Contents

INTRODUCTION

As a long-standing advocate for open standards in network audio, L-Acoustics remains a committed member of the Avnu Alliance and promotes Milan-AVB as our primary solution for transporting audio streams. Milan-AVB, a natural progression of Ethernet technology, relies on AVB/TSN—a set of IEEE standards meticulously crafted to ensure reliable and deterministic transmission of time-sensitive traffic over Ethernet (IEEE802.3). The benefits of AVB/TSN directly enhance audio quality for line sources, particularly when utilizing our Wavefront Sculpture Technology (WST).

Implementing a Milan-AVB network requires that all the network bridges involved in transmitting AVB streams are Avnu-certified and have AVB options or profiles enabled. In certain projects, typically in theme parks or cruise ships, an installed sound system may have the constraint of relying on pre-existing, non-AVB layer-3 network infrastructure.

While Milan-AVB remains the first choice for network audio, L-Acoustics wishes to provide an alternative solution that continues our commitment to open standards and supports the use cases mentioned above. This is why L-Acoustics has added AES67 to a selection of its installation products.

This implementation is focused on supporting AES67 streams generated by the Q-SYS platform, which is the typical audio processing core found on the targeted installation use cases.

PURPOSE OF THIS DOCUMENT

The present Practical Guide for using AES67 with L-Acoustics and Q-SYS covers the following topics:
- The requirements for using AES67 in an L-Acoustics ecosystem
- Setting up the network, the AES67 senders and the AES67 receivers
- Setting options and connecting AES67 streams
- Effective resulting latency
- Handling AES67 connections in case of maintenance
- Network setup recommendations and tested network topologies
- Troubleshooting AES67 stream reception

TERMINOLOGY

DEVICE: L-Acoustics AES67-capable product.
AMP: L-Acoustics AES67-capable amplified controller
CONVERTER: L-Acoustics AES67-capable media converter

> **i** This symbol indicates additional detail that may be useful to the user.

> ⚠ This symbol indicates important information that must be checked or implemented by the user.

BEFORE STARTING

REQUIREMENTS

Hardware Requirements

L-Acoustics AES67 capable devices:
- LA7.16i Amplified Controller (minimum firmware version 2.13.3)
- LC16D Network Audio Converter (reciever/listener only - minimum firmware 2.14.0)

Third-party:
- Q-SYS Core
- Windows PC for Q-Sys Designer
- Windows or macOS PC for LA Network Manager (can be the same Windows PC used for Q-SYS Designer)

Software Requirements

L-Acoustics
- LA Network Manager: minimum version is 3.8. (For seamless redundancy version 3.9 or above is required)
- Q-SYS plugins: Amplified Controller 16 channels (version 1.8.0 or above), LC16D (version 1.1.0 or above)

Third-party
- Q-SYS Designer Software: minimum version 8.1.0; the current Long Term Support version of Q-SYS Designer is version 9.4.6, at the time of writing the present guide.
- Web Browser application: for setup of LC16D through its Web UI

> **i** Q-SYS Cores used by L-Acoustics during internal testing:
>   Q-SYS Core 110f ; Q-SYS Core 8 Flex ; Q-SYS Core Nano
> Versions of Q-SYS Designer used by L-Acoustics during internal tests:
>   8.1.0 ; 9.4.5 ; 9.9.0 ; 9.10.0

OPERATING MODES AND LIMITATIONS

Supported AES67 stream formats

A DEVICE supports AES67 streams with the following characteristics:

SAMPLING RATE	48 000 Hz
ENCODING	L24 (PCM 24 bit); L16 (PCM 16 bit)
NUMBER OF CHANNELS PER STREAM	From 1 to 8 channels
PACKET TIME	1 ms; 0.333 ms
DISTRIBUTION MODE	Multicast only
ADVERTISING PROTOCOLS	SAP/SDP; Manual

LA7.16i and LC16D can recieve up to 16 distinct AES67 streams, with up to 8 channels per stream

> **i** When LC16D is configured in AES67 mode, it only converts AES67 network audio streams to AES/EBU and MADI outputs. The conversion of AES/EBU and MADI input signals to AES67 network audio disabled.

USING L-ACOUSTICS WITH Q-SYS & AES67
PRACTICAL GUIDE

Daisy-chaining

The DEVICES can be daisy chained when their network mode is set to Normal.

In this situation the DEVICE is acting as a PTP Boundary Clock between its two Ethernet ports..

Supported redundancy schemes

Redundancy Scheme	Description	Supported	Audio interuption
Seamless network redundancy (as offered by Q-LAN 'Live A/B' or Milan-AVB)	Audio transmitted on both networks simultaneously using a pair of redundant streams	Yes*	No
Q-SYS Core redundancy	Redundant Q-SYS Core takes over in case of Primary Core failure, transmitting on the same multicast addresses	Yes	Yes (2 to 4 seconds)
Fallback to XLR	The amplified controller automatically switches to analog or AES/EBU auxiliary input in case of stream reception error	Yes	Yes (less than 1 second)
Spanning Tree redundancy	The Spanning Tree Protocol recalculates the spanning tree in case of network topology changes	Yes	Yes (dependant on the network size and STP convergence time)

ℹ * The minimum L-Aoustics firmware version that supports AES67 dual network redundancy is 2.14.0

Supported PTP clock modes

When the DEVICE is in AES67 mode, the media clock is directly derived from the PTP network clock.

The DEVICE is a PTP ordinary clock and can either be PTP Slave or elected PTP Clock Master, depending on the result of the BMCA or 'Best Master Clock Algorithm'.

