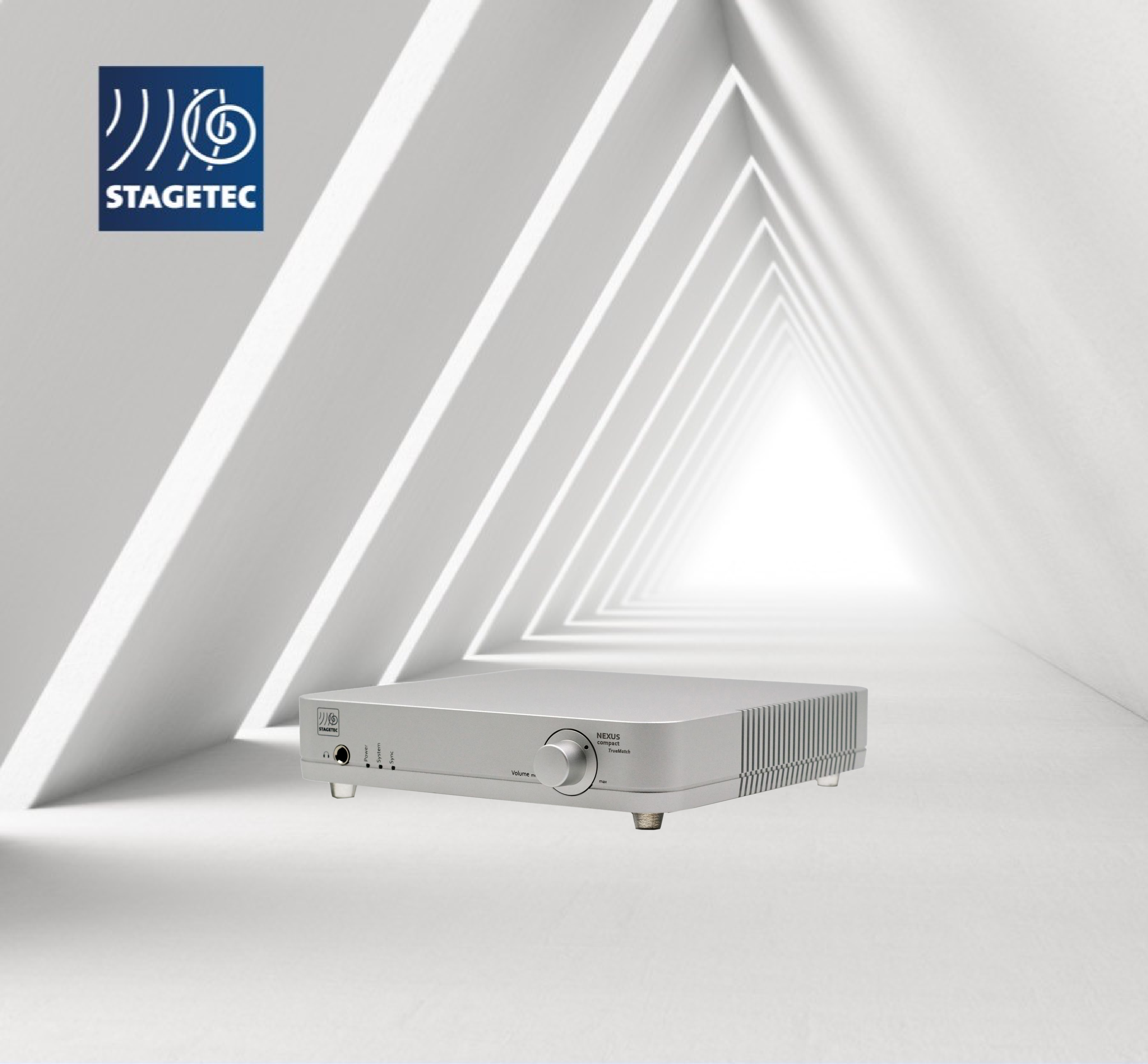


NEXUS compact // AUDIO INTERFACES

Operators Manual V 2.0

02/08/2024

TABLE OF CONTENTS

1 // GENERAL INFORMATION 4

1.1 // Introduction 4

1.1.1 Manufacturer 4

1.1.2 Warranty 4

1.1.3 Support 4

1.1.4 Target Audience 4

1.2 // General safety instructions 4

1.2.1 Power Supply 4

1.2.2 Operational conditions 4

1.3 // Cleaning 5

1.4 // Software 5

1.5 // Declaration of conformity 5

2 // SYSTEM DESCRIPTION 6

2.1 // Key features 6

2.1.1 The TrueMatch Technology 6

2.2 // Hardware 9

2.2.1 Front panel 9

2.2.2 Rear panel 10

2.2.3 AoIP Options (AES67 / Dante) 10

3 // WEB-GUI 11

3.1 // Menu elements / layout 11

3.1.1 Top menu bar 11

3.1.2 Right menu bar 13

3.2 // DASHBOARD page 13

3.2.1 Mic Split panel 14

3.2.2 Digital Inputs panel 15

3.2.3 Processing panel 16

3.2.4 Outputs panel 17

3.3 // DETAILED- and SIMPLE-view 18

3.4 // MIC IN page 18

3.5 // INPUTS page 20

3.6 // OUTPUTS page 21

3.7 // PROC / MIX page(s) 23

3.7.1 Input Processing Channels 23

3.7.2 Mix Busses 24

3.7.3 Copy / Paste 26

3.7.4 MIX SEND view 28

3.7.5 Detail views 29

3.7.5.1 Equalizer Detail View 30

3.7.5.2 Compressor Detail View 31

3.7.5.3 Expander Detail View 32

3.7.5.4 Limiter Detail View 33

3.7.5.5 Delay Detail View 33

3.8 // XY ROUTING page 34

3.9 // System page 35

3.9.1 General 35

3.9.1.1 Network panel 36

3.9.1.2 Audio panel ... 37
3.9.1.3 System panel ... 37
3.9.1.4 Power On State panel ... 38
3.9.1.5 Generator panel ... 39
3.9.2 GPIO ... 40
3.9.3 Snapshots ... 41
3.10 // Headphone panel ... 42
3.11 // Snapshot menu bar ... 43
3.11.1 Creating a new snapshot ... 43
3.11.2 Editing an existing snapshot ... 44
3.11.3 Snapshot parameter override ... 44
3.11.4 Loading snapshots ... 44
4 // SETUP ... 45
4.1 // Connecting to the Web-GUI ... 45
4.2 // Configuring AoIP-Interfaces ... 45
4.2.1 Setting up the network / AoIP networking requirements ... 45
4.2.1.1 System planning ... 46
4.2.1.2 Device configuration ... 50
4.2.1.3 Check for proper synchronization (PTP) ... 50
4.2.1.4 Stream configuration ... 51
4.2.2 Out-Of-Band management ... 59
4.2.3 Configuring AES67-ports ... 59
4.2.4 Configuring Dante-ports ... 59
4.3 // Setting up AES67 audio streams - Direct Out GUI operation ... 60
4.3.1 Introduction (Direct Out) ... 60
4.3.2 Synchronisation ... 60
4.3.3 IGMP ... 62
4.3.4 Quality of Service (QoS) ... 62
4.3.5 Multicast/Unicast ... 63
4.3.6 Stream Parameters ... 63
4.3.7 Stream Discovery ... 64
4.3.8 Input Stream Settings - Stream Discovery ... 64
4.4 // Setting up Dante – Dante controller operation ... 65
4.5 // Setting up Ember+ control ... 66
4.5.1 Identifying the "path" / Ember+ viewer ... 66
4.5.2 Expample: Setting up "Companion" ... 68
4.6 // Redundancy options ... 70
5 // APPENDIX ... 71
5.1 // Blockdiagram ... 71
5.2 // SUB-D pinout schematics ... 72
5.3 // Technical data ... 73
5.4 // Glossary ... 75

1 // GENERAL INFORMATION

1.1 // INTRODUCTION

NEXUS compact is a high quality product for professional use. In order to prevent material damage and personal injury, please read this manual carefully and follow the safety regulations noted below before powering up the device. Incorrect handling can lead to damage or compromised functionality.

1.1.1 Manufacturer

STAGETEC GmbH
Tabbertstrasse 10-11, 12459 Berlin/Germany
Phone: +49 30 639902-0, Fax: +49 30 639902-32
Email: office@stagetec.com[1],

Website: http://www.stagetec.com

1.1.2 Warranty

Technical changes by the user are generally not permitted. There are no user serviceable parts inside the unit. Unauthorized opening will void the warranty.

1.1.3 Support

Even after the purchase, our support is available to you. Do you have any questions that the documentation cannot answer? Or perhaps technical issues? You can reach us through various means:

- Internet: http://www.stagetec.com
- Email: office@stagetec.com[2]
- Phone: +49 30 639902-0
- Fax: +49 30 639902-32

1.1.4 Target Audience

NEXUS compact is a high quality device for professional sound processing. It should be operated by experienced sound engineers only.

1.2 // GENERAL SAFETY INSTRUCTIONS

To ensure a safe operation of the NEXUS compact, it is essential to observe the following safety instructions. Please read them carefully before
use. Disregarding the safety instructions can lead to damage, operation failure or personal injury.

1.2.1 Power Supply

Inside the unit and the separate power supply there may be parts that carry high voltages and currents. The devices shall only be opened by qualified service personnel. Do not insert any objects through the openings of the housing. Power and network cables must not be bent or placed near sharp edges. Defective cables have to be replaced immediately.

1.2.2 Operational conditions

The device may only be operated within the specified temperature range which reaches from 0° to 50° C. Do not expose the unit to direct heat from sunlight, stage spotlights or other sources. Don`t cover the top of the device with objects that could affect proper cooling. When installing the NEXUS compact in a rack, make sure to leave enough space for a sufficient airflow below and above the device. Do not expose the device to rain or moisture and make sure to operate it only in a dry state. In case of condensing moisture the unit must dry out adequately before being switched on. During operation, the air humidity should not exceed 90% (non-condensing). For a long lifetime the unit should only be operated in a smoke-free environment.

1 mailto:office@stagetec.com
2 mailto:office@stagetec.com

1.3 // CLEANING

Please adhere the following rules when cleaning your NEXUS compact:

- Avoid using cleaners containing solvents, alcohol, petroleum ether or acetone since they can damage the surface of the device.
- Do not use oily substances or aggressive household cleaning agents.
- To clean the surface we recommend using a soft, clean brush or a soft cloth slightly dampened with water. In case of heavier soiling you can add a minimal amount of mild dishwashing detergent.
- Under normal conditions the inside of the unit doesn`t need cleaning. The device should not be opened by the user. If cleaning of the inside should become necessary, please send the unit to STAGETEC.

1.4 // SOFTWARE

The NEXUS compact can be controlled by a web-based user interface. The web GUI is compatible to every modern internet browser running on Windows or MacOS computers as well as Android or iOS mobile devices.

1.5 // DECLARATION OF CONFORMITY

STAGETEC GmbH | Tabbertstraße 10-11 | 12459 Berlin | Germany

STAGETEC

CE – DECLARATION OF CONFORMITY

Herewith the manufacturer

STAGETEC GmbH
Tabbertstrasse 10-11
12459 Berlin - GERMANY
Phone: +49 (0) 30/639902-0

declares that the device

Audio Routing System, Types NEXUS compact TrueMatch

complies with the following directives and harmonized standards:

- **2014/30/EU, EMC Directive**
- **EN 55032:2015+A11:2020; EN55035:2017+A11:2020**
- **2014/35/EU, Low Voltage Directive**
- **EN 62368-1:2014**
- **2009/125/EC, ErP Directive**
- **2001/95/EC, GPS Directive**
- **EN 62368-1:2014+A11:2017**
- **2011/65/EU, RoHS Directive**

Further information:

In case of unauthorized changes to the device this declaration will become invalid. This declaration is only effective under conditions of correct application and installation, particularly with respect to wiring. The instruction manuals need to be observed strictly.

Signed for and on behalf of STAGETEC GmbH, Germany:

Berlin, October 2023

Thomas Gmeiner, Managing Director

2 // SYSTEM DESCRIPTION

Congratulations on the purchase of your NEXUS compact. For the first time this highly professional device makes the legendary STAGETEC audio quality available within a small, convenient footprint. With 8 TrueMatch 32 bit microphone inputs, 8 high quality analog outputs, 4 stereo AES3 I/Os, GPIO connections and a modern audio-over IP infrastructure sporting either Dante or AES67 protocols, the NEXUS compact defines a new standard in audio distribution. All control parameters are easily acccessible via an intuitive web interface, which runs on every computer or mobile device without the requirement of any software installation – a standard web browser is all you need.

2.1 // KEY FEATURES

The following key features make NEXUS compact perfectly suited for a wide range of applications: by experienced sound engineers only.

- 8 x TrueMatch 32 bit microphone inputs with 4 splits each.
- 4 x AES3 stereo inputs.
- 4 x AES3 stereo outputs.
- 8 x Analog outputs 24 bit.
- Stereo headphones.
- 64 x Dante / AES67 I/Os (option based on module configuration).
- 6 x GPI / 5 x GPO. GPIs in pairs are also usable as encoders.
- 16 processing channels with Delay, Expander, EQ and Compressor
- 16 mix buses with EQ, Compressor, Limiter
- All inputs of the NEXUS compact can be assigned to processing channels
- All outputs of the NEXUS compact can be assigned to mix buses

Possible applications include:

- Reference A/D and D/A converter for recording studios with highest sonic demands for both input (microphone/line) and output signals.
- Connecting camera-mounted microphones with simultaneous operation of monitor/comm/tally.
- Small stageboxes for rehearsal stages, foyer sound reinforcement, backstage.
- Small workstations deployment with local low latency monitor mixing.
- On-site mixing for small remote productions with connection to local audio sources.

2.1.1 The TrueMatch Technology

Outstanding audio quality thanks to unprecedented converter technology

Compared to conventional A/D converters, the patented TrueMatch technology provides for enhanced dynamics. Today, converters with 64 or 128 times oversampling are almost exclusively used in modern studios. They feature an excellent linearity that depends merely on the accuracy of the utilized downsampling filters and is therefore considered as practically perfect.

However, linearity curves in A/D-converter specifications still show errors that increase as levels reduce. These errors are essentially caused by the following factors:

- Noise: A positive difference indicates that the digitized value corresponds to an input voltage higher than the actually applied voltage. The quantization noise is added to the input voltage.
- Insufficient bit resolution: Both positive and negative variations from linear results indicate truncation errors – the digitized value does not represent the input signal adequately. Such errors are also referred to as »quantization noise«.

Consequently, today's conversion technology is perfectly suitable for digitizing high-level audio signals. Conventional converters achieve the hypothetically best digitization accuracy just below the clipping threshold. Today, THD&N values of better than –100 dBFS (i. e. <0.001 %) are almost standard.

Low-level Signals

Significantly worse results are achieved, however, with low-level signals. For example, when a signal level is 60 dB below the clipping threshold, conventional converters will accomplish THD&N values no better than 0.1…0.5 % (i. e. –60…45 dB@60 dBFS), meaning that such signals are digitized with a maximum of 8…10 utilized bits!

Hence, it is not only the high levels that need to be considered. (No engineer will complain about a harmonic distortion of 1 % at full-scale level as this is already produced by speakers.) Low-level signals, too, must be as lowdistortion and low-noise as possible.

Dynamic Enhancement

As our acoustic environment has a large dynamic range and loud signals must never exceed the clipping threshold (a sufficient headroom must be considered), audio technology has always called for improved dynamics, or, in terms of the digital age, for more »bits«. Consequently, methods using better and better resolutions have been developed but have always been based on theoretical dynamic values: 16-bit A/D converters feature a THD&N of –96 dBFS, 20-bit converters of –120 dBFS, and 24-bit converters even of –144 dBFS (approximate values). However, analog technology (or, to be more precise, the fact that harmonic distortion increases with rising levels due to – analog – input stages) has never been seriously taken into account.

Converters in Real Life

The figure below shows the interdependence between THD&N and level. The theoretical value of a 24-bit converter at full-scale level (0 dBFS) is approximately –144 dBFS. A signal with a very low level of less than 144 dBFS cannot be processed by such a converter anymore as the quantization noise predominates (see the broken line). The right part of the illustration shows the additional distortion produced by the real converter as the level increases.

The Idea Behind TrueMatch

The factors described made us develop the TrueMatch technology. The objective was to achieve excellent THD&N values at high and low levels. The idea is to run a standard 24-bit converter in its optimum operating range by deploying a variable preamp function; the converter must, however, never clip. TrueMatch achieves this by using multiple preamplifiers for various amplification ranges, each featuring its own A/D converter. When a signal is applied, a processor selects the most appropriate of the available converters on the basis of the signal level. This so-called »gain-ranging« technique is well known; however, it has one shortcoming: All gain factors of the individual preamps and the errors of all converters must be known in order to compensate them. And it is particularly the intermediate ranges between any two converters that may present difficulties.

TrueMatch Logic

STAGETEC has invested many years of hard work into the development of a method that does not only convert audio signals but also permanently monitors all amplifiers and A/D converters involved; all relevant values are constantly being measured with extreme accuracy. Gain and offset errors, even phase shifts, and pulse-dependent distortion between the individual signal paths are detected and immediately corrected. The corrections are much more exact than the employed A/D converters, meaning that correction errors are far below the converters' noise floor and cannot be verified by measuring.

For instance, phase shifts between the individual paths can now be corrected to a value better than 0.01° at 20 kHz!

Consequently, the corrections do neutralize not only component tolerances and aging inside the used amplifiers but also eliminate exemplary dispersion of the individual converters and parameter shifting caused by temperature changes. Thus, it is possible to use an A/D converter for digitizing a signal that is being output by another converter - and even the noise floor of the other converter can be determined! The converters are truly matched!

The Implementation of the TrueMatch Converter

The preamp gain is distributed in a way that exactly one A/D converter is operated in its optimum range just below the clipping threshold. Thus, for achieving optimum THD&N values over the entire dynamic range, the processor »just« has to find and select the A/D converter(s) with the most accurate conversion result and to perform all necessary corrections. Therefore, it is practically possible to make converters with a resolution of more than 24 bits and a dynamic range of better than 144 dB. The 32-bit microphone A/ D converter manufactured by STAGETEC accomplishes a utilizable dynamic range of more than 150 dB!

Correction values	
More than 45 correction parameters (some fixed, some variable) are determined individually for each converter stage inside STAGETEC's 28-bit converters. All these parameters are then considered during the conversion process. They include:	
Permanent gain-quotient verification	$gain_2/gain_1$ $gain_3/gain_2$ $gain_4/gain_3$
Permanent offset-voltage variation verification	$U_{offset2}-U_{offset1}$ $U_{offset3}-U_{offset2}$ $U_{offset4}-U_{offset3}$
Permanent phase-shift verification	$phase_2-phase_1$ $phase_3-phase_2$ $phase_4-phase_3$
Total-gain calibration (performed once)	V_{signal} $V_{generator}$

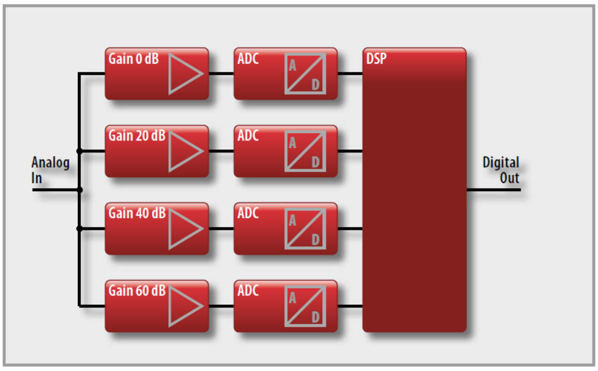


Fig. 1 TrueMatch converter

The Analog Side

Extreme demands like never before are put on the analog components, too. The amplifiers must be able to process voltage ranges of more than eight decades! They must work both extremely low-noise and low-THD. Analog and digital signals from converters currently not in use must not falsify the utilizable signal by crosstalk.

Despite of the enormous overhead needed, for example, by a fourfoldstacked 28-bit A/D converter, the STAGETEC converter requires only marginally more supply voltage than conventional, monolithically structured A/D converters.

TrueMatch has been patented in most countries and has also been licensed to other manufacturers. STAGETEC uses the system in all high-quality converters, for example, for the XAD and XMAD/XMIC boards.

2.2 // HARDWARE

2.2.1 Front panel

Fig. 2 NEXUS compact - front panel

Since all parameters of the NEXUS compact are controlled via the web GUI, the front panel contains just a few elements. From left to right you will find:

- A 6,3mm TRS headphone jack with a high quality headphone amplifier which will easily fulfil most demanding requirements. Controlled by the web GUI, the headphone jack can monitor any signal source within the system.
- A power LED that indicates proper power distribution.
- A system LED that shows proper system performance.
- A Sync LED which indicates proper synchronization to the AES67 or Dante network.
- A volume control for the headphone output.

2.2.2 Rear panel

Fig. 3 NEXUS compact - rear panel

The rear panel gives you all the connectors you need to integrate NEXUS compact into your audio system:

Connector	Description
D/A- Connector	This female Sub-D connector provides 8 balanced analog outputs in standard Tascam format. Maximum output level for 0 dB fs is + 24dBu (adjustable). Please make sure to use shielded breakout cables.
AES3 I/O - Connector	This female Sub-D connector provides 4 stereo AES3 in- and outputs in standard Tascam format.
MIC- Connector	This female Sub-D connector provides 8 balanced analog microphone inputs (32 bit TrueMatch) in standard Tascam format. Maximum input level for 0 dB fs is + 24 dBu (adjustable). Please make sure to use shielded breakout cables.
GPIO- Connector	This female Sub-D connector provides 6 GPIs and 5 GPOs in standard Tascam format. GPIs in pairs are useable as encoders.
DC 12-24V- Connector	Attach the optional PSU here and turn the connector slightly clockwise to lock it securely. You can run the NEXUS compact powered via PoE+ (Power over Ethernet) as well. If you use both options simultaneously a redundant power distribution is established automatically. In case of a power failure the unit will seamlessly switch to the other power source. Notice: Power distribution via PoE+ is possible on the Control Port only. Primary and Secondary AoIP Ports are not applicable. For a redundant operation, power supply via PoE and by an external PSU is required. Connecting more than one port to PoE will not provide power supply redundancy. The PoE power source as to fulfil at least the standards according to 802.3at Type2, PoE+ (30 Watt max., 600mA, 50-57V).
CTRL- Connector	RJ-45 port to connect the control-PC. Via the web GUI the control port can be mirrored to one of the AES67 / DANTE ports, so only one connection is needed for both control and audio transmission.
SEC- Connector	Dante / AES67 secondary port (option based on module configuration).
PRI- Connector	Dante / AES67 primary port (option based on module configuration).

