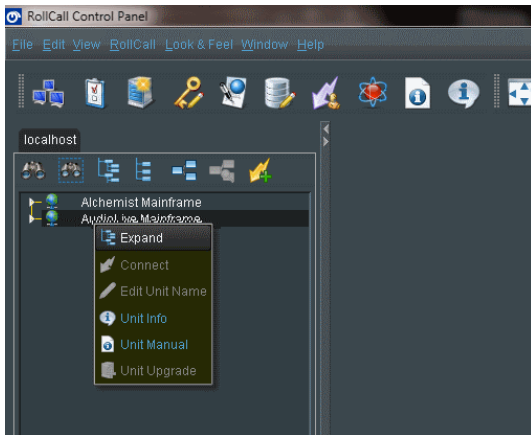
Select **OK**.

Audio Mainframe will appear in the RollCall connection pane:

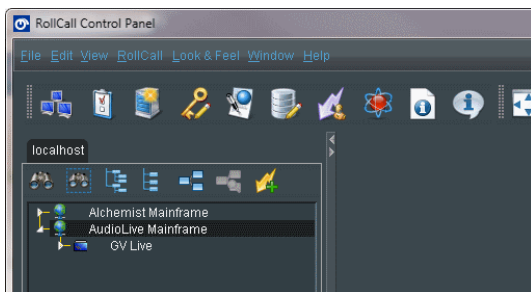


3. Right click on **Audio Mainframe**.

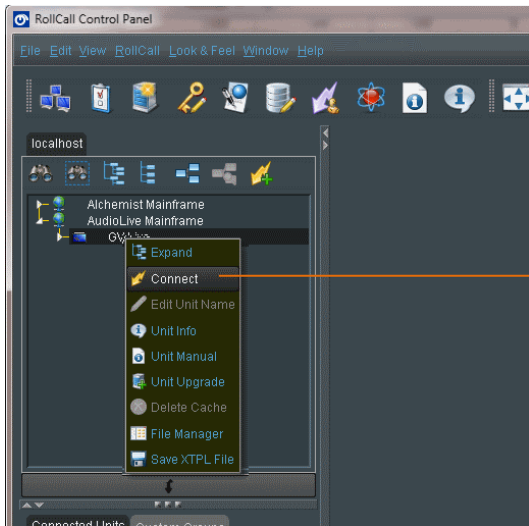
A drop-down list will appear. Select Expand.



GV Live will now appear:

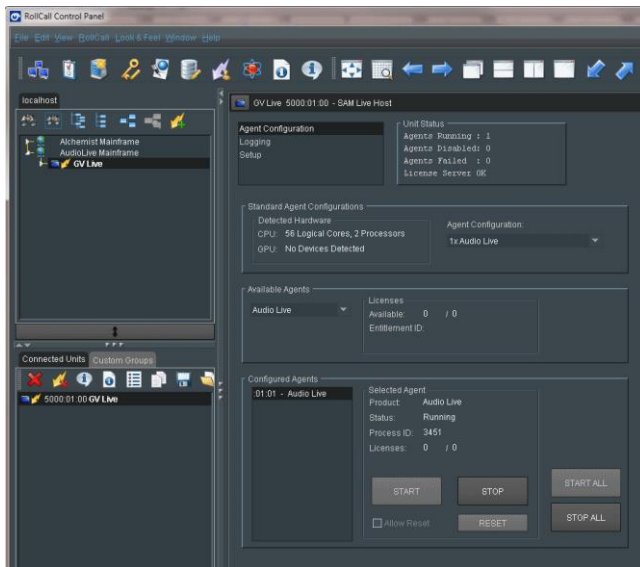


4. Right click on **GV Live**. A drop-down list will appear. Select **Connect**.

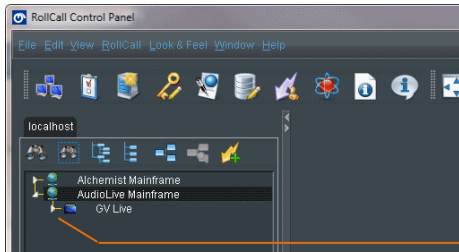


Right click to bring up the sub menu shown and select **Connect**

The **GV Live** template will now load.

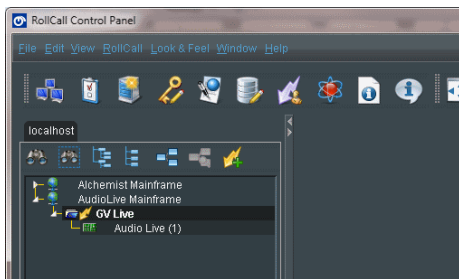


- In the RollCall Connection pane, expand the tree by selecting the small white arrow to the left of the **GV Live** icon:

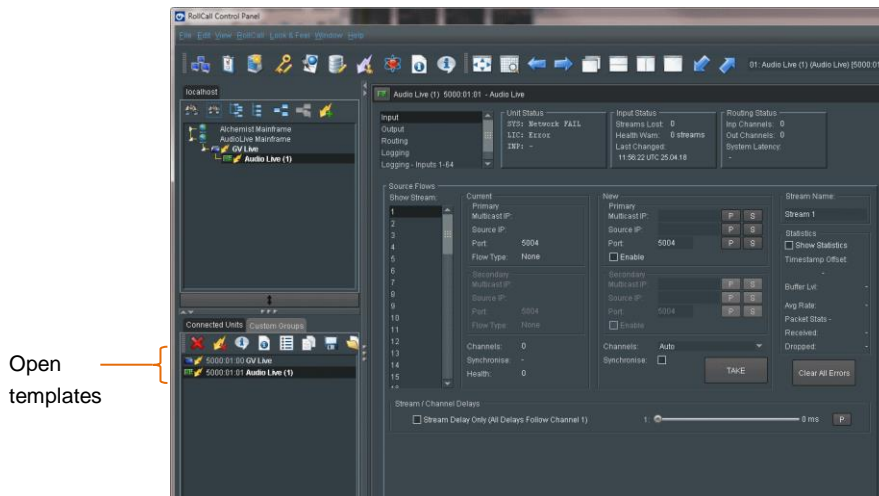


Click to expand the tree

Audio Live will appear:



- Double click on the **Audio Live** listing, and the Audio Live RollCall Template will load:



Open templates

Both the GV Live and Audio Live templates are now loaded and maybe configured.

Confirmation of the loaded templates is shown under the **Connected Units** tab.

It is possible to quickly change between templates, by simply clicking on the required template, in the **Connected Units** pane.

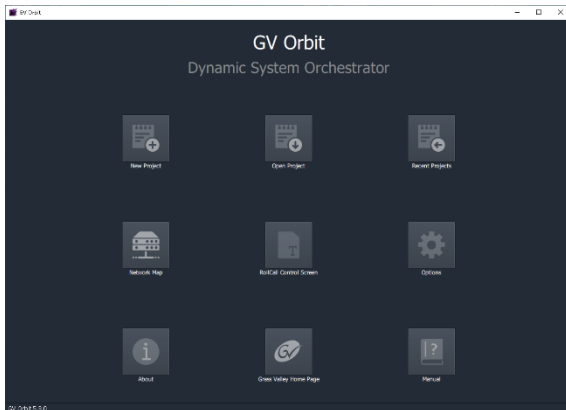
8. Using RCP in Orbit

RCP can be run as a component of Orbit.

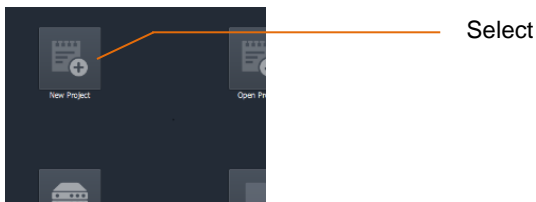
Note that Orbit is a paid-for Automated Control and Monitoring application. For customers who already own Orbit, RCP can be run as a component within Orbit.

Procedure

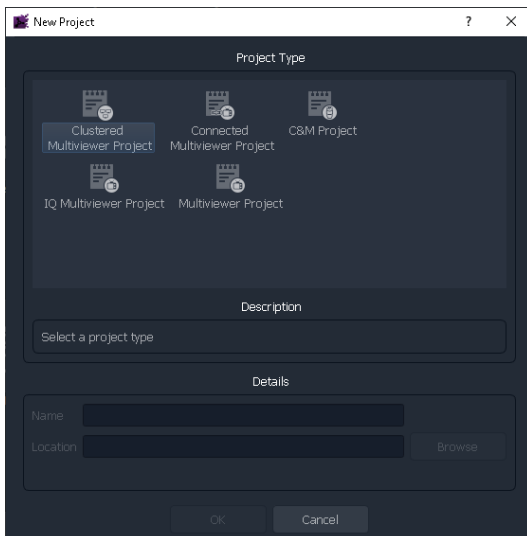
1. Open the Orbit control panel.



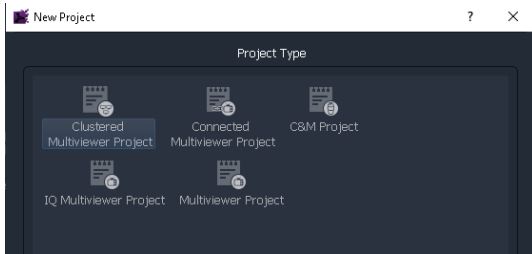
2. Select the **New Project** button:



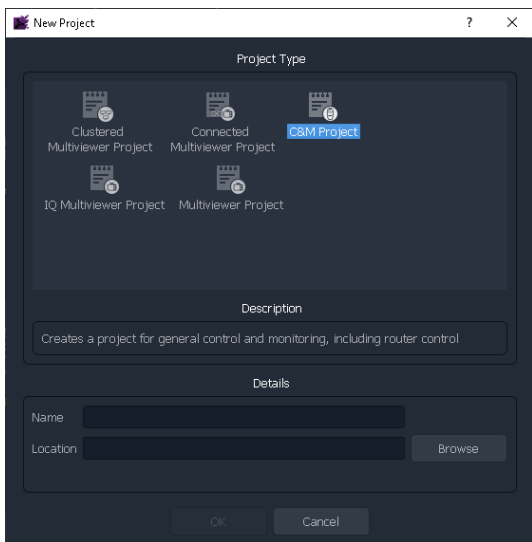
A new window will open called **New Project**



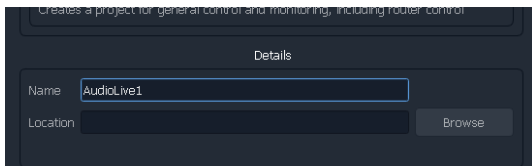
- In the Project type pane, select **C&M Project**



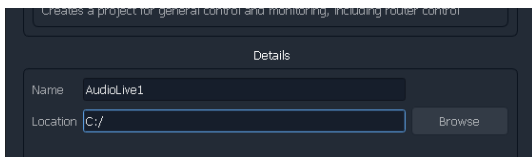
The New Project Window will update:



- Enter a name for the new C&M Project. This can be anything the User chooses, but should be something to make future reference easier.

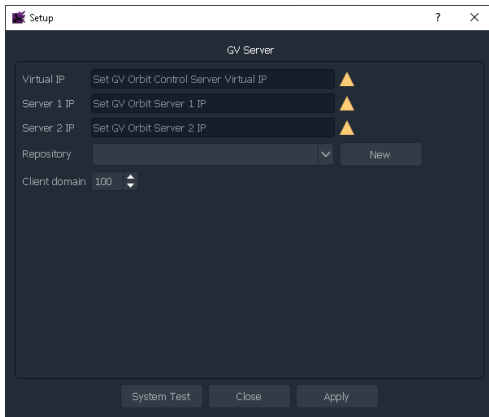


- Enter a location where the new C&M Project is to be stored. The URL path can be entered manually, or by use of the Browse button:



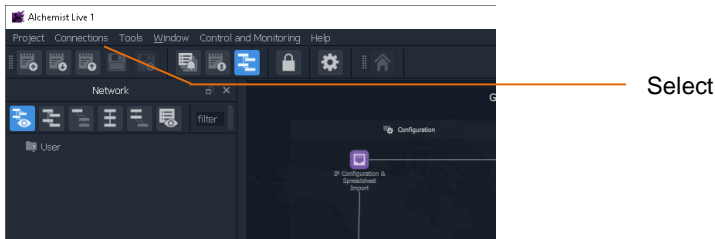
In this example the location is the root directory C:/

Click OK. This will prompt a new window called Setup to appear:

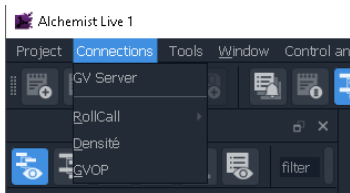


This can be simply closed.

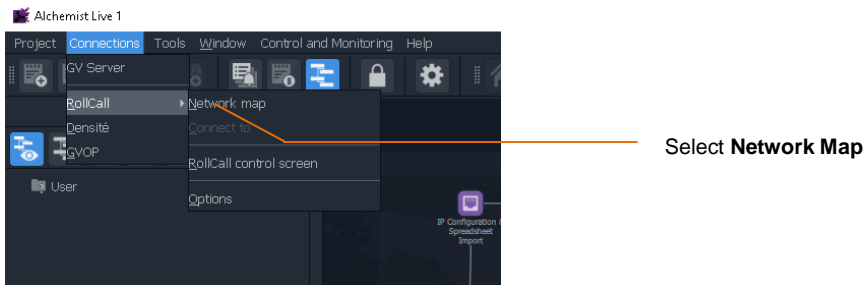
6. At the top of Orbit Client, click on **Connections**



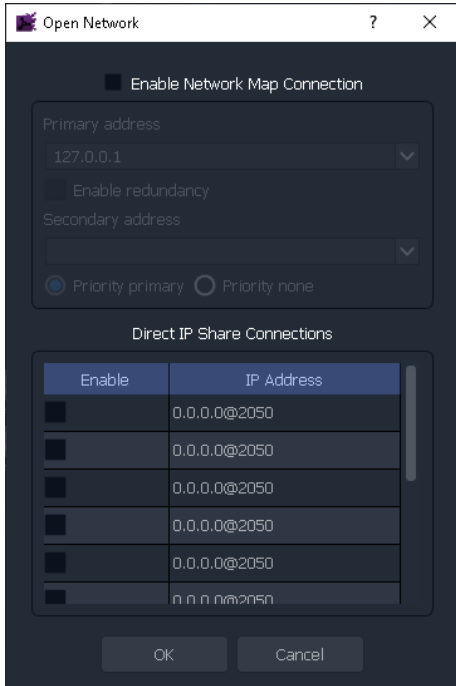
Clicking **Connections** will expose a drop down menu:



7. In the Connections dropdown menu, select RollCall. This will expose a new dropdown menu:

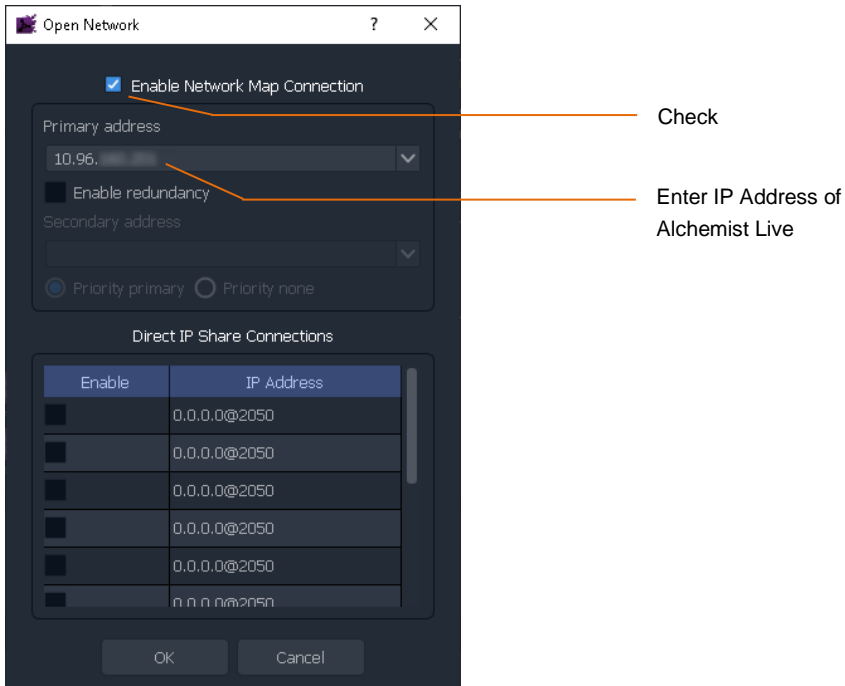


A new window called **Open Network** will appear:

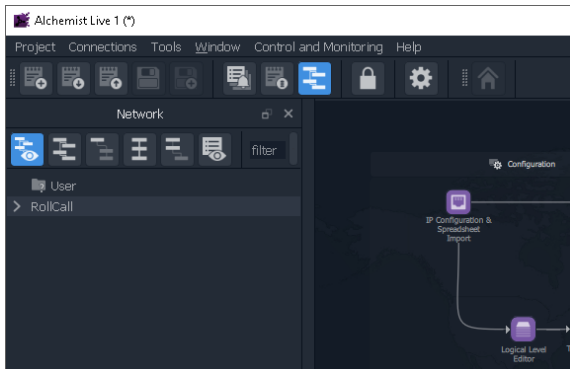


- In the **Open Network** window, check the **Enable Network Map Connection** checkbox.

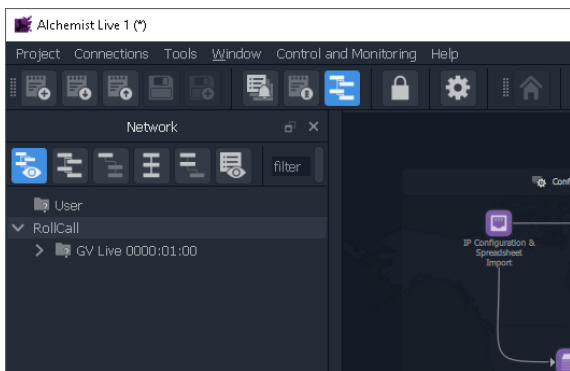
This will make the **Primary address** field active. Enter the IP Address of the Alchemist Live that is to be controlled:



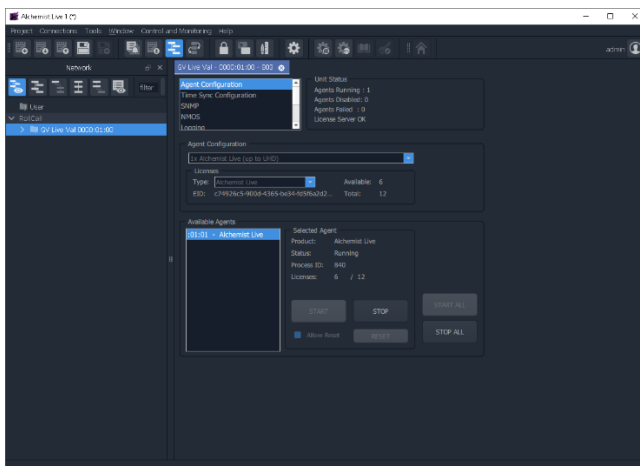
9. In the left hand pane of the Orbit Client a RollCall link will appear:



10. Click on the arrow adjacent to RollCall to expand the RollCall Tree
The GV Live host Machine will appear:

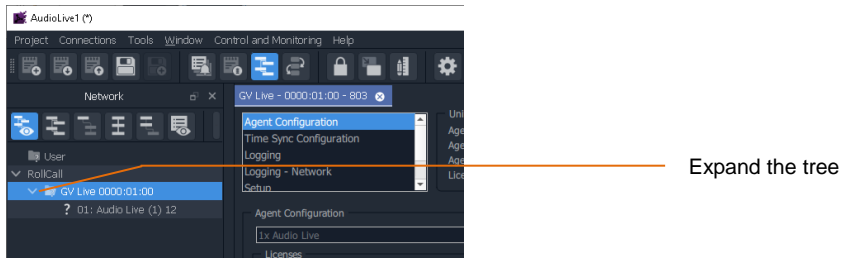


Double click on the GV Host. The GV Live Host template will now open:



11. Click on the arrow adjacent to the GV Live to expand the tree.

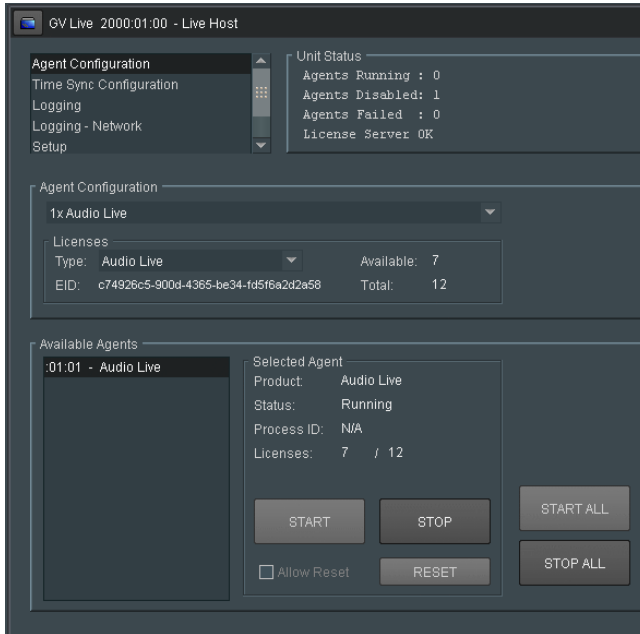
This will expose the Alchemist Live



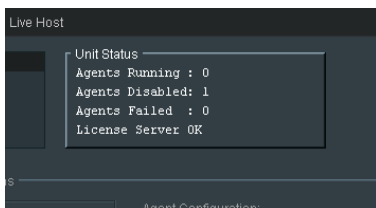
Double click on the Alchemist Live to open the template.