See for details on how to manage the PTP Master Clock.

IGMP Snooping

The DEVICE is compatible with IGMP snooping as it answers IGMP queries for PTPv2 and subscribed AES67 multicast groups.
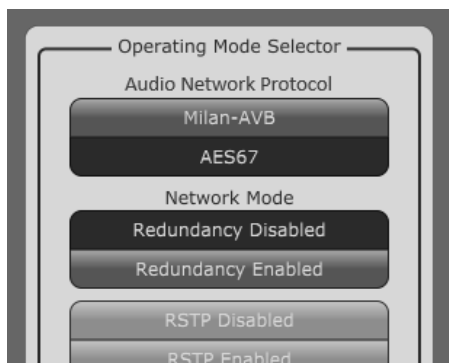
SETUP

Audio Network Protocol in LA Network Manager and Q-SYS plugin

Setting up a DEVICE in AES67 Audio Network Protocol

By default (at unboxing or after a reset to factory defaults), AES67 is disabled on the DEVICE(S) . For AMP(S) use LA Network Manager to configure them to receive AES67 streams. For CONVERTER(S), use the Web UI to set the network selection:

1. Add the AMP(S) to LA Network Manager session or connect to the AMP(S) using an existing session file.
2. In the Setup page, select the desired unit(s) and open the Operating Mode Selector panel.
3. Click the 'AES67' button in the Audio Network Protocol section.

> The DEVICE must be rebooted after changing the Audio Network Protocol.
>
> When the DEVICE is set to AES67 Audio Network Protocol, Milan-AVB does not operate on the DEVICE anymore. These modes are mutually exclusive.
>
> The Audio Network Protocol and AES67 stream connections of AMP(S) are saved with the LA Network Manager session. This setting may cause a device type conflict or require a reboot of the device when synchronizing virtual and physical units.
>
> The Audio Network Protocol and Network Mode settings are preserved on DEVICE firmware update.
>
> In case of difficulties at this step, contact L-Acoustics at nm@l-acoustics.com

Setting up the L-Acoustics Q-SYS plugin for AES67 control and monitoring

The L-Acoustics Q-SYS plugins allow users to control and monitor the AES67 streams reception on the DEVICE.

Note: This example shows LA7.16i for reference

1. Add a new instance or select an existing instance of the Q-SYS plugin controlling the DEVICE.
2. Edit the properties of the plugin:
   - Set the 'Audio Network Protocol' to 'AES67'.
   - If necessary, adjust the number of 'AES67 Streams' to be displayed on the plugin's UI, according to the project needs.
   - When dual network redundancy is implemented enable 'AES67 Redundancy'

Setting up a DEVICE in AES67 Audio Network Protocol (cont)

Q-Sys plugin property	Allowed values	Description
Audio Network Protocol	Milan-AVB; AES67 (Milan-AVB by default)	Updates the UI[1] and features of the Q-SYS plugin either for using Milan-AVB or AES67
AES67 Streams	1 to 16 (2 by default)	Adjusts the number of AES67 stream receivers to monitor and control from the plugin
AES67 Redundancy	Yes; No	Extends plugin controls for setup and monitoring of AES67 redundanct streams

1: This has no effect on the network mode of the connected device. For AMP(S) only LA Network Manager can set the device to AES67 mode, use the Web UI for CONVERTER(S)

> When connected to the DEVICE, if there is a mismatch between the DEVICE and the Q-SYS plugin regarding Audio Network Protocol selection, the plugin raises an error in the main status control, encouraging the user to adjust either the plugin or the DEVICE Audio Network Protocol (using LA Network Manager).
>
> The plugin design-time properties only have an impact on user interface and features access. Setting incorrect values will not interrupt audio or interfere with the DEVICE operation when IP connection is established between the plugin and the DEVICE.

AES67 page of the L-Acoustics Amplified Controller 16 channels plugin, while receiving one AES67 stream and its redundant replica from Q-SYS Core.

6

PTP Clock Master selection

The Best Master Clock Algorithm of PTPv2 allows the user to influence the election of the PTP Clock Master on the network.

This is done using PTP priority settings. The PTP device with the lowest priority numbers wins and becomes PTP Clock Master.

The PTP priority settings of the Q-SYS Core can be adjusted in the Design Properties dialog at design time.
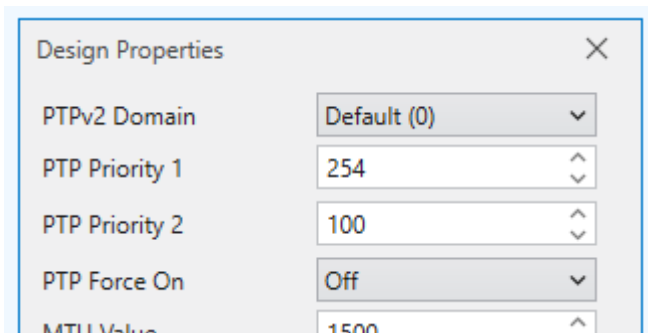
The PTP priority settings of the DEVICE can be adjusted at runtime using the controls in the AES67 page of the L-Acoustics Q-SYS plugin while it is connected to the device.
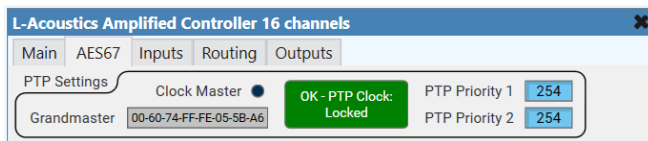
The plugin reports the Grandmaster ID and indicates if the DEVICE currently has the Clock Master role.