2.2.3 AoIP Options (AES67 / Dante)

NEXUS compact is available with either a Dante module or an AES67 module.

3 // WEB-GUI

Fig. 4 Web GUI

The Web-GUI of the NEXUS compact consists of 6 panels:

Panel#	Name	Description
1	Top menu bar	The menu bar provides information about the current state of the NEXUS compact. Further, it provides a fast access to the snapshot queue.
2	Right menu bar	The right menu bar is used to navigate through the various pages of NEXUS compact, such as [MIC IN], [INPUTS], [...]
3	Mic Split panel	The Mix Split panel provides access to the most important parameters of the NEXUS compacts' TrueMatch input-converters
4	Digital inputs panel	The digital inputs panel provides access to the most important parameters of the NEXUS compacts' AES3- and AES67/Dante inputs
5	Processing panel	The Processing panel provides access to the input-metering of the 16 processing channels, as well as access to the most important parameters of the 16 mix busses of the NEXUS compact
6	Outputs panel	The outputs panel provides access to the most important parameters of the NEXUS compacts' analog line-outs, as well as to the AES3- and AES67/Dante outputs

In the following chapters the detailed functionalities of these panels will be explaines in greater detail.

3.1 // MENU ELEMENTS / LAYOUT

When starting up, the web GUI displays the Dashboard page that gives you a quick overview of all functions. Every screen of the web GUI has two menu bars at the top and the right side, which are always visible.

3.1.1 Top menu bar

On its left side, the top menu bar shows important system informations. Depending on the selected function page the right side can additionally provide buttons to switch between channel blocks.

Element	Function
ID nxct-000068	Shows the device ID of the NEXUS compact currently connected.
	Shows proper power distribution via PoE (power over ethernet) when backlit in green.
	Shows proper power distribution via external power supply when backlit in green.
1 2 3 4 5 6 7 8	When backlit in green, the 8 numbers next to this symbol indicate that 48 V phantom power is active on the corresponding mic input channels 1 - 8. This feature ensures that you can always see where phantom power is present, which can be important to prevent damage to other equipment like ribbon microphones.
50 °C	Shows the current temperature of the main board. The temperature should not exceed 50° C.
48 kHz	Shows the current sample rate.
	Shows the state of the Mic split system.
PRE	Sets the metering point of the outputs pre fader.
POST	Sets the metering point of the outputs post fader.

In all views (with the exception of the [SYSTEM]-view), the top menu shows snapshot/queue-functionalities on the right side:

Element	Function
1 - START	Snapshot-queue selection ("1-START" = example snapshot). Shows the currently loaded snapshot. A push on the [area] shows a dropdown with other snapshots assigned to the queue according to their position in the queue .
AUTO INCR AUTO INCR	[AUTO INCR(rement)] snapshot queue. Activate or deactivate the auto increment function of the snapshot queue. White background: active, black background: inactive
LOAD	[LOAD] the currently selected snapshot.

Top menu items in the [SYSTEM]-View:

Element	Function
GENERAL	Opens the [GENERAL]-setup
GPIO	Opens the [GPIO]-setup
SNAPSHOTS	Opens the [SNAPSHOTS]-setup (snapshot-queue)

3.1.2 Right menu bar

The right menu bar contains 7 buttons that provide access to the corresponding function pages. The button representing the currently selected function page is backlit in white. An additional panel at the bottom of the right menu bar controls the headphone output, signal generator and snapshot automation.

Element	Function
DASHBOARD	Opens the [DASHBOARD] page, which gives you a quick overview of all in- and output parameters.
MIC IN	Opens the [MIC IN] page that gives you control of the 8 TrueMatch 32 bit microphone inputs.
INPUTS	Opens the [INPUTS] page that gives you control of all digital inputs.
OUTPUTS	Opens the [OUTPUTS] page that gives you control of all analog and digital outputs. Furthermore, the OUTPUTS page is used to configure the signal routing.
PROC / MIX	Opens the [PROC/MIX] page that provides access to the processing channels and the mix-matrix
XY ROUTING	Opens the [XY ROUTING] page that gives you control over the input- and output routing.
SYSTEM	Opens the SYSTEM Page which provides access to network settings, basic functions, GPIO setup and snapshot automation.

3.2 // DASHBOARD PAGE

When the web GUI is started, the DASHBOARD is the first screen to be displayed. Furthermore, you can access the DASHBOARD from any page of the web GUI by clicking [DASHBOARD] on the right menu bar.

Fig. 5 Dashboard page

The DASHBOARD is divided into four panels representing microphone inputs, digital inputs, processing channels and outputs. Each panel has level meters and controls for eight channels. If the corresponding channel type has more than eight channels, they can be switched by tab buttons which are located above the panel.

3.2.1 Mic Split panel

Each microphone input has four splits named A - D, allowing signal to be sent to four destinations with different parameter settings. As it directly affects the input stage, the setting of the phantom power parameter (48V) is always identical to all four split points. By clicking the tab buttons [MIC SPLIT A] to [MIC SPLIT D] you can reach the parameters of the corresponding split points.

Fig. 6 Mic split panel

The following parameters are available for each microphone input and split point individually.

Element	Function
	Activates 48 V phantom power. This parameter will always be set identically for all four split points. If phantom power is activated, the button-backgroung will turn "red".
	Inverts the polarity of the input signal (phase reverse). If polarity is reversed, the background will change to "white".
	Activates the input limiter. If the limiter is active, the background will change to "blue"
	When backlit in green, this symbol indicates that the low cut filter is active. The low cut filter is adjustable from linear to 155 Hz. To activate the filter, navigate to the MIC IN screen by pressing [MIC IN] on the right menu bar.
	The left meter shows the current input level after the digital gain stage. The red triangle indicates the last maximum level. You can reset this peak hold function by clicking on the triangle. The right meter shows the current gain reduction applied by the limiter.
	Mutes the microphone input
	This rotary control provides a digital level adjustment from -20 dB to +70 dB. Due to the exceptionally high resolution of the 32 bit TrueMatch converter no analogue gain stage is required. It is not possible to overload the input stage with microphone signals. This gain stage adjusts the level of the digital signal in order to ensure a proper alignment with the following equipment. Please note that it is possible to exceed the level of 0 dBfs (maximum digital resolution) so clippings may occur when high amounts of gain are applied.
	By clicking in the label field on the bottom of every strip you can enter a custom name for that input / split.

3.2.2 Digital Inputs panel

On the “Digital Inputs panel” you have access to level meters and key parameters of eight digital input channels simultaneously. By clicking on the tab buttons above the panel you can select the digital inputs you want to be displayed.

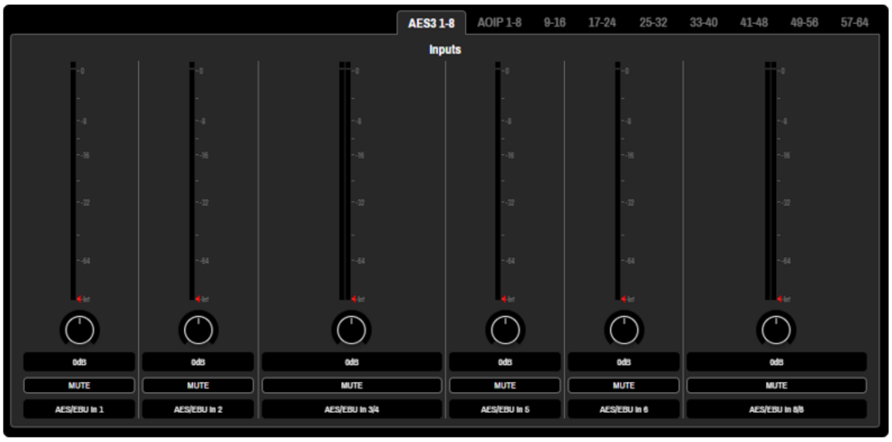
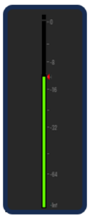


Fig. 7 Digital inputs panel

The “Digital Inputs panel” provides a fast access to the most important parameters of:

- 8x AES3 Inputs
- 64 AoIP Inputs (AES67 / Dante, depending on model)

Element	Function
	Shows the current input level after the gain stage. The red triangle indicates the last maximum level. You can reset this peak hold function by clicking on the triangle.
	Mutes the digital input.
	This rotary encoder provides a digital level adjustment from -20 dB to +20 dB. Please note that it is possible to exceed the level of 0 dBfs (maximum digital resolution) so clippings may occur when applying high amounts of gain.
	By clicking in the label field on the bottom of every strip you can enter a custom name for that input.

3.2.3 Processing panel

If a source is selected for the processing channels of the NEXUS compact, the "processing panel" provides level meters for the assigned inputs. It also provides access to the most important parameters of the mix-buses, such as

- Metering
- Mute
- Trim
- Polarity

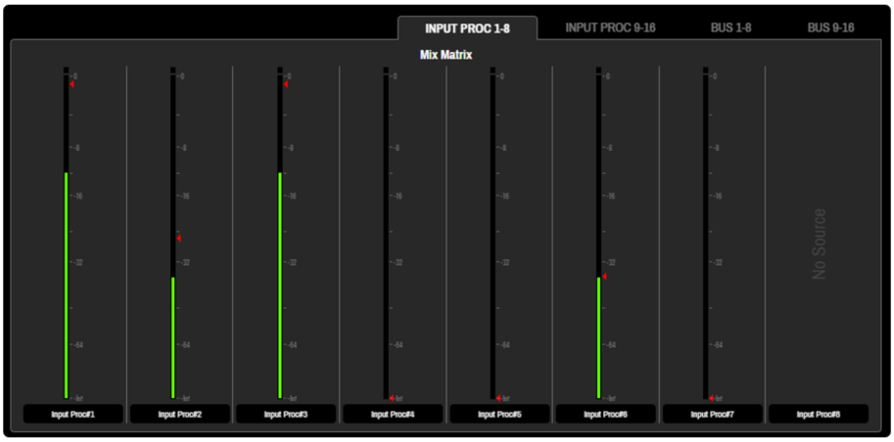
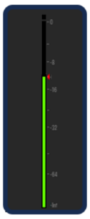


Fig. 8 Processing panel

<u>Note</u>: Processing-inputs of the mix-matrix of the NEXUS compact do not have a dedicated master-fader on the input side. Furthermore, all processing inputs can be mixed to any of the 16 mix buses individually.

Element	**Function**
	The metering shows the level of the mix bus. The red triangle besides the meter indicates the last maximum level. You can reset the peak hold function by clicking on the triangle. *<u>Note:</u> processing inputs only show metering.*
	Mutes the mix bus.
	Inverts the polarity of the mix bus (phase reverse). If polarity is reversed, the background will change to "white".
	The label field on the bottom of every strip shows the name of the mix bus.
	Adjust the output level of the mix bus from -144db (off) to +15db

3.2.4 Outputs panel

On the “outputs panel” you have access to level meters and key parameters of all outputs of the NEXUS compact:

- 8x Analog out
- 8 AES3 out
- 64x AoIP out (AES67 or Dante, depending on model)

By clicking on the tab buttons above the panel you can select the outputs you want to be displayed.

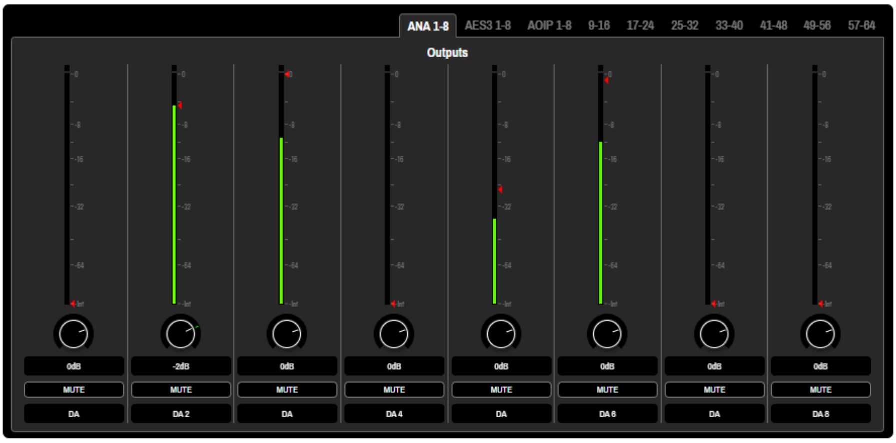
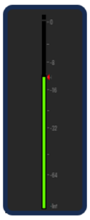


Fig. 9 Outputs panel

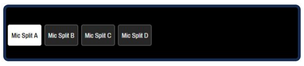
Element	Function
	Shows the current output level. The metering point can be switched between pre and post fader by clicking [PRE] or [POST] in the top menu bar. This setting affects the metering points of all outputs simultaneously. The red triangle next to the meter indicates the last maximum level. You can reset the peak hold function by clicking on the triangle.
	Mutes the output.
	This rotary control represents the output level fader.
	By clicking in the label field on the bottom of every strip you can enter a custom name for that output.

3.3 // DETAILED- AND SIMPLE-VIEW

When selecting one of the following pages in the right menu bar,, the “top menu bar” is extended with the button [DETAILED]/[SIMPLE]:

- [MIC IN]
- [INPUTS]
- [OUTPUTS]
- [PROC/MIX]
- [XY ROUTING]

This button toggles between the following modes:

Button	Top bar	Explaination
		The [SIMPLE] view shows 4 buttons only, that will load the pages for the microphone splits.
		The [DETAIL] view extends the visible items by blocks showing metering, gain, mute state and name of each channel. Tap/click on a [BLOCK] to load the corresponding page. Hold a block and scroll vertically to navigate to other blocks that might be hidden on the right hand side.

3.4 // MIC IN PAGE

The [MIC IN] page provides convenient access to all parameters of the eight TrueMatch microphone inputs. Each input has four split points which can be sent to individual destinations with different parameter settings. You can access the individual parameter sets of the four split points by pressing the buttons [SPLIT A] to [SPLIT D] on the right side of the top menu bar.

<u>Note:</u> Since it directly affects the input stage, the setting of the phantom power parameter (48V) will always be identical to all four split points.

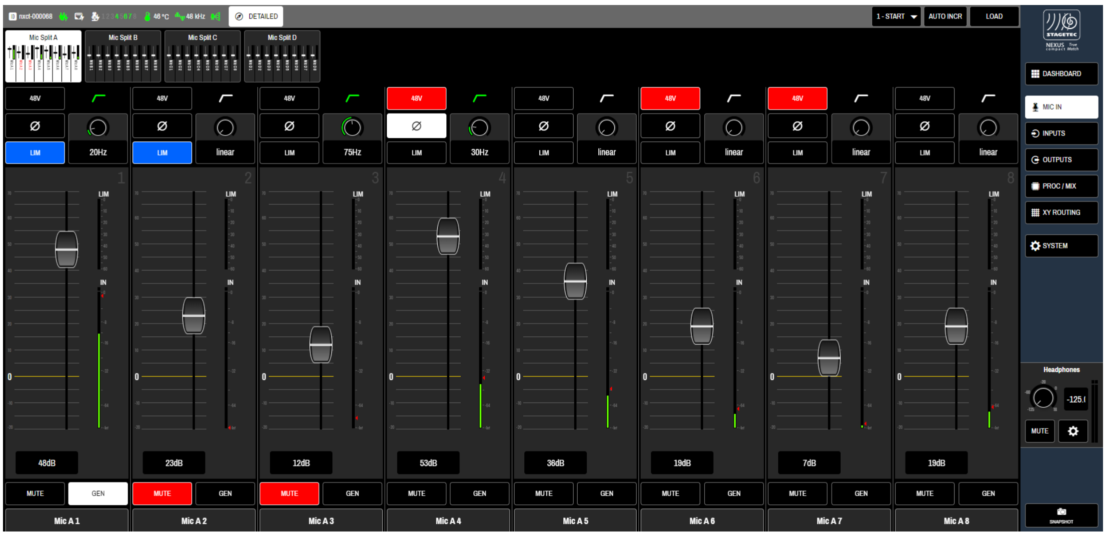
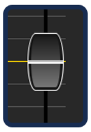
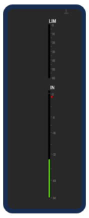
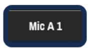


Fig. 10 MIC IN page

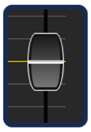
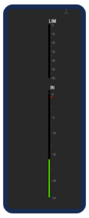
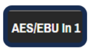
Each microphone input provides the following parameters:

Element	Function
	Activates 48 V phantom power. This parameter is always set identically for all four split points.
	Inverts the polarity of the input signal (phase reverse).
	Activates the input limiter
	The low cut filter can be adjusted from linear to 155 Hz by the rotary encoder. The current frequency is displayed in the field below the encoder where you can also enter the desired frequency manually. To deactivate the filter turn the encoder fully counterclockwise until the display reads LIN. In this case, the low cut filter symbol will change its colour from white to grey.
	The gain fader provides a digital level adjustment from -20 dB to +70 dB. Due to the exceptionally high resolution of the 32 bit TrueMatch converter no analogue gain stage is required. It is not possible to overload the input stage with microphone signals. This gain stage adjusts the level of the digital signal in order to ensure a proper alignment with the following equipment. Please note that it is possible to exceed the level of 0 dBfs (maximum digital resolution) so clippings may occur when high amounts of gain are applied.
	Shows the current input level after the digital gain stage on the bottom, and the current gain reduction applied by the limiter on the top. The red triangle indicates the last maximum level. You can reset this peak hold function by clicking on the triangle.
	Mutes the microphone input
	Activates the signal generator. For safety reasons this button must be pressed for two seconds to become active.
	By clicking in the label field on the bottom of every strip you can enter an individual name for that input / split.

3.5 // INPUTS PAGE

The INPUTS page provides access to the parameters of eight digital inputs simultaneously. Use the navigation in the top menu bar to select the desired digital input banks from AES 1-8 to AOIP 57-64. To enter the INPUTS page click on the [INPUTS] button on the right menu bar.

Fig. 11 INPUTS page

Element	Function
	Adjusts the input level by a maximum +- 20 dB. Please note, that it is possible to exceed the level of 0 dBfs (maximum digital resolution) so clipping may occur when applying high amounts of gain. The desired value can also be entered directly into the display field below the fader.
	Shows the current input level after the digital gain stage. The red triangle indicates the last maximum level. You can reset this peak hold function by clicking on the triangle.
	Couples two adjacent channels to stereo.
	When backlit in white, this button indicates that two adjacent channels are stereo coupled. Press the button once again to switch back to two mono channels.
	Mutes the input.
	By clicking in the label field on the bottom of every strip you can enter a custom name for that input.

3.6 // OUTPUTS PAGE

The OUTPUTS page provides access to the parameters of eight outputs simultaneously. This is also where all signal routing takes place. By clicking the tab buttons on the right of the top menu bar you can select the desired output banks from LINE 1-8 to AES 3 to AOIP 57-64. To enter the OUTPUTS page click on the [OUTPUTS] button on the right menu bar.

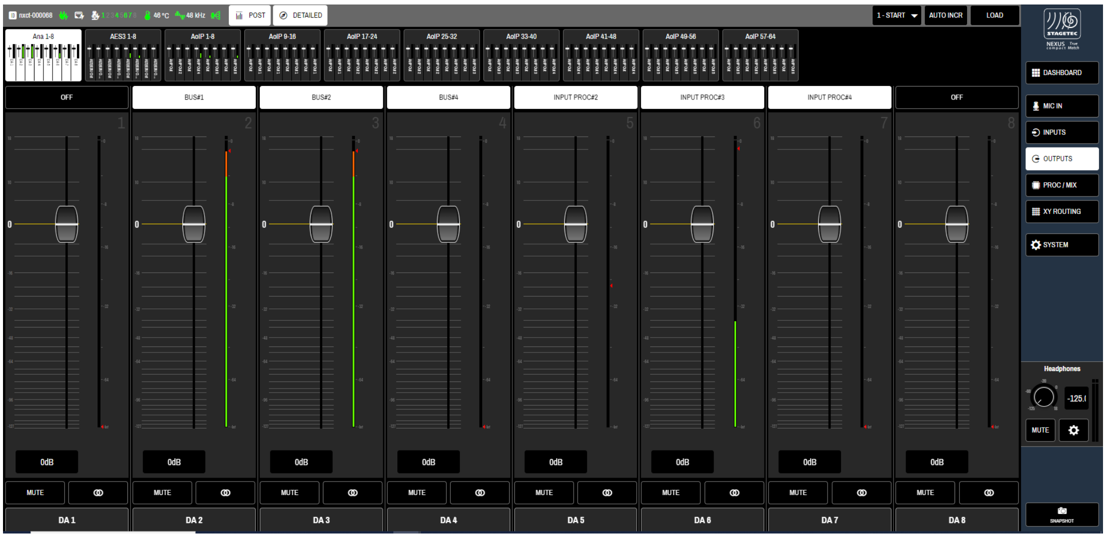
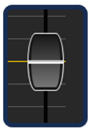
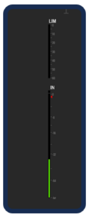


Fig. 12 OUTPUTS page

Elements of the OUTPUTS page:

Element	Function
	The routing-button is located at the top of the fader/meter. By clicking on this routing button the I/O assignment-dialog will appear. You can select the signal source for this particular output. If the button is black, no signal source is selected.
	Adjusts the output level. Please note, that it is possible to exceed the level of 0 dBfs (maximum digital resolution) so clipping may occur when applying high amounts of gain. The desired value can also be entered directly into the display field below the fader.
	Shows the current output level. The metering point can be switched between pre and post fader by clicking [PRE] or [POST] in the top menu bar. This setting affects the metering points of all outputs simultaneously. The red triangle besides the meter indicates the last maximum level. You can reset the peak hold function by clicking on the triangle.
	Couples two adjacent output channels to stereo.
	When backlit in white, this button indicates that two adjacent channels are stereo coupled. Press the button once again to switch back to two mono channels.
	Mutes the output.
	By clicking in the label field on the bottom of every strip you can enter a custom name for that output.