9. GV Live Template

9.1 Host Agent Configuration

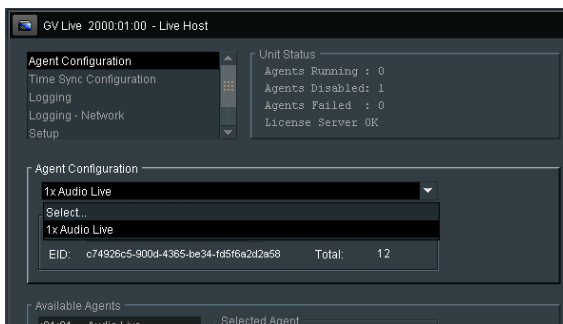


Note that no **Audio Live Agent** is currently running, hence the **Unit Status** pane will show an Agent disabled:



9.1.1 Agent Configuration

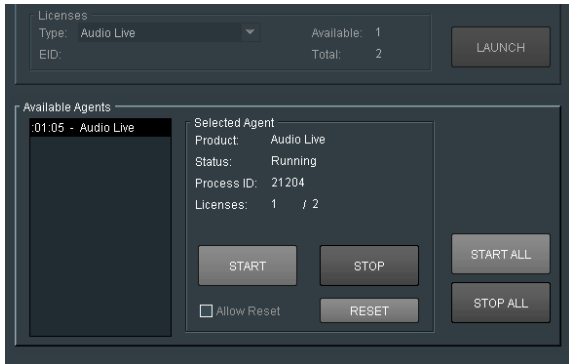
Here the Agent type can be configured using the drop-down menu:



Under **Licenses**, choose the Agent type using the dropdown menu. In this example **Audio Live** is selected:

9.1.2 Available Agents

This pane will now report that an Audio Live Agent is running. Select the audio Live in the list of available agents and press the **START** button to start the selected agent.

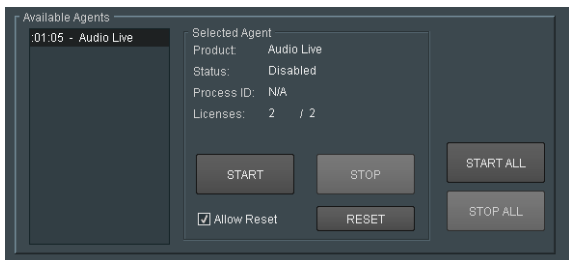


Note that the Agent can be 'stopped' and 'started' using the appropriate button.

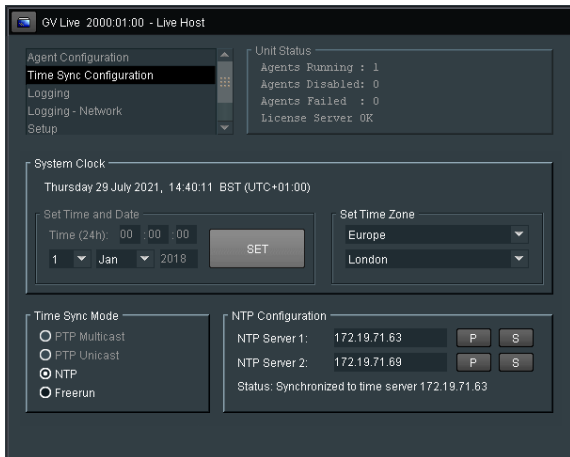
RESET

The Agent can be reset to its factory default condition, by pressing the **RESET** button.

Note that the **RESET** button will only be active if the Agent is stopped and the **Allow Reset** box is checked:

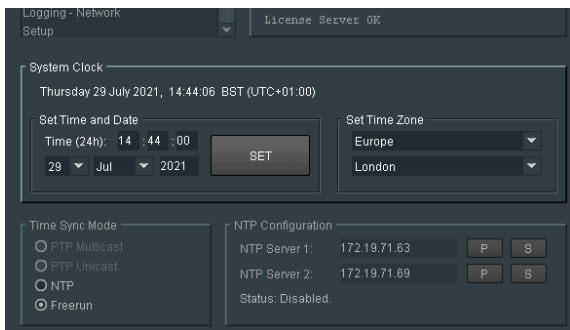


9.1.3 Time Sync Configuration



System Clock

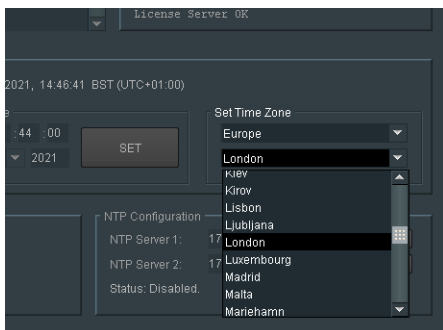
In the **System Clock** pane, the **Set Time and Date** menu will only be active if the **Time Sync Mode** is set to **Freerun**.



The time and date can be manually entered. Time to the nearest second can be configured and Synchronized to an external clock by pressing **SET** at a specific time.

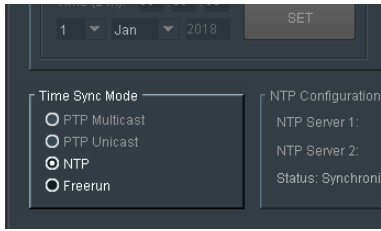
When **NTP** mode is selected, the time stamp will referenced to an external NTP server and so there is no requirement to set the system time.

Set Time Zone



The relevant Time Zone can be set using the drop-down menu.

Time Sync Mode

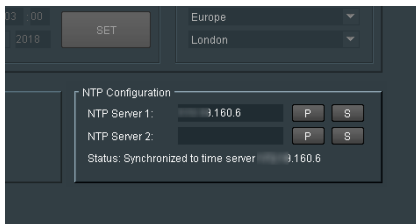


Four **Time Sync Modes** are offered, however, at the time of writing the PTP modes are not available and these options are grey'd out.

The Time Sync Mode can be set to be either:

- NTP
- Freerun

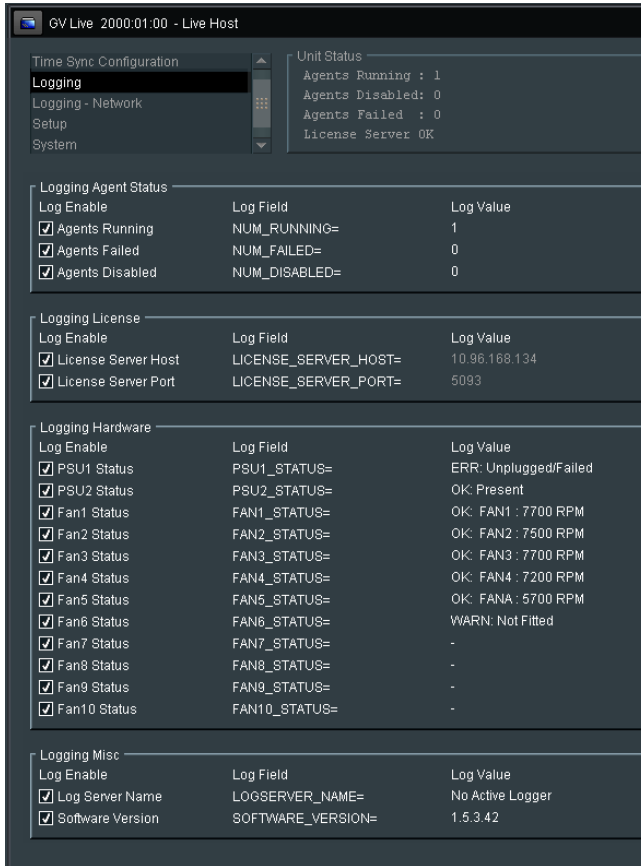
NTP Configuration



Here the NTP timestamp source can be defined. The IP Address of the primary NTP server can be entered and set by pressing 'S'.

A secondary NTP Server can also be defined. Should something happen the primary NTP Server making it unavailable, the secondary NTP server will act as back up.

9.1.4 Logging



In the **Logging** window, log fields maybe be disabled or enabled. Log fields are sent to the defined Log Server (see section 9.2.4 below).

The **default** configuration is everything **enabled**.

9.1.5 Logging - Network

GV Live 2000.01.00 - Live Host

Agent Configuration
Time Sync Configuration
Logging
Logging - Network
Setup

Unit Status
Agents Running : 1
Agents Disabled: 0
Agents Failed : 0
License Server OK

Logging Network Interface 1

| Log Enable | Log Field | Log Value |
|-------------------------------------|-------------------|---|
| <input checked="" type="checkbox"/> | Iface Name | LAN_PORT_1_NAME= eno1 |
| <input checked="" type="checkbox"/> | Iface Speed | LAN_PORT_1_SPEED= 1 Gbps |
| <input checked="" type="checkbox"/> | IP Address | LAN_PORT_1_IPADDRESS= 10.96.160.177 |
| <input checked="" type="checkbox"/> | MAC Address | LAN_PORT_1_MACADDRESS= AC:1F:6B:00:E6:E2 |
| <input checked="" type="checkbox"/> | Link State | LAN_PORT_1_LINK_STATE= OK: RUNNING |
| <input checked="" type="checkbox"/> | Switch Name | LAN_PORT_1_SWITCH_NAME= Leaf2 |
| <input checked="" type="checkbox"/> | Switch Chassis ID | LAN_PORT_1_SWITCH_CHASSIS_ID= F4:0F:1B:6F:A7:C8 |
| <input checked="" type="checkbox"/> | Switch Port ID | LAN_PORT_1_SWITCH_PORT_ID= mgmt0 |
| <input checked="" type="checkbox"/> | Switch Port VLAN | LAN_PORT_1_SWITCH_PORT_VLAN= 0 |

Logging Network Interface 2

| Log Enable | Log Field | Log Value |
|-------------------------------------|-------------------|---|
| <input checked="" type="checkbox"/> | Iface Name | LAN_PORT_2_NAME= eno2 |
| <input checked="" type="checkbox"/> | Iface Speed | LAN_PORT_2_SPEED= 1 Gbps |
| <input checked="" type="checkbox"/> | IP Address | LAN_PORT_2_IPADDRESS= 192.168.123.142 |
| <input checked="" type="checkbox"/> | MAC Address | LAN_PORT_2_MACADDRESS= AC:1F:6B:00:E6:E3 |
| <input checked="" type="checkbox"/> | Link State | LAN_PORT_2_LINK_STATE= OK: RUNNING |
| <input checked="" type="checkbox"/> | Switch Name | LAN_PORT_2_SWITCH_NAME= Leaf2 |
| <input checked="" type="checkbox"/> | Switch Chassis ID | LAN_PORT_2_SWITCH_CHASSIS_ID= F4:0F:1B:6F:A7:F9 |
| <input checked="" type="checkbox"/> | Switch Port ID | LAN_PORT_2_SWITCH_PORT_ID= Ethernet1/42 |
| <input checked="" type="checkbox"/> | Switch Port VLAN | LAN_PORT_2_SWITCH_PORT_VLAN= 199 |

Logging Network Interface 3

| Log Enable | Log Field | Log Value |
|-------------------------------------|-------------------|---|
| <input checked="" type="checkbox"/> | Iface Name | LAN_PORT_3_NAME= ens1f0 |
| <input checked="" type="checkbox"/> | Iface Speed | LAN_PORT_3_SPEED= 40 Gbps |
| <input checked="" type="checkbox"/> | IP Address | LAN_PORT_3_IPADDRESS= 10.96.164.177 |
| <input checked="" type="checkbox"/> | MAC Address | LAN_PORT_3_MACADDRESS= 24:8A:07:B1:B7:DC |
| <input checked="" type="checkbox"/> | Link State | LAN_PORT_3_LINK_STATE= OK: RUNNING |
| <input checked="" type="checkbox"/> | Switch Name | LAN_PORT_3_SWITCH_NAME= Leaf2 |
| <input checked="" type="checkbox"/> | Switch Chassis ID | LAN_PORT_3_SWITCH_CHASSIS_ID= 88:1D:FC:5B:52:31 |
| <input checked="" type="checkbox"/> | Switch Port ID | LAN_PORT_3_SWITCH_PORT_ID= Ethernet2/4 |
| <input checked="" type="checkbox"/> | Switch Port VLAN | LAN_PORT_3_SWITCH_PORT_VLAN= 164 |

Logging Network Interface 4

| Log Enable | Log Field | Log Value |
|-------------------------------------|-------------------|---|
| <input checked="" type="checkbox"/> | Iface Name | LAN_PORT_4_NAME= ens1f1 |
| <input checked="" type="checkbox"/> | Iface Speed | LAN_PORT_4_SPEED= 40 Gbps |
| <input checked="" type="checkbox"/> | IP Address | LAN_PORT_4_IPADDRESS= 10.96.166.177 |
| <input checked="" type="checkbox"/> | MAC Address | LAN_PORT_4_MACADDRESS= 24:8A:07:B1:B7:DD |
| <input checked="" type="checkbox"/> | Link State | LAN_PORT_4_LINK_STATE= OK: RUNNING |
| <input checked="" type="checkbox"/> | Switch Name | LAN_PORT_4_SWITCH_NAME= Leaf2 |
| <input checked="" type="checkbox"/> | Switch Chassis ID | LAN_PORT_4_SWITCH_CHASSIS_ID= 88:1D:FC:5B:52:37 |
| <input checked="" type="checkbox"/> | Switch Port ID | LAN_PORT_4_SWITCH_PORT_ID= Ethernet2/10 |
| <input checked="" type="checkbox"/> | Switch Port VLAN | LAN_PORT_4_SWITCH_PORT_VLAN= 166 |

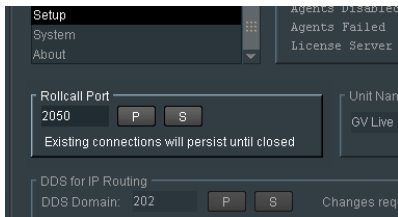
In the **Logging - Network** window, Log fields relevant to the Network Adaptors may be disabled and enabled as required. Log fields are sent to the defined Log Server (see 9.2.5 below).

The **default** configuration is everything **enabled**.

9.2 Setup



9.2.1 RollCall Port



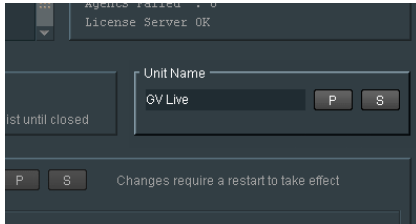
The default **RollCall Port** is 2050.

In normal circumstances the **RollCall Port** is left in its default setting. However, an alternative RollCall port can be configure here, if required.

A different port may be configured if required. Pressing **S** will take the new port number entered.

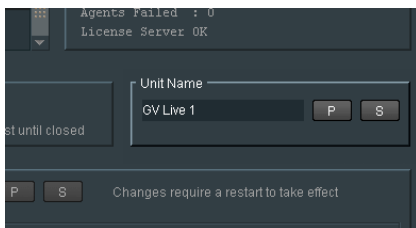
Pressing **P** will reset the port number to the default value.

9.2.2 Unit Name



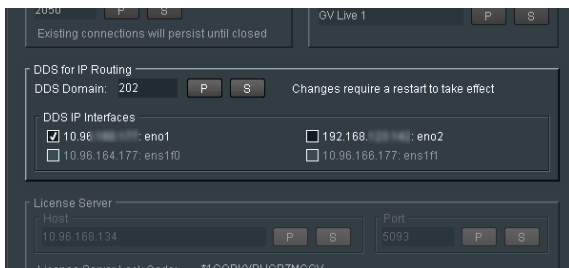
This is the identifying name of the GV Live product/unit/application. If several GV Live products are running, it may be desirable to give them individual names:

To edit the unit name, simply type the desired name in the box and press 'S' to save.



Pressing the 'P' button will return the name to its default setting: **GV Live**

9.2.3 DDS for IP Routing



Here the relevant DDS Domain can be defined.

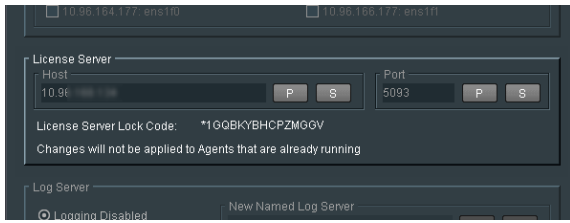
In this example the DDS Domain has been defined to be 101.

DDS IP Interfaces

Here the relevant Interface can be selected for DDS traffic.

DDS traffic cannot be transmitted via the media interfaces ens11f0 and ens11f1, hence these are greyed out and cannot be selected.

9.2.4 License Server



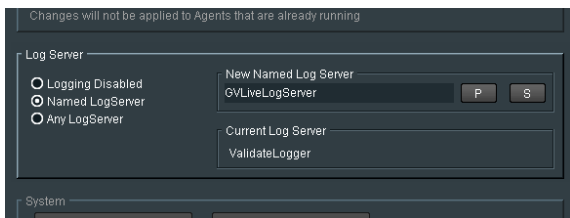
This field details the relevant License Server of the Audio Live. In normal circumstances the License Server runs on the Product/Turnkey host machine, hence the **Host** is defined as **Localhost**.

The default **Port** is 5093. In normal circumstances the default Port would be used. If desired the Port number can be changed, by simply typing the required port number in the box and pressing 'S' (save).

Pressing 'P' (Preset) will return the Port number to its default value.

Lock Code: The Audio Live license is locked to the hardware that the License Server is running on. The lock code is used to generate the license.