Additionally, the plugin reports the current PTP clock status on the device.

> **i** PTP priorities are set to 254 by default on the DEVICE.
>
> The PTPv2 domain of the DEVICE is always 0.

OPERATION

Connecting to an AES67 stream by name

The Stream Name control of the Q-SYS plugin exposes the list of detected and compatible AES67 streams on the Q-SYS Core and on the network.

The plugin detects streams of two kinds:
- AES67 Transmitters used in the current design that are named components or for which the *Script Access* property is set to either *Script* or *All*
- AES67 streams advertised on the network using the SAP/SDP protocol

Selecting a stream in the list automatically fills all the controls with the detected parameters of the stream.

It is possible to change the stream selection during reception. The reception is then automatically restarted using the new stream parameters.

Selecting the **<clear>** item resets all parameters to default value and stops stream reception
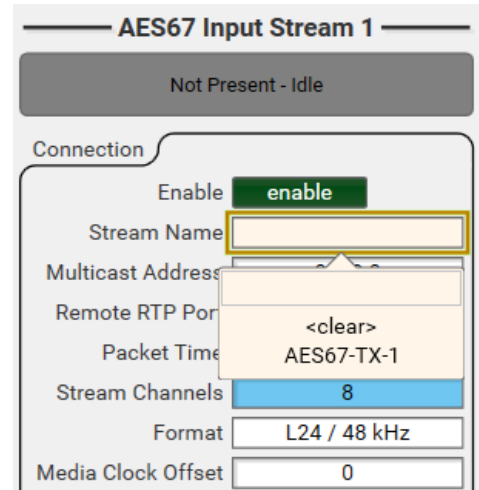
Connecting to an AES67 stream manually

If the desired AES67 stream is not listed in the Stream Name listbox, leave the Stream Name control blank and manually enter all stream information.

If the entered parameters exactly match one of the detected streams, the Stream Name control is then automatically filled with the associated stream name.

It is possible to change stream parameters when stream reception is enabled. The reception is then automatically restarted using the new stream parameters.

## Setting up AES67 network redundancy

Requirements

- Minimum L-Acoustics firmware version: 2.14.0
- The DEVICE is set in Redundancy Network Mode (AMP(S): Using LA Network Manager 3.9 or above)
- The primary Ethernet port of the DEVICE must be connected to Q-SYS LAN A and the secondary Ethernet port must be connected to the Q-SYS LAN B network.

Setting up AES67 network redundancy (cont)

Q-SYS design setup

1. Set the 'AES67 Redundancy' property of the L-Acoustics Q-SYS plugins to 'Yes'
2. Set the 'PTP Force On' design property parameter to 'Both' (this is not needed is the Q-SYS design already has network redundant Q-SYS devices configured)
3. Duplicate all AES67 transmitters that need to do network redundancy
   - Use the exact same property values (channel count, connection mode)
   - Route the exact same audio signals to each redundant pair of transmitters
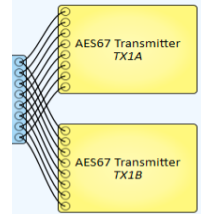
Runtime AES67 transmitters configuration

1. Set the value of the 'Interface' control to 'LAN A' on the first transmitter and to 'LAN B' for the second transmitter of each redundant pair
2. Mirror all audio parameters
   - Invert
   - Mute
   - Gain
3. Mirror all stream parameters*
   - Packet Time
   - Media Clock Offset
   - Network Tx Buffer

   inconsistent audio parameters will create audio artifacts
4. The only parameters that can have distinct values are:
   - Stream Name
   - Multicast address
   - Remote RTP Port
   - Remote RTCP Port
   - Interface
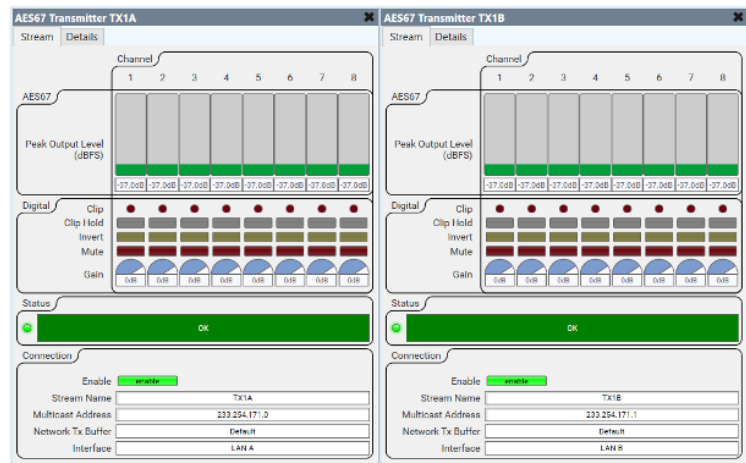
Runtime L-Acoustics Q-SYS plugins configuration

1. Select the appropriate stream names for both primary and secondary sections (or enter stream parameters manually if required)
2. Enable stream reception
3. Recommended *:
   - In the 'IP Connection' section of the 'Main' page, enter both the primary and secondary IP addresses of the DEVICE
   - Enable the 'Automatic Failover' button

   By carrying out these additional steps, in the scenario where one network is failing the plugin automatically reconnects the DEVICE via the secondary network and maintains control and monitoring.