I/O-assignment dialog

The I/O-assignment dialog allows you to quickly assing sources (inputs or processing channels) to outputs, without the need to leave the current page in order to open the XY ROUTING.

Fig. 13 I/O-assignment dialog

To assign a source to an output, click on the routing button above the fader/meter. The I/O-assignment dialog opens. Simply click on the source you want to assign to the output (background will highlight "white") and close the dialog.

3.7 // PROC / MIX PAGE(S)

NEXUS compact provides a mix-matrix that enables all inputs to be mixed to 16 mix busses which then can be routed to any output of the NEXUS compact. Additionally, 16 input processing channels are available. Any input of the NEXUS compact can be assigned to any of the 16 processing channels which are also part of the mix matrix and can be sent to any of the mix busses.

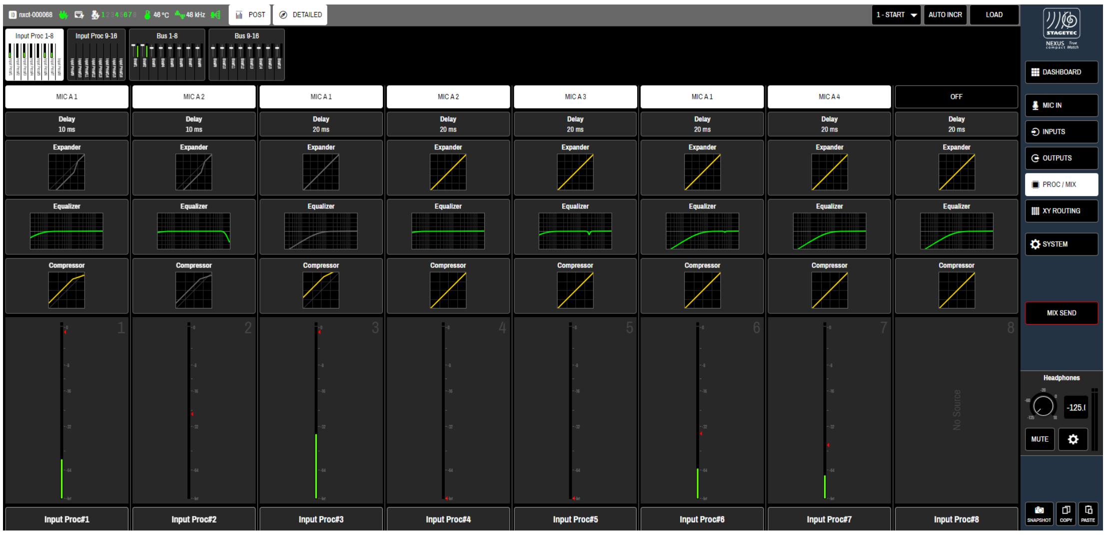


Fig. 14 PROC/MIX page

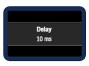
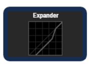
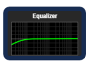
<u>Note:</u> When opening the [PROC/MIX] page, the right menu bar is extended with a [MIX SEND]-button. This button will swap the web interface into mix mode. Also, in the right menu bar, next to the [SNAPSHOT] button the [COPY]/[PASTE] function will become visible.

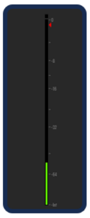
<u>Tip:</u> Long-press (2 seconds) on a processing miniature to change which miniature shall be displayed.

3.7.1 Input Processing Channels

Processing channels are not to be confused which a "traditional" mixing console. Furthermore, they serve as inputs to the mix-matrix and can also be routed to outputs directly. This allows a greater flexibility as processing channels can be used to process individual inputs and outputs independently to the mix-matrix as well (e.g. for adding processing to a PA output in case the NEXUS compact is used as a breakout box). In standalone-applications, the processing channels can be used as "Input-Channels" of the mix-matrix, wich is similar to a mixing console application.

Each processing channel provides the following parameters:

Element	**Function**
	The routing-button is located at the top of theprocessing channel strip. By clicking on this routing button the I/O assignment-dialog will appear. You can select the signal source for this particular processing channel. If the button is black, no signal source is selected.
	The [DELAY] is located at the top of the channel and is adjustable from 0ms to 4sec. To adjust the delay, click on [DELAY] to open the "detail view".
	The [EXPANDER] is located between the delay and the equalizer. To adjust the expander, click on [EXPANDER] to open the "detail view".
	The [EQUALIZER] is located between the expander and the compressor. To adjust the equalizer, click on [EQUALIZER] to open the "detail view".
	The [COMPRESSOR] is located at the bottom of the processing channel. To adjust the compressor, click on [COMPRESSOR] to open the "detail view".

Element	Function
	Shows the current level of the processing channel at the channel input (source level).
	The label field on the bottom of every strip shows the name of the input processing channel.

3.7.2 Mix Busses

Mix Busses are the summing busses of the NEXUS compact. These summing-busses can be assigned to any output of the NEXUS compact. To navigate to the mix buses, go to the [PROC/MIX]-page and select the “BUS BLOCKS” in the top menu bar.

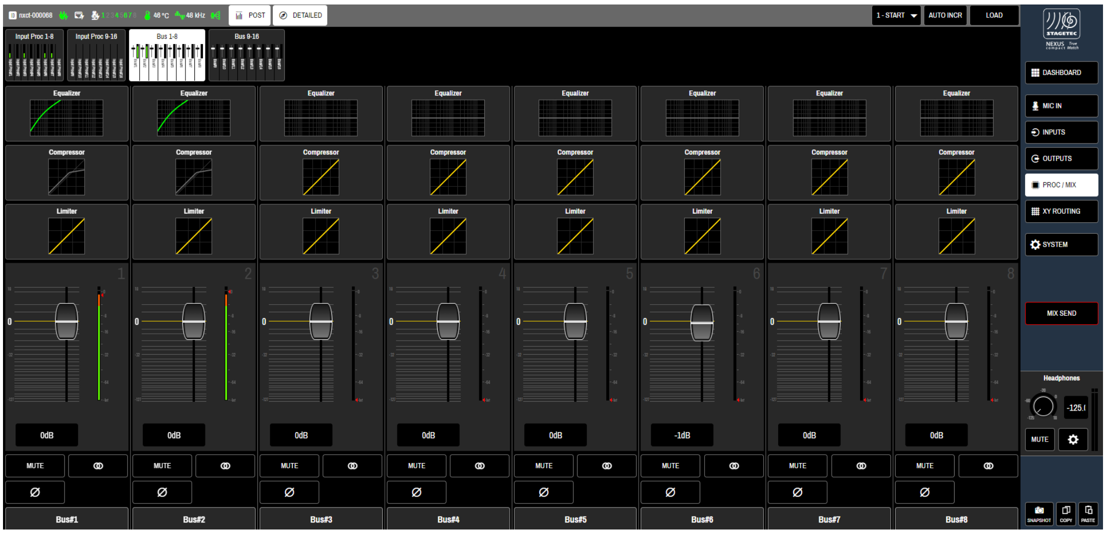
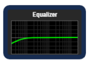
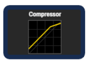
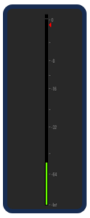
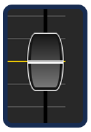


Fig. 15 Mix bus view

Each bus provides the following parameters:

Element	Function
	The routing-button is located at the top of theprocessing channel strip. By clicking on this routing button the I/O assignment-dialog will appear. You can select the signal source for this particular processing channel. If the button is black, no signal source is selected.
	The [EQUALIZER] is located at the top of the bus. To adjust the equalizer, click on [EQUALIZER] to open the “detail view”.
	The [COMPRESSOR] is located between the equalizer and the limiter. To adjust the compressor, click on [COMPRESSOR] to open the “detail view”.
	The [LIMITER] is located between the compressor and the delay. To adjust the equalizer, click on [LIMITER] to open the “detail view”.
	The [DELAY] is located at the bottom of the bus and is adjustable from 0ms to 4sec. Currently, the delay is only accessible trhough the detail view pages.

Element	Function
	Shows the current level of the bus at the channel output. This pick-off point of the metering can be selected [PRE/POST] in the top menu bar.
	Adjusts the output level. Please note, that it is possible to exceed the level of 0 dBfs (maximum digital resolution) so clipping may occur when applying high amounts of gain. The desired value can also be entered directly into the display field below the fader.
	Couples two adjacent busses to stereo.
	When backlit in white, this button indicates that two adjacent busses are stereo coupled. Press the button once again to switch back to two mono busses.
	Mutes the bus.
	Inverts the polarity of the bus output (phase reverse).
	The label field on the bottom of every strip shows the name of the mix bus.

3.7.3 Copy / Paste

Input processing channels and mix busses provide a copy/paste function. Once the [PROC/MIX] page is selected, these functions become available a the bottom of the right menu bar:

Button		Explanation
	Copy	Press [COPY] to activate the parameter selection mode
	Paste	Press [PASTE] to apply the parameters to the selected targets

<u>Copy parameters</u>

Once the copy-mode is activated, the GUI switches into the parameter-selection mode.

<u>Note:</u> Currently only the parameters of processing are available. In this section we refer to "channels", but the same functionality also applies to busses.

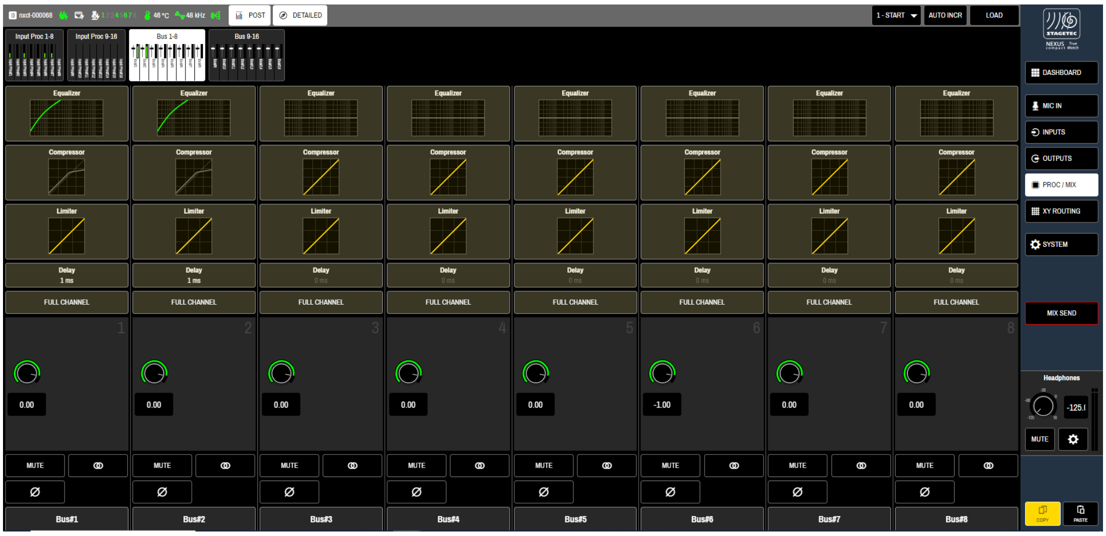


Fig. 16 GUI in "copy"-mode

All parameters (processing-items) that can be selected will start flashing. Select the desired parameters. Selected parameters will stop flashing and be highlighted permanently. You can either select individual processing blocks or select [FULL CHANNEL], which will copy all processing blocks of the entire channel.

Once all parameters are selected, press [PASTE] to switch to the "paste"-mode.

Paste parameters

After selecting the parameters as previously describes, press [PASTE] to switch to the paste-mode.

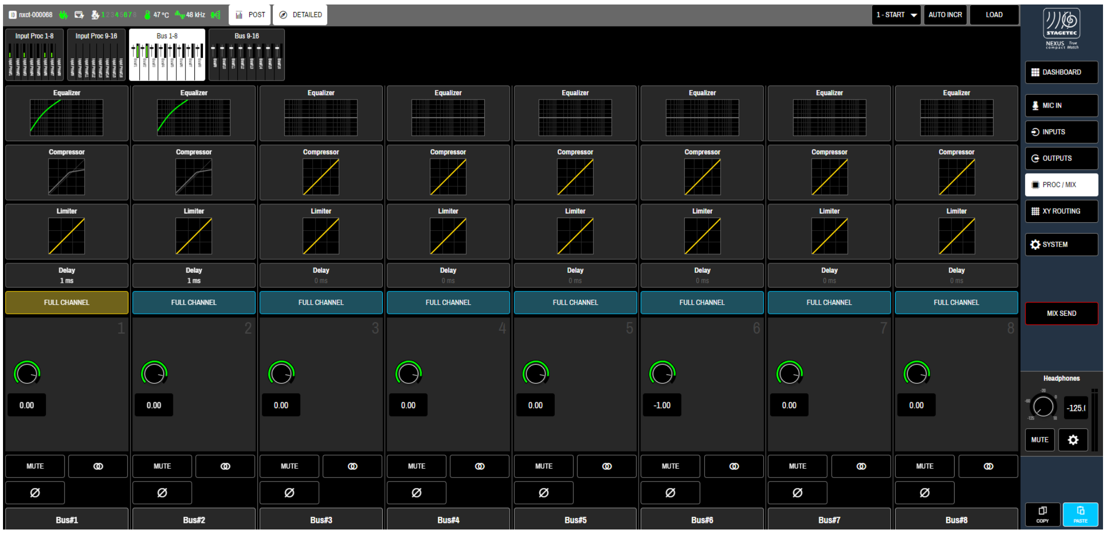


Fig. 17 GUI in "paste"-mode

All possible targets (according to the copy-selection) will highlight in "blue". Clicking on a possible target will immediatly paste the copied parameters to the target. Press [PASTE] again to the leave the copy/paste mode.

The GUI will return to the normal page view.

3.7.4 MIX SEND view

The [MIX SEND] view is located in the [PROC/MIX] page and provides access to the mixing functionality of the NEXUS compact. All available inputs (incl. the processing inputs) can be sent to mix-busses. To enter the mix-matrix, press [MIX SENDS] in the right menu bar.

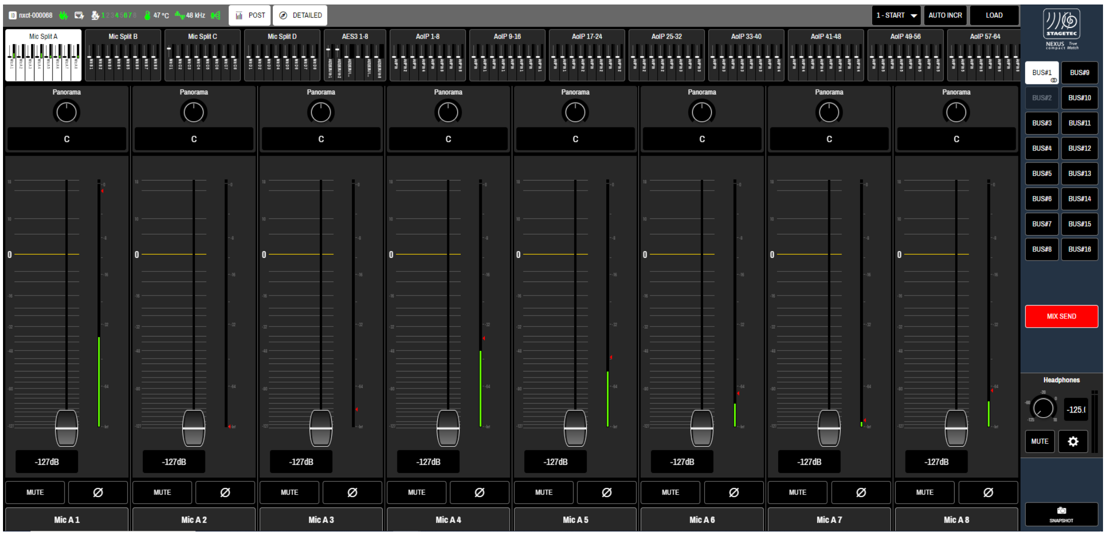
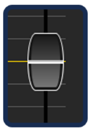
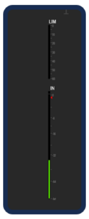
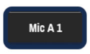


Fig. 18 MIX SEND view

The right menu bar will now change the layout and you will find the mix-busses in the upper section of the right menu bar. The faders of the inputs now represent the send levels. To send an input to a mix, select the target-[BUS] on the right side and adjust the fader to the desired value. In order to select other inputs, use the top menu bar to select the "Block" that holds the input. The faders will now populate to the content of the selected block. To exit the [MIX SEND] view, press [MIX SEND] again.

Element	Function
	Panorama of the send to the selected bus. Panorama is only available, if the selected bus is a stereo bus.
	Adjusts the send level. Please note, that it is possible to exceed the level of 0 dBfs (maximum digital resolution) so clipping may occur when applying high amounts of gain. The desired value can also be entered directly into the display field below the fader.
	Shows the current input level of the channel/input. The metering point can be switched between pre and post fader by clicking [PRE] or [POST] in the top menu bar. This setting affects the metering points of all outputs simultaneously. The red triangle besides the meter indicates the last maximum level. You can reset the peak hold function by clicking on the triangle.
	Mutes the input. This affects all sends.
	Inverts the polarity of the input signal (phase reverse).
	By clicking in the label field on the bottom of every strip you can enter an individual name for that input.

3.7.5 Detail views

Clicking on a processing block will take you to the detail view of the selected processing item.

Common elements of all detail views:

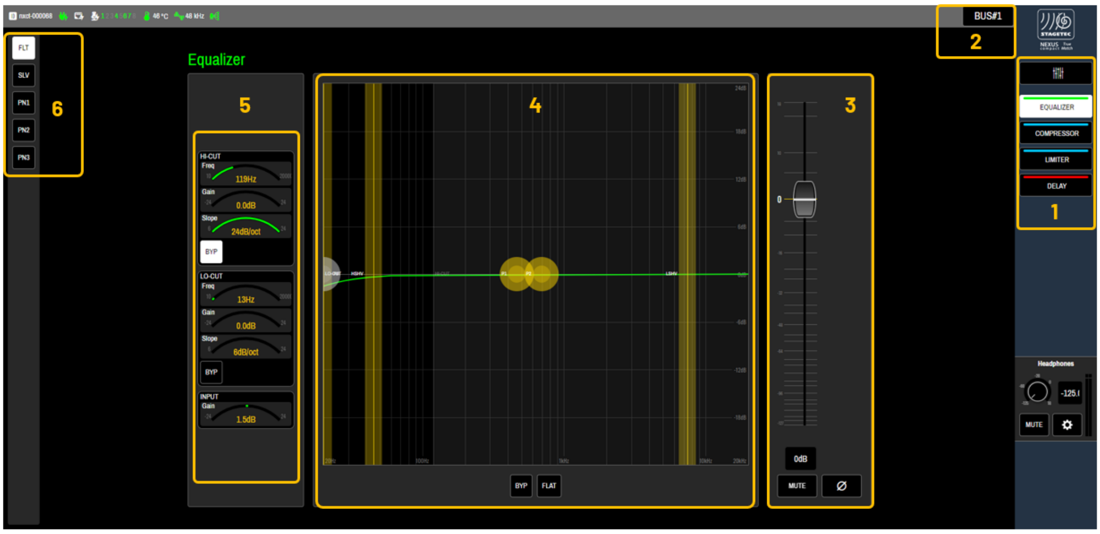


Fig. 19 Detail view page

The detail view page is divided into six important areas that are common across all details views:

Area#	Name	Description
1	Right menu bar	Navigate through the processing of the selected channel/bus by clicking on • [EQUALIZER] • [COMPRESSOR] • [LIMITER] • [EXPANDER] • [DELAY] Note: not all items are available on every channel type. Klick on [MIXER (Icon)] to return to the [PROC/MIX] page
2	Channel/Bus selection	Clicking on the [NAME] opens a dialog where you can pick any other available channel/bus to be shown. *<u>Tip:</u> Use this function to navigate between channels/busses, there is no need to return to the [PROC/MIX] page only to select another channel/bus.*
3	Channel parameter area	This area always provides access to the channels/busses main parameters such as Gain, Mute, Polarity. Not all Items are available on every type.
4	Graphical representation area	Shows a graphical representation of the currently selected processing item. Change parameters directly in the graph my touch or mouse control Simply drag the parameter to the desired value. Further, common parameters (such as [BYP] bypass or [FLAT] are located in this area. *<u>Note:</u> Parameters are surrounded by one, sometimes two yellow circles. If an item has multiple parameters (e.g. EQ: Gain, Frequency and Q), the outer ring operates the other parameter (similar to AVATUS).*
5	Arc indicator area	The Arc indicator area represents individual parameters in greater detail. To modify a parameter, either click/swipe directly on the arc, or tap on the arc to open a +/- window.
6	Left menu bar	If a selected processing item has more parameters than conveniently could be displayed, buttons will populate in the left menu bar to switch between parameter sets. The selected sets will then populate in der arc indicator area.

3.7.5.1 Equalizer Detail View

The equalizer of the NEXUS compact is an AVATUS-grade 10 band EQ with HI-CUT / LO-CUT, HI-Shelve / LO-Shelve and six parametric bands, each switchable between peak filter or notch filter.