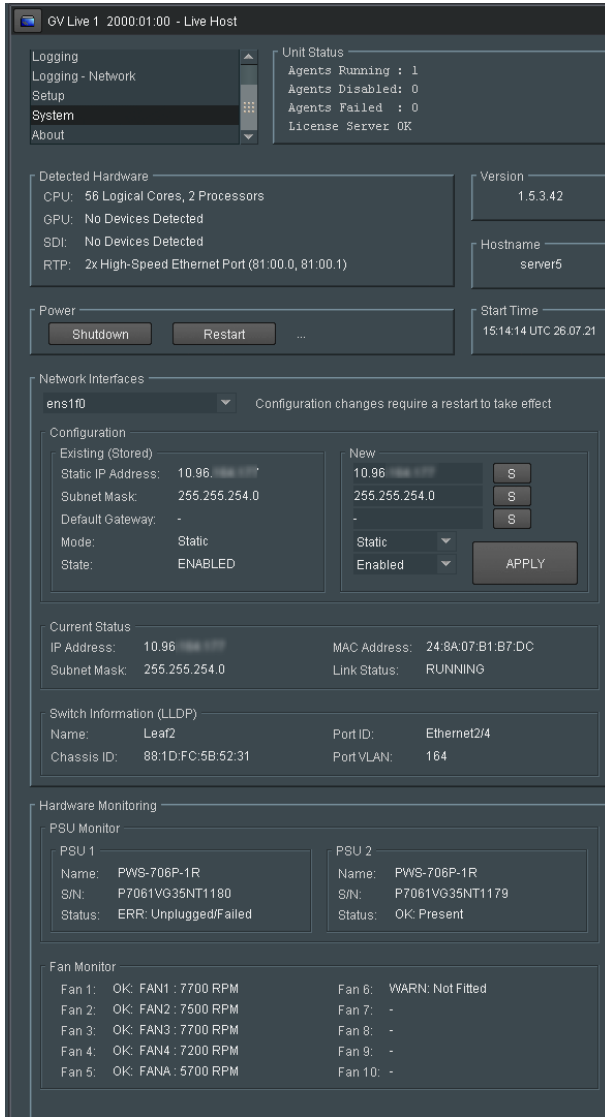
9.2.5 Log Server



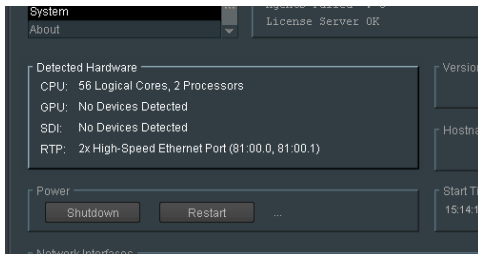
Here the desired **Log Server** can be:

- Disabled,
- Specifically defined
- Connect to any available Log Server.

9.3 System



9.3.1 Detected Hardware



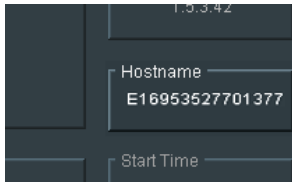
This field reports the CPU, GPU, SDI Interface & RTP Interface resource of the GV Live Host Machine. With respect to Audio Live, GPU & SDI Interface resource are not required.

9.3.2 Version



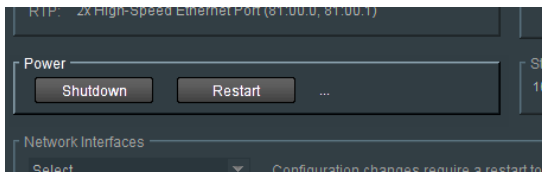
Reports the software version of the Audio Live.

9.3.3 Hostname



The host name reported is the serial number of the Host Machine.

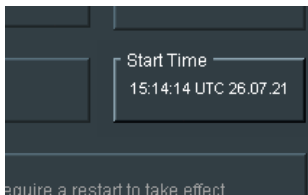
9.3.4 Power



The Power controls allow the Host Machine to be either shutdown, or restarted, remotely. Note that if the Host Machine is shut down remotely, RollCall will disconnect and it is not possible to restart the system remotely via RollCall.

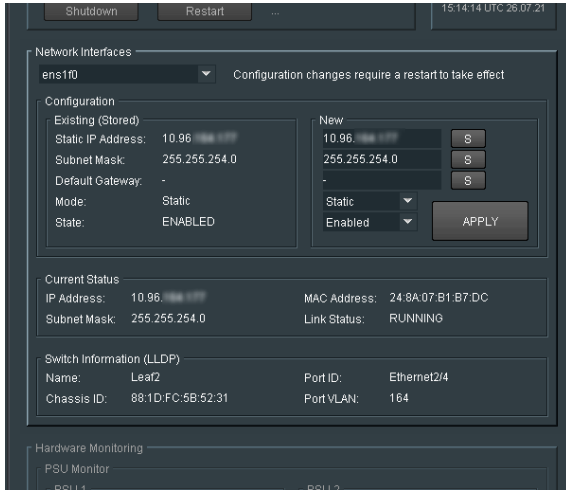
The system can be restarted locally either using the power button, or the console (see section 5.2 above). Or the system can be restarted remotely via the IPMI Port (see Appendix A below).

9.3.5 Start Time



Reports the start time that the Audio Live system was last started up.

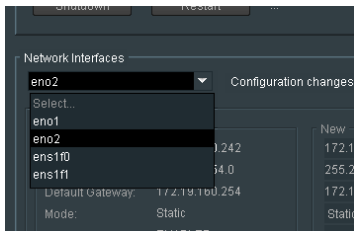
9.3.6 Network Interfaces



The Audio Live Host Machine has two ports **eno1** and **eno2** used for general network coms.

The other two ports **ens1f0** and **ens1f1** are used to interface the audio streams two and from the Audio Live Host Machine.

Each NIC can be selected from the drop-down menu:

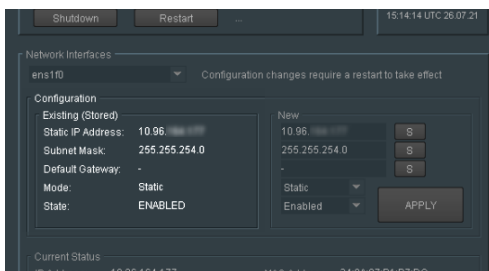


Once the appropriate NIC has been selected, the IP connection details can be edited.

Note that, by design, the NIC **eno1** can only be configured with a Static IP address.

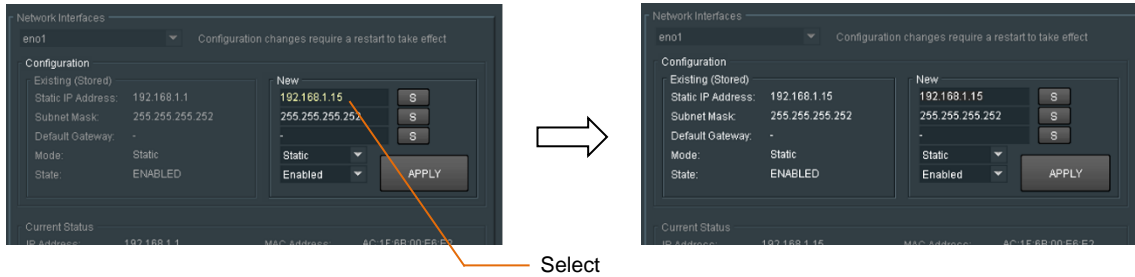
Configuration

Existing (Stored)



Once the NIC has been selected, the current connection details, if configured, will be displayed in the **Existing (Stored)** field.

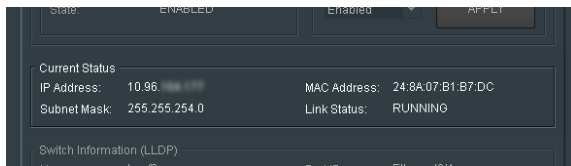
If a new connection is to be configured, the details should be entered in the **New** field:



Select

When the **APPLY** button is selected, the configured network under the **New** pane will migrate to the **Existing (stored)** pane.

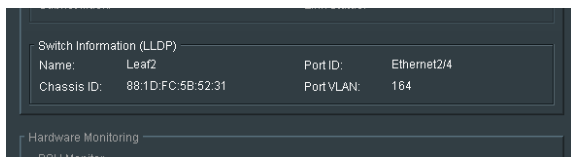
Current Status



This field will display the status of the current NIC (as selected above).

Conformation of the current **IP Address**, the **MAC Address** and **Link Status** are displayed in the **Current Status** field.

Switch Information (LLDP)



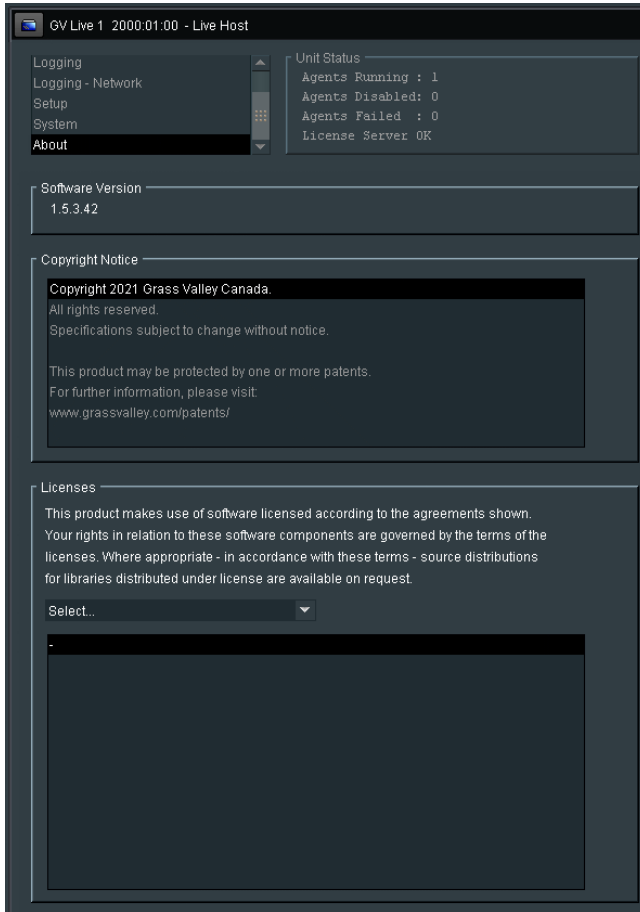
LLDP enables the reporting of upstream network connections if the neighbours are LLDP enabled.

The details of the network connection are reported back to the Alchemist Live.

This could include the switch Port ID of the Network Switch Alchemist Live is connected to, and if capable, the associated VLAN Port.

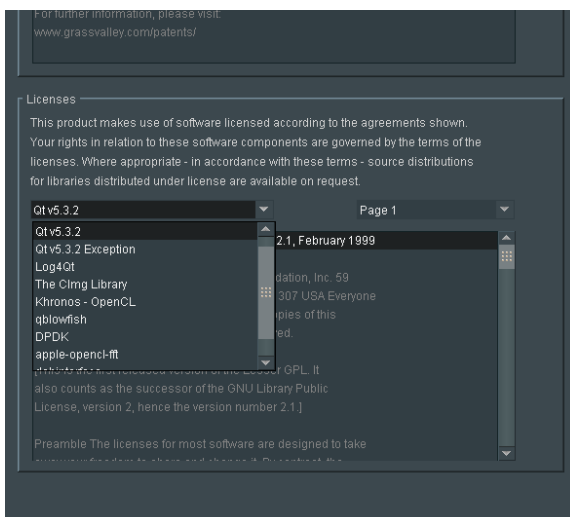
Note: **LLDP** is an acronym of: **Link Layer Discovery Protocol** (see **Table 1: Table of Terminology** section 1 above).

9.4 About

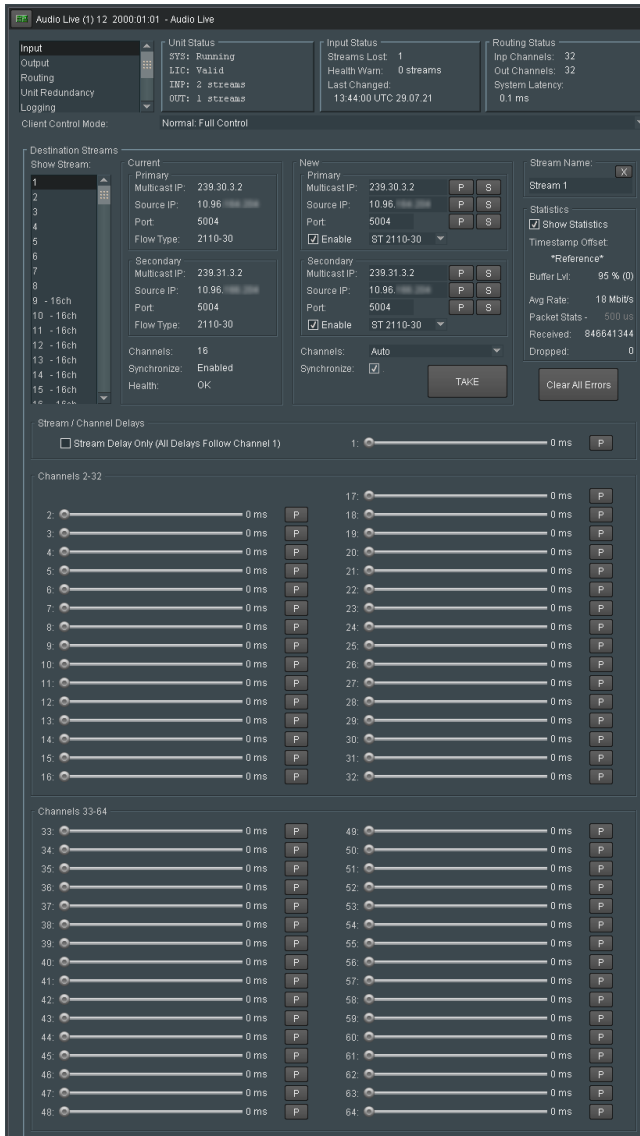


The **About** page details the current software version, the GV Live copyright and the various license agreements.

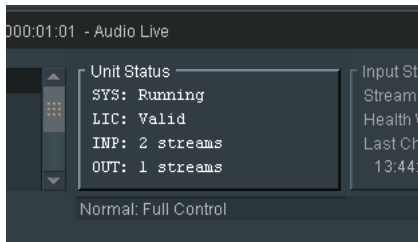
The various licenses applicable to the GV Live product, maybe viewed using the drop down menu, within the **Licenses** pane.



10. Audio Live Template



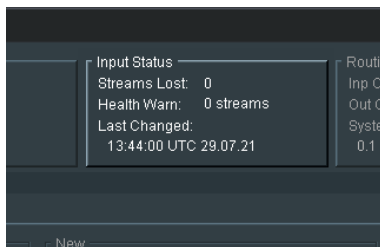
10.1 Audio Live Unit Status



Here confirmation is given that the:

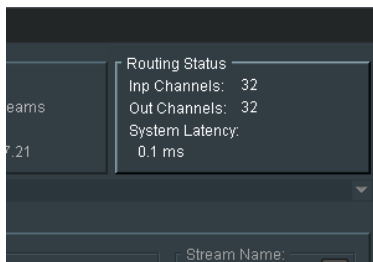
- **System** (SYS) is running.
- **License** (LIC) is valid and present.
- **Input** (INP) reports the number of incoming audio streams.
- **Output** (OUT) reports the number of output audio streams.

10.2 Audio Live Input Status



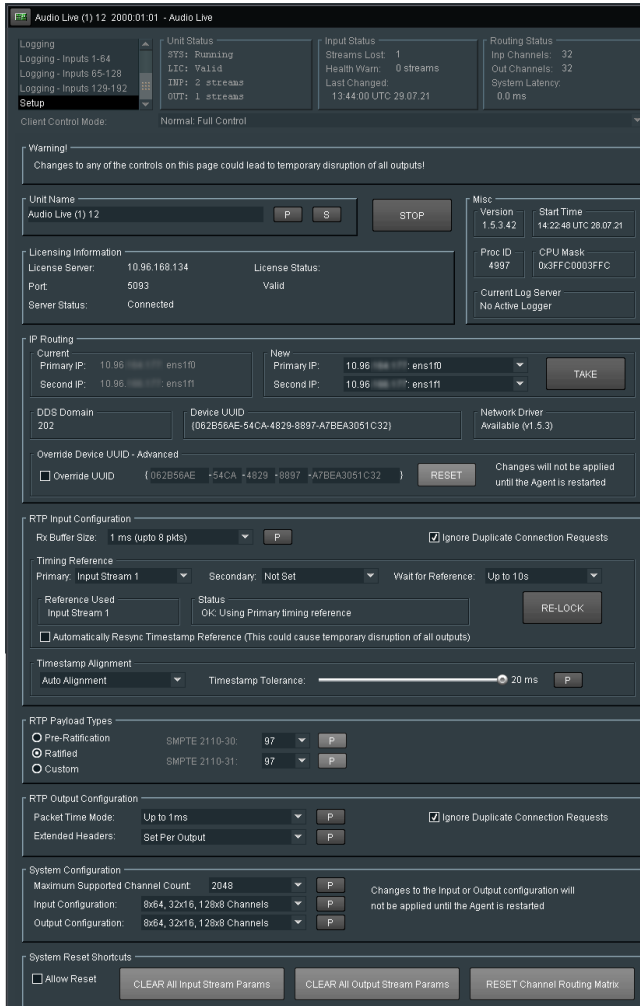
Here the status of the incoming audio is reported.

10.3 Audio Live Routing Status



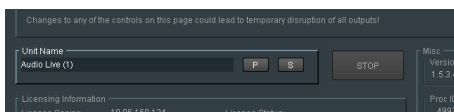
Here the quantity of Input and Output channels is reported, with an indication of system latency.

10.4 Audio Live Setup

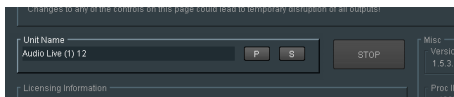


10.4.1 Unit Name

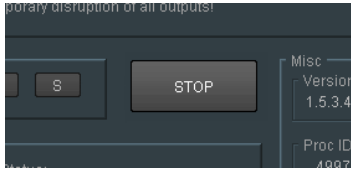
The default name is **Audio Live**:



The unit name can be modified to something more meaningful, if required.



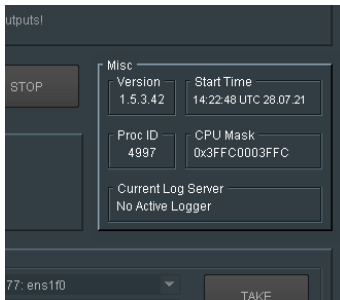
10.4.2 Stop Agent



The **STOP** control can be used to stop the Audio Live agent.

Note that if the agent is stopped, the Audio Live template will close. The agent must be restarted from the GV Live template.

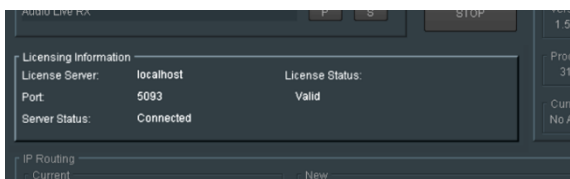
10.4.3 Misc



Reported here is:

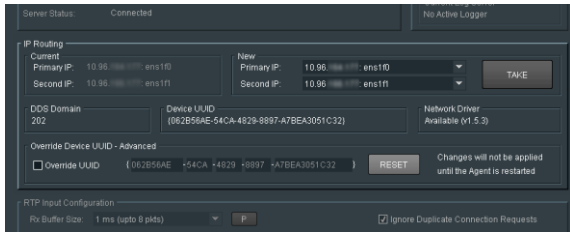
- Software Version.
- Start Time (the time the Audio Live Agent was started).
- Proc ID
- CPU Mask
- The current Log Server

10.4.4 License Information

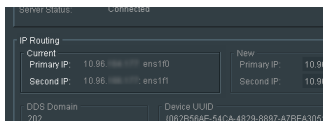


Reports the details of the **License Server**.

10.4.5 IP Routing



Current



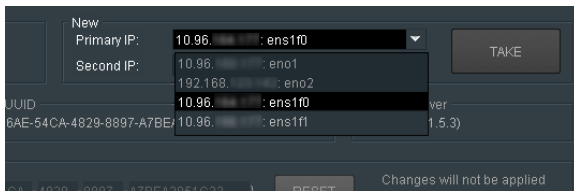
The **Current** pane details the current IP addresses of the Mellanox network adaptors and describes which the Primary is and which the Secondary is.

Note – the Mellanox interface has two network ports. The Primary and the Secondary. The function of the Secondary port is to act as a redundant interface, should something adverse happen to the Primary. It is not a mandatory requirement that the Secondary port is configured.

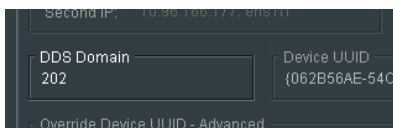
In normal circumstances Network Port **ens11f0** is defined as the Primary and **ens11f1** as the Secondary.

New

The designation of Primary and Secondary can be modified using the dropdown menu.