• l-acoustics.com • l-isa.l-acoustics.com •
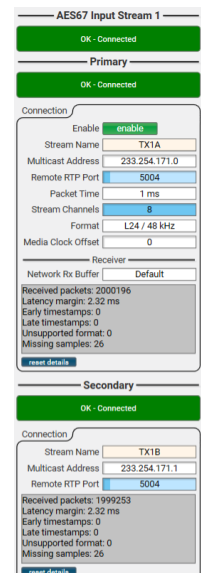
Network Rx buffer, Packet Time and Latency

By default, the DEVICE applies a network latency of 3 packet times. It is possible to extend the latency to 4 packet times in cases audio packets arrive too late due to network forwarding time exceeding the default maximum.

> **i** The DEVICE is compatible with packet times of 0.333 ms and 1 ms

Use the Network Rx Buffer control to choose between 'Default' (3 packet times) and 'Extra 1 packet time' (4 packet times).
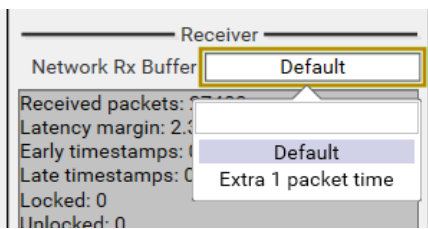
Network Rx buffer, Packet Time and Latency (cont)

Network Rx Buffer is an internal setting of the DEVICE's AES67 reception module, and it is the only parameter that does not have to match the AES67 stream description.

Network latency		Packet Time	
		0.333 ms	**1 ms**
Network Rx Buffer	Default	1 ms	3 ms
	Extra 1 packet time	1.333 ms	4 ms

> **i** In the Details text indicator located below the Network Rx Buffer control, one of the listed values is named 'Latency Margin'. This value indicates how early the audio packets arrive in the DEVICE prior to their playing time and helps choosing the optimal latency setting.
>
> If the latency margin is too short, increase the Network Rx Buffer to reduce the risk of dropping late audio packets.
>
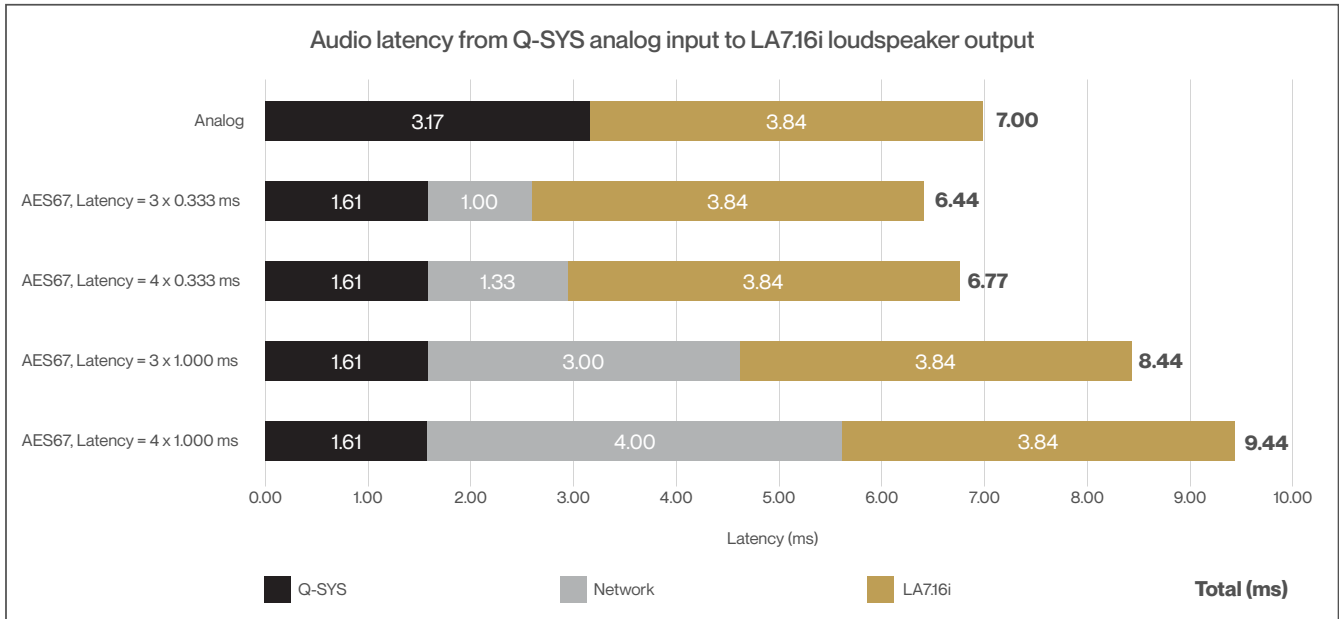> If the latency margin is comfortably high, consider reducing the packet time in case shrinking the overall latency is an interesting benefit for the installation.

> ⚠ AES67 latency parameters <u>MUST</u> be absolutely identical for all Amplified Controllers that drive a given line source. In general, it is highly recommended to use identical latency parameters within a given sound system for simplicity.

Network Rx buffer, Packet Time and Latency (cont)

The chart below shows the measured total latency from a Q-SYS analog input to the loudspeaker output of LA7.16i, using various Packet Time and Network Rx Buffer settings.

The network latency is added to the processing time induced by the Q-SYS Core and the LA7.16i DSP stage.

Audio latency from Q-SYS analog input to LA7.16i loudspeaker output

Setting	Q-SYS	Network	LA7.16i	Total (ms)
Analog	3.17		3.84	7.00
AES67, Latency = 3 x 0.333 ms	1.61	1.00	3.84	6.44
AES67, Latency = 4 x 0.333 ms	1.61	1.33	3.84	6.77
AES67, Latency = 3 x 1.000 ms	1.61	3.00	3.84	8.44
AES67, Latency = 4 x 1.000 ms	1.61	4.00	3.84	9.44

The latency between Q-SYS analog ingest and Q-SYS AES67 packetization is smaller than the constant analog-to-analog 3.17 ms latency through a Q-SYS system.