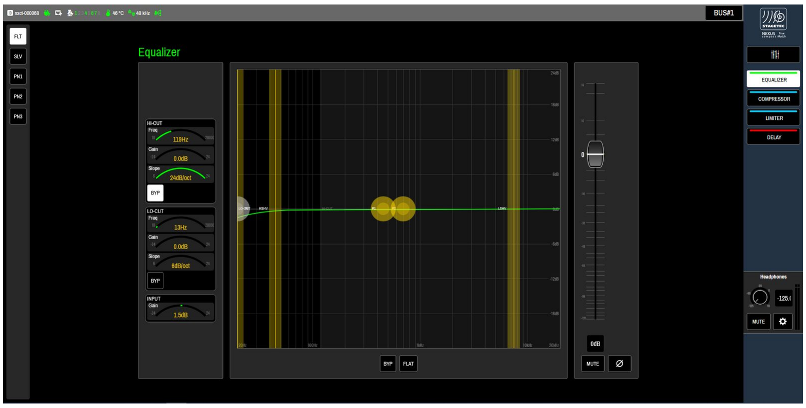


Fig. 20 Equalizer Detail View

Left menu bar	Arc indicator assignments	Graphical Representation
FLT	• Filters (Hi-Cut / Lo-Cut) with bypass and • Freq (10Hz - 20kHz) • Gain (+/- 24db) • Slope (6,12,18,24) • Input Gain	• [BYP] EQ bypass • [FLAT] reset the EQ *<u>Note:</u> Bypassed parameters will not be displayed in the graphical representation.*
SLV	• Shelving bands with bypass and • Freq (10Hz - 20kHz) • Gain (+/- 24db) • Slope (6,12,18,24) • Input Gain	
PN1	• Peak/Notch-Filter 1 and 2 with • Function switch (Peak/Notch) • Freq (10Hz - 20kHz) • Gain (+/- 24db, only on peak filters) • Q (0.125-20) • Input Gain	
PN2	• Peak/Notch-Filter 3 and 4 with • Function switch (Peak/Notch) • Freq (10Hz - 20kHz) • Gain (+/- 24db, only on peak filters) • Q (0.125-20) • Input Gain	
PN3	• Peak/Notch-Filter 5 and 6 with • Function switch (Peak/Notch) • Freq (10Hz - 20kHz) • Gain (+/- 24db, only on peak filters) • Q (0.125-20) • Input Gain	

3.7.5.2 Compressor Detail View

The compressor of the NEXUS compact is an AVATUS-grade high quality compressor with sidechain filter and additional parameters such as soft clip, wet/dry, rms-window.

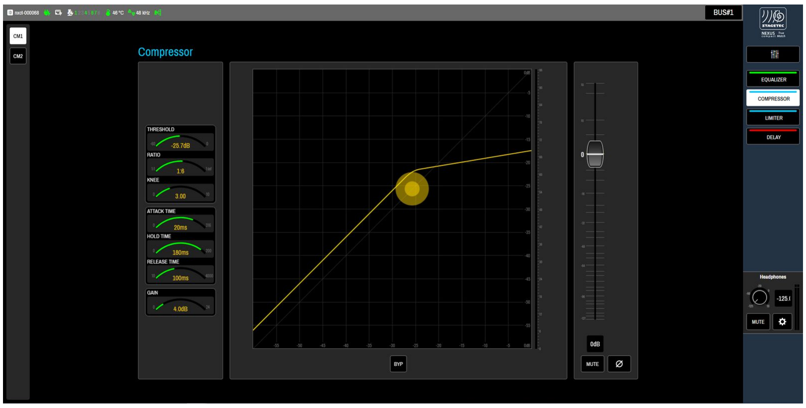


Fig. 21 Compressor Detail View

Left menu bar	Arc indicator assignments	Graphical Representation
	• Threshold (-60 - 0dB) • Ratio (1:1 - 1:inf) • Knee (0-10dB) • Attack Time (0 - 316ms) • Hold Time (0 - 200ms) • Release Time (10 - 4000ms) • Gain (0 - 24dB)	• [BYP] EQ bypass • [FLAT] reset the compressor *<u>Note:</u> If the compressor is bypassed, the reference-lines in the graphical representation will turn "grey".*
	• Sidechain High-Pass (20 - 5000Hz) • Sidechain Impact (0 - 100%) • RMS window (0 - 10ms) • soft clip (0 - 0.3) • wet/dry (0-100%) • [DYNAMIC MODE] switches between log/lin behavior	

3.7.5.3 Expander Detail View

The expander of the NEXUS compact is an AVATUS-grade high quality expander with sidechain filter, wet/dry mix and "DYNAMIC MODE"-option.

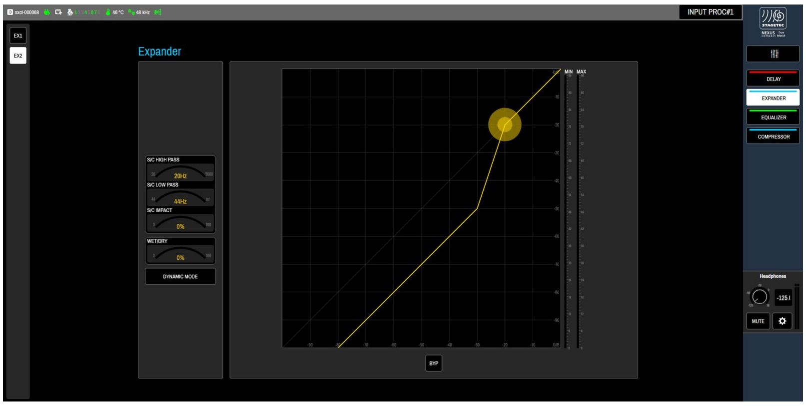


Fig. 22 Expander Detail View

Left menu bar	Arc indicator assignments	Graphical Representation
	• Threshold (-100 - 0dB) • Ratio (1:1 -1:20) • Knee (0 - 10dB) • Limit (-100 - 0dB) • Attack Time (0 - 316ms) • Hold Time (0 - 200ms) • Release Time (10 - 4000ms)	• [BYP] Bypass the expander • Min/Max Meter *<u>Note:</u> If the expander is bypassed, the reference-lines in the graphical representation will turn "grey".*
	• Sidechain High-Pass (20 - 5000Hz) • Sidechain Lo-Pass (44 - inf. Hz) • Sidechain-Impact (0 - 100%) • Wet/Dry (0 - 100%) • [DYNAMIC MODE]	

3.7.5.4 Limiter Detail View

The compressor of the NEXUS compact is an AVATUS-grade high quality limiter.

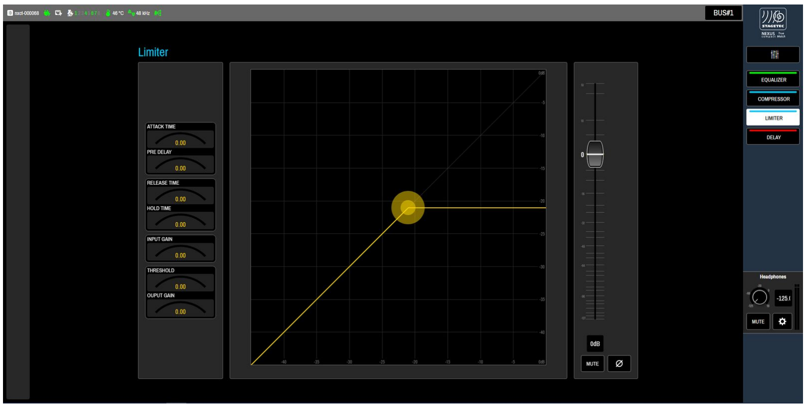


Fig. 23 Limiter Detail View

Left menu bar	Arc indicator assignments	Graphical Representation
-	• Attack Time • Pre Delay (Look-Ahead) • Release Time • Hold Time • Input Gain • Threshold • Output Gain	

3.7.5.5 Delay Detail View

The delay of the NEXUS compact can be switched between samples, mx, m, ft and frames(25).

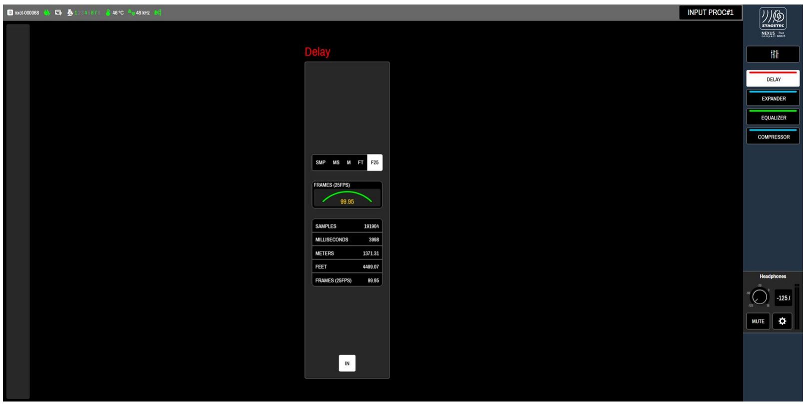


Fig. 24 Delay Detail View

Left menu bar	Arc indicator assignments / Graphical Representation
-	• Display mode switch • Samples • ms • meter • feet • frames (25) • Delay time (0 - 4000ms) • Delay display

3.8 // XY ROUTING PAGE

In addition to the direkt source/target selection in the Input/Output pages, NEXUS compact provides a convenient XY routing matrix. The [XY ROUTING] displays sources on the left, and destinations on the bottom of the page.

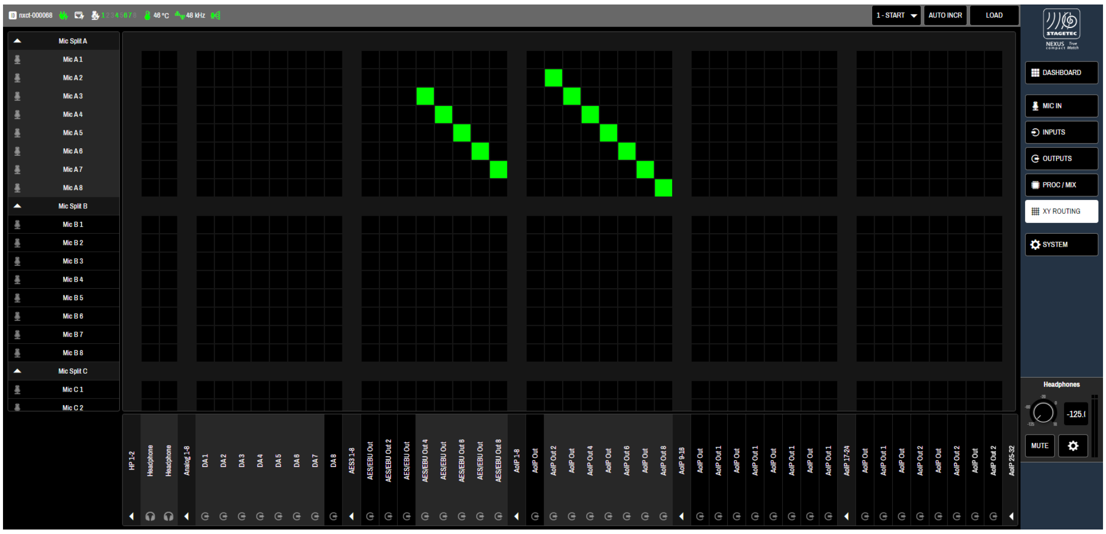


Fig. 25 XY ROUTING page

For better navigation, the I/O-types are grouped. Groups can be expanded or collapsed by pressing the [TRIANGLE]-symbol next to the group.

Input	Output
Mic Split A	Headphones
Mic Split B	Analog 1-8
Mic Split C	AES3 1-8
Mic Split D	AoIP 1-8 ... AoIP 57-64
AES3 1-8	Input Proc 1-8, Input Proc 9-16
AoIP 1-8 ... AoIP 57-64	
Bus 1-8, Bus 9-16	
Input Proc 1-8, Input Proc 9-16	
Miscellaneous	
Generator	

To establish a connection/routing, simply click on the crosspoint of source and destination. Inactive crosspoints are indicated with a "black" background, while active crosspoints are indicated with a "green" crosspoint.

Note: Unlike NEXUS, NEXUS compact always executes the routing immediately (direct mode).

3.9 // SYSTEM PAGE

On the SYSTEM page allows you to configure the network ports, save and load system information and set basic audio parameters. Furthermore you have access to the GPIO funktionality and snapshot automation. Enter the SYSTEM page by selecting [SYSTEM] from the right menu bar. The SYSTEM page is divided into 3 sub pages: GENERAL, GPIO and SNAPSHOTS. You can toggle between these sub-pages using the tab buttons at the right of the top menu bar.

3.9.1 General

The GENERAL page allows you to configure the Network ports, save and load system information and set basic audio parameters. To acces the GENERAL page, select [SYSTEM] on the right menu bar, followed by [GENERAL] from the top menu bar.

Fig. 26 SYSTEM page / general

3.9.1.1 Network panel

On the network panel you set the parameters of the three network ports.

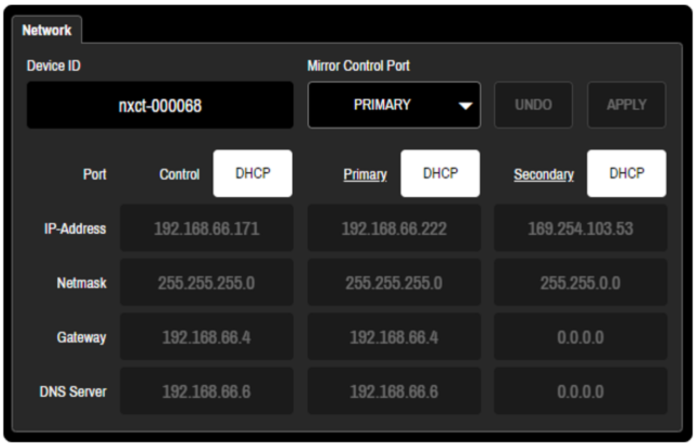
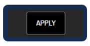
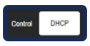
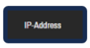
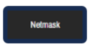
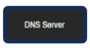


Fig. 27 Network panel

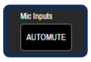
Element	Function
	Shows the device ID (serial number) of the NEXUS compact currently connected
	This popup menu allows you to select one of the audio-over-IP-ports (primary or secondary) to be used as control port.
	Discards the latest changes on the network panel and sets network parameters to the last saved state.
	All changes made to the network panel require the [APPLY] button to be pressed to become active. Please note that the device will reboot when [APPLY] is pressed and audio interruptions are likely to occur.

Port	Element	Function
The following parameters are available for CONTROL PRIMARY and SECONDARY ports respectively.		If the [DHCP] button is active (backlit in white) all network parameters are set automatically by a DHCP server.
		If [DHCP] is not active you can enter an IP adress here.
		If [DHCP] is not active you can enter a netmask here.
		If [DHCP] is not active you can enter a gateway here.
		If [DHCP] is not active you can enter a DNS here.

3.9.1.2 Audio panel

On the audio panel you set basic audio parameters.

Fig. 28 Audio panel

Element	Function
	These buttons are used to define the audio clock source. You can choose between PTP (audio over IP) or one of the AES3 ports. Samplingrates from 44.1/48kHz to 88.2/96kHz and 176.4/192kHz are possbile.(Note that the channelcount on the AoIP side can decrease with high samplingrates)
	When activated, [AUTOMUTE] automatically mutes the mic inputs when cables are connected or disconnected. This function can help to avoid unwanted clicks, especially when 48 V phantom power is applied.
	This rotary encoder allows you to set the analogue reference level for 0 dB fullscale in the digital domain. The sensivity of the converters can be set from 0 to +24 dBu. Please note, that the analogue output level at 0 dBfs is also affected.
	active/inactive: deactivate the front panel encoder for the Headphone Output

3.9.1.3 System panel

The system panel provides access to basic system information.

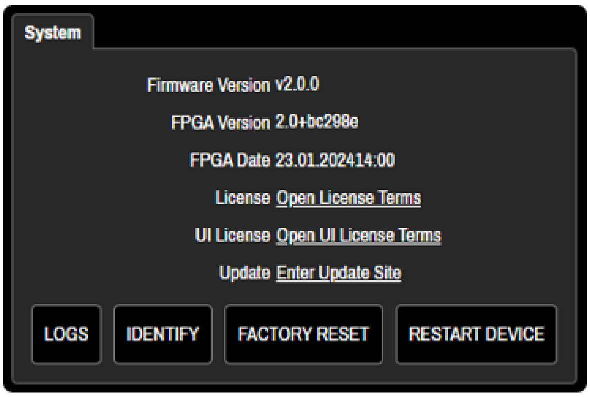


Fig. 29 System panel

Element	Function
Firmware Version	Shows the firmware version of the NEXUS compact currently connected.
FPGA Version	Displays the FPGA version of the NEXUS compact currently connected.
FPGA Date	Shows the FPGA build date of the NEXUS compact currently connected.
[License]	Show license information
[UI License]	Show license information
[Update]	Open the Update page to perform system updates
LOGS	Saves the logfiles of the NEXUS compact to the harddrive of your computer.
IDENTIFY	Pressing [IDENTIFY] will make the system LED of the NEXUS compact blink for four seconds. This feature can help to identify the NEXUS compact currently controlled by this specific web GUI.
FACTORY RESET	Restore the factory setting
RESTART DEVICE	Restarts the NEXUS compact and loads the current power on state. Please note that audio interruptions are likely to occur.

3.9.1.4 Power On State panel

This panel is used to manage, save and recall the initial status of the NEXUS compact when power is switched on.

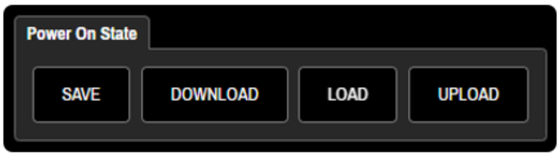


Fig. 30 Power On State panel

Element	Function
SAVE	Saves the current parameter settings as power on state.
DOWNLOAD	Saves the current power on state to the hard drive of your computer. Please note, that [DOWNLOAD] saves the power on state, which will not necessarily be the current parameter setting.
LOAD	Loads the parameter settings of the power on state.
UPLOAD	Loads an *.xml file containing a power on state from your computer`s hard drive.

3.9.1.5 Generator panel

The Generator panel provides access to the parameters of the signal generator, which can be routed to any output of the NEXUS compact.

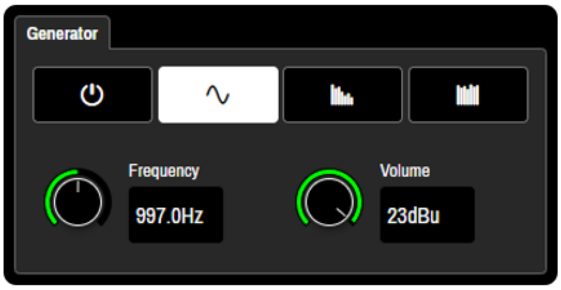


Fig. 31 Generator panel

Element	Function
	Select the generator output • On/Off • Sine wave • Pink noise • White noise
	Set the frequency of the generator
	Set the output of the generator

<u>Note</u>: To assign the signal generator to a specific output, select „Generator" in the corresponding routing popup menu on the OUTPUTS page. To assign the signal generator to a specific mic input, press the [GEN] button in the corresponding strip on the MIC IN page.

3.9.2 GPIO

You can access the GPIO page by pressing the [GPIO]-button in the top menu bar. The GPIO page is used to configure the GPI remote control of the NEXUS compact. Furthermore you can activate the GPO remote functionality. To acces the GPIO page, select [SYSTEM] on the right menu bar, followed by [GPIO] from the top menu bar.

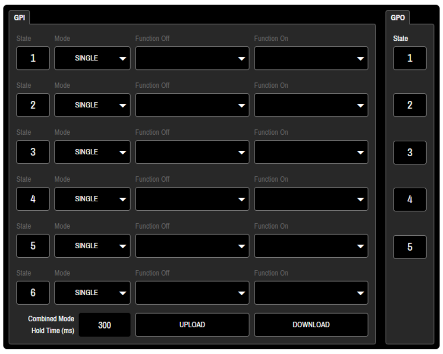
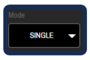
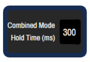
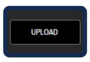


Fig. 32 GPIO panel

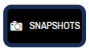
While the GPI panel configures the GPI remote of the NEXUS compact, the GPO panel simply triggers the five GPOs available in order to check the functionality of external equipment using the appropriate GPO.

Element	Function	
	[State](GPI): By clicking on the „State" button you can test the functionality of the corresponding GPI . If the button is black, the GPI / GPO is set to „LOW". If it is backlit in white, the GPI / GPO state is „HIGH" .	
	[State](GPO):The GPO „State" buttons activate 5V control voltage on the corresponding GPO output. The GPO is set to „HIGH" (5V present) when the button ist backlit in white.	
 Mode: This popup menu lets you choose the basic behaviour of the GPI	[SINGLE]	When a GPI is pulled „HIGH", the ON snapshot is loaded. If the GPI is pulled „LOW", nothing happens.
	[DUAL]	Acts like [SINGLE] mode, but additionally loads the OFF snapshot when the GPI is pulled „LOW".
	[LATCHED]	When a GPI is pulled „HIGH", the ON and OFF snapshots are loaded alternately. If the GPI is pulled „LOW", nothing happens.
	[COMBINED]	[COMBINED] is a mixed mode of [DUAL] and [LATCHED]. If the GPI is pressed briefly it acts like [LATCHED]. When being held „HIGH" for a longer time it acts as [DUAL]. This behaviour can be useful for PFL / Talk functions. The hold time can be specified at the bottom of the GPI panel.
	[ENCODER]	Sets the GPI to encoder functionality. When a GPI ist set to [ENCODER], no GPI snapshots can be used.
	„Combined Mode Hold Time": Click in the text box to enter the desired minimum hold time to achieve [DUAL] behaviour in [COMBINED] mode.	
	This popup menu is used to select a GPI snapshot for the corresponding GPI. Please note, that in [ENCODER] mode no GPI snapshot can be used.	
	Loads a set of GPIO configurations from he hard drive of your computer	
	Saves the current GPIO configuration to the hard drive of your computer.	