DDS Domain

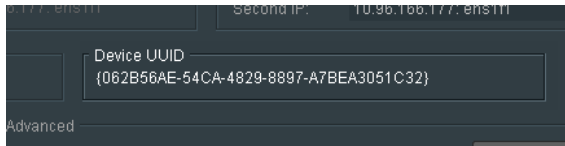


Here the **DDS Domain** is confirmed.

In this example the **DDS Domain** has been confirmed to be: **202**

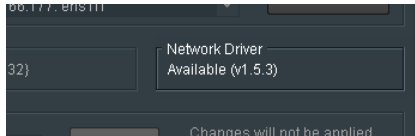
Note: The **DDS Domain** is configured at Host level (see section 9.2.3 above).

Device UUID



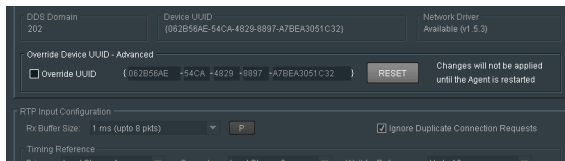
Reports the UUID (Universal Unique Identifier) of the host machine.

Network Driver



Reports the Network Driver

Override Device UUID - Advanced

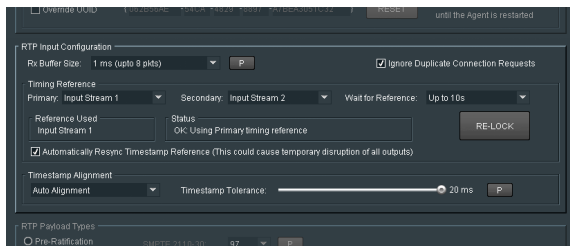


In normal circumstances the UUID is randomly generated the first time the audio Live is launch and this UUID will then remain.

There may be circumstances where the User wishes to define the UUID. The **Override Device UUID – Advanced** feature allows the User to define an alternative UUID.

The RESET button may be used to revert the UUID back to the originally generated value.

10.4.6 RTP Input Configuration



Here controls pertinent to the RTP inputs can be configured, *please note that adjusting these controls will cause temporary disruption to all outputs:*

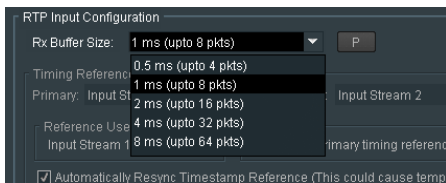
- **Rx Buffer Size**
- **Timing Reference**
- **Timestamp Alignment**

Rx Buffer Size:

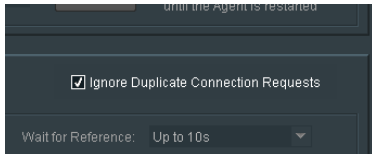
The RX Buffer is a short queue of RTP packets for each of the Audio Live inputs; it is here that packet redundancy and packet re-ordering are handled. Packet redundancy makes use of dual-redundant RTP streams to minimise the probability of a dropped packet in the network fabric having a material impact on the audio quality. Either of the redundant streams can supply a packet with a given RTP sequence number to the RX Buffer with the packet arriving first being used downstream. If the redundant streams are not perfectly aligned, this can be managed by the buffer as long as it has sufficient capacity.

Increasing the size of the RX Buffer incurs additional latency in the signal path but increases the range over which the re-ordering of out of sequence RTP packets can be successfully corrected. It also allows for a greater misalignment between the two RTP streams of a dual-redundant configuration.

The received packet buffer size can be set using the dropdown menu:



Default setting is: **1 ms (up to 8 Packets).**

Ignore Duplicate Connection Requests

This feature is enabled by default.

This feature is new in V1.5.3.42. This may be disabled so as to match pre-V1.5.3.42 behaviour.

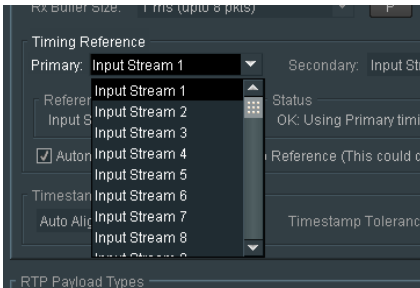
Timing Reference:

The Audio Live uses the RTP packets arriving at its inputs to establish the rate at which it must transmit output packets and to dictate the RTP/PTP (Precise Time Protocol) timestamp reference embedded in its RTP output streams. As part of this process, one of the input streams must be selected as a timing reference. This selection can be guided using the Primary/Secondary reference controls (recommended) or it can be automatic, in which case Audio Live will select a stream that appears suitable from an analysis of RTP packets received.

Once a reference stream has been selected, it will then be used to control the rate at which the entire system responds to incoming RTP packets, regardless of the coming and going of other RTP input streams. This will not change unless the timing reference is lost or timing reference controls are re-configured.

If primary and secondary reference streams have been specified, the loss of one will automatically trigger a fail-over to the other and normal operation will continue uninterrupted. If both primary and secondary are lost, or the system has been configured to automatically select a reference stream, Audio Live will attempt to continue by free-running the RTP packet timestamps from the last reference point at a rate dictated by the fastest running of its current inputs.

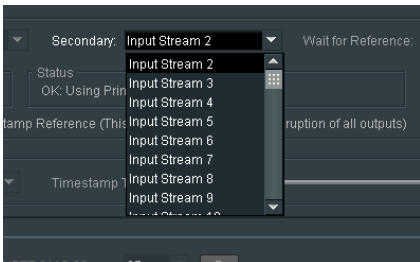
Primary Reference



A **Primary Reference** can be defined using the drop-down menu.

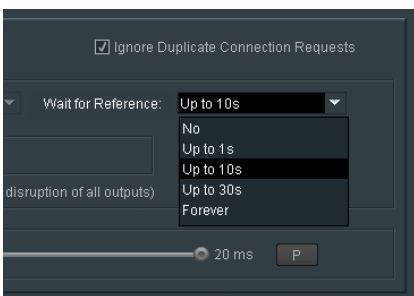
Secondary Reference

In case of failure of the **Primary Reference** stream, a **Secondary Reference** stream can also be defined:



Note that a **Secondary Timing Reference** can only be set if a **Primary Timing Reference** has been defined. If the **Primary Timing Reference** is set to **Auto**, the menu to define the **Secondary Timing Reference** will be greyed out.

Wait for Reference

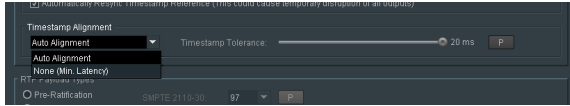


The **Wait for Reference** control allows the User to define how long the system waits after experiencing the loss of both the Primary and Secondary References.

Should the Primary and Secondary References not return within the described period, then the system will fall back to the input stream with the earliest timestamp.

Timestamp Alignment

The Timestamp Alignment mode can be configured using the dropdown menu:

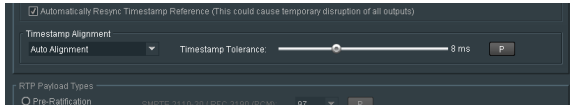


Auto Alignment:

In this mode, the system's timestamp reference that has been established from the timing reference stream is used to automatically align all inputs prior to channel routing.

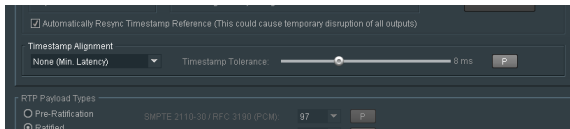
In order to achieve this, the necessary alignment must fall within a window defined by the **Timestamp Tolerance** control. This control has default setting of 6ms. The range of this control is dictated by the Input/Output Configuration (**System Configuration**, paragraph 10.4.9).

Increasing this control beyond the value necessary for input alignment is not recommended as it will add latency to all RTP input streams.

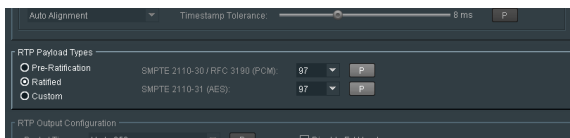


None (Min. Latency):

In this mode, no timestamp alignment will be attempted and the latency imposed on all RTP input streams is kept to a minimum. However, the exact relationship between audio channels originating from different inputs can no longer be guaranteed at the output to Audio Live.



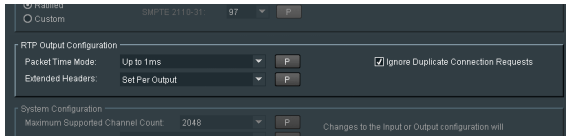
10.4.7 RTP Payload Types



The RTP Payload Type is a 7 bit code in the RTP Packet Header. It indicates the format of the payload and determines its interpretation by the application.

- Pre-Ratification – RTP payload type IDs used by some GV equipment prior to SMPTE ratification.
- Ratified – RTP payload type IDs agreed and ratified by SMPTE.
- Custom – Fully customizable RTP payload type IDs for specific user applications.

10.4.8 RTP Output Configuration



Packet Time

The RTP output stream packet time describes the frequency of transmission of packetized audio samples, essentially the packet period. To transmit 64 pcm audio channels per stream, a packet time of 125us is required. If the packet time is increased, the number of audio channels per stream, will decrease.

The type of audio (PCM or AES) will affect the number of channels supported:

Table 3: Max number of channels supported per stream

| | PCM | AES3 |
|--------|-----|------|
| 125 us | 64 | 60 |
| 250 us | 32 | 30 |
| 500 us | 16 | 14 |
| 1ms | 8 | 6 |
| 4ms | 2 | - |

Note:

PCM employs 3 Bytes per internet packet.

AES employs 4 Bytes per internet packet.

Because AES requires more Bytes per packet, the maximum number of audio channels that can be carried is less than PCM.

Audio Live supports the following SMPTE 2110-30 profiles:

A: 48 kHz streams with:

- 1 to 8 channels at packet times of 1 ms

B: 48 kHz streams with:

- 1 to 8 channels at packet times of 1 ms,
- 1 to 8 channels at packet times of 125 µs

C: 48 kHz streams with:

- 1 to 8 channels at packet times of 1 ms
- 1 to 64 channels at packet times of 125 µs

Levels **AX**, **BX** and **CX** are not covered as these require support for 96 kHz sampling.

The packet time can be set for each output stream but the options are limited by the setting selected from the RTP Output Configuration Packet Time drop-down menu:

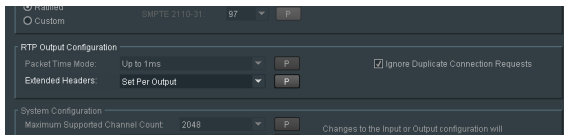


There are two possible settings:

- Up to 250us
- Up to 1ms

If packet times of up to 1ms are allowed, the propagation delay imposed on all audio channels passing through Audio Live will increase by approximately 1ms, irrespective of the associated output packet time option selected (see **Output Packet Time** below – section 10.6.1).

Extended Headers



These proprietary extensions contain additional metadata about the stream contents.

Using the **Extended Headers** menu, it is possible to configure the extended headers on each output stream.

There are three possible settings accessed via the drop-down menu:



- **On for all Outputs**
- **Off for all Outputs**
- **Set Per Output**

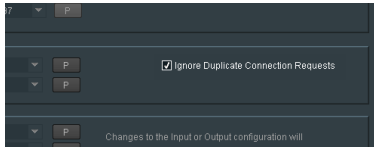
On for all Outputs: when set, extended headers will be enabled on all output streams.

Off for all Outputs: when set, extended headers will be disabled on all output streams.

Set Per Output: when set, it will be possible to enable, or disable, extended headers on each individual output stream. Note that when this setting is configured, the **Disable Ext Headers** control in the Output Menu (see section 10.6.1, below) will become active and hence the extended header in that output stream, can be enabled, or disabled.

The default setting is: **On for all Outputs**

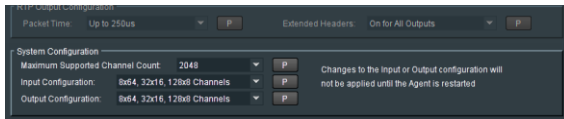
Ignore Duplicate Connection Requests



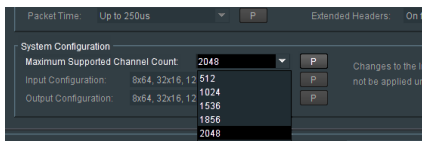
This feature is enabled by default.

This feature is new in V1.5.3.42. This may be disabled so as to match pre-V1.5.3.42 behaviour.

10.4.9 System Configuration

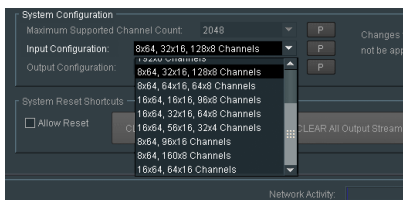


Maximum Supported Channels Count



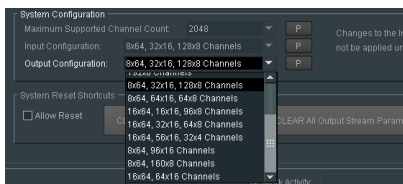
This control is used to define the **Maximum Supported Channels Count**. The desired value can be selected from the drop down list.

Input Configuration



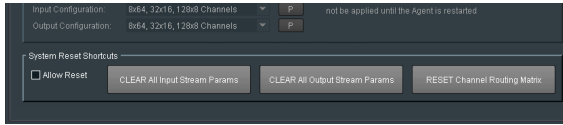
Here the Input Audio Matrix can be defined, by selecting the appropriate option from a dropdown list.

Output Configuration

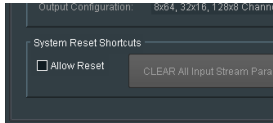


Here the Output Audio Matrix can be defined, by selecting the appropriate option from a dropdown list.

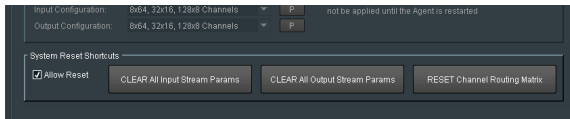
10.4.10 System Reset Shortcuts



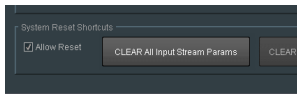
Allow Reset



For the **CLEAR** and **RESET** buttons to become active, the Allow Reset box must be checked:

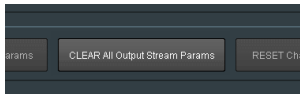


CLEAR All Input Stream Params



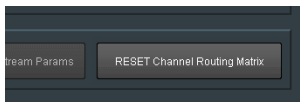
Pressing **CLEAR All Input Stream Params** will clear all the currently configured Input Stream settings.

CLEAR All Output Stream Params



Pressing **CLEAR All Output Stream Params** will clear all the currently configured Input Stream settings.

RESET Channel Routing Matrix



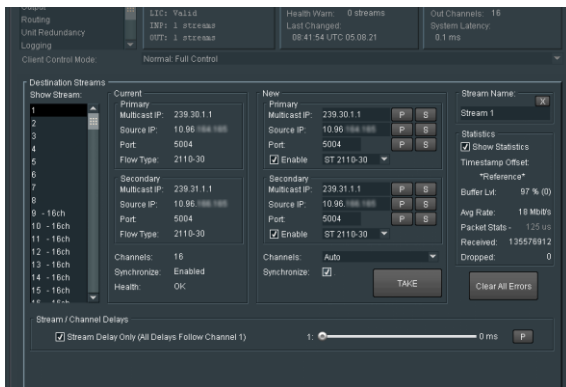
Pressing **RESET Channel Routing Matrix** will return the Routing Matrix to its default condition.

10.5 Audio Live Input



10.5.1 Destination Flows

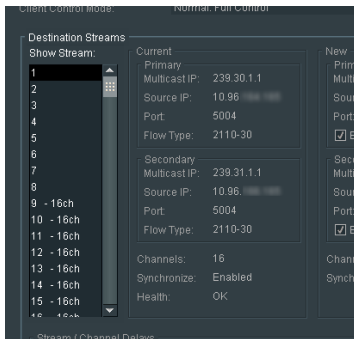
In the **Destination Flows** field, the relevant details of each destination audio stream can be entered.



Note that in this example, the source **Stream 1** configured has been identified as carrying 16 audio channels.

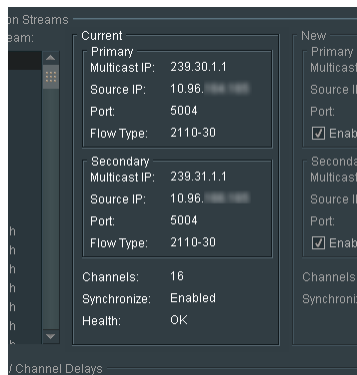
Destination Flows

Note: The incoming RTP Flows to Audio Live are referred to as Destination Flows. The IP details entered will be output side of the source device i.e. the upstream source device's destinations.



In the **RTP Dest Flows** pane, any of the source streams can be selected. Once selected, the details of the stream are reported in the **RTP Dest Flows** menu.

Current

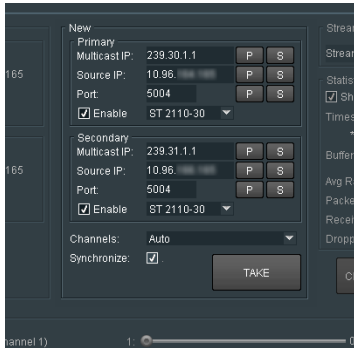


This field details the connection details of the specific input stream selected.

Confirmation is given of:

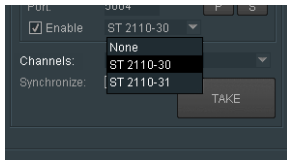
- the number of source channels associated with the selected stream.
- the state of the Synchronizer.
- the health of the incoming stream.

New



Here the connection details of a particular source stream can be modified. Once the new connection details have been configured, pressing the **TAKE** button will action the changes.

Note that the input audio type can be defined. Both PCM and AES are supported. The required option can be configured using the drop-down menu.



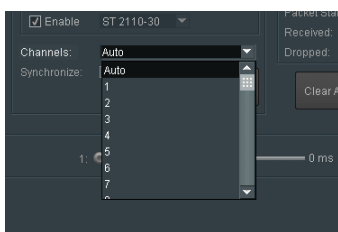
PCM is defined by setting: **ST2110-30**

AES is defined by setting: **ST2110-31**

Note: The input can be defined as **None**. This will effectively disable the port.

Channels

Note that the **Channels** control default is **Auto**. However, it is possible to define the number of audio channels present in an audio stream, via the drop-down menu:



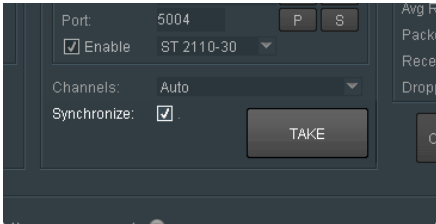
Note: Audio live can support up to 64 channels of pcm (ST 2110-30), and 60 channels of AES3 (ST2110-31), per stream.

Synchronize

A feature of Audio Live is the Synchronizer. This allows up to 16 asynchronous audio streams to be synchronized with respect to the Timing Reference, as defined in part 10.4.6 above.

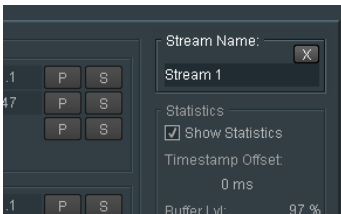
The **Synchronize** control is only active when the relevant audio stream is disabled. If the stream is enabled, uncheck the **Enable** controls of both the Primary and Secondary connections and press **TAKE**.

Once disabled the Synchronizer control will become active and can now be configured:

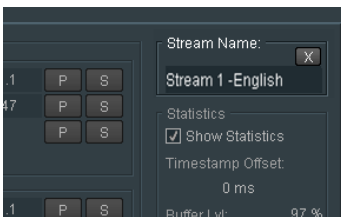


Once the Synchronizer has been configured the stream can be reactivated by checking the **Enable** control and pressing **TAKE**.