In case time alignment is needed between analog and AES67 audio paths, add Delay audio components upstream of analog or AES67 outputs in Q-SYS Designer.
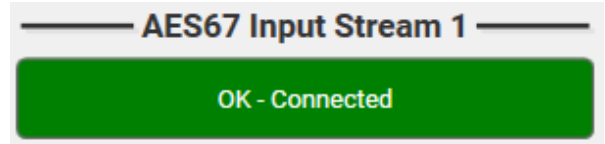
Leave the 'Network Tx Buffer' parameter of the Q-SYS AES67 Transmitter to 'Default'. Otherwise, there is a risk of early timestamp errors and audio packets being dropped on the receiver.

• l-acoustics.com • l-isa.l-acoustics.com •

Monitoring AES67 stream reception

The Status control of each AES67 stream reports the overall status of stream reception.

AES67 stream status	Description
Not Present - Idle	The stream receiver is disabled
Initializing - Connecting	After the stream reception started, waiting for audio packets to be received.
OK - Connected	Audio packets are received, and no error detected
Fault – The received timestamps are early or late	Audio packets are received but cannot be played because timestamps are too early or too late compared to the running PTP media clock
Fault - The payload size of received packets doesn't match the sink parameters	Audio packets are received but the size of the payload does not match the expected data size.
Fault - The stream has been disconnected	The DEVICE does not receive audio packets
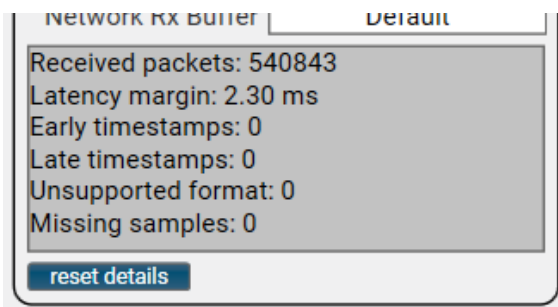
The AES67 receiver exposes a list of counters and statistics to help with troubleshooting.
(The list of values may vary depending on firmware version or device type):

Click the 'reset details' button to reset all counters to zero.

RECOMMENDATIONS

General recommendations

- If multiple Amplified Controllers are driving enclosures contributing to the same line source, then connect these Amplified Controllers to the same network switch. This reduces the risk of time misalignment due to PTP jitter or PTP clocking offset.

- Force one Q-SYS Core as PTP Grandmaster using the PTP Priority 1 parameter in the Q-SYS Design Properties window, to minimize the risk of PTP Grandmaster recalculation.

- Leaving default PTP priority settings on Q-SYS Cores and L-Acoustics DEVICES could result in one of the L-Acoustics DEVICES becoming PTP Grandmaster because of their default PTP Priority 1 parameter being set to 254. See: **"Supported PTP clock modes", page 4** for more details.

Network setup recommendations

- Update the network switches to their latest available firmware.
- Configure the network switch which has the lowest IP address as IGMP Querier

Network switch configuration examples

The following network switch configurations were used during internal tests of L-Acoustics AES67 implementation.

Luminex Gigacore range

- **Tested models**:
  - GigaCore 26i (firmware 3.0.2)
  - GigaCore 10 (firmware 3.0.2)

- **AES67 Group settings**:
  - IGMP
    - Snooping: enabled
    - Querier: enabled
  - PTP (before firmware 3.0.2)
    - PTPv2: enabled
    - Mode: E2E
    - Domain: 0

Netgear Pro AV range

- **Tested models**:
  - M4250-10G2F-PoE+ (firmware 13.0.4.19)
  - M4350-24G4XF (firmware 14.0.2.18)

- **VLAN Network Profiles**:
  - Audio AES67
  - Audio Q-SYS (in cases where the VLAN is shared with Q-LAN streams)
- **Other settings**
  - PTP transparent clock (PTP residency time stamping) enabled