Notice: For a detailed description of the workflow with GPI snapshots refer to chapter GPI snapshot menu bar

3.9.3 Snapshots

You can access the snapshots queue page by pressing the [SNAPSHOTS]-button in the top menu bar:

Element	**Function**
	Opens the Snapshots queue page

On the Snapshot Page you can define a queue of up to 32 snapshots. This determines the order in which the snapshots are retrieved by the SNAPSHOTS panel on the right menu bar.

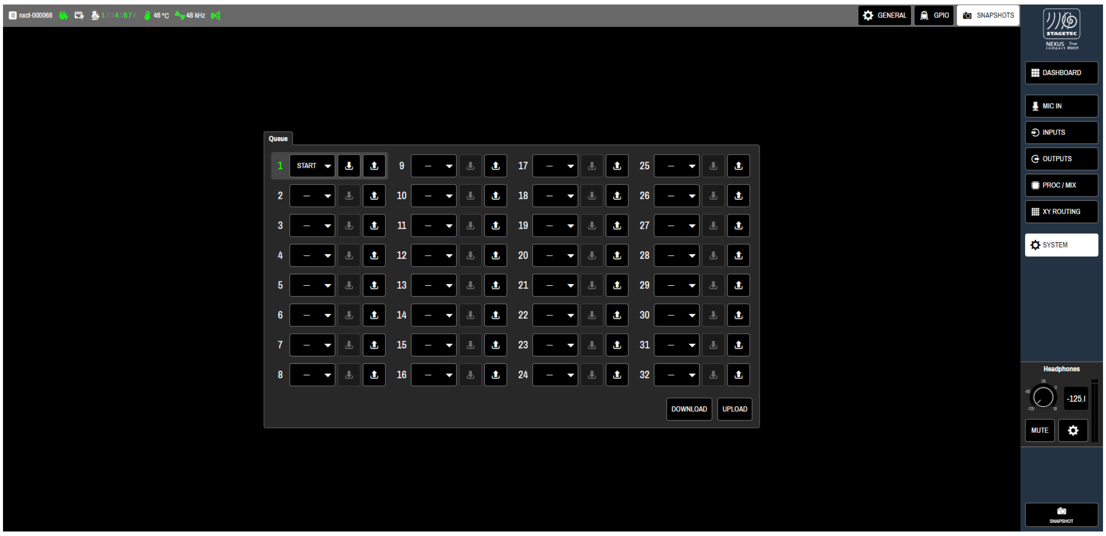
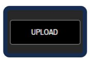
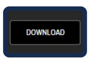


Fig. 33 Snapshots page

Element	**Function**
	Queue-position number. Defines the order in which the corresponding snapshots are loaded. If a snapshot is assigned to that specific number, the display turns green. Otherwise the number remains grey. Snapshot numbers without snapshots are omitted from the recall order.
	Click on the popup menu to load a snapshot to the corresponding slot.
	Saves the snapshot from this slot to the hard drive of your computer.
	Loads a snapshot from the hard drive of your computer into the corresponding slot.
	Loads a recall queue from he hard drive of your computer.
	Saves the current recall queue to the hard drive of your computer.

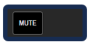
Notice: For detailed informations on creating and recalling snapshots refer to chapter Snapshot menu bar (see page 43)

break

3.10 // HEADPHONE PANEL

Below the function buttons on the right menu bar you will find a control panel for headphone output. This panel is directly accessible from all function pages.

Fig. 34 Headphone panel

Element	Function
	Adjust the volume of the headphones
	Mute the headphone output
	Open the headphones-settings dialog

The Headphone Settings Dialog

The headphone settings dialog provides a fast way to select the headphone source:

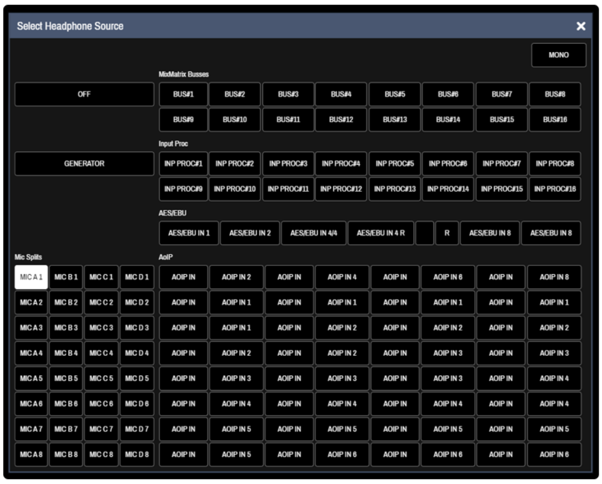


Fig. 35 Headphone settings dialog

To select a source, simply click on the [SOURCE]-button. Additionally, you can set the headphone output to [MONO].

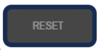
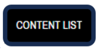
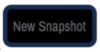
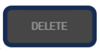
3.11 // SNAPSHOT MENU BAR

At the bottom of the right menu bar you can find the [SNAPSHOT]-button that will open the snapshot menu bar at the bottom of the page, which provides access to snapshot creation and editing.

Element	Function
	Opens the snapshot-menu bar

The Snapshot menu bar provides all elements necessary to create and edit snapshots.

Fig. 36 Snapshot menu bar

Element	Function
	[RESET] deselects certain functions from the list of parameters to be stored. Pressing [RESET] briefly will backlight the button in purple. Now you can remove parameters from your selection by simply clicking on them. Press [RESET] again to confirm your choice. Pressing and holding [RESET] for two seconds will remove all parameters.
	Creates a full snapshot containing all parameters of the NEXUS compact. When pressing [FULL], a confirmation popup window will appear. Pressing [OK] selects all parameters, while [CANCEL] will revive the previous parameter selection.
	Opens a list of all parameters stored in the currently selected snapshot.
	Loads the parameter settings of the selected snapshot into the NEXUS compact.
	Selects a snapshot to be edited or loaded into the device. To create a new snapshot, select the entry called „New Snapshot" and type a name for the snapshot you want to generate.
	Saves a new snapshot. Please note, that you have to enter a name in the Snapshot popup menu first.
	Deletes the currently selected snapshot. Briefly pressing [DELETE] will bring up a confirmation popup window while holding the button for at least two seconds will immediately delete the selection.
	Closes the snapshot menu bar. For safety reasons a confirmation popup window will appear.

3.11.1 Creating a new snapshot

To create a new snapshot, choose "New Snapshot" from the snapshot popup menu. Navigate to the relevant pages and select the parameters you wish to save by clicking on them. They will be highlighted with a purple frame. Now you can adjust them to the desired values as usual. Pressing [SAVE] will create the new snapshot.
To remove parameters from your selection press [RESET]. The button will be backlit in purple. You can now deselect all parameters you wish to remove. Press [RESET] again to confirm your choice and return to normal mode. Holding [RESET] for two seconds will deselect all parameters.
To create a full snapshot containing all parameters of the NEXUS compact, press [FULL].

3.11.2 Editing an existing snapshot

To edit an existing snapshot first select it from the snapshot popup menu. You can now select and deselect parameters as described above. Furthermore you can change values of selected parameters as desired. To save your changes, press [SAVE] and click [OK] in the confirmation popup menu.

Notice: While the snapshots menu bar is active, all parameters highlighted with purple frames display
the values of the selected snapshot. This is a preview mode and does not represent the current state of the NEXUS compact hardware. To load the parameter settings of the selected snapshot into the NEXUS compact, press [UPDATE DEVICE].

3.11.3 Snapshot parameter override

When a snapshot is loaded while snapshot parameters have been previously changed, parameter collisions may occur. In this case, the web GUI displays a popup window that allows you to choose how conflicts are handled.

Element	Function
[OVERRIDE COLLISIONS]	Loads all snapshot parameters and overrides local changes.
[DROP COLLISIONS]	Preserves local changes. The snapshot values of the corresponding parameters are not loaded.
[REPLACE CURRENT]	Behaves like [OVERRIDE COLLISIONS]. Additionally, all parameters not contained in the snapshot wil be deselected. Only parameters previously selected in the snapshot will be saved.
[CANCEL]	Cancels the snapshot recall.

3.11.4 Loading snapshots

There are two ways to load snapshots: Via the snapshot menu bar or by using the snapshot queue.
To load a snapshot without using the snapshot queue, navigate to the snapshot menu bar and select the desired snapshot from the snapshot popup menu. Pressing [UPDATE DEVICE] will load the snapshot into the NEXUS compact hardware.

To use the snapshot queue, navigate to the SNAPSHOTS tab on the SYSTEM page. Load the snapshot you want to recall in one of the slots using the corresponding popup menu. Select the slot in the SNAPSHOTS panel of the right menu bar and press [LOAD].

4 // SETUP

4.1 // CONNECTING TO THE WEB-GUI

For the initial setup of the NEXUS compact follow the steps below:

Web-GUI connection within an existing DHCP/DNS intrastructure

1. Connect the Control-Port of the NEXUS compact to a 1GbE switch within your network (control/management network)
2. Connect your computer to the same network
3. Open a web-browser and type "http:// nxct-<serial#>.local" to open the NEXUS compact configuration page. Use the serial number of your NEXUS compact. If the serial number is 000078, the URL would be „http://nxct-000078.local"
4. Proceed according to chapter Configuring AoIP-Interfaces (see page 45)

Web-GUI connection within a network environment without DHCP/DNS

1. Connect the Control-Port of the NEXUS compact directly to your computer or to a 1GbE switch
2. When using a switch, connect your computer to the same network
3. Make sure the IP-configuration of your computer is set to „DHCP/self assigned". Your computer and the NEXUS compact will connect automatically using zeroconf/self assigned addresses
4. Open a web-browser and type "http:// nxct-<serial#>.local" to open the NXCT configuration page. Use the serial number of your NEXUS compact. If the serial number is 000078, the URL would be „http://nxct-000078.local"

Proceed according to chapter Configuring AoIP-Interfaces (see page 45)

Troubleshooting
If it is not possible to access the web GUI, you can verify the connection by using the ping command. Within an existing DHCP infrastructure, type "ping nxct-<serial#>.<yourDomain>". Use the serial number of your NEXUS compact. If the serial number is 000078, your command would be "ping nxct-000078.sampledomain". If no active DNS-server is running, use the ping command "ping nxct-<serial#>.local". In both cases, if you get a timeout-message, refer to your network administrator for a proper network setup.

4.2 // CONFIGURING AOIP-INTERFACES

PRECAUTIONS:

- In some cases the web GUI can already be accessible while the NEXUS compact is still starting up and the AoIP-modules haven't been read out yet, don't press the [APPLY] button in the NETWORK panel of the SYSTEM screen, while you still see orange frames and no valid IP-addresses are available, since this also can cause the AoIP-modules to fail.
- For the DANTE version of the NEXUS compact, using the secondary port with gateway/DNS is only possible in DHCP configuration. With static IP-addresses no gateway or DNS can be defined or configured.

4.2.1 Setting up the network / AoIP networking requirements

Source: https://ravenna-network.com/wp-content/uploads/2020/02/AES67-Practical-Guide-1.pdf

Before wiring up a system, careful planning is advised. Configuration, system monitoring and debugging will be much more efficient if the general system layout and other vital aspects have been given ample thoughts.

4.2.1.1 System planning

Network infrastructure

Managed switches

In most cases, an AES67 system requires an administrable network (due to QoS and multicast requirements), which mandates the deployment of managed switches. Managed switches provide means for accessing the switch configuration, which, in most cases, is achieved by an internal web browser providing user-friendly access through any web browser. Other switches (mostly enterprise-grade switches) may offer a command line interface ("CLI") for more complex configuration tasks. While most switches have a useful out-of-the box default configuration, it is always advisable to check and verify the required settings.

Topology

While AES67 is strictly based on IP and can thus run on any "standard" network topology, it is always a good rule to minimize the number of switches ("hops") any particular stream will navigate in the final network. A small network may consist of only one switch, which of course makes configuration relatively easy. As the network becomes larger, star or tree topologies11 come in to play. In larger corporate networks spanning multiple subnets, it can be essential to have a deterministic route for any given connection – in this case a leaf-spine architecture12 would be the most preferred topology.

Bandwidth

In any case, it needs to be assured that ample bandwidth on any given path is available. While individual devices may have more than enough bandwidth available on a 100 Mbit/s Fast Ethernet (FE) port, the total required bandwidth to accommodate all streams on a backbone link may easily require Gigabit speed (GbE). It is a good idea to use GbE switches exclusively for your infrastructure. If you need to accommodate several hundred channels of audio, particularly if you plan to share your network with other IT services, you may even consider upgrading your backbone infrastructure to higher speeds (i.e. 10 GbE or above).

Despite the nominal link rate of the switch ports it may also be advisable to check for the maximum switching capacity. Some switches (specifically at the lower cost end) may offer a large number of ports, but won't be able to cope with the total traffic when all ports are heavily loaded. Check for terms like "backplane speed" or "non-blocking switch fabric" etc. if you expect a high load on your switch.

"Green" is evil

While preserving energy is usually a good idea, it impedes proper operation of any low latency realtime audio over IP technology. With the energy-saving function switched on, most switches will not forward single incoming packets immediately, but will wait for a few more packets to be sent down a specific link. This will result in an increased packet delay variation (PDV) which directly affects the PTP operation (end nodes fail to settle into a stable sync condition or exhibit a large time jitter). Simple disable any energy-saving functions on all switches.

Cabling

This may sound like odd advice, but ensure that you are always using quality patch cables. The required grade for GbE is Cat5e, but it doesn't hurt to use Cat6 or Cat7 cabling, especially if you need longer runs close to the maximum allowed Ethernet cable length (~ 125 m). Special care needs to be taken with mobile installations where cables often come on a drum for multiple uses: cable quality will degrade over time as twisted pairs tend to slacken inside the cable. This may lead occasionally to dropped packets despite signaling an otherwise proper link status.

IP addressing

Even if you are planning a small or medium-sized installation running on a single LAN, IP addressing is required. In general, there are three methods to assign IP addresses (and every device, including the switches, requires an IP address):

- DHCP: an automatic IP address assignment which requires the presence of a DHCP server; in most cases this can be one of the switches, if a dedicated DHCP server is not present. While this method is very convenient and you don't have to fiddle with address administration, subnet and gateway configuration, the disadvantage is that the assigned IP addresses are not immediately known (however, a device GUI will reveal its current IP address in most cases) and that devices may not receive the same IP address again once repowered or reconnected to the network.
- Zeroconf: an automatic IP address assignment which doesn't require a DHCP server. Devices entering the network assign themselves an available IP address in the pre-defined zeroconf IP address range 169.254.0.0/16.While this is also a convenient method for device network configuration in small LAN setups, it exhibits the same disadvantages as DHCP (you will get different IP addresses each time), plus one can't even select the IP subnet range.
- Manual / static IP configuration: This method requires devices to be configured individually, and IP addresses are assigned on an administrative basis. While this is quite a lot of work, especially in larger environment, it provides full control on how subnets and devices are configured. Since IP addresses remain unchanged after repowering or reconnecting to the network, a device can be safely preconfigured offline. A spreadsheet or a device database is essential to manage the network configuration.

Multicast

In order to avoid multicast packet flooding, your switches need to be configured for proper multicast traffic registration and forwarding by activating IGMP. Three versions of the IGMP protocol exist; AES67 requires IGMPv2 to be supported by the network. You can also configure your switches to support IGMPv3; they will, by definition, automatically revert to version 2 once any device is issuing IGMPv2 messages.

Next, the IGMP snooping function needs to be activated, and forwarding of unregistered multicast traffic needs to be disabled.

In order for IGMP snooping to work properly, an IGMP querier needs to be present on the network. This function can usually be invoked on any managed switch. Although a network can accommodate multiple IGMP queriers (and will automatically select one), it is safer to have only one IGMP querier enabled , preferably on a switch sitting close to the root of your network topology.

On larger networks or when employing enterprise-class switches, further multicast traffic management configuration may be required: some switches can be configured to forward any incoming multicast to a so-called multicast router port; this may or may not be desirable, depending on your network situation.

QoS

Since clock and audio traffic require high forwarding priorities, AES67 end nodes support DiffServ QoS and assign certain DSCP tags to those IP packets. The switches need to be configured to support DiffServ QoS and prioritized forwarding. Most switches have layer 2 CoS QoS enabled by default; this needs to be changed to layer 3 DiffServ QoS. Once enabled, check the priority assignments – a managed switch usually has at least 4 priority queues per egress port and AES67 operating with the recommended / default parameters requires this configuration:

- DSCP EF (46) (clock traffic)
- highest priority queue (4)
- DSCP AF41 (34) (audio packets)
- second-highest priority queue (3)
- All other DSCP values (remaining traffic)
- lowest priority queue (0)

Note: On some networks running other important / prioritized traffic other priority configuration may be required; however, it is advised, that PTP traffic always receives highest priority treatment. RAVENNA and Dante use other DSCP defaults (CS6 (48) for PTP, EF (46) for audio), but unlike Dante, most RAVENNA implementations allow DSCP reconfiguration at the end nodes to match the AES67 defaults (or any other desired configuration). For guidelines on how to interoperate AES67 with Dante devices in AES67 mode, refer to the respective chapter later in this guide.

Finally, check that the forwarding policy for the egress scheduler is set to "strict priority forwarding" (at least for the PTP traffic class, but also recommended for the audio traffic class).

Note that on larger / corporate networks, specifically if stretching across WAN connections, DSCP tags may not be respected by edge routers (they may be configured to not trust the DSCP markings originating from the local subnets and may even delete them). This will break the tight priority forwarding requirements and may lead to increased packet delay variations, resulting in longer latencies and degraded clock accuracy. Furthermore, after traversing any WAN link employing this "DSCP no-trust" policy, the DiffServ priority mechanism may be irreparably broken for any subsequent local network segments, eventually resulting in AES67 ceasing to work at all after traversing a WAN link. You may have to consult with your network administrator to discuss options to remove or bypass this constraint, if it exists.

PTP

Planning for PTP deployment is a topic on its own which may exhibit many complex facets, especially if your network is larger and stretches several subnets. Larger networks in most cases require PTP-aware switches (Boundary or Transparent Clocks) in strategic positions in the network. Due to the complexity which may be involved in configuration of such networks, we limit the discussion of PTP planning to a single LAN segment without PTP-aware switches.

PTP parameters

In most cases, PTP-aware switches are not required in LAN segments up to a medium size (several tenths of end nodes). With standard COTS switches, proper QoS configuration should result in a decent PTP performance. However, there are a few parameters of choice:

- Domain number: unless required for certain reasons, leave the domain number to the default value (0).
- SYNC message interval: all AES67 devices are required to operate with the PTP Default profile which has a default sync message interval of 1 second (2^0). Other choices under the Default profile are 2^1 and 2^-1 - we recommend setting the SYNC message rate to 2^-1 for faster settlement and better stability. AES67 also defines its own PTP profile, the Media profile. If all AES67 devices on the network support this profile (this is not a requirement), you can reduce the SYNC message interval down to 2^-4 – we recommend that you keep the SYNC message interval at the Media profile default value of 2^-3.
- ANNOUNCE message interval: ANNOUNCE messages are required to establish the best master clock currently available on the network. We suggest that you keep the ANNOUNCE message interval at the default value of 2^1 (applies both for the PTP Default and Media profiles) and the ANNOUNCE message timeout interval at 3. Note: it is very important, that ALL devices have the same setting, otherwise the BMCA may not work as expected and devices may not synchronize properly.
- DELAY REQUEST intervals: no need to deviate from the default values (2^0) either (unless you know what you are doing). Keep the delay measurement mode configured to end-toend (E2E) delay measurement.

BMCA parameters

For best synchronization results, you want to make sure that the best available master clock on the network is actually taking this role. If you have a dedicated Grandmaster device, all settings are usually in place by default to let this device become Grandmaster.

However, if you experience that this is not the case or if no dedicated Grandmaster device is present, you may have to dig a bit further into the BMCA parameter configuration in order to resolve any problems or make sure that only those devices qualify for BMCA competition which exhibit a decent PTP Grandmaster functionality by design (usually a device with a very precise and stable internal clock circuitry or which can be connected to an external reference signal, i.e. a word clock or a black-burst input).

The BMCA is an exactly specified algorithm that each devices has to follow to come to the same conclusion on the best available master clock on the network; any failure to fully and correctly implement the BMCA (even in end nodes which never can become Grandmaster at all) may result in improper synchronization results (yes, we have seen this). The BMCA relies on the ANNOUNCE messages being distributed in the network. The ANNOUNCE messages contain certain parameters about the clock quality which are compared in certain precedence:

1. Priority 1 Field: This is a user settable value. The lowest number wins. Normally this is set at 128 for master-capable devices and 255 for slave-only devices. However, if you want to overrule the normal selection criteria you can change Priority 1 and create any pecking order you wish.
2. Clock Class: This is an enumerated list of clock states. For example, a clock with a GPS receiver locked to Universal Coordinated Time (UTC) has more class than one which is free running and set by hand to your wrist watch. There are also states for various levels of holdover when a clock which had a GPS receiver lost the connection.
3. Clock Accuracy: This is an enumerated list of ranges of accuracy to UTC, for example 25-100 ns.
4. Clock Variance: This is a complicated log-scaled statistic which represents the jitter and wander of the clocks oscillator over a SYNC message interval.
5. Priority 2 Field: You guessed it, another user-settable field. The main purpose at this low end of the decision tree is to allow system integrators to identify primary and backup clocks among identical redundant Grandmasters.
6. Source Port ID: This is a number which is required to be unique, and is usually set to the Ethernet MAC address. Essentially this is a coin toss which is guaranteed to break a tie.