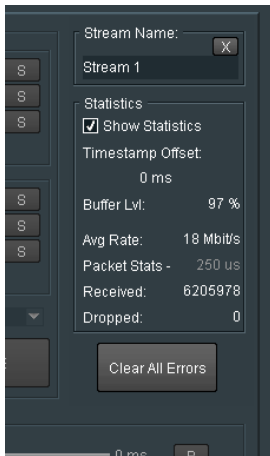
Stream Name



The Stream Name can be either left as default (in this case Stream 1), or changed to something more meaningful:



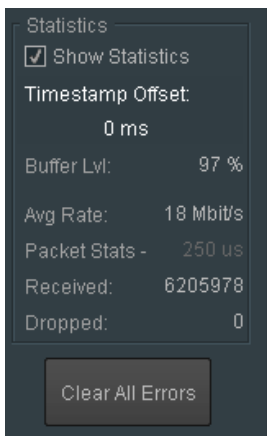
Statistics



The Statistics pane shows the statistics for the relevant stream

Statistics can be configured to be either shown, or hidden.

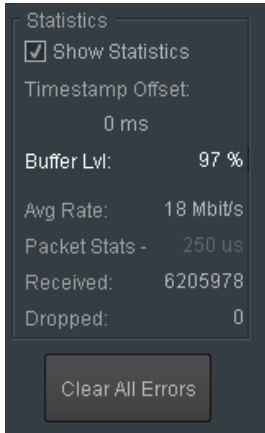
Timestamp Offset



Displays the timing difference between the relevant input stream and the reference.

Note – if the **Timestamp Offset** is a fixed value, then this indicates that the particular input stream is synchronous with respect to the reference. In such a circumstance, this particular input does not need to be synchronized.

However, if the **Timestamp Offset** is changing, this indicates that this particular input is indeed asynchronous and the hence the Synchronizer should be enabled.

Buffer Lvl:

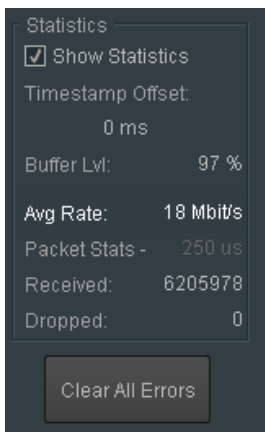
In order to achieve timestamp alignment for all input streams, it is necessary to buffer audio samples before they are processed.

The degree of buffering is defined by **the Timestamp Tolerance** requested (see section 10.4.6 page 55).

The reported buffer level indicates how close to the tolerance level the buffer occupancy is running.

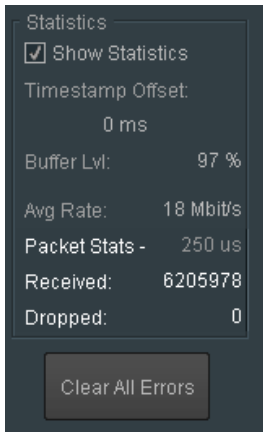
For example, if the tolerance is set to 5ms, an input reporting a buffer level of 100% will have buffered 5ms of audio.

Synchronous input streams would be expected to have a level below 100% but asynchronous inputs (that are not being corrected with synchronisation) could well report an occupancy well above.

Avg Rate:

Indicates the average bitrate of the incoming audio stream.

Packet Stats



This reports the packet time for the incoming stream and defines the max number of channels that the stream can carry:

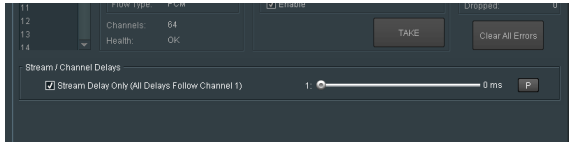
See Table 3 above (section 10.4.8)

Received: details the number of audio packets received.

Dropped: details the number of audio packets dropped.

Clear All Errors: when pressed, the reported 'received' and 'dropped' packet stats will be reset to zero.

10.5.2 Input Stream / Channels Delays

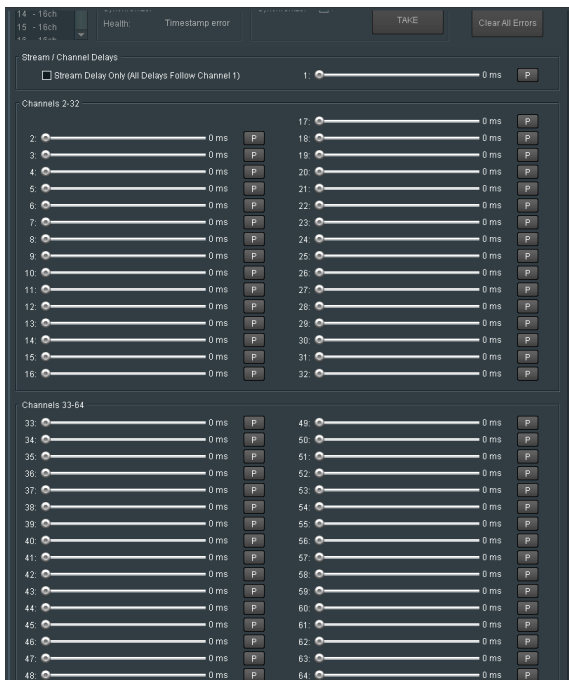


When the control **Stream Delay (All Delays Follow Channel 1)** is checked (enabled), a generic delay is applied to all audio channels.

The range of the control is 0 to 2000mS (2 seconds).

Default value is 0mS.

When the control **Stream Delay (All Delays Follow Channel 1)** is unchecked, the extended delay menu will appear:



It is now possible to configure unique delay values to each channel.

Each control has a range of 0 to 2000ms

Default is 0ms.

10.6 Audio Live Output

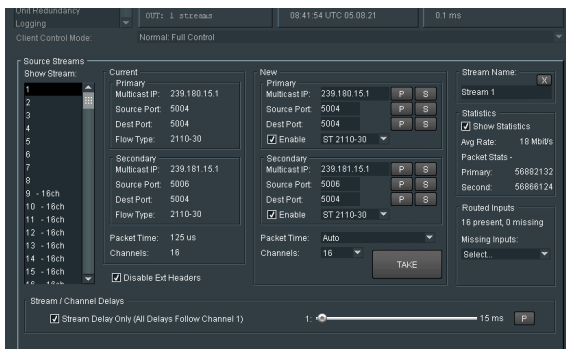


10.6.1 Source Streams

Source Streams

Note: The output audio streams of the Audio Live are referred to as **Source Streams**. This is because the IP details of the Audio Live output represent the Source Streams of the downstream destination.

In the **Source Streams** field, the relevant details of each destination audio stream can be configured.

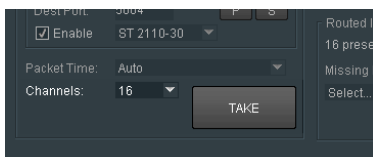


Note that in this example, that **Stream 1** has been defined to carry 64 PCM audio channels.

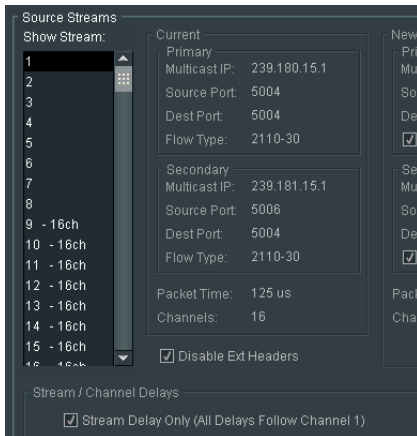
The default number of output audio channels is 8.

In the example below, the output stream has been configured to have 16 channels.

Clicking on **Take** will execute the change.



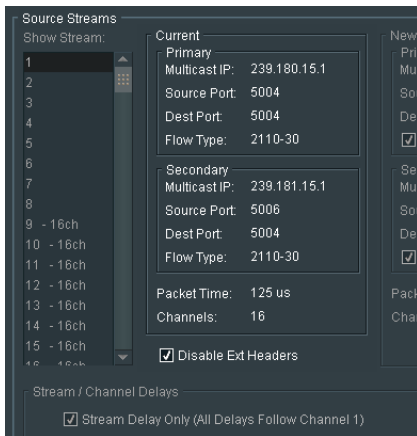
Show Stream



In the **Show Streams** pane, any of the output streams can be selected.

Once selected, the details of the stream are reported in the **RTP Source Flows** menu.

Current



This field details the connection details of the specific output stream selected.

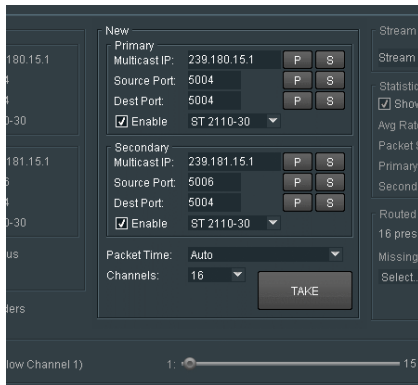
Packet time

Reports the current packet time setting of the specific stream.

Channels

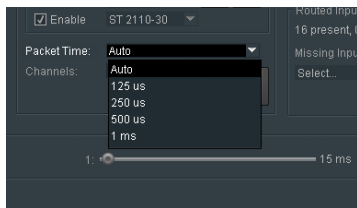
Confirmation is given of the number of destination channels associated with the selected stream.

New

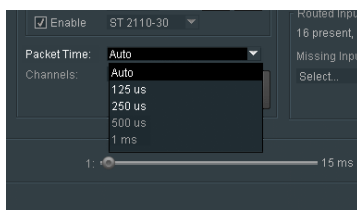


Here the connection details of a particular destination stream can be modified. Once the new connection details have been configured, pressing the **TAKE** button will action the changes.

Packet Time



The packet time of a specific output stream can be defined from the drop-down menu. Note that the options that are available will be affected by the setting of the **Packet Time** control in the **RTP Output Configuration** menu (part 10.4.8 above):

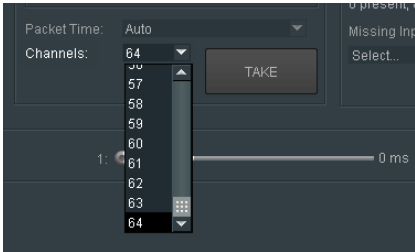


If the **Auto** option is selected, the output stream packet time will be automatically set to the most appropriate value for the requested channel count (see below). If however the packet time is forced, the available channel count options for the stream will be restricted. For example, selecting a packet time of 1ms will limit the maximum number of channels to 8.

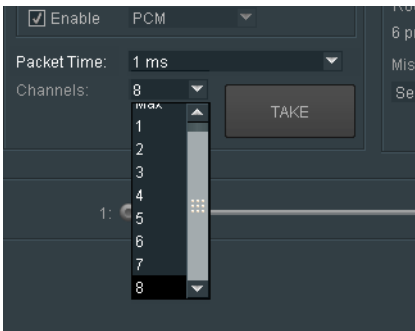
Channels

This control allows the number of output channels that the selected output audio stream can support.

Configuration is via a drop-down menu:



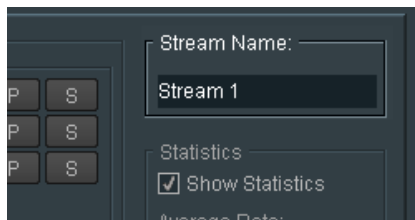
Depending on the **Packet Time Configuration**, the number of output channels that can be set will be affected.



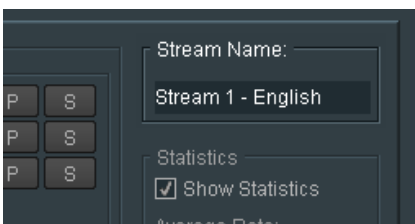
The maximum number of output audio channels that can be defined per stream, is determined by the defined **Packet Time** setting.

See **Table 3** above (section 10.4.8)

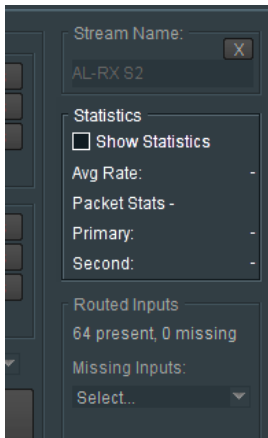
Stream Name



The **Stream Name** can be either left as default (in this case **Stream 1**), or changed to something more meaningful:

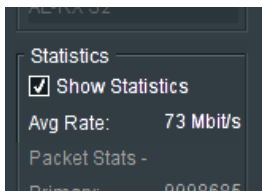


Statistic



The **Statistics** pane shows the statistics for the relevant output stream. Statistics can be configured to be either shown, or hidden.

Avg Rate



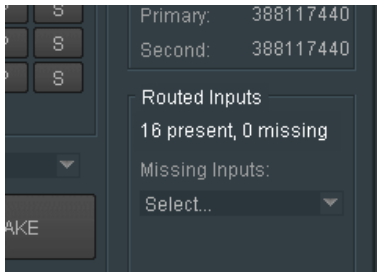
Reports the average bit rate of the selected output stream.

Packets Stats



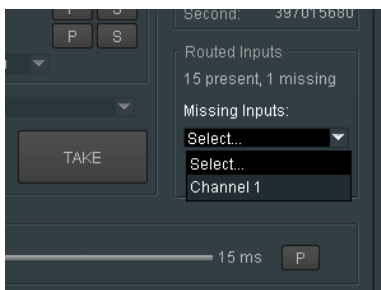
Reports the number of packets sent, of the selected output audio stream, on both the Primary and Secondary connections.

Routed Time



Reports the number of input channels that are routed through to the Output Stream.

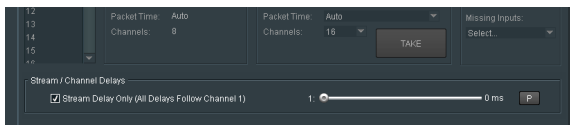
Missing Inputs



In circumstances where input channels are missing, using the drop-down menu, the actual missing channel(s) is/are reported.

In the example above, it can be seen that input Channel 1 is missing in the selected output stream.

10.6.2 Output Stream / Channel Delays

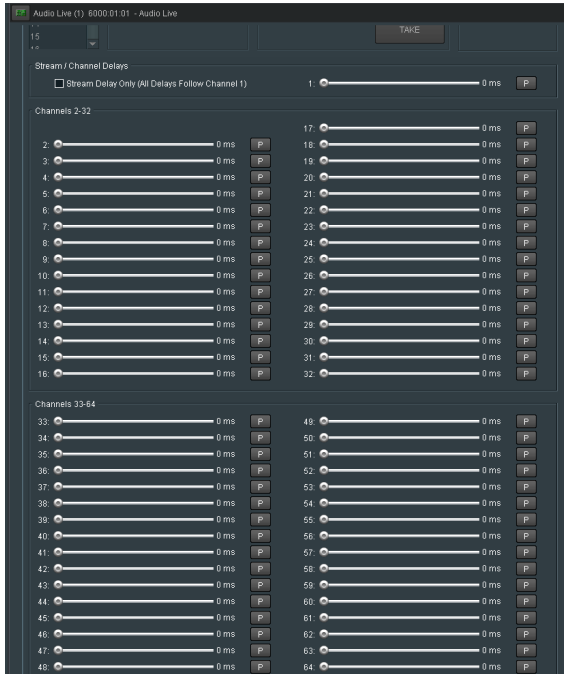


When the control **Stream Delay (All Delays Follow Channel 1)** is enabled, a generic delay is applied to all audio channels.

The range of the control is 0 to 5000ms (5 seconds).

Default value is 0ms.

When the control **Stream Delay (All Delays Follow Channel 1)** is unchecked, the extended delay menu will appear:



It is now possible to configure unique delay values to each channel.

Each control has a range of 0 to 2000ms

Default is 0ms.

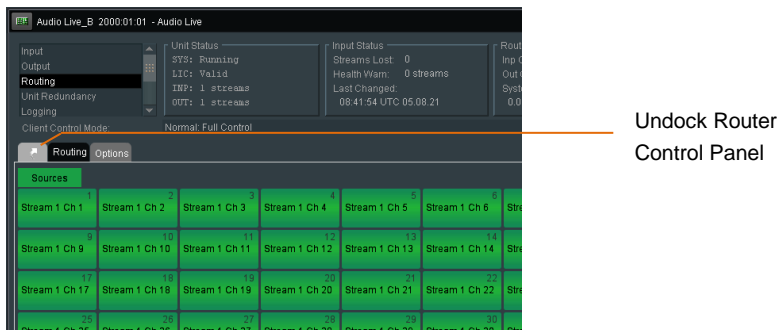
10.7 Audio Live Routing Control

In the **Audio Live** main menu, the **Routing** template is displayed when **Routing** is selected.

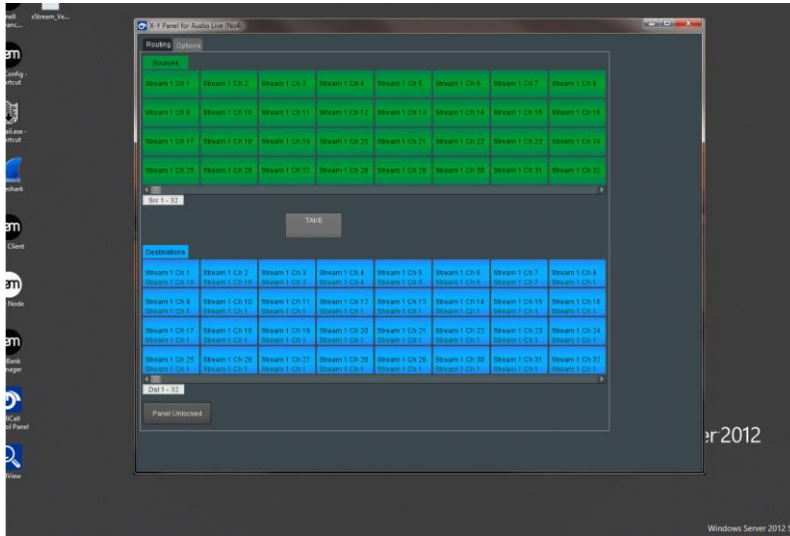


10.7.1 Router - undocking

The Routing Control Panel maybe undocked from the RollCall Control Panel by clicking on the **Undock** button.



Undocked example:



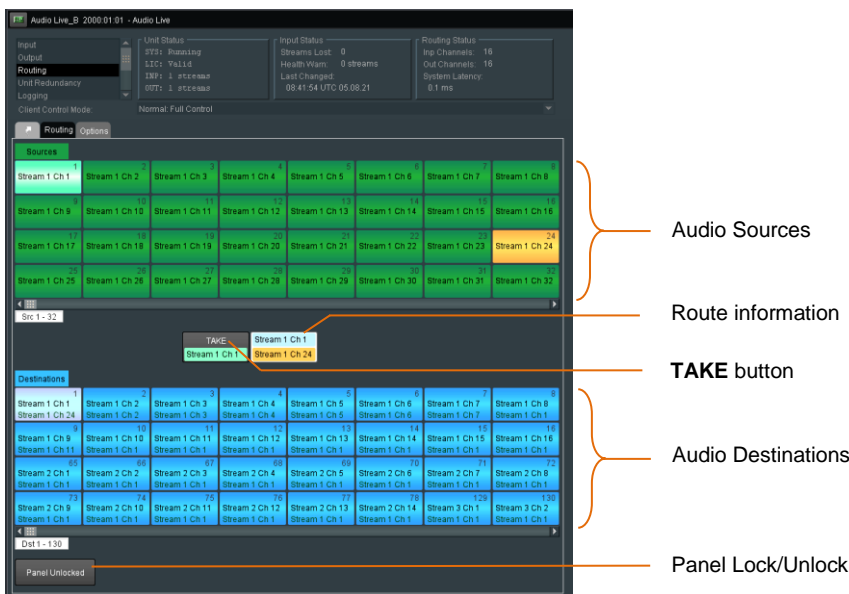
The properties of the Pop-out Display maybe adjusted under the options tab - see section 10.7.7 below.