Example network topologies

Star topology #1

LA7.16i

NETWORK SWITCH #2

LA7.16i
LA7.16i
⋮ 8 units
LA7.16i
LA7.16i

Q-SYS Core 110f
AES67 sender

LAN A

NETWORK SWITCH #1

LA7.16i setup
LA7.16i setup

NETWORK SWITCH #3

LA7.16i
LA7.16i
⋮ 7 units
LA7.16i
LA7.16i
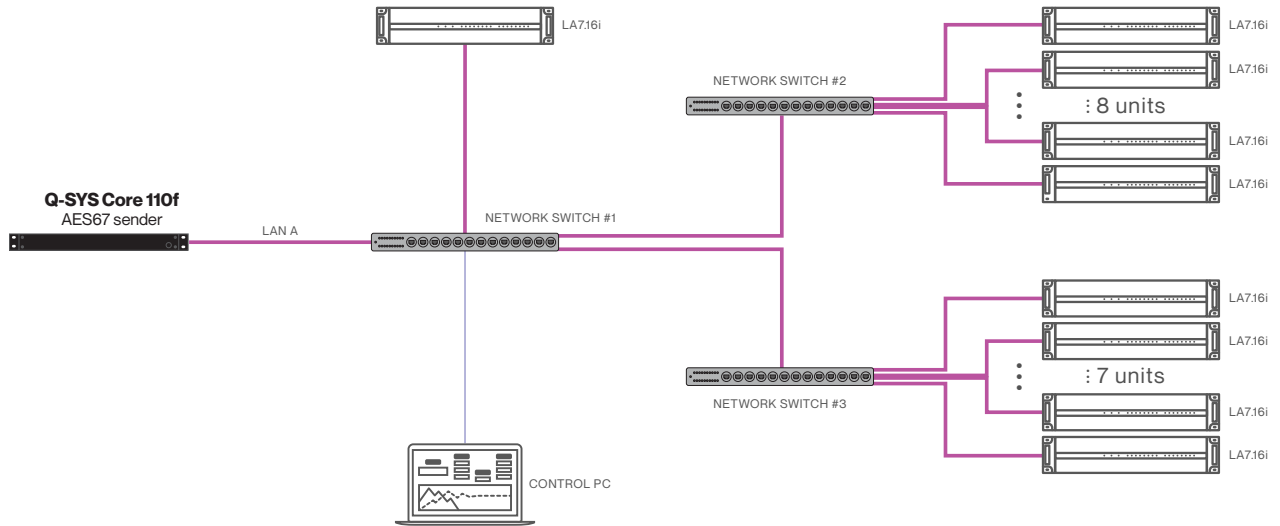
CONTROL PC

Q-SYS Core setup
- PTPv2 grandmaster
- 16 x AES67 output streams
  - 8 channels
  - 48 kHz
  - Packet time 1 ms

PC setup
- Q-SYS Designer
- LA Network Manager
  (connected to the amplified controllers)

Network switch setup
- QoS configured for AES67
- PTPv2 transparent E2E clock
- IGMP Snooping & 1 Querier

LA7.16i setup
- PTPv2 slave
- 2 x AES67 input streams
  (connected from Q-SYS core)
- Normal network mode

Star topology #2

AES/EBU + MADI Outputs

LC16D

Q-LAN
Reciever

NETWORK SWITCH #5

NETWORK SWITCH #4

LA7.16i

NETWORK SWITCH #3

LA7.16i

Q-SYS Core 110f
AES67 sender

LAN A

NETWORK SWITCH #1

NETWORK SWITCH #2

LA7.16i

LA7.16i

Q-SYS Core setup
- PTPv2 grandmaster
- 1 x AES67 output stream
  - 8 channels
  - 48 kHz
  - Packet time 1 ms
- 15 x Q-LAN output streams
  - 8 channels

Network switch setup
- QoS configured for
  Q-LAN + AES67
- IGMP Snooping & 1 Querier

LA7.16i & LC16D setup
- PTPv2 slave
- 1 x AES67 input stream
  (connected from Q-SYS core)
- Normal network mode
- Default latency
  (3 packet times)

Ring topology

Q-LAN Reciever

LA7.16i

NETWORK SWITCH #4

LA7.16i

NETWORK SWITCH #5

REDUNDANCY

NETWORK SWITCH #3

LA7.16i

Q-SYS Core 110f AES67 sender

LAN A

NETWORK SWITCH #1

LA7.16i

NETWORK SWITCH #2

LA7.16i

Q-SYS Core setup
- PTPv2 grandmaster
- 1 x AES67 output stream
  - 8 channels
  - 48 kHz
  - Packet time 1 ms
- 15 x Q-LAN output streams
  - 8 channels

Network switch setup
- QoS configured for Q-LAN + AES67
- IGMP Snooping & 1 Querier

LA7.16i setup
- PTPv2 slave
- 1 x AES67 input stream (connected from Q-SYS core)
- Normal network mode
- Default latency (3 packet times)

Redundant topology

AES/EBU + MADI Outputs

LA7.16i

LC16D

LAN B NETWORK SWITCH #2

LA7.16i

LA7.16i

LA7.16i

LAN B NETWORK SWITCH #1

Q-SYS Core 110f AES67 sender

LAN B
LAN A

:12 units

LAN A NETWORK SWITCH #1

LA7.16i

LA7.16i

LA7.16i

CONTROL PC

LAN A NETWORK SWITCH #2

Q-SYS Core setup
- PTPv2 grandmaster
- Redundant pair of AES67 output streams
  - 1 on LAN A & 1 on LAN B
  - 8 channels
  - 48 kHz
  - Packet time 1 ms