For practical purposes, the Priority 1 field is the most important. Start with keeping the value at the device default setting (should be 128 for devices which can become GM and 255 for devices which are slave-only). If you don’t have a dedicated GPS-referenced GM device in the network you may either decrease the Prio1 for certain devices you want to become preferred GMs, or decrease the Prio1 field for those devices, which should become GM only in case there is absolutely no better GM available on the net.

In any case, and regardless of the intended size of your network, always make sure that the PTP distribution results in the desired accuracy in any particular network segment before proceeding with setting up any stream traffic. A good indicator is the clock offset (calculated time offset from PTP master) indication offered by most end nodes. Indicators may vary between devices, most feature at least a status indicator or a numerical offset display. If you see a "green" light or see offset numbers in the single-digit microseconds or sub-microseconds range, you are usually good. Remember to check those indicators from time to time during regular operation.

Discovery

As described in the introduction, session description data is required to connect to an available stream and decode its content. While the parameters required and their proper line-up are defined by the session description protocol (SDP), AES67 does not define a mandatory method to transport the data; hence, manual read-out and entry is assumed as the minimal common ground.

Most AES67 systems or devices provide means of discovering available streams on the network and support protocol-based communication of these SDP parameters. The methods and protocols supported usually relate to the native networking solution those devices adhere to; RAVENNA, Livewire and Dante all offer discovery and connection management functionality, which of course includes the transfer of SDP data.

Unfortunately, they all use different methods and protocols, rendering them incompatible with each other:

- RAVENNA uses DNS-SD20 for discovery and RTSP21 for SDP transfer
- Livewire uses a proprietary protocol, but also supports the RAVENNA method
- Dante uses different methods – a proprietary method based on mDNS22 for native stream operation and SAP23 for AES67 formatted streams.

Since Dante devices don't have means for manual read-out or entry of SDP data, there is no practical way to establish connections between Dante devices with activated AES67 mode and any other AES67 device. For this reason, some device manufacturers have decided to include SAP support.

RAV2SAP

ALC NetworX has released the RAVENNA-2-SAP converter (RAV2SAP), a freeware tool24 to convert between RAVENNA and Dante discovery method. It translates selected stream announcements from one side to the other and makes the SDP data available accordingly. It also features manual SDP data entry and read-out and can thus help to diagnose any connection problems or integrate any devices which do not support RAVENNA or SAP

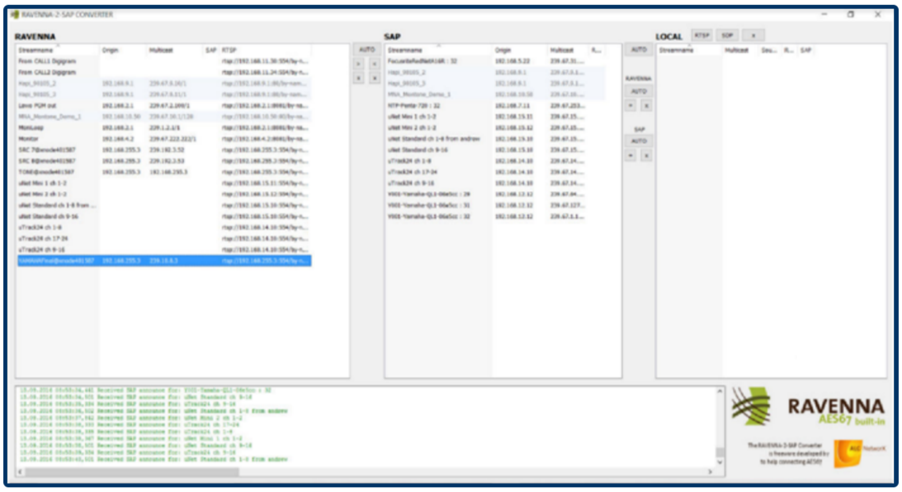


Fig. 37 RAV2SAP SCREEN SHOT

RAV2SAP is a Windows application which needs to run on a PC which is connected to the audio network. RAV2SAP only monitors and transmits discovery and SDP-related data traffic, no audio is passed through the PC (unless your PC also hosts an AES67-capable virtual sound card).

RAV2SAP is available on the RAVENNA web site - https://www.ravenna-network.com/usingravenna/support/downloads/

4.2.1.2 Device configuration

IP configuration

Select the method of IP assignment: DHCP, Zeroconf, manual / static. In case of static IP assignment, make sure you don't assign any IP address twice and that the subnet mask matches your intended network configuration. In some cases, a gateway needs to be configured (even if it won't be used). If no gateway is present, just enter the IP address of one of your switches. An Excel spreadsheet helps tracking IP configuration. If you prefer automatic IP configuration, check if IP parameters have been properly assigned through DSCP or Zeroconf.

PTP configuration

Check / configure all relevant PTP parameters the device offers; follow the guidelines given in section PTP (see page 48)

Device-specific configuration

Some devices need further configuration to interoperate properly with other AES67 gear. Here are a few commonly observed settings which may have to be configured individually:

AES67 mode

Some devices require you to activate the AES67 mode (i.e. Dante devices), other devices support AES67 natively (RAVENNA, Livewire).

Multicast address range

Despite not being fully AES67-compliant, some devices only support a limited range of multicast addresses for AES67 interoperation (i.e. Dante devices). The range needs to be configured properly with all devices; note that this may even affect devices which don't exhibit this limitation, as AES67 streams would only be identified / accepted when their multicast address is within that configured range. As some devices don't have a general device-level configuration for multicast address range (they can work with any valid multicast address in the range 239.x.y.z), this may be have to be respected when configuring individual streams.

Discovery (device specific)

While most devices also use their native discovery method for announcement of AES67 streams, some devices offer to enable other discovery options on demand (i.e. enable SAP support).

Audio-related configuration

Some devices support different sampling rates, but only one may be selected at any given time (usually because a device only has one clock circuitry). AES67 calls for support of 48 kHz, but other sample rates may be used; make sure you select the desired sample rate. Further device-specific parametrization may be required, check with the operator's manual.

4.2.1.3 Check for proper synchronization (PTP)

Once all devices have been configured, check for proper synchronization. This is important because all devices on the network derive their locally generated media clocks from the network clock distributed with PTP.

Grandmaster selection

is advised that you select a device as the preferred master beforehand and set every other device to slave-only mode. Follow the steps under BMCA parameters (see page 48) . Once properly configured, all devices should indicate that they are listening to the same Grandmaster (IP address and / or GM-ID should be identical). If you see different GM-IDs, the BMCA did not work as intended and at least one device is assuming a false GM role. Here's a quick checklist:

- Check if (PTP) multicast traffic is forwarded to all nodes (nodes need to receive the ANNOUNCE messages from all other devices for proper BMCA execution). Although the PTP multicast address (224.0.1.129) is a well-known multicast address which should be forwarded by a switch by default, an IGMP request may need to be issued to activate forwarding in certain switches.
- Check priority 1 values of devices which assume a GM role unexpectedly and compare with the settings of the designated GM. You may have to lower the priority (increase the priority 1 value) of that particular device or assign "slave-only" operation. Alternatively, increase the priority (lower the priority 1 value) of the designated GM.
- It may also help (even just for analysis) to select a different device to become GM by adjusting the priority 1 fields accordingly, or by temporarily removing suspected devices from the network.

- If you have PTP-aware switches in the network, it may help to switch PTP support off to diagnose the situation. If the situation corrects after switching off PTP support, you need to carefully check all PTP-related settings in the PTP-aware switches.

PTP accuracy

Check PTP accuracy on all nodes – slave devices generally inform about proper sync status. They either have a sync indicator (traffic light or any other graphical means) or they indicate the current offset from master numerically; in most cases single-digit microseconds are usually sufficient, submicroseconds are perfect.

If you don't have proper sync on all end nodes, you have to resolve this situation before proceeding any further (i.e. configuring streams). You may check on these potential issues:

- SYNC message rate too low: some devices require a certain sync message rate in order to reach a stable locking situation. Try to decrease the SYNC message interval at the chosen Grandmaster (i.e. try a SYNC message interval of 2^-2 or 2^-3).
- QoS not properly configured: PTP traffic needs to receive the highest forwarding prioritization. Check if PTP packets are marked with a proper DSCP value25 and if all participating switches in the network are configured to store packets with this DSCP value in their highest priority queue.
- Removing traffic load: If you are unsure about properly configured QoS you may also try to remove any foreign traffic on the network to reduce the bandwidth utilization (i.e. to remove potential network overload). The simplest approach would be to unlink devices or network segments which are not relevant to AES67. You may also start building your network from scratch by incrementally plugging in devices and check each time for proper synchronization.
- Mixed switch configuration (FE / GbE)26: In some cases a mixed use of switches with different network link speeds may cause synchronization issues. In most cases, this will result in a permanent offset from master only without necessarily affecting the synchronization stability. The node may settle into a synchronized condition, but most likely larger latency settings will be required for streams coming from / going to this node due to a permanent displacement between local and network time. It is a good advice to only use GbE switches in the network and connect end nodes with FE interfaces directly to the GbE switch ports.
- PTP-aware switches: as mentioned earlier, PTP-aware switches are certainly valuable (or even required) to improve synchronization (especially in larger networks), but they may make things more complicated and require deeper knowledge for proper configuration. Check if PTP-aware switches are part of your network and try switching PTP support (temporarily) off. Make sure that all configuration requirements for COTS switches are in place (i.e. QoS, IGMP etc.).

4.2.1.4 Stream configuration

Once your network is prepared as described above, you are ready to configure streams. While AES67 calls for support of multicast and unicast transport, we will focus on multicast streaming only as this is the method commonly available on all AES67 devices. Configuring and connecting to multicast streams generally always follows these two basic steps:

1. Configure and start a multicast stream on the sender node
2. Make the related SDP data available to the desired receiving node
3. Connect to the selected stream

Execution of these steps usually varies between individual devices; in this guide we use screenshots from the RAVENNA Virtual Sound Card (RVSC) which is based on the RAVENNA framework developed by ALC NetworX. Consult the respective Operating Manuals of other devices to execute these steps accordingly.

AES67 stream format

Since the main focus of AES67 is on interoperability, the stream format variations to be supported by all devices are pretty narrow:

- Sample rate: 48 kHz
- Data encoding: linear PCM with 16- and 24-bit (L16 / L24)
- Number of channels per stream: 1..8
- Packet time (number of samples per packet): 1 ms (48 samples per channel per packet)

Other variations are recommended, but not required to be supported. Therefore, we focus on a typical AES67 stream setup: a stereo stream with L24 encoding running at 48 kHz with 1ms packet time.

Creating AES67 streams

Invoke the stream creation function ("create Tx stream", "create session source" or alike) and fill in the parameters as required:

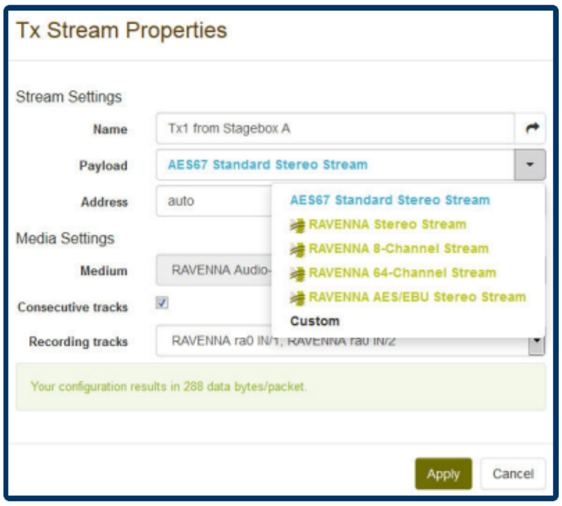


Fig. 38 RVSC screen shot: Tx Stream Properties (payload format selection)

- *Name*: assign a meaningful name for this stream (not required by AES67, but helps to identify this stream when discovery is used)
- *Payload*: select from pre-defined stream formats (here: AES67 standard stereo)
- *Address*: enter desired multicast address27 or leave at "auto" for automatic assignment

The most essential choice is of course on the number of channels in the stream and which channels to incorporate from the individual device. Select the desired audio channel pair from the drop-down menu:

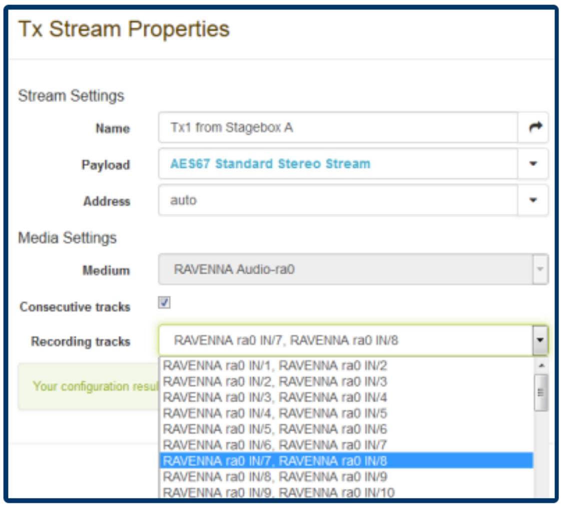


Fig. 39 RVSC screen shot: Tx Stream Properties (audio channel assignment)

Once selected, hit "Apply". This will create an AES67 stereo stream in L24 / 48 kHz data format from audio channels 7 + 8 with an automatically assigned multicast address. The stream is immediately started and available on the network. The stream packets will reach the first switch where they are dropped unless another device has registered to this stream by IGMP.

Fig. 40 RVSC screen shot: Overview with one TX stream created

If you need to access other stream configuration options, you can select "Custom" from the payload format drop-down menu which opens all available parameter fields:

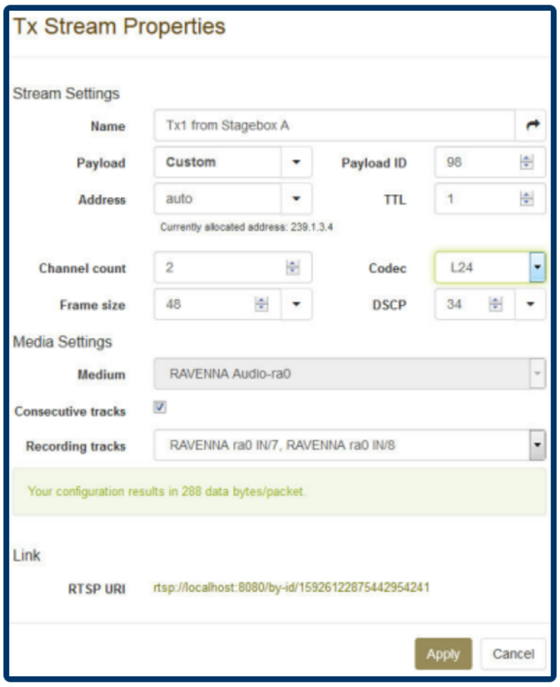


Fig. 41 RVSC screen shot: TX stream extended properties

You can now change the number of channels in the stream, use a different encoding format (i.e. L16 or AM824), change the packet time (frame size) or assign another DSCP value to this particular stream.

Accessing the SDP data

In order to connect to this stream, the desired receiver device needs access to the respective SDP data. While AES67 specifies the required SDP data, it does not mandate for a specific method to convey this data. Most AoIP solutions offer means for advertising / discovering available streams and transporting the SDP data automatically. If no common method is available between sender and receiver device, manual SDP data transfer is assumed. Alternatively, the RAV2SAP converter (see RAV2SAP (see page 49)) may be used to translate between different discovery methods and / or aid manual SDP data transfer.

SDP data transfer by a common discovery method

The streams available on the network will be directly visible in the desired receiver and the SDP data will be transferred when setting up the connection; no further action has to be taken at this stage

Manual transfer of SDP data

If manual transfer is required, the SDP data can be copied (and later pasted) by opening the related SDP data record. The RVSC provides a link to the SDP data set when creating the Tx stream:

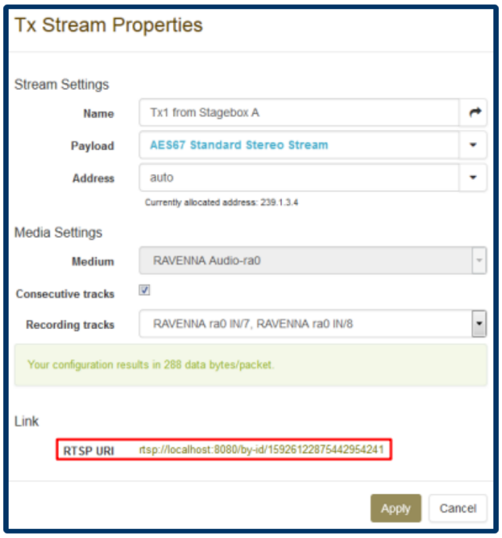


Fig. 42 RVSC screen shot: TX stream SDP link

The link allows direct access to the SDP data by any device supporting RTSP. Since RTSP is very similar to HTTP and the SDP data is formatted in ASCII text, the link can also be used to access the SDP data with any browser; simply copy and paste the link into your browser address field and replace “rtsp://…” with “http://…”. (see page 54)

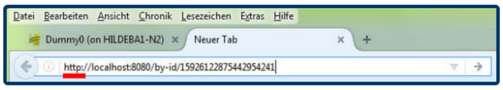


Fig. 43 Accessing SDP data with browser (by HTTP)

This will open the Windows text editor (or any application linked to text files):

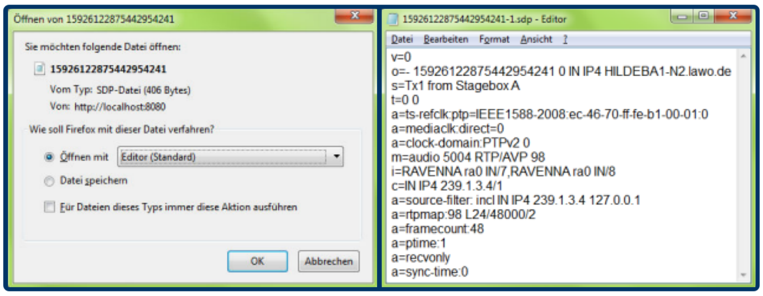


Fig. 44 Opening SDP data with editor

The SDP data can now be copied (or saved) and used for setting up a connection to this stream at the desired receiver.

Receiving an AES67 stream

Connecting to an existing AES67 stream requires inputting the SDP data of the respective stream to the desired receiver. This can either be done manually or with support of a discovery & connection management method.

Connecting to an AES67 stream with discovery support

If a common discovery method is supported by both the sender and receiver, connecting to a stream should be as easy as identifying the desired stream in the receiver's user interface (usually by name) and executing the connection function.

In the RVSC, hit open the Rx creation dialog ("Connect Rx Stream"):

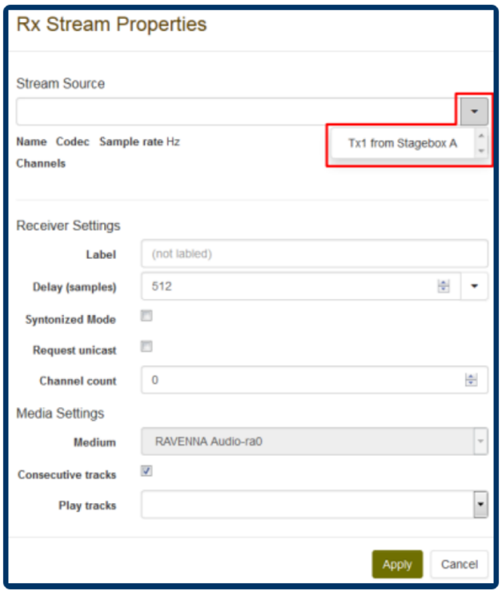


Fig. 45 RVSC screen shot: RX stream properties (stream source selection)

In the “Stream Source” drop-down box, select the desired stream from the list. The related SDP file will automatically be accessed and all relevant parameters are filled-in:

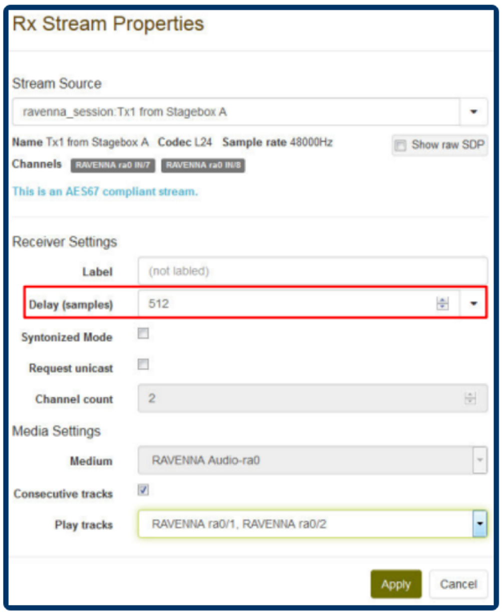


Fig. 46 RVSC screen shot: RX stream properties (stream parameters filled in)

Next, you can select the desired latency by adjusting the delay value accordingly. The RVSC provides the ability to enter the latency individually per stream to accommodate for any relevant packet delay variation, individual stream setup and correlation between any streams on the network, if desired. The number indicates the desired playout delay of an individual audio sample with respect to its sampling time at the sender. The configured number must be large enough to cover the original packet time plus any jitter the packet may experience while being transported across the network. Since the packet time in AES67 is 1 ms, the delay needs to be larger than 48 samples plus sufficient delay to cope with the packet jitter.

Other devices or AoIP systems may offer predefined (sometimes even system-wide) latency classes like low / medium / high or the equivalent.