10.7.2 Routing procedure

1. In the Audio Live menu, select **Routing**. The Routing template will load.

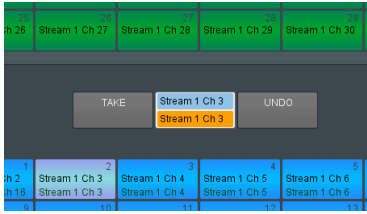


The Routing menu (under the Routing tab) is used to set up routes from sources to destinations. Note that the default view is display an 8x8 matrix of Audio Sources, and 8x8 matrix of Audio Destinations. The number of rows and columns can be modified (see section 8.5.9 below). In the following example, the number of rows displayed has been reduced to 4. Note that the **Route Information** field will only appear when a **Destination** is selected.



Routes can be set up to either occur immediately as soon as the route is made, or upon activating the **TAKE** button.

An **Undo** button can be added to the display so that a route may be undone. This can be configured in **Options** in **Router Functions** (see part 10.7.5 below).



- To modify the audio routing, first select an Output Audio Destination. In this example **Stream 1 Ch 5** is selected.



Note that when **Stream 1 Ch 5** is selected, the current input channel routed to **Stream 1 Ch 5**, will be highlighted in the Sources matrix.

In the above example **AUD 1 IN1 Ch 5** is routed to **Stream 1 Ch 5**.

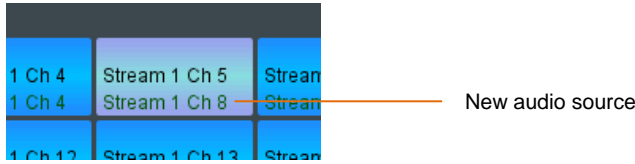
- Select the desired input that is to be routed to **Stream 1 Ch 5**.



Note that the **Take** button has been modified with the pending routing configuration.

In this example Audio Source **Stream 1 Ch 8** has been selected.

- Pressing the **Take** button will make the change permanent.
The output Routing button **Stream 1 Ch 5** will update with the new Input details.



10.7.3 Navigating the Router Page

Use the slider bars under the source and destination panels to navigate forward and reverse through the pages.

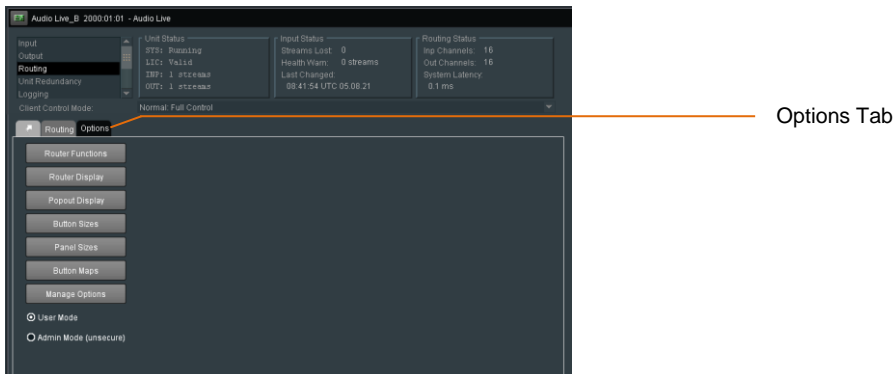


Navigating to another page of sources or destinations, displays a button indicating on which page the currently selected source or destination is on.

- Click the button to navigate to the page of the currently selected source or destination, as applicable.

10.7.4 Routing Options

Clicking on the Options tab will bring up the options menu.



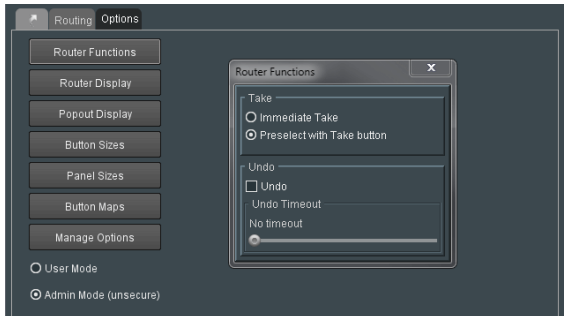
Note - in **User Mode** the **Options** buttons are not accessible (greyed out).

The **Options** buttons will become active when **Active Mode (unsecure)** is selected:



10.7.5 Router Functions

Selecting **Router Functions** will bring up the **Router Function** window.



Here it is possible to define **Take** and **Undo** buttons to be included in the Routing page. Check the **Preselect with a Take button** checkbox to add a **Take** button to the Routing page (default settings).

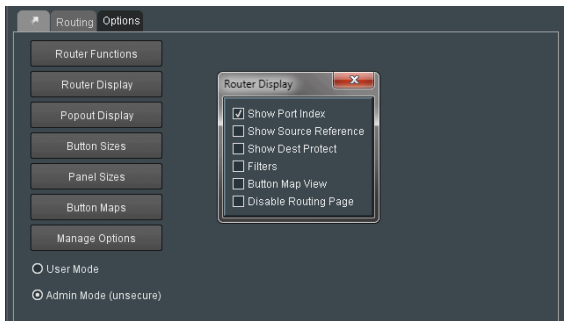
If no **Take** button is added, routing takes place immediately on selection of a source. Check the **Undo** checkbox to add an **Undo** button to the Routing page.

Add a timeout to the **Undo** function, by moving the slider. The time set is the period after performing a **Take** operation (or without a **Take** operation, if **Immediate** is set above) in which an **Undo** operation is possible, returning to the previous route before the **Take**.

Once the timeout time has elapsed the **Undo** operation is no longer possible.

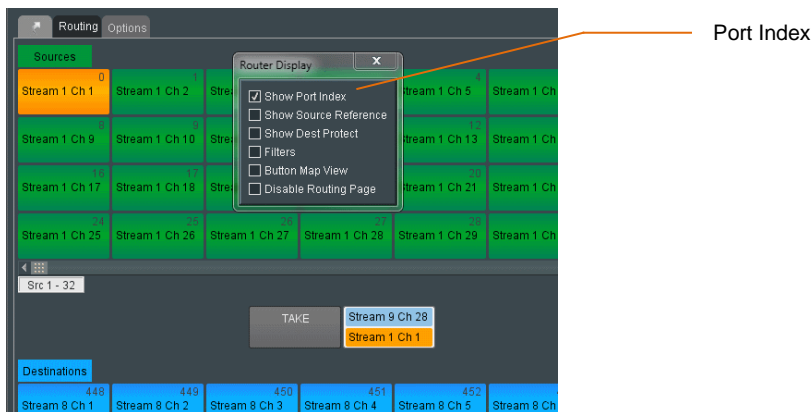
10.7.6 Router Display

Selecting **Router Display** will bring up the **Router Display** window.

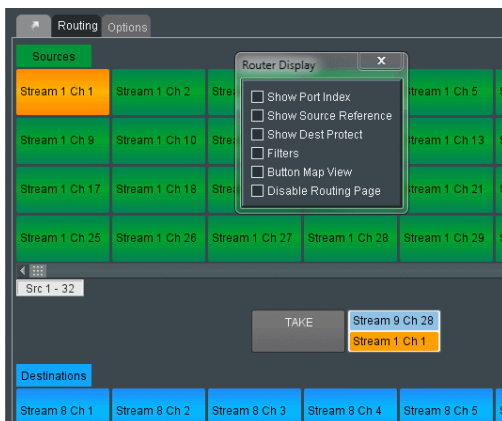


Show Port Index

When selected, the Port Index associated with each button, will be displayed:

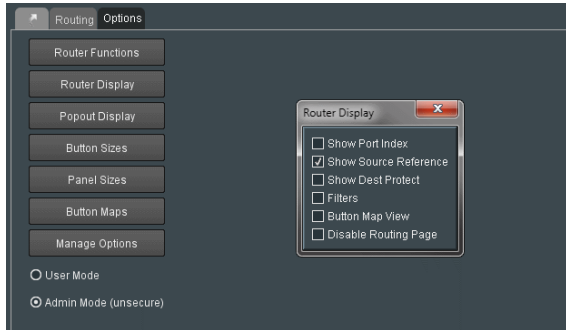


When disabled, the Port Index will not appear:



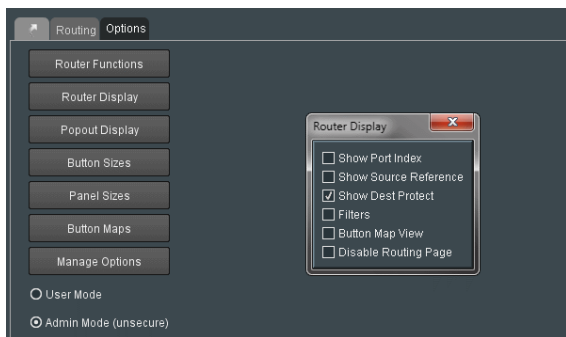
Default setting is **Enabled**.

Show Source Reference



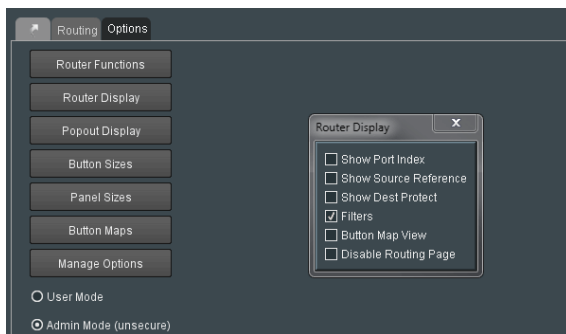
This control has no effect when associated with a GV Live Audio Routing process.

Show Dest Protect



This control has no effect when associated with a GV Live Audio Routing process.

Filters



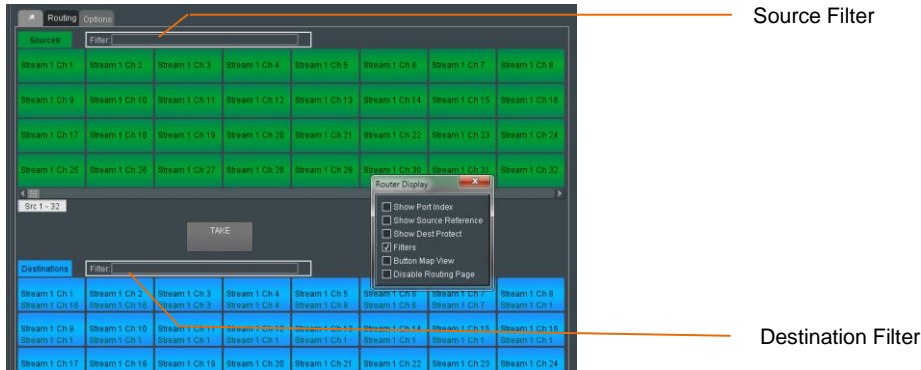
The Router Display **Filters** feature, when enabled, displays the filtering function on the **Routing** page.

In the example above, there is only a single input stream of 16 channels, routed to a single output stream of 16 channels. This makes the Filtering feature somewhat redundant as relevant audio channels can be easily found.

However, it should be considered that the GV Live Audio router can support up to 4096 audio channels (up to 64 streams, each up to 64 channels), hence locating a particular audio channel that is required to be configured, can be quite awkward in large routing matrix.

The filter feature allows a channel to be located by name.

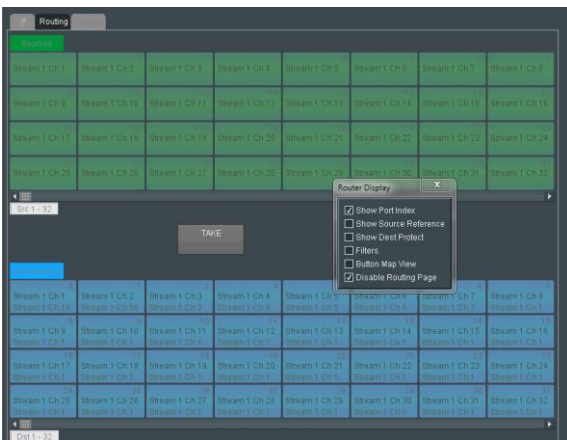
When **Filters** is enabled in the Router Display window, the Filtering functions will appear in the Routing menu.



Example of the input filter in use:



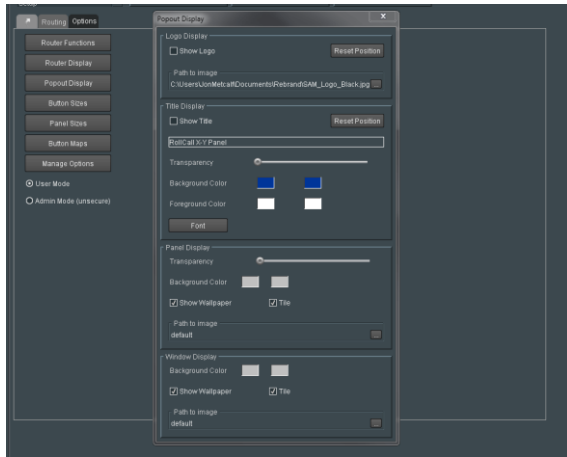
Disable Routing Page



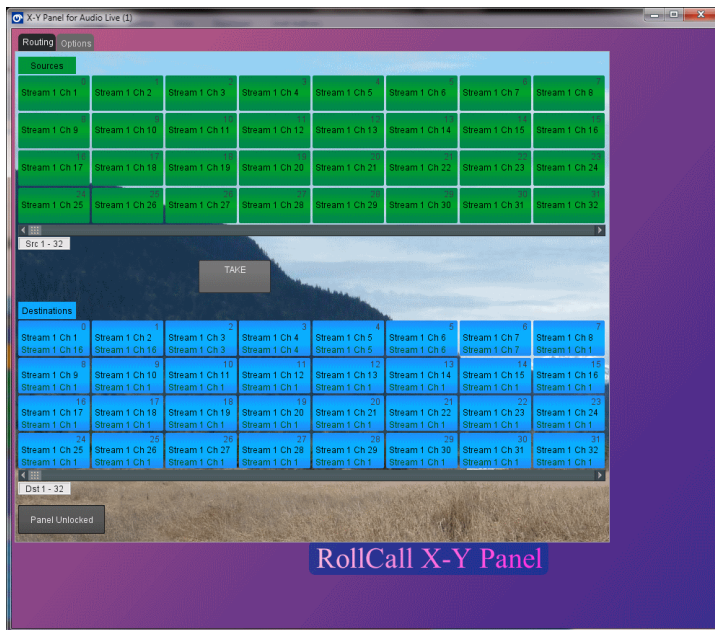
When selected, the Router Panel will be disabled.

10.7.7 Popout Display properties

The properties of the undocked Router Control Panel can be adjusted aesthetically if desired using the **Popout Display** menu.

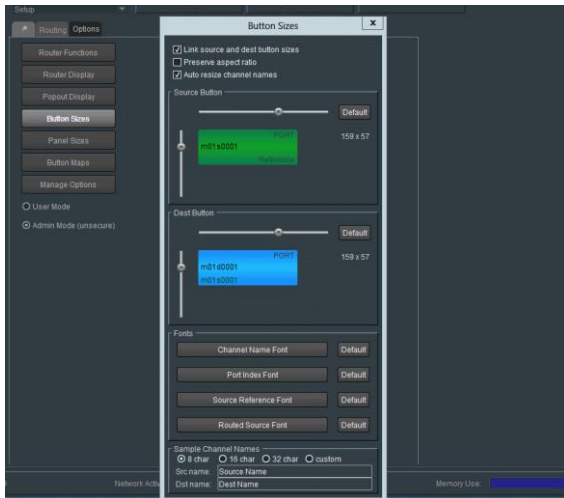


Example of a custom Popout Menu:



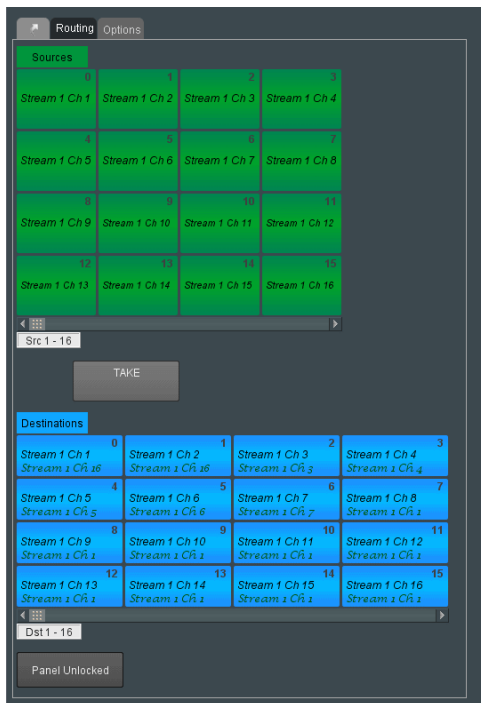
Using the **Popout Display** menu, the look of the Popout display can be customised; the background colors maybe adjusted, or an image (.jpg, .png, bitmap, etc) can be configured as the background, if desired.

10.7.8 Router Panel Button Sizes

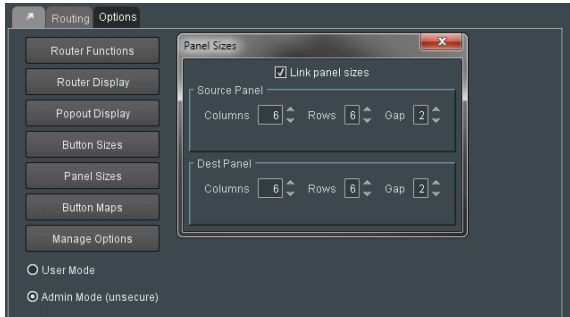


The size of buttons on the Router Panel may be adjusted using the **Button Sizes** menu. Custom fonts may also be configured.

Example of custom button size

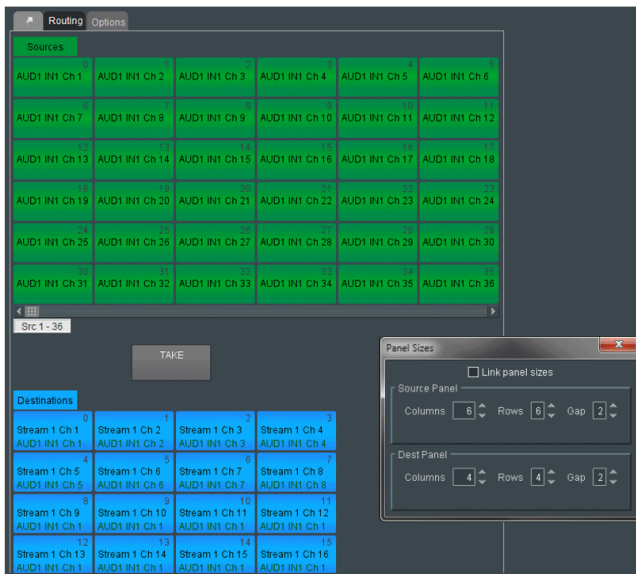


10.7.9 Panel Sizes



The Router Panel button matrix can be configured to have a specific number of rows and columns (Columns between 1 and 12, Rows between 1 and 8). Source buttons can be linked, resulting in the same number of Destination buttons as Source buttons.

Alternatively, Source and Destination can be unlinked meaning different button configurations can be set for Source, and Destination.



The GAP control adjusts the space between individual buttons:

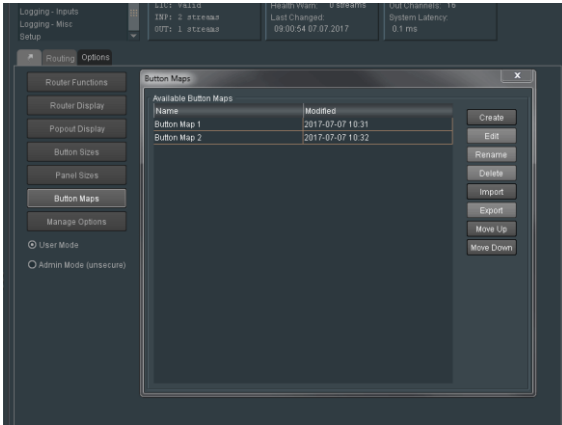


10.7.10 Button Maps

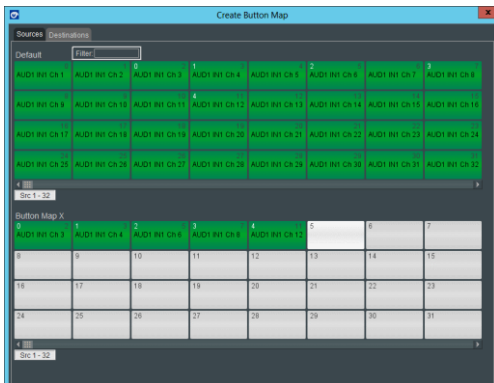
Use button maps to define alternative mappings of the source and destination buttons.