PC setup
- Q-SYS Designer
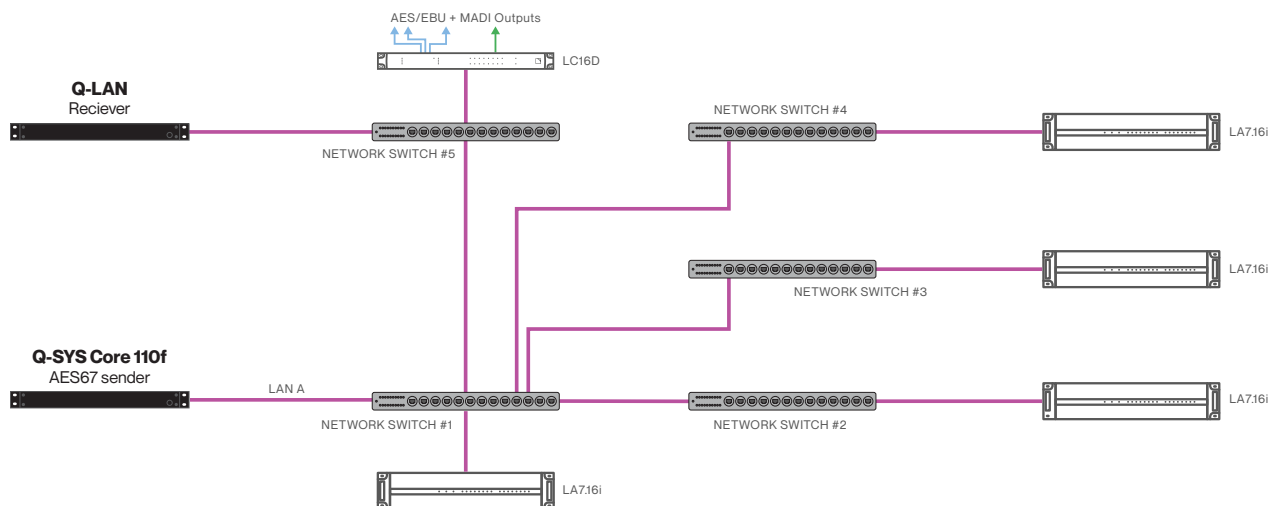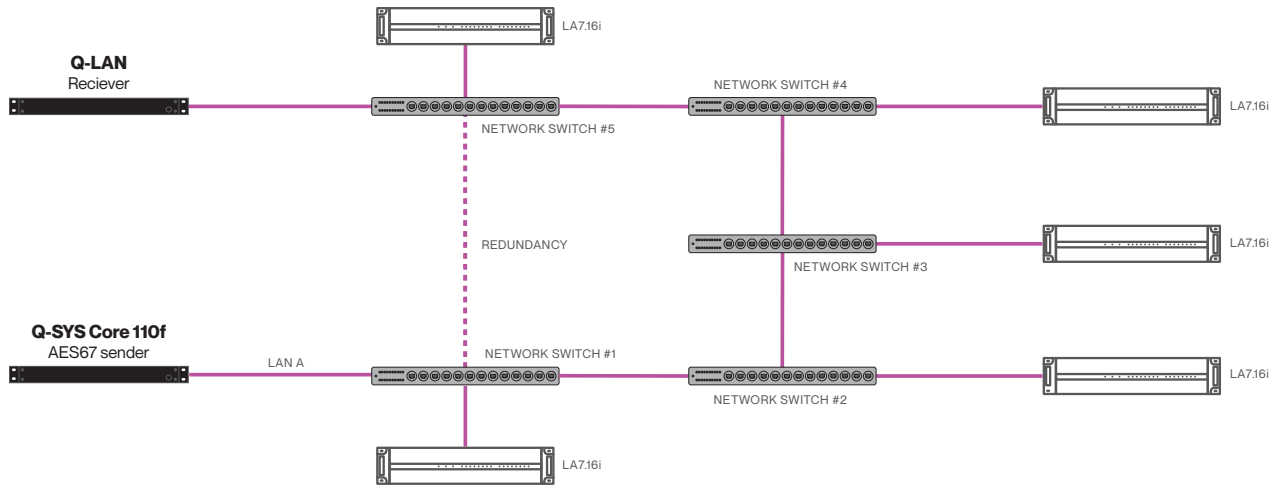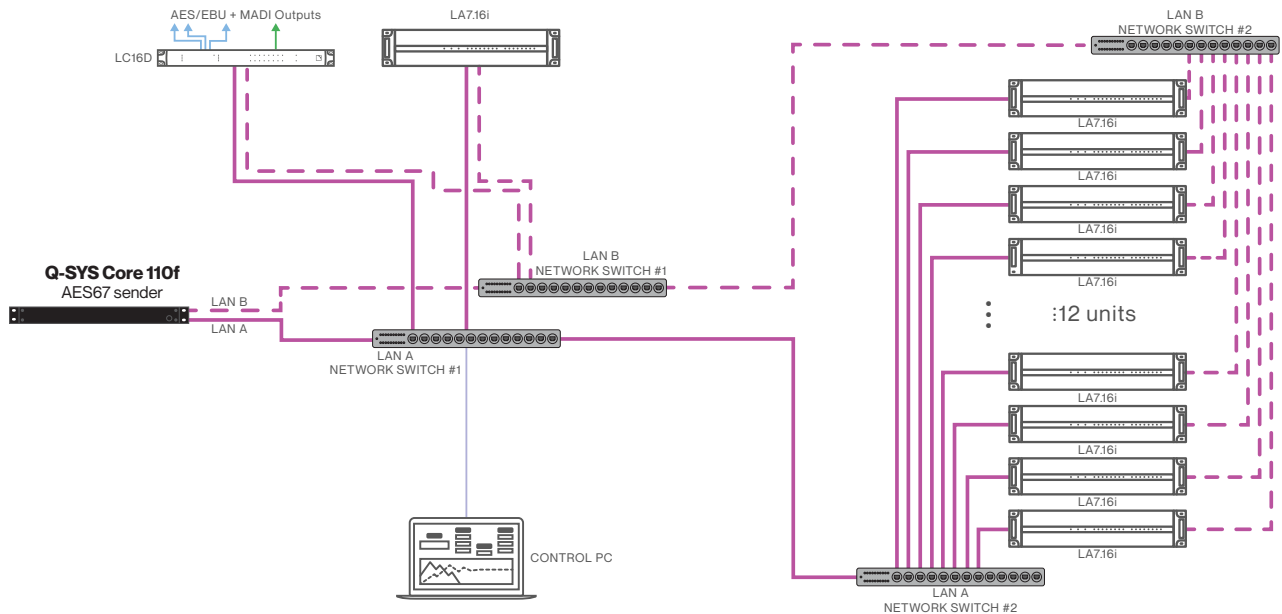- LA Network Manager (connected to the amplified controllers)

Network switch setup
- QoS configured for Q-LAN + AES67
- IGMP Snooping & 1 Querier

LA7.16i & LC16D setup
- PTPv2 slave
- Redundant pair of AES67 input streams (connected from Q-SYS core)
- Redundant network mode
- Default latency (3 packet times)

MAINTENANCE

DEVICE firmware updates

On firmware update of the DEVICE, the Audio Network Protocol setting is preserved, but the AES67 connection parameters are reset to default.