The final step is to assign the channels being transported in the stream to the desired output channels of the device:

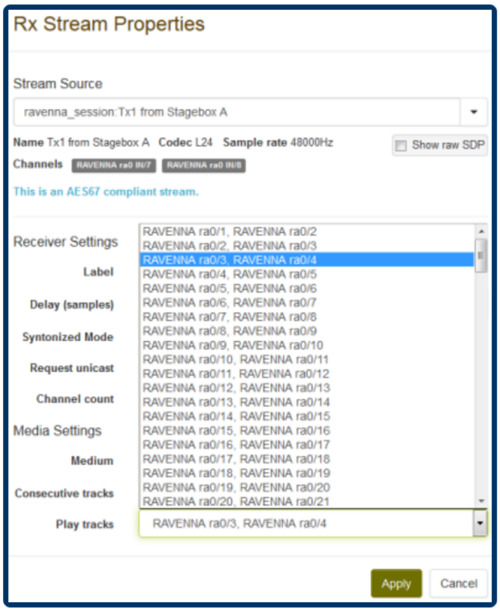


Fig. 47 RVSC screen shot: RX stream properties (channel assignment)

Once assigned from the drop-down list, hit "Apply" and the receiver will connect to the selected stream.

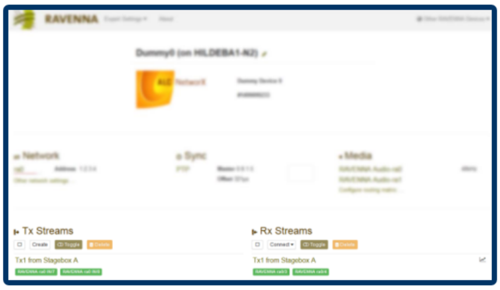


Fig. 48 RVSC screen shot: overview with one RX stream connected

The RVSC offers statistic displays where the current packet jitter can be visualized (other devices may offer numerical values to indicate current PDV):

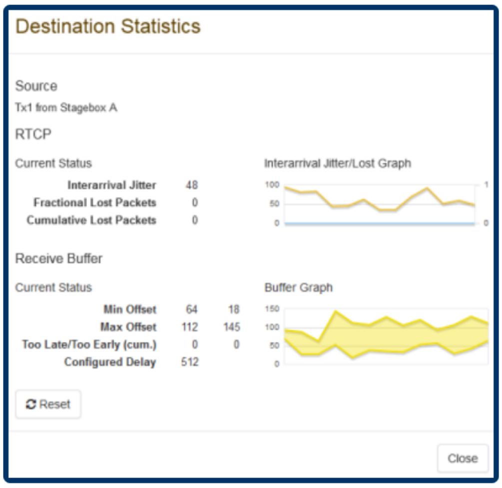


Fig. 49 RVSC SCREEN SHOT: statistics with packet jitter and receiver buffer utilization

Connecting to an AES67 stream manually

If a common discovery method is not available, the SDP data has to be entered manually into the receiver. The means on how to enter the data varies among devices, a device may either accept the SDP data record as a whole (effectively using copy & paste) or it offers a form with individual parameter fields (which will then look similar to the Tx creation screen where you have to manually type in the respective values).

The RVSC offers the option to type in or paste a complete SDP data set. Open the Rx creation dialog ("Connect Rx Stream"), select "Show raw SDP" and double-click into the large empty field which just opened up. This field is now in edit mode and ready to accept the SDP data input. Simply paste the copied SDP data provided by the sender into this field and modify if necessary (i.e. you may assign a different name by changing the "s=..." line:

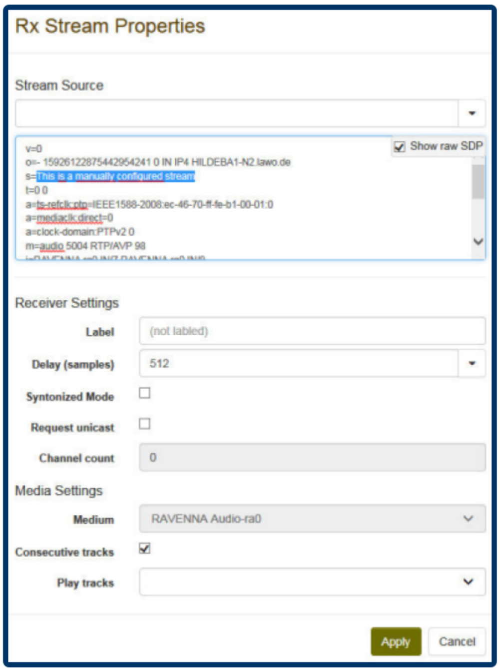


Fig. 50 RVSC screen shot: RX stream properties with pasted SDP data

Hit the "Show raw SDP" field again, apply desired latency setting and assign channels as in the previous example and hit "Apply". The stream is now being connected to and it shows up with the edited name in the Rx section of the overview screen:

Fig. 51 RVSC screen shot: overview with 2 RX streams connected

4.2.2 Out-Of-Band management

By default, the control surface of NEXUS compact is configured for „Out-of-Band management". This results in the configuration page only being available on the "control port", while the AoIP setup has to be carried out on the primary or secondary audio-over-IP ports respectively. To switch to In-Band management, open the Web GUI and navigate to the setup page by pressing [SYSTEM] on the right sidebar of the screen. On the NETWORK panel you will find a popup-menu called „Mirror Control Port", where you can choose the AoIP-port you want to mirror the control port to. Finally you have to confirm your changes by pressing [APPLY]. Please be aware that this will reboot the NEXUS compact and audio interruptions are likely to occur.

Fig. 52 Mirror control port

4.2.3 Configuring AES67-ports

The built-in AES67 module has to be configured via the primary or secondary audio-over-IP ports respectively. To access the web interface of the AES67 module follow the steps below:

1. Navigate to the SYSTEM page of the NEXUS compact Web GUI by pressing [SYSTEM] on the right sidebar of the screen.
2. On the NETWORK panel note the IP-address of the port you want to configure. (Hint: if you deactivate the [DHCP] button, you can copy the address. Make sure to activate the button again if necessary.)
3. Connect your computer or your network switch to the AoIP port you want to configure.
4. Open your web browser and navigate to „http://<IP-address> of AoIP-Port>"
5. The web interface of the AES67 module opens. Here, all functions of the AES67 module can be controlled, including setting-up the AES67-streams.
6. For the operation manual of the AES module refer to „https://www.directout.eu/download/operating-instructions%E2%80%9C and download the file „software-manual-rav2-oem.pdf".

4.2.4 Configuring Dante-ports

DANTE ports are configured via the Dante Controller application. To proceed, follow the steps below:

1. Go to https://www.audinate.com/products/software/dante-controller download the Dante Controller application and install it.
2. Connect your computer or your network switch to the AoIP port you want to configure.
3. Start the Dante Controller application and set the network port parameters as needed for your environment.
4. Also use the Dante Controller application to configure the required audio connections.

4.3 // SETTING UP AES67 AUDIO STREAMS - DIRECT OUT GUI OPERATION

4.3.1 Introduction (Direct Out)

Source:

https://www.directout.eu/de/download/info-aes67-stream-setup/?wpdmdl=7437&refresh=656a279a7b0ee1701455770&ind=16438240625661&filename=info_aes67_stream_setup.pdf

AES67 is an interoperability standard for high-performance Audio-over-IP workflows. Originally published in 2013, it provides common ground for already existing AoIP technologies such as Dante, Livewire, Q-LAN and RAVENNA enabling interoperability between compliant devices. It defines the synchronous transport of PCM data using RTP packets in IP networks. With MONTONE.42 and RAV.IO DirectOut offers products which are fully compliant with the AES67 standard. This guide intends to help you establishing AES67 streams with either of the units. It is based on and refers to AES67 Practical Guide by ALC NetworX and Merging Technologies. The web UIs of both devices MONTONE.42 and RAV.IO are nearly identical, hence screenshots are only taken from one of the devices.

4.3.2 Synchronisation

As within any other digital audio environment, a media network capable of AES67 transport needs a common clock, AES67 specifies PTPv2 (IEEE1588-2008) as common clock source which all participants use to create their specific media clock. PTPv2 uses the BMCA (Best Master Clock Algorithm) to automatically determine the grandmaster for the network. If the PTP-Mode “auto” is selected on the “Advanced” tab of the web UI, the BMCA will define the grandmaster. If you would like to clock your network to an external sync signal such as Word clock or MADI, select “preferred master” on the “Advanced” tab of the unit that receives the external sync signal. It is crucial that the externally synced device becomes grandmaster for the network. All other units should be set to “auto”. At the same time you need to select “extern” from the Clock master drop down list on the “Status” tab.

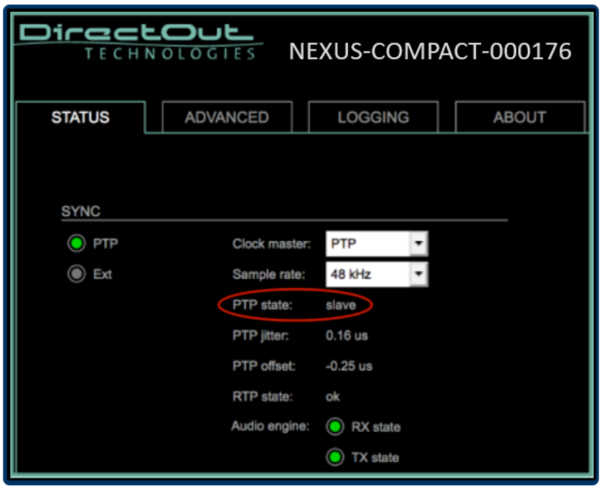


Fig. 53 PTP Master or slave status

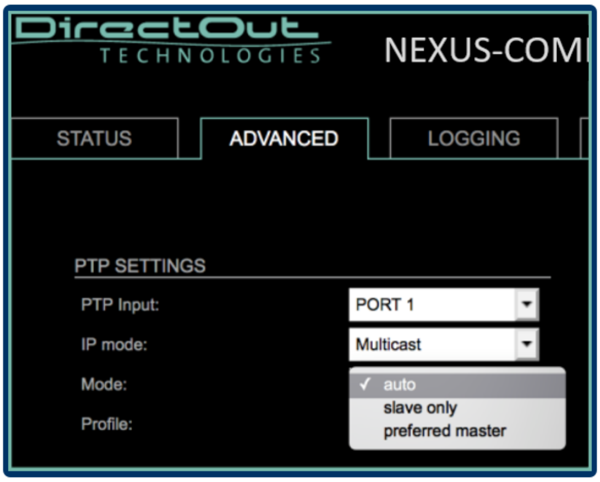


Fig. 54 PTP-Mode

Fig. 55 PTP extern

On the “Advanced” tab select “default E2E” or “Media E2E” PTP profile if you do not use switches which act as boundary clock in the network. AES67 recommends to use the Media profile. If you have switches in your network that do not support PTP, you can set the total amount of switches which are Non-PTP-aware under PTP clock settings.

This will reduce the PTP jitter but increase the time until the device actually locks to the grandmaster. DirectOut recommends to use PTP-aware switches for all professional AoIP setups.

If your network is built with boundary clocks you can use the PTP profile “default P2P” or “media P2P”.

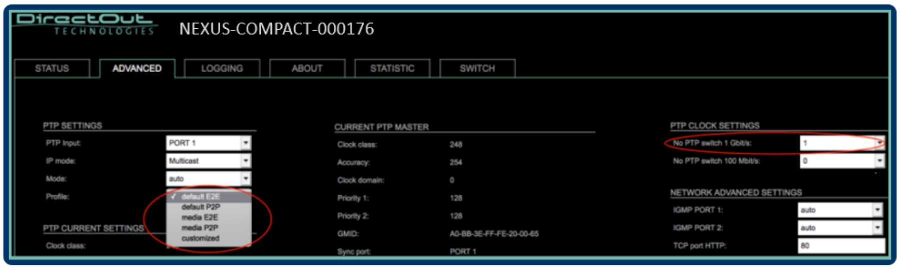


Fig. 56 PTP Profile and Clock Settings

Adjust the clock domain to match your grandmaster. It is recommended to use a domain different from 0.

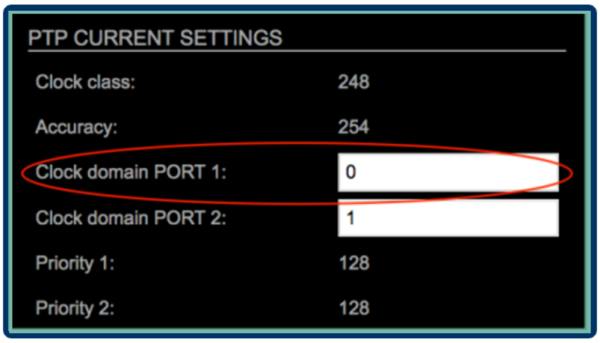


Fig. 57 Clock domain

PTPv2 is not compatible with PTPv1 (IEEE1588-2002) which is used in e.g. closed Dante networks. A Dante interface supporting AES67 is capable of listening to PTPv2 and acting as translator to PTPv1.

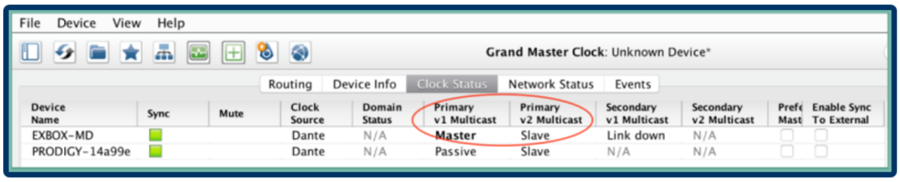


Fig. 58 Dante controller - unit as PTP translator between PTPv1 and v2

4.3.3 IGMP

The Internet Group Management Protocol (IGMP) ensures that devices in an AES67 media network do not get flooded by streams without subscribers. The management protocol only forwards streams to ports of the switch where the device subscribed for the relevant stream. Switches used in a network infrastructure ready for AES67 transport require the support of IGMPv2 to avoid multicast packet flooding. IGMP comes with a negotiation functionality to adopt all clients to the oldest version in the network, meaning if only one unit supports IGMPv1 whilst all other units do support IGMPv3, parts of the network or the whole network will fall back to IGMPv1.

DirectOut MONTONE.42 and RAV.IO do support IGMPv3 which includes by definition IGMPv1 and IGMPv2.

4.3.4 Quality of Service (QoS)

Switches used in an AES67 capable infrastructure should support DiffServ to provide correct prioritising of PTP and RTP packets. A node supporting AES67 shall tag PTP and RTP packets with different values which support priority settings for the data transport. AES67 default DSCP tags:

PTP (clock packets)	DSCP value: EF/46/0x2E
RTP (audio packets)	DSCP value: AF41/34/0x22

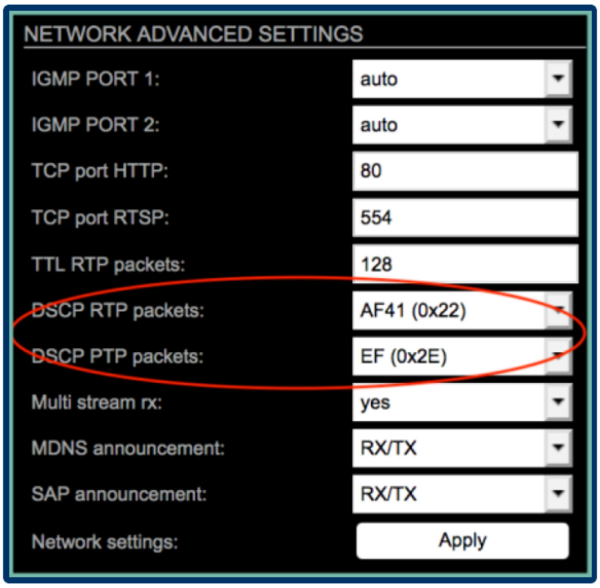


Fig. 59 DSCP values PTP and RTP packets AES67

Attention when using QoS: Some Dante devices with older firmware tag PTP packets with the DSCP value CS6/48/0x30 and RTP packets with the DSCP value EF/46/0x2E. This can lead to queues in switches with RTP packets from Dante devices and PTP packets from other AES67 compliant devices mixed in the same queue with identical prioritisation! MONTONE.42 and RAV.IO offer the possibility to adjust the tags to the same values as Dante devices. This adjustment affects all streams which are created in that device, single streams cannot get tagged individually. Under *Network Advanced Settings* on the "Advanced" tab you will find drop down lists for those adjustments.

Fig. 60 DSCP values Dante AES67

4.3.5 Multicast/Unicast

AES67 devices must support both multicast and unicast streaming. Whilst stream exchange between AES67 devices usually works with multicast using the administratively scoped multicast IP address range 239.xxx.xxx.xxx the second octet of the multicast IP needs to be fixed for Dante compatible AES67 streams e.g. 239.69.xxx.xxx. The user can define the value for the second octet in Dante Controller for each unit.

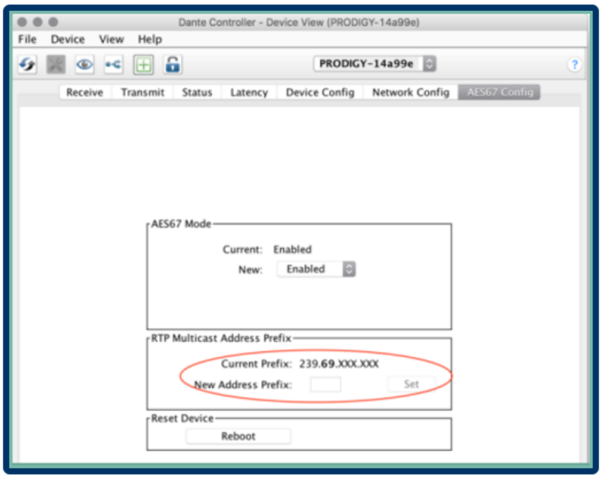


Fig. 61 Dante Controller AES67 Config

4.3.6 Stream Parameters

The following default parameters must be supported by an AES67 compliant device (minimum requirements)	
Sample Rate	default 48kHz (44.1kHz and 96kHz also possible)
Audio Format/ Payload Encoding	linear PCM 16 bit/24 bit (L16/L24)
Channel Count	1-8 mono audio channels
Packet Time	1ms (48 samples per audio channel per Ethernet frame)
Frame size	48 samples/channel
RTP payload ID	any value between 96 and 127 (Dante uses dynamic payload assignments on a perstream basis)
RTP Port	default 5004 (other values are allowed)

On the „Status" tab you will find the configuration windows for Output Streams. Define a stream name and set the above mentioned parameters as well as the multicast IP address of the stream.

Fig. 62 Output Stream Settings Dante compatible AES67 stream

Activate the stream and continue at the receiving end.

4.3.7 Stream Discovery

AES67 defines that the above parameters are stored as SDP (session description protocol) file. AES67 does not define a way how to announce or share the SDP information. It can be obtained via URL or RTSP/SAP protocols or entered manually into the receiving device. Dante devices require SAP to exchange the SDP data, manual configuration is not possible in Dante Controller.

For unicast streams SIP is mandatory for SDP exchange

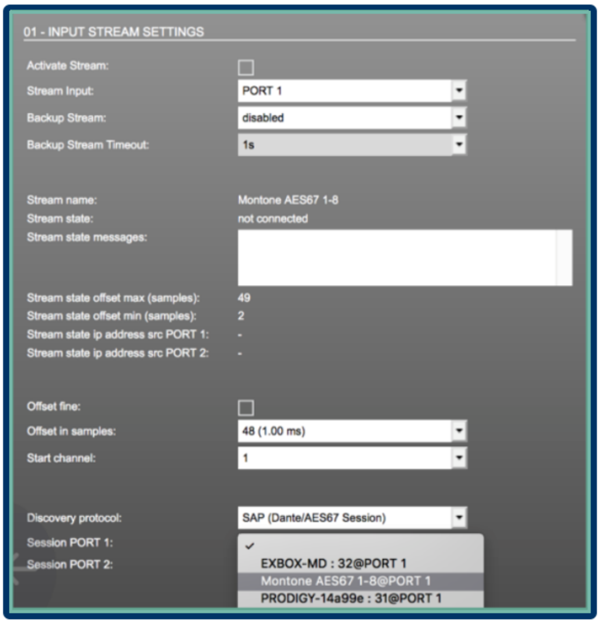


Fig. 63 Input Stream Settings - Stream Discovery

4.3.8 Input Stream Settings - Stream Discovery

The stream offset/delay at the receiving side needs to be set to a higher value (about factor 2) than the packet time at the sender (e.g. 2.67ms/128 samples if the packet time at the sender is set to 1ms/48 samples). DirectOut MONTONE.42 and RAV.IO show a warning under Stream State Messages such as *RTP time stamp* out of bound if the offset value is too low and packets arrive too late to be played out on time.

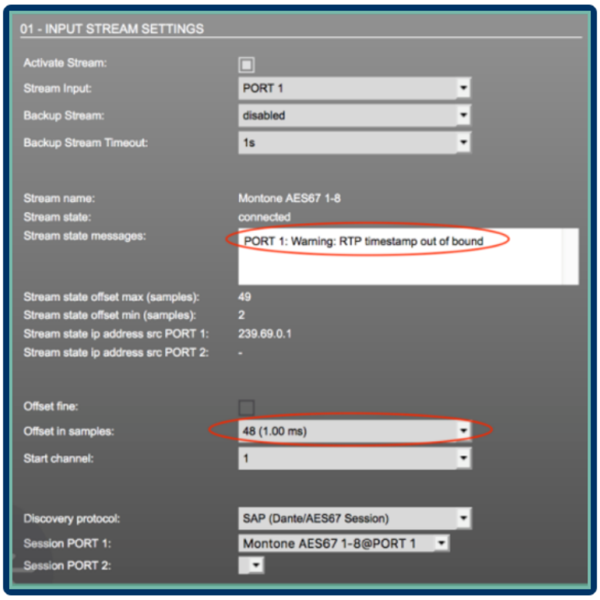


Fig. 64 Input Stream Settings – RTP Error

Increasing the offset at the receiver increases the total latency of the signal transport and at the same time resolves timing issues.