Map some or all of the buttons to limit the number of source and destination buttons visible on the Router displays, and/or the order in which they are displayed.

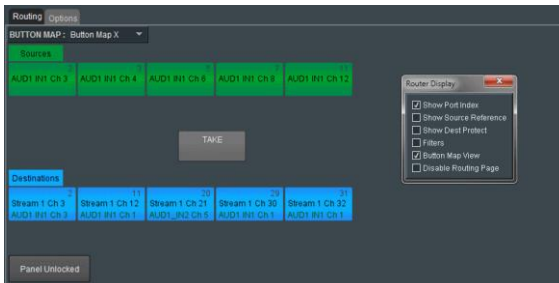
Several different button maps can be created, according to requirements.



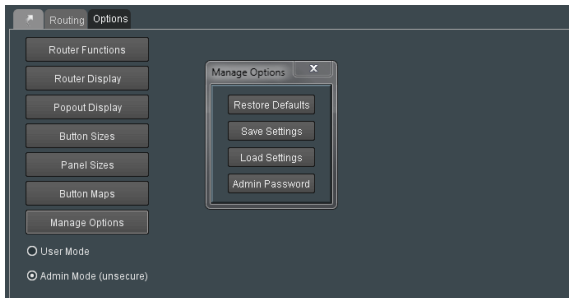
Button Maps may be created for both source and destination buttons:



The resultant Button Map can be viewed by selecting **Button Map View** in the **Router Display** menu.



10.7.11 Manage Options

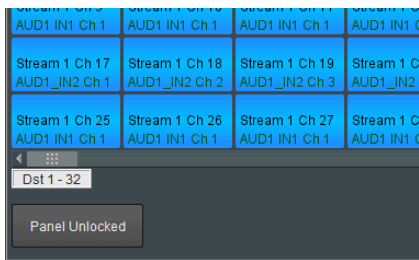


Here current configurations can be saved, or previously saved configurations can be recalled.

Default settings can be recalled.

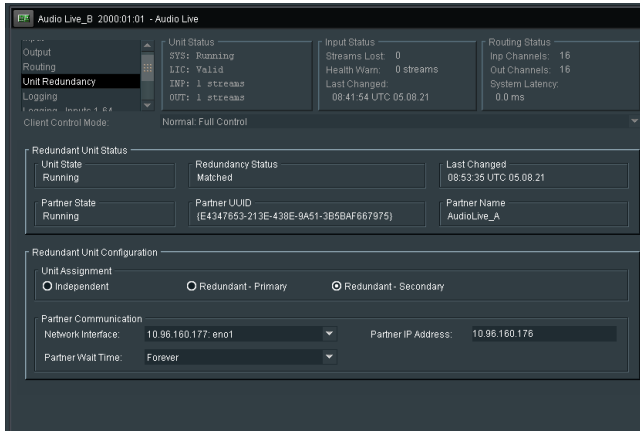
An **Admin Password** can be configured.

10.7.12 Panel Locked/Unlocked



The Router Panel can be locked using the **Panel Lock** feature.

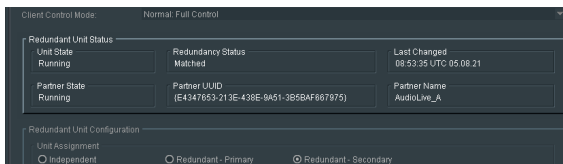
10.8 Unit Redundancy



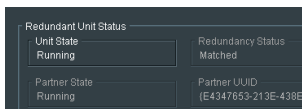
Unit Redundancy is a new feature added in version V1.5.3.42.

This new feature allows two Audio Lives to be paired and effectively run with redundant protection.

10.8.1 Redundant Unit Status



Unit State



Unit Status, indicates the Status of the Audio Live. There are two possible states:

- **Initializing**
- **Running**

Initializing

This indication may be seen upon launching the Audio Live template, if connection to the partner Audio Live has yet to complete.

During the initializing period, no audio packets will be output

Running

When operating in redundant mode, the **Running** status indicates either:

- successful connected and synchronised to the partner Audio Live,
- that the **Partner Wait Time** (see below) has expired, resulting in the Audio Live operating independently, with no redundancy.

Redundancy Status

| | | |
|--------|--|------------------------------|
| Status | Redundancy Status Matched | Last Changed 13:33:10 UTC |
| | Partner UUID (E4347853-213E-438E-9A51-3B5BAF667975) | Partner Name AudioLive_ |

The **Redundancy Status** will report either:

- **Matched**
- **Unknown**
- **Error**

Matched: When running in redundant mode, the **Matched** status indicates that the Audio Live is successfully connected to its partner Audio Live, and both are running as a redundant pair.

Unknown: When running in redundant mode, an **Unknown** status is an indication that the connection to the partner Audio Live has been lost.

Error: When running in redundant mode, an **Error** status is an indication that, although the connection to the partner Audio Live is successful, redundancy is not possible. There can be several causes of such a condition – see **Redundancy Notes** below.

Last Changed

| | |
|--------------|-----------------------|
| Last Changed | 09:44:23 UTC 10.08.21 |
| Partner Name | AudioLive_B |

Reports the time that the Redundant Audio Lives were last successfully paired and **Matched**.

Partner State

| | | | |
|---------------|---------|-------------------|--|
| Unit State | Running | Redundancy Status | Matched |
| Partner State | Running | Partner UUID | {062B56AE-54CA-4829-8897-A7BEA3051C32} |

Reports the status of the partner Audio Live system. A **Running** state must be present to allow redundancy.

Partner UUID

| | |
|--------------|--|
| Matched | 09:56:4 |
| Partner UUID | {062B56AE-54CA-4829-8897-A7BEA3051C32} |
| Partner Name | AudioLive_ |

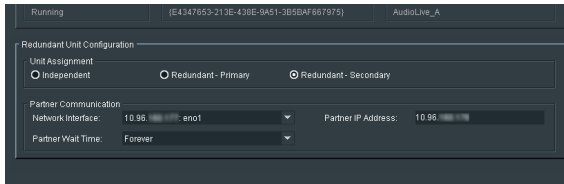
Reports the UUID of the Partner Audio Live.

Partner Name

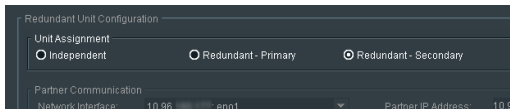
| | |
|--------------|-------------|
| Partner Name | AudioLive_B |
|--------------|-------------|

Reports the name of the Partner Audio Live.

10.8.2 Redundant Unit Configuration



Unit Assignment



There are three possible settings of the **Unit Assignment**:

- **Independent**
- **Redundant – Primary**
- **Redundant – Secondary**

Independent:

This setting should be configured if the Audio Live is to be used in a non-redundancy mode.

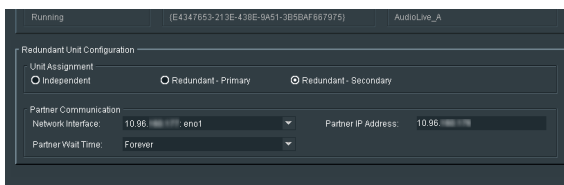
Redundant – Primary

When used with a partner Audio Live as a redundant pair, one unit must be defined as the Primary and one as the Secondary.

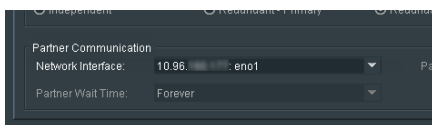
Redundant – Secondary

This setting should be configured if the partner Audio Live is designated as the Primary.

Partner Communication



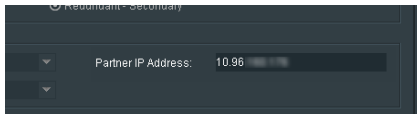
Network Interface



Allow for the configuration of the local interface employed for communication with the partner Audio Live. This may be set using the drop-down menu.

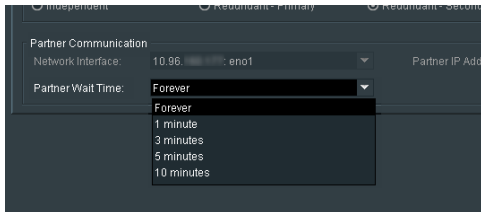
In this example, **eno1** has been configured.

Partner IP Address



The IP Address of the partner Audio Live is required to be entered in the **Partner IP Address** field.

Partner Wait time



If communication is lost with the partner Audio Live, certain circumstances could exist that would result in one Audio Live waiting for its partner that may not return resulting in no audio output from the redundant pair (see **Redundancy Notes** below).

The Partner Wait Time allow the User to define the maximum time that an Audio Live waits for its partner Audio Live to connect. Should the time expire, the Audio Live will revert to operating in isolation, without the protection of redundancy.

10.8.3 Redundancy Notes

Considering two Audio Live systems, Audio Live A and Audio Live B, configured as a redundant pair, where Audio Live A is configured to be the Primary and Audio Live B is configured to be the Secondary.

1. In the scenario where Audio Live A is operating correctly, but Audio Live B fails, Audio Live A will continue to operate in isolation and no loss of audio will be experienced downstream.
2. In the scenario described above in part '1', when Audio Live B returns, it will undergo an initializing period, during which it will copy the configuration of Audio Live A. During this initializing period, it will not output any audio packets.
3. In the scenario where both Audio Live A and Audio Live B fail maybe due to a common power failure, when an Audio Live returns (say audio Live A) it will try to connect to its partner Audio Live. During this period it will not output any audio packets.

However, should for any reason, the partner Audio Live (in this example Audio Live B) not return, this could potentially result in Audio Live A being in a permanent state of inactivity.

To prevent such a scenario, the control **Partner Wait Time** can be configured, where a period of time can be defined to wait for the partner Audio Live to return. Should this time period expire, then Audio Live A will commence to operate in isolation and will begin to output audio packets.

If Audio Live B returns at a later point in time, it will initialize, taking its configuration from Audio Live A and the Audio Lives will then continue to operate with redundant protection.

4. In the scenario that Audio Live A (Primary) fails, Audio Live B (Secondary) will continue to operate in isolation. At the point that Audio Live A returns, it will configure itself to the configuration of Audio Live B, despite Audio Live A being the Primary.
5. **Redundancy Status** may report an **Error** because of:
 - Both partner Audio Lives are configured to be the Primary, or both are configured to be the Secondary. For a Matched status, one must be configured as the Primary, and one as the Secondary.
 - If the partner is configured with the **Unit Assignment** set to: **Independent**.
 - If the configuration of the Input and Output spigots, of the partner Audio Lives, are different.

10.9 Logging

The **Logging** fields will appear. Default settings will enable all logged items.

The screenshot shows the 'Logging' configuration window in the Audio Live software. The window title is 'Audio Live_B_2000.01.01 - Audio Live'. The interface is organized into several sections, each with a 'Log Enable' checkbox and a table of log fields.

| Section | Log Field | Log Value |
|--------------------|---------------------------------|---|
| Logging Input | Max Number of Inputs | MAX_NUM_INPUTS= 188 |
| | Number of Active Inputs | NUM_ACTIVE_INPUTS= 1 |
| | Total Number of Input Channels | TOTAL_INPUT_CHANS= 16 |
| | Max Number of Synchro Inputs | MAX_SYNCHRO_INPUTS= 16 |
| | Number of Synchro Inputs | NUM_SYNCHRO_INPUTS= 3 |
| | Number of ST 2110-31 Inputs | NUM_ST31_INPUTS= 0 |
| Logging Output | Max Number of Outputs | MAX_NUM_OUTPUTS= 188 |
| | Number of Active Outputs | NUM_ACTIVE_OUTPUTS= 1 |
| | Total Number of Output Channels | TOTAL_OUTPUT_CHANS= 16 |
| | Number of ST 2110-31 Outputs | NUM_ST31_OUTPUTS= 0 |
| Logging Reference | Reference Source | REFERENCE_SOURCE= Input Stream 1 |
| | Reference State | REFERENCE_STATE= OK: Using Primary timing reference |
| | Reference Last Changed | REFERENCE_LAST_CHANGED= 09:56:18 UTC 10:08:21 |
| Logging Health | Network State | NETWORK_STATE= OK |
| | Processing Health | PROC_HEALTH= OK |
| | System Latency | SYS_LATENCY= 0.0 ms |
| Logging License | License Server Host | LICENSE_SERVER_HOST= 10.96.168.134 |
| | License Server Port | LICENSE_SERVER_PORT= 5093 |
| | License Server State | LICENSE_SERVER_STATE= Connected |
| | License State | LICENSE_STATE= Valid |
| Logging Redundancy | Redundancy Unit State | RDNDCY_UNIT_STATE= Running |
| | Redundancy Status | RDNDCY_STATUS= Matched |
| | Redundancy Last Changed | RDNDCY_LAST_CHANGED= 09:56:21 UTC 10:08:21 |
| | Redundancy Partner State | RDNDCY_PARTNER_STATE= Running |
| | Redundancy Partner UUID | RDNDCY_PARTNER_UUID= (E4347653-213E-438E-9A51-3B5BAF667976) |
| | Redundancy Partner Name | RDNDCY_PARTNER_NAME= AudioLive_A |
| | Redundancy Assignment | RDNDCY_ASSIGNMENT= Redundant - Secondary |
| Logging Misc | Log Server Name | LOGSERVER_NAME= No Active Logger |
| | Software Version | SOFTWARE_VERSION= 1.5.3.42 |

Normal practice would be to leave the Logging fields as default, but Users may disable any fields they so desire, by simply un-checking the relevant check-box.

10.10 Logging – Inputs 1-64

The **Logging Inputs** field will appear. Default settings will enable all logged items.



10.11 Logging – Inputs 65-128

The **Logging Inputs** field will appear. Default settings will enable all logged items.

The screenshot displays the Audio Live software interface. At the top, there are four status panels: Unit Redundancy (SYS: Running, ITC: Yes/No, IRT: 1 at/ceas, IOT: 1 at/ceas), Unit Status (SYS: Running, ITC: Yes/No, IRT: 1 at/ceas, IOT: 1 at/ceas), Input Status (Streams Lost: 0, Health Warn: 0 streams, Last Changed: 09:56:18 UTC 10.08.21), and Routing Status (Inp Channels: 16, Out Channels: 16, System Latency: 0.0 ms). Below these is the Client Control Mode set to Normal: Full Control.

The main section is titled 'Logging Inputs' and contains a table with the following columns: Chan, INPUT_#_STATE, INPUT_#_NUM_CHAN, INPUT_#_HEALTH, and INPUT_#_LAST_CHANGED. The table lists channels from 65 to 128. Each row shows a channel number, a 'WARN: Disabled' status, a value of 0 for the number of channels, a 'WARN: Disabled' health status, and a timestamp of 09:56:12 UTC 10.08.21 for the last change.

| Chan | INPUT_#_STATE | INPUT_#_NUM_CHAN | INPUT_#_HEALTH | INPUT_#_LAST_CHANGED |
|------|----------------|------------------|----------------|-----------------------|
| 65 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 66 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 67 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 68 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 69 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 70 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 71 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 72 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 73 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 74 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 75 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 76 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 77 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 78 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 79 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 80 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 81 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 82 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 83 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 84 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 85 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 86 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 87 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 88 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 89 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 90 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 91 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 92 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 93 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 94 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 95 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 96 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 97 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 98 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 99 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 100 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 101 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 102 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 103 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 104 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 105 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 106 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 107 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 108 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 109 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 110 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 111 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 112 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 113 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 114 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 115 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 116 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 117 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 118 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 119 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 120 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 121 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 122 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 123 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 124 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 125 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 126 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 127 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |
| 128 | WARN: Disabled | 0 | WARN: Disabled | 09:56:12 UTC 10.08.21 |

10.12 Logging – Inputs 129-192

The **Logging Inputs** field will appear. Default settings will enable all logged items.



10.13 IPMI

Introduction

The **Intelligent Platform Management Interface (IPMI)** is a set of computer interface specifications that provides management and monitoring capabilities independently of the host system's CPU, firmware (BIOS or UEFI) and operating system.

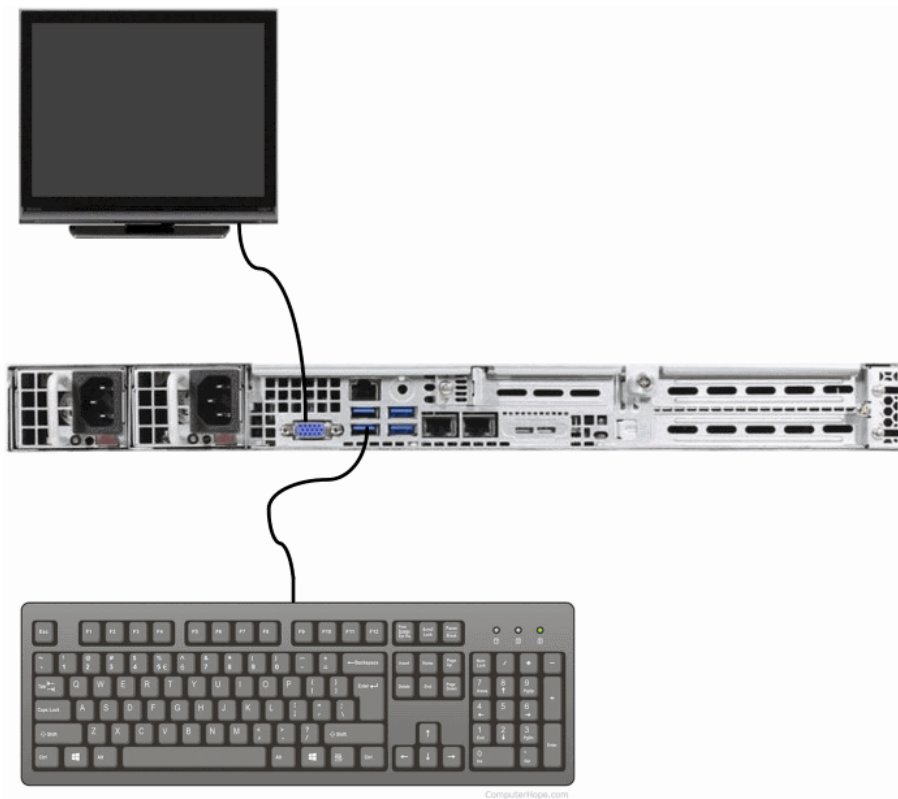
Practical uses of this application in a GV Live environment, is that it allows:

- Remote access to the host machine.
- Remote power cycling.
- Remote access to the BIOS of the host machine.

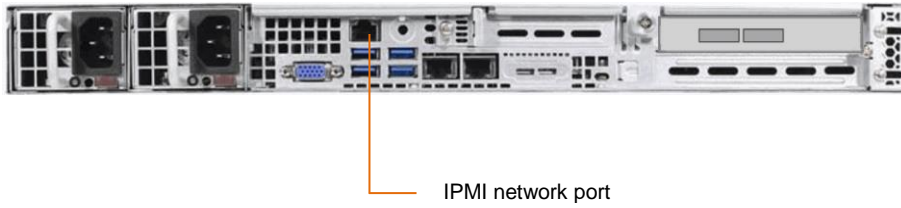
IPMI IP Address Configuration

The initial configuration of the IPMI network address will require local access to the GV Live host machine.