For AMP(S) LA Network Manager can restore previously configured AES67 connections by loading a session file containing the desired AES67 connection parameters.

DEVICE reset to factory defaults

When the DEVICE is reset to factory default settings, its Audio Network Protocol is reset to Milan-AVB and its AES67 connections and erased and reset to default values.

For AMP(S) LA Network Manager can restore previously configured Audio Network Protocol and AES67 connections by loading a session file containing the desired AES67 parameters.

TROUBLESHOOTING

In case this troubleshooting guide does not help fixing the encountered issue, contact L-Acoustics using the following email addresses:
- avcontrol@l-acoustics.com for topics related to Q-SYS, PTP and AES67.
- nm@l-acoustics.com for topics related to LA Network Manager.

Issue / Symptom	Potential cause	Resolution / Actions
The desired stream is not listed in the Stream Name listbox	The AES67 Transmitter cannot be detected by the plugin because it does not have the correct 'Script Access' setting	Change the 'Script Access' setting of the AES67 Transmitter component to either 'Script' or 'All', and save to Core and run. See: **"Connecting to an AES67 stream by name", page 8**
		Manually setup the stream parameters. See: **"Connecting to an AES67 stream manually", page 8**
	The desired stream is an external AES67 sender and is not advertised via SAP/SDP	Manually setup the stream parameters. See: **"Connecting to an AES67 stream manually", page 8**
The input stream status is OK but there is no audio signal	The device is in Standby mode. In Standby mode the DSP level meters are disabled	Take the DEVICE out of Standby mode
	The AES67 stream is not routed to one of the device DSP inputs	Update the 'Source' and 'AES67 Channel' parameters in the 'Inputs' page of the plugin to map AES67 channels to DSP inputs
	Fallback to Analog-AES/ EBU AUX input is active	Reset or disable Fallback to AUX in the 'Inputs' page of the plugin
The stream status is 'Connecting' or 'Error' and the 'Received packets' counter is not incrementing	The AES67 Transmitter is disabled	Enable the AES67 Transmitter
	The AES67 reciever of the DEVICE is stuck with obsolete stream identifiers	Reboot the DEVICE
	The Multicast Address parameter is incorrect	Check the stream parameters. See: **"Connecting to an AES67 stream by name", page 8**
	IGMP snooping blocks the multicast traffic	• Make sure there is an IGMP Querier on the network. • Disable IGMP snooping (this may cause an unwanted increase or flooding in network traffic)
	The network is filtering the PTP or AES67 multicast traffic	Contact your network administrator to allow multicast traffic for PTP and RTP

Troubleshooting (Cont)

Issue / Symptom	Potential cause	Resolution / Actions
The stream status is 'Connecting' or 'Error' and the 'Early timestamps' counter is incrementing	The 'Media Clock Offset' parameter is incorrect	Set the 'Media Clock Offset' parameter to 0 (this should always be the case with Q-SYS AES67 Transmitters). See **"Connecting to an AES67 stream by name", page 8**
	The 'Network Tx Buffer' of the Q-SYS AES67 Transmitter is set on 'Extra [x]ms'	Set the 'Network Tx Buffer' of the Q-SYS AES67 Transmitter to 'Default'
	The AES67 sender and receiver are not on the same PTP clock	Check the Q-SYS Core and device PTP grandmaster and PTP clock status. See: **"PTP Clock Master selection", page 7**
The stream status is 'Connecting' or 'Error' and the 'Late timestamps' counter is incrementing	Audio packets arrive too late because of network forwarding time and/or network traffic	• Increase the receiver latency by adjusting the 'Network Rx Buffer' parameter. See: **"Network Rx buffer, Packet Time and Latency", page 10** • Increase the packet time parameter of the AES67 stream. • Contact your network administrator to adjust the network architecture or QoS settings to reduce audio packets latency
	The 'Media Clock Offset' parameter is incorrect	Set the 'Media Clock Offset' parameter to 0 (this should always be the case with Q-SYS AES67 Transmitters). See **"Connecting to an AES67 stream by name", page 8**
	AES67 sender and receiver are not on the same PTP clock	Check the Q-SYS Core and device PTP grandmaster and PTP clock status. See: **"PTP Clock Master selection", page 7**
The audio signal is distorted	The stream encoding parameters are incorrect	Check that the receiver's 'Packet Time', 'Stream Channels' and 'Format' parameters match the sender's stream parameters
	The PTP clock is not stable	Check the Q-SYS Core and device PTP grandmaster and PTP clock status. See: **"PTP Clock Master selection", page 7**
Unexpected PTP Clock Master	The current PTP Clock Master has higher priority than the expected Clock Master device	Adjust the PTP priority settings of the involved devices. See: **"PTP Clock Master selection", page 7**
	The Multicast Address parameter is incorrect	Check the stream parameters. See: **"Connecting to an AES67 stream by name", page 8**
The PTP Clock status is not Locked	The device does not receive PTP clock properly or in time	Review the network setup and traffic forwarding rules for PTP traffic
		Contact L-Acoustics if network configuration is set properly but the PTP clock status remains unstable or faulty