1. Connect a Monitor, Keyboard and Mouse to the Audio Live Host machine:



2. Connect the IPMI network port to the LAN.



3. Boot up the GV Live host machine.

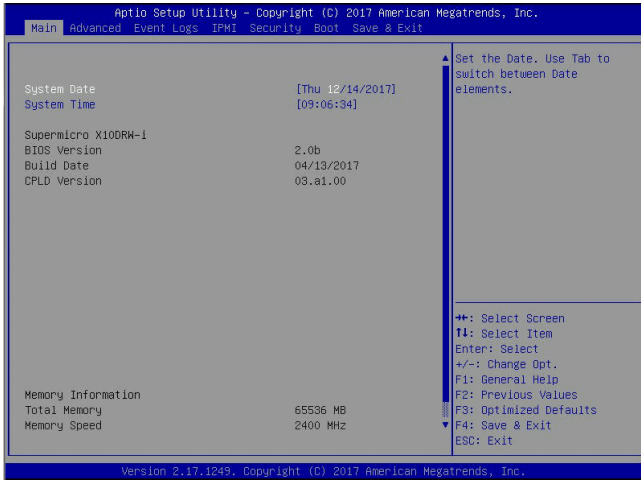
Note that the current IPMI IP Address will be displayed on the opening splash screen:



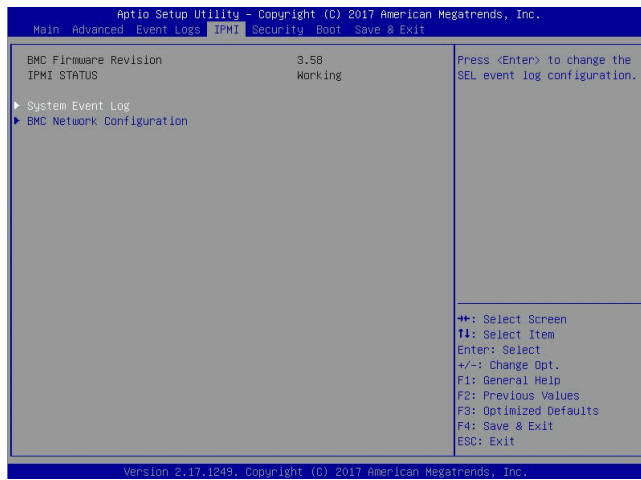
4. Upon seeing the splash screen, press `DEL` to enter the BIOS Menu:



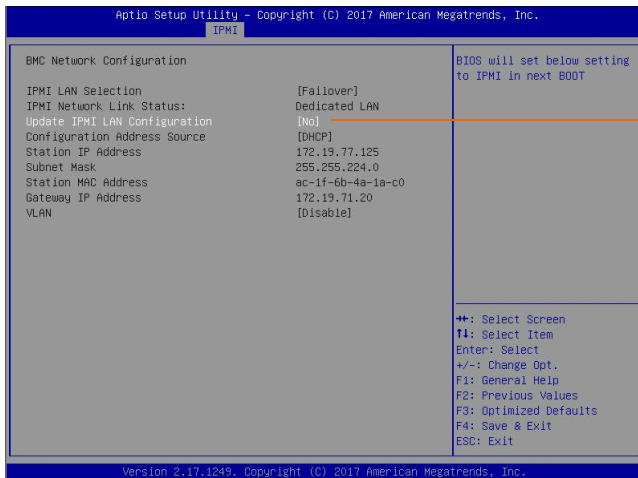
The BIOS menu will now appear:



5. Navigate to the IPMI menu:



- Enter the **BMC Network Configuration** submenu.

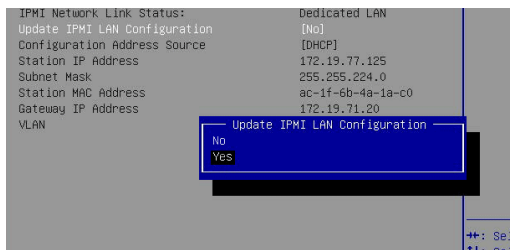


Update IPMI LAN Configuration
Default setting = [No]

Note that it is possible to lockdown the **BMC Network Configuration** menu, to deter unauthorised configuration. This is the Factory Default condition of this menu.

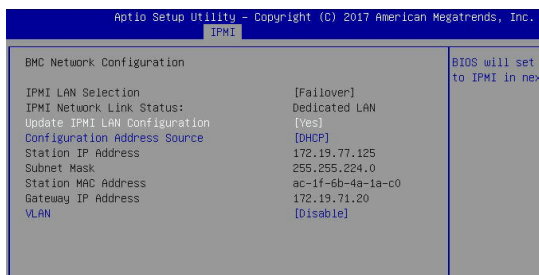
If the menu is in lockdown mode, the menu item **Update IPMI LAN Configuration** will be set to **No**, as shown above.

In order to interact with the **BMC Network Configuration** menu, it must be unlocked. This is done by setting **BMC Network Configuration** menu to **Yes**.

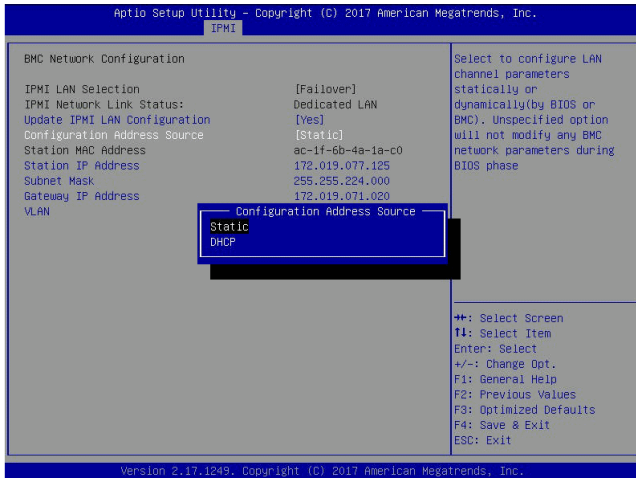


- The **BMC Network Configuration** menu will now be active.

In this example the **Configuration Address Source** is set to DHCP.



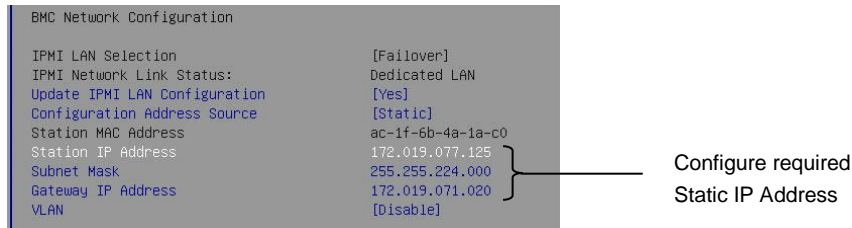
8. If it is desired to configure the address source to be a Static IP Address, select the menu item **Configuration Address Source**. A submenu will appear called **Configuration Address Source**:



The menu items:

- Station IP Address
- Subnet Mask
- Gateway IP Address

These items will all now become active and the desired Static IP Address details maybe configured.



Save and exit the BIOS menu and the GV Live host machine will continue to boot normally.

10.14 Remote Access via Web Browser

Remote access to the Alchemist Live, via the IPMI port, is possible from a remote PC. This procedure should work on a PC running Windows, Linux or Apple Mac OS.

The following example details remote access from a Windows PC (procedure should also work for Mac, or Linux)

It is a requirement that Java be installed on the remote PC.

BMC Login

Alchemist Live architecture is based on a Supermicro workstation. Supermicro workstations are pre-configure with unique BMC login details. The login details can be found attached to the shipping carton.

Please refer to the Supermicro document: BMC Unique Password for more information.

This can be downloaded from the Supermicro website, here:

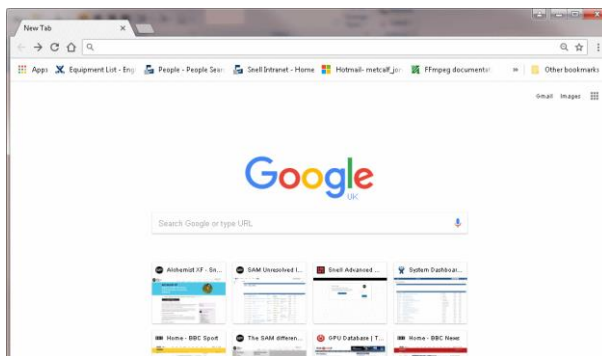
https://www.supermicro.com/support/BMC_Unique_Password_Guide.pdf

1. Open an Internet Browser

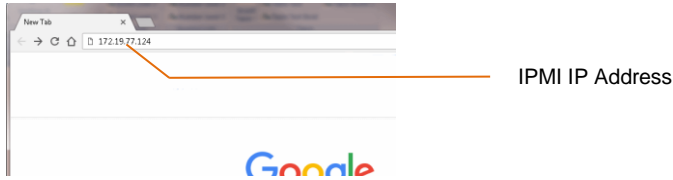
The following have been tested by GV.

- **Windows:** Google Chrome/Mozilla Firefox/Internet Explorer
- **IOS:** Safari

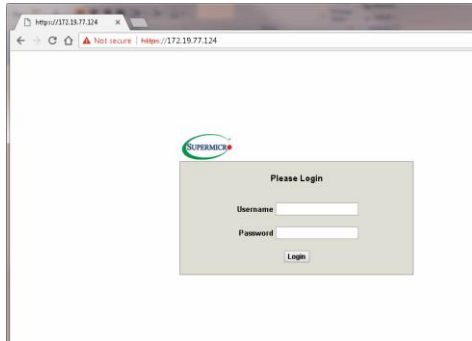
In this example, on a Windows system Google Chrome is the browser of choice:



- In the Address Bar, enter the IPMI IP Address:

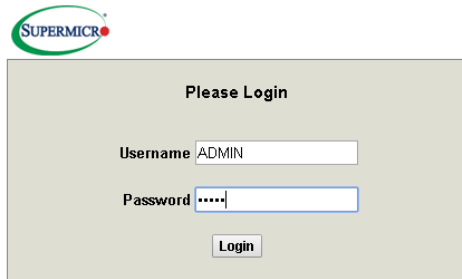


A login page will appear



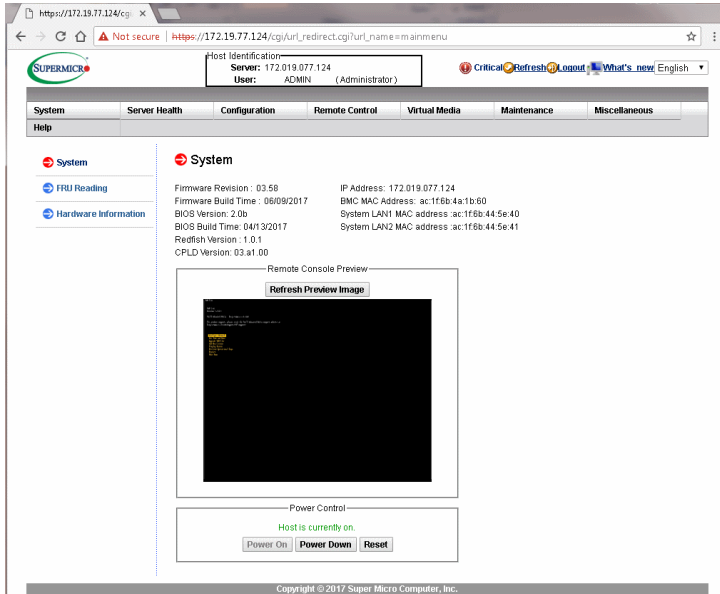
- Enter the User Name and Password.

Note: the login details are unique to the Alchemist Live host machine.
These are factory set (see preamble above).

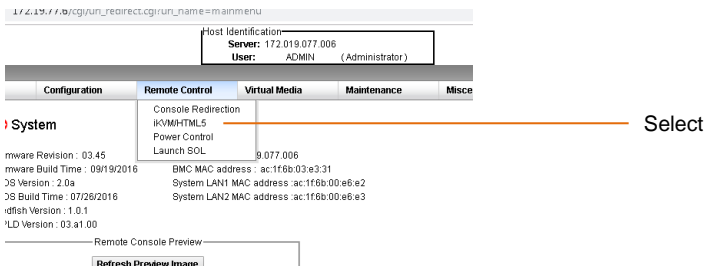


Press *return* to continue.

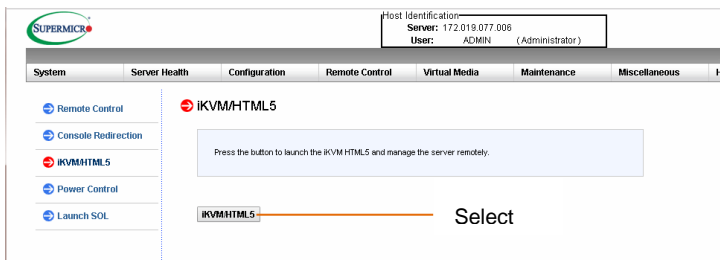
The Supermicro System menu will load:



- 4. In the option bar, hover mouse over **Remote Control**. A drop down menu will appear. Select **IKVM/HTML5**:



The **IKVM/HTML5** page will appear.



Select: **IKVM/HTML5**

The Console should now be displayed

```

GU Live
Version 1.5.3.42
Grass Valley - http://www.grassvalley.com/

For product support, please visit the Grass Valley support website at
http://www.grassvalley.com/support/

License Lock Code: *19C PP2M 6TXF 2ZHL
Chassis Host Name: server5

Configure Network - Standard (Recommended)
Configure Network - Custom (Advanced)
Network Diagnostics
Set Time and Date
Upgrade GU Live
Add New License
Display License Info
Display Status
Retrieve Operational Logs
Restart
Shut Down

```

If the following is displayed, press **Alt F2** to recover the console.

```

[~/bin/KVM View (1280x1024) (Resolution 1280x1024)]
[View: Home | Window | About | System | User Ctrl | Capture | Power Control | Exit]
[Error] Failed to start Load Kernel Modules.
See 'systemctl status systemd-modules-load.service' for details.
[OK] Found device 022220 from by (connect=4 led...)
[OK] Started Monitoring of LAMP servers, snapshots etc. using dmesgtool or progress polling.
[OK] Reached target Local File Systems (fs).
Mounting /mnt/auge...
[OK] Mounted /mnt/auge.
[OK] Listening on Load/Save RF Kill Switch Status /dev/rfkill1 Match.
[OK] Found device /dev/disk/by-uuid/787D-104B.
Mounting /boot/efi...
[OK] Mounted /boot/efi.
[OK] Reached target Local File Systems.
Starting Set console font and keymap...
Starting openhd - configure wireless devices...
Starting TeLL Pipepath To Write Out Runtime Data...
Starting LSB: ntpd.service initialization...
Starting Create Volatile Files and Directories...
[OK] Started TeLL Pipepath To Write Out Runtime Data.
[OK] Started Create Volatile Files and Directories.
[OK] Started Update UHF about System Boot/Shutdown...
[OK] Started Set console font and keymap.
[OK] Started Update UHF about System Boot/Shutdown.
[OK] Created slice system-serial.slice.
[OK] Started LSB: ntpd.service initialization.
[OK] Reached target System Initialization.
Starting Socket activation for snappy daemon.
[OK] Listening on PF2D Listen Socket.
[OK] Started Bolly Cleanup of Temporary Directories.
[OK] Started Time to automatically refresh installed snaps.
[OK] Listening on P-Bus System Message Bus Socket.
[OK] Started PF2D Events Check.
[OK] Reached target Paths.
Starting LD - unix socket.
[OK] Started ifup for ens1f0.
[OK] Started ifup for ens1.
[OK] Started ifup for ens2.
[OK] Started ifup for ens3f1.
[OK] Listening on Socket activation for snappy daemon.
[OK] Listening on LX0 - unix socket.
[OK] Reached target Sockets.
[OK] Reached target Basic System.
[OK] Started Regular background program processing daemon.
Starting To load the AHA driver...
[12:57:26.9] aha@sc version 13.0: Rohini loaded
Starting System Logging Service...
[OK] Started PF2D event daemon.
[OK] Started Deferred execution scheduler.
[OK] Started nls_interface.service - configure ens1f1.
Starting Login Service...
Starting To display the GUI Live console on thep...
[OK] Started PUSP Filenames for LAC.
Starting LX0 - container startup/shutdown...
Starting LSB: Record successful boot for GRUB...
Starting Thermal Daemon Service...
[OK] Started nls_interface.service - configure ens1f0.
[OK] Started D-Bus System Message Bus.
[OK] Started Thermal Daemon Service.
Starting LSB: fd monitoring daemon...
[OK] Started Snappy daemon.
Starting Auto Import assertions from block devices...
[OK] Started System Logging Service.
[OK] Started To load the AHA driver.
[OK] Started LX0 - container startup/shutdown.
Ubuntu 16.04.3 LTS: E157495C7900262: tty1
E157495C7900262 login: _

```

Appendix A. FAQ's

Q1: What happens if the reference input stream is removed?

A1: Nothing, Audio Live will lock on boot, from then on it will remain locked even if the input stream is removed.

Q2: What happens if all input streams are removed?

A2: If all inputs are removed the system will relock once the inputs are reapplied.

Q3: Does Audio Live except inputs with different packet times?

A3: Yes Audio Live can receive streams with different packet times:

Input: 125us, 250us, 500us, 1ms & 4ms (provided the packets size is within 1500 MTU).

Output: 125us, 250us, 500us, 1ms.

Note: Nominated input reference stream should be $\leq 1\text{ms}$

Q4: Does Audio Live require PTP Extended Headers to be present?

A4: It used to be the case that PTP Extended Headers were a requirement, but this is no longer the case. PTP Extended Headers are not required.

Q5: What are the packet timing options on output streams?

A5: For streams with ≤ 32 channels, the packet time will be 250us. For > 32 channels the packet time will be 125us.

Q6: Is it possible to control Audio Live via SWP08?

A6: Yes, via the IP Routing Control Server.

Q7: Does Audio Live accept asynchronous sources?

A7: Yes, up to 16 streams can be synchronized to the nominated reference stream.

Q8: Does Audio Live offer support for -7 style redundancy?

A8: Yes, on both input and output streams.

Q9: Can Audio Live realign input packets which are presented out of order?

A9: Yes

Appendix B. Input Timestamp Tolerance

The following table details the relationship between the in/out configuration to the Input Timestamp Tolerance and the influence this has upon the maximum audio delay that can be applied to an audio stream/channel:

Table 4: Input Timestamp Tolerance

| Input Configuration | Input Timestamp Tolerance |
|---------------------|---------------------------|
| 32 x 64 | 1s |
| 64 x 64 | 1s |
| 64 x 32 | 1s |
| 128 x 16 | 1s |
| 192 x 8 | 20ms |
| 8x64, 32x16, 128x8 | 20ms |
| 8x64, 64x16, 64x8 | 20ms |
| 16x64, 16x16, 96x8 | 100ms |
| 16x64, 32x16, 64x8 | 100ms |
| 16x64, 56x16, 32x4 | 100ms |
| 8x64, 98x16 | 100ms |

Appendix C. Input / Output Stream Channel Delay

The following table details the relationship between the input and output configuration on the associated input and output max Stream / Channel delay:

Table 5: Input / Output Delay

| Input Configuration | Input / Output Max Stream Channel Delay |
|---------------------|---|
| 32 x 64 | 5s |
| 64 x 64 | 5s |
| 64 x 32 | 5s |
| 128 x 16 | 2s |
| 192 x 8 | 2s |
| 8x64, 32x16, 128x8 | 2s |
| 8x64, 64x16, 64x8 | 2s |
| 16x64, 16x16, 96x8 | 2s |
| 16x64, 32x16, 64x8 | 2s |
| 16x64, 56x16, 32x4 | 2s |
| 8x64, 98x16 | 2s |

Appendix D. Agent – Input Logging Descriptions

| Message | Description |
|------------------------------------|--|
| OK | All good – none of the below |
| WARN: Disabled | Input is disabled |
| WARN: Lost | No valid audio channels detected |
| WARN: Timestamp error | The timestamp of the input packets are outside of the allowed tolerance from the reference input timestamps |
| WARN: <i>xxxx</i> pkt drops | The input stream has discontinuities in its packet sequence numbering – packets could have been lost at source, in the network fabric, or in the Audio Live packet handling |
| ERR: Unsupp (<i>xx ch</i>) | The detected input channel count exceeds the restriction placed on the spigot |
| ERR: Buffer overflow | The input buffer has reached the limit of its storage; it is not being emptied fast enough because the samples are arriving faster than the reference rate or there is a problem with the processing rate of the unit (a system error) |
| FAIL: Illegal rate (<i>xx</i> Hz) | The audio sample rate of the input flow doesn't match 48kHz (the only supported sample rate) |
| FAIL: No Reference | There is no reference input available so no inputs can be processed |