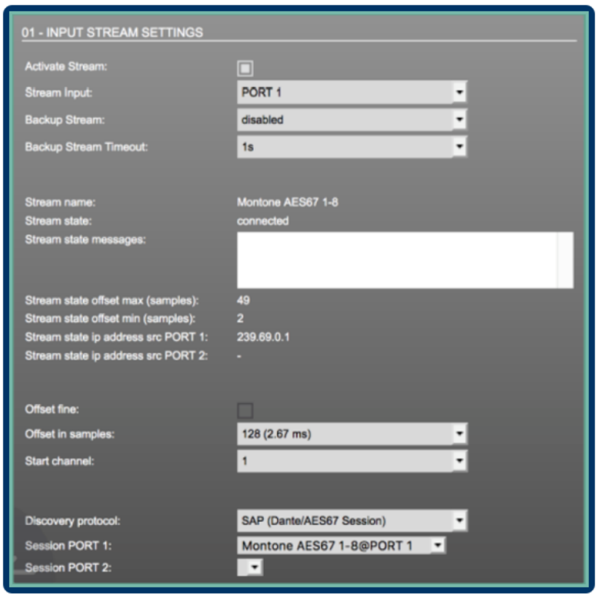


Fig. 65 Input Stream Settings - no warning

Activate the stream.

For further instructions how to map audio channels etc., please take a close look into the manual of MONTONE.42 and RAV.IO

4.4 // SETTING UP DANTE – DANTE CONTROLLER OPERATION

For information on how to setup Dante-networks, please find documentation here: https://my.audinate.com/resources/technical-documentation

4.5 // SETTING UP EMBER+ CONTROL

Ember+ is a control protocol designed to provide extensive control prossibilities with miminal implementation efforts.

4.5.1 Identifying the "path" / Ember+ viewer

To get an inside view of the control options of NEXUS compact, download the latest version of the Ember+ viewer: Download Ember+ viewer here[3]

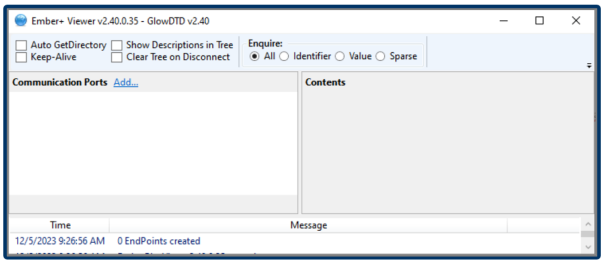


Fig. 66 Ember+ viewer

To connect to a NEXUS compact, ensure your computer is located within the control network of the NEXUS compact. Click on [Add...] and enter the IP-Address or the hostname of the NEXUS compact, Port is 9000. Confirm with [OK].

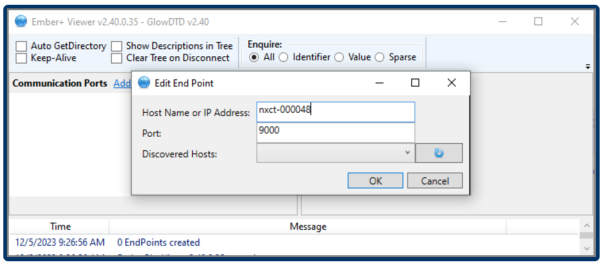


Fig. 67 Ember+ viewer: establishing a conntection

If the connection is established, the Ember+ tree of the NEXUS compact will populate. The tree is devided intro three sections:

- System
- Audio
- GPIO

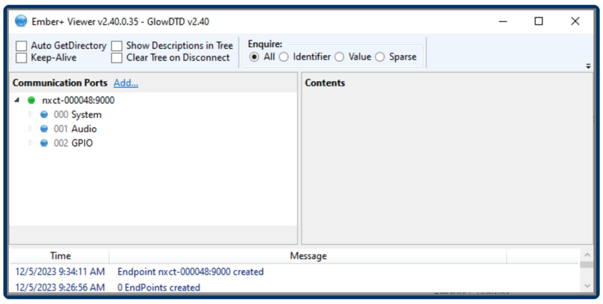


Fig. 68 Ember+ viewer: connection established

3 https://github.com/Lawo/ember-plus/releases

Sections of the Ember+ tree	
System	This section provides access to the majority of system settings and information, such as "power on state", "firmware information", "network settings"...
Audio	This section provides access to all audio-related parameters for inputs, outputs, processing and the mix-matrix
GPIO	This sections provides access to the state of the GPIO's of the NEXUS compact.

To control a parameter via Ember+, the parameters' "Path" is required. To identify the "Path", use the Ember+ viewer to navigate to the parameter you want to control. The following example shows the parameter "Mic 1 - Split 1 → Mute" within the Ember+ tree:

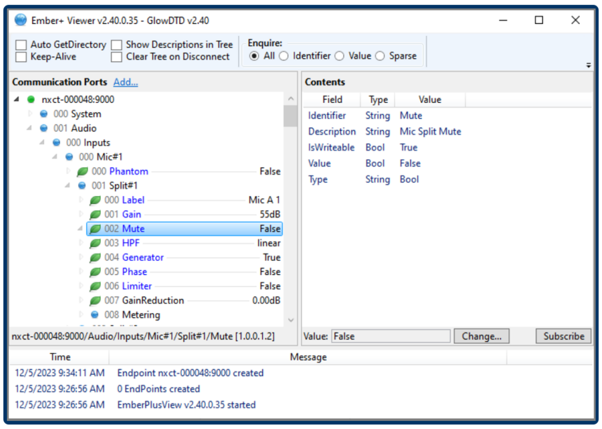


Fig. 69 Ember+ viewer: parameter selected

Once a parameter is selected, the Ember+ viewer provides detailled information about this parameter, it's value and it's data-type. In the left bottom corner you will find the "Path" for this parameter, in our expamle: 1.0.0.1.2

To change the mute-state in this example, you need to send the parameter "true/false" with the data-type "Bool" to the path 1.0.0.1.2

This can be done via the Ember+ viewer for testing purposes (select the parameter, click on [Change...] or by any control software supporting Ember+.

4.5.2 Expample: Setting up "Companion"

“Companion” is a free software that allows devices being controlled via Ember+ (and other control protocols). It provides a Web-GUI as well as support for hardware devices (3rd Party).

Companion can be downloaded here: Download "Companion"[4]

After installation, click on [Launch GUI] to open the web interface:

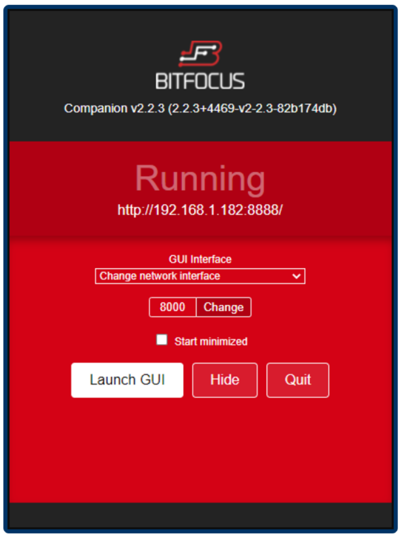


Fig. 70 Companion: Startup dialog

In the web gui, click on [Connections] and search for “Ember+” in the right side search pane. Click [Add] to add “Generic Ember+” as connection:

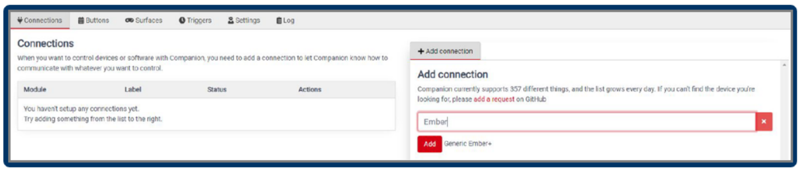


Fig. 71 Companion: Adding generic Ember+

Next, enter the IP of the NEXUS compact and a Label for the connection in the setup-dialog, the “Target Port” is 9000. Click on [Save].

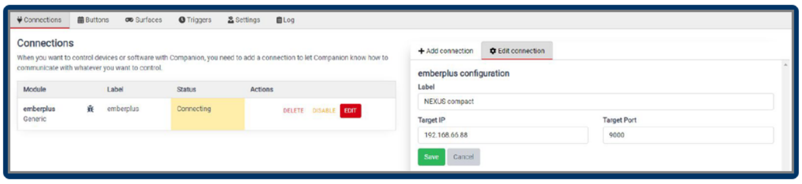


Fig. 72 Companion: Connection parameters

4 https://bitfocus.io/companion

In this (basic) example, we will assign the “Mute function” of “Mic 1 - Split 1” to two virtual buttons. Navigate to [Buttons], select the first button and click on [Regular Button] on the right side. Companion will now provide settings for this button:

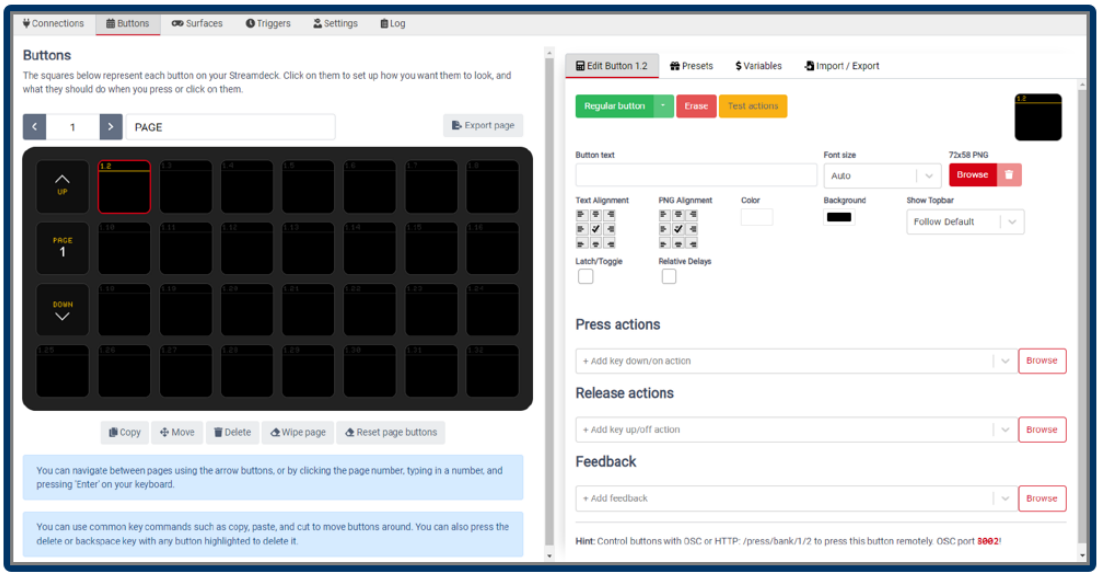


Fig. 73 Companion: Button parameters

Enter a “Button Text” (e.g. Mic 1 Mute) and a button color (e.g. red). Go to “Press Actions” and press on [Browse]. A new dialog will open, select your “NEXUS compact” connection, a dropdown of options will appear:

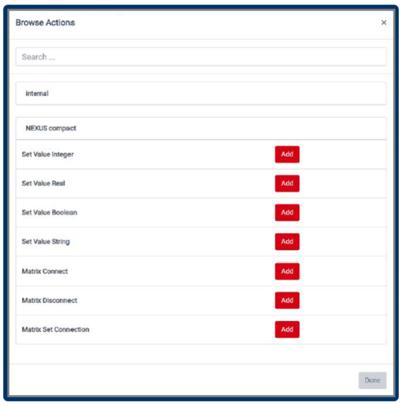


Fig. 74 Companion: adding a press action

As identified earlier using the Ember+ viewer, the path for “Mic 1 - Split 1 → Mute” is 1.0.0.1.2 and the type is “Bool”. Therefore, select “Set Value Boolean (true)” and press [Add]. Enter these values to the “Press action”:

Fig. 75 Companion: Press action details

Repeat these steps with a second button (use value "false") to unmute "Mic 1 - Split 1 → Mute". On the left side of the web gui, use [Web Buttons] or [Emulator] to access your buttons:

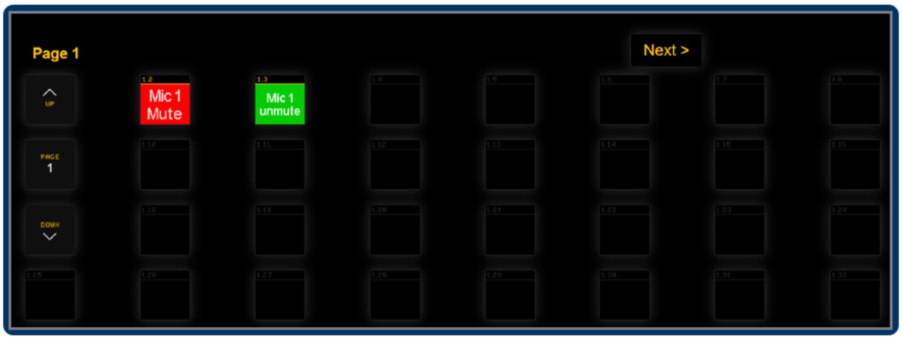


Fig. 76 Companion: Web buttons

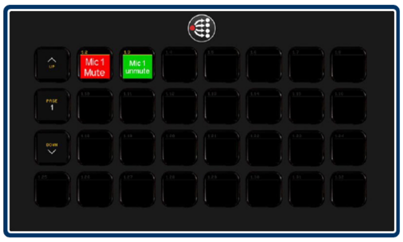


Fig. 77 Companion: Emulator

<u>Note:</u> sometimes it is necessary to disable and enable the Ember+ connections in Companion after intial connection.

4.6 // REDUNDANCY OPTIONS

NEXUS compact provides the following redundancy options:

<u>Power supply:</u>

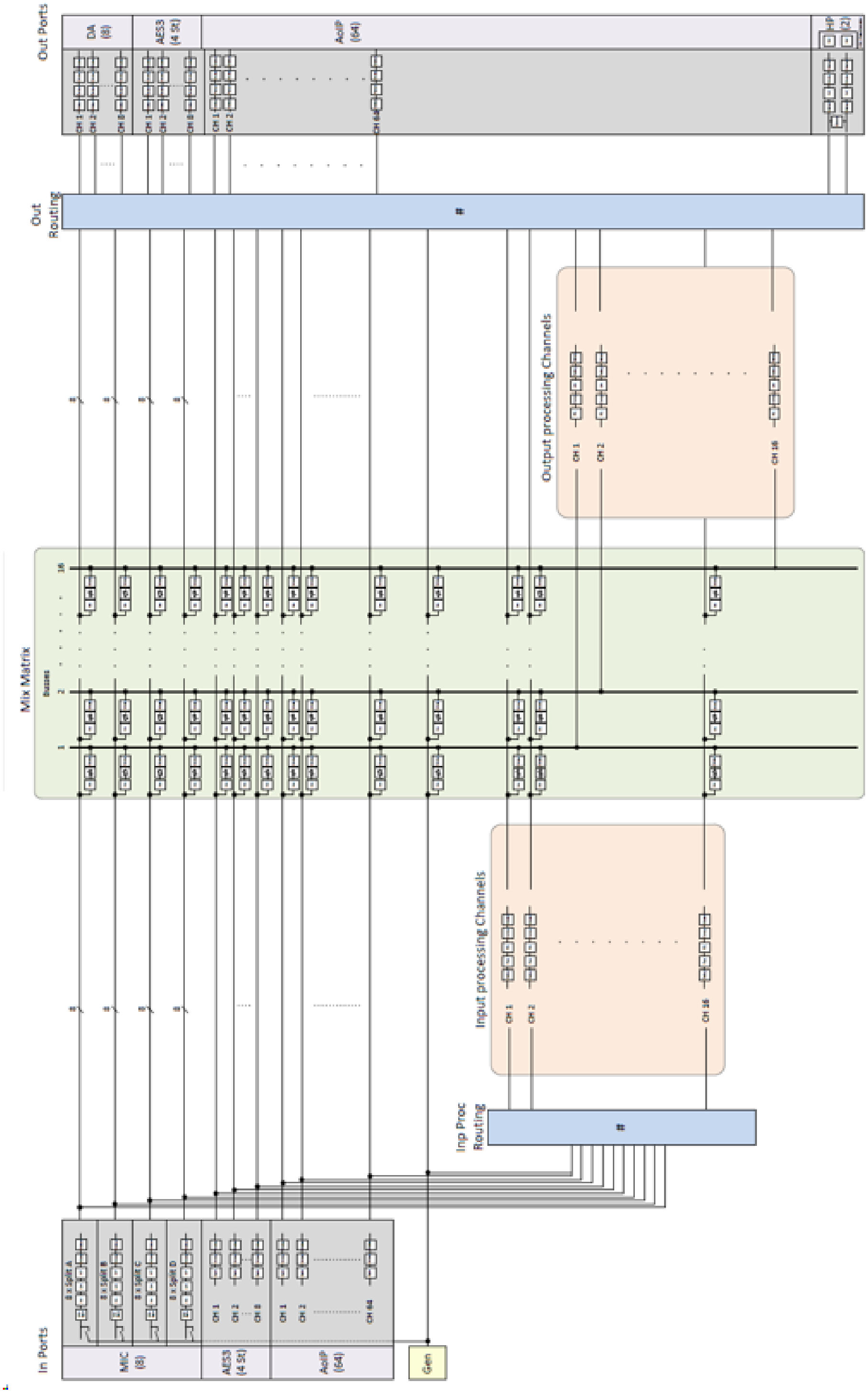
- By using PoE and the external PSU, redundant power supply can be achieved.

<u>AoIP:</u>

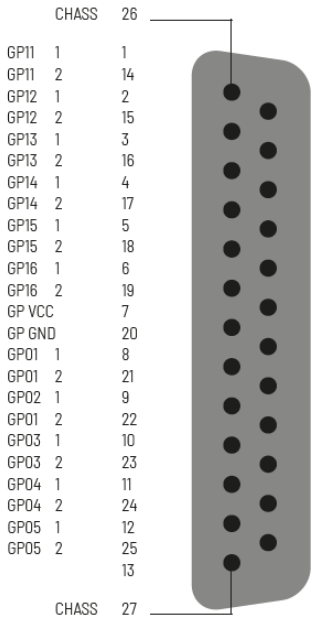
- Through the use of the primary and secondary AoIP port, network redundancy can be achieved.
- NEXUS compact supports SMPTE 2022-7 ("Seamless Protection Switching") Concept: A SMPTE 2022-7-enabled transmitter duplicates the input stream and sends it via two different paths to the destination receiver. The receiver (also SMPTE 2022-7 enabled) combines the streams from both paths and reconstructs the original stream. If a packet was lost on path 1, the packet is taken from path 2. In case path 1 is completely gone, the entire stream is taken from path 2 and vice versa.

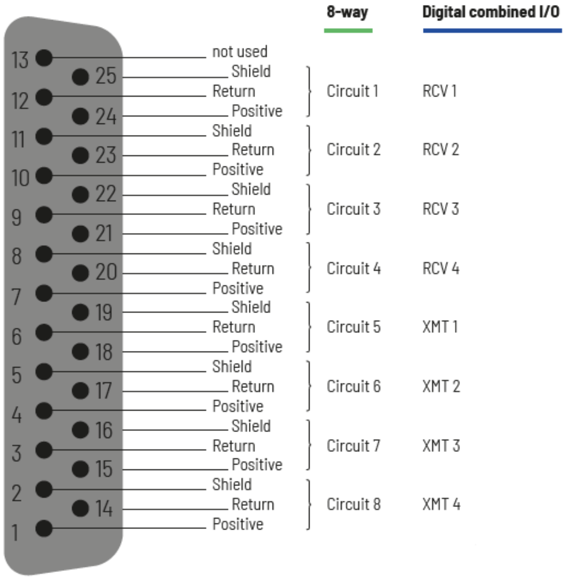
5 // APPENDIX

5.1 // BLOCKDIAGRAM



5.2 // SUB-D PINOUT SCHEMATICS





GP IN+OUT
D-Sub 25, f

MIC IN
AESS9, f,
8-way

DA OUT
AESS9, f,
8-way

AES3 IN+OUT
AESS9, f,
Digital combined

5.3 // TECHNICAL DATA

Microphone inputs	
Dynamic range	157.6 dB (A) Converter dynamic range >144 dB (RMS) resp. 147 dB (A) empty channel noise
Distortion factor (THD+N)	typ. 0.003 % at 24 dBu guaranteed < 0.004 % typ. 0.003 % in the range -50...0 dBu
Amplitude response	20...20.000 Hz: < 0,05 dB (typ. < ±0,1 dB) at 20 Hz: typ. -3 dB (below 18 dB/oct. drop according to IRT specifications)
Equivalent noise level in input	<- 129.5 dBu(A), at 200 Ohm source impedance <- 126 dBu(RMS), at 200 Ohm source impedance <- 133.6 dBu(A), at 0 Ohm source impedance <- - 115 dBqp CCIR1K, at 200 Ohm source impedance
Input level	max. 24 dBu balanced, with phantom power switched off, unbalanced sources also allowed
Signal delay	395 μs (at 48 kHz sample rate)
Crosstalk attenuation	140 dB (20...20.000 Hz) typ. > 170 dB at 1 kHz typ. > 150 dB at 20 kHz
Gain	digital up to 70 dB, continuously adjustable in 1 dB increments

Analog outputs	
D/A conversion	24 bit, 128-fold oversampling
Dynamic range	typ. 131 dB (A) typ. 128 dB RMS at 0 dBFS = 22 dBu
Output level	0...24 dBu at load > 600 Ohm (at 300 Ohm up to 15 dBu); adjustable in 1 dB steps
Distortion factor (THD+N)	typ. 0,003 % at 24 dBu typ. < 0.006 % in the range -20...+24 dBu, guaranteed < 0.02 % typ. < 0,0006 % at 4 dBu –68 dB bei -60 dBFS
Amplitude response	20...20,000 Hz (+0 dB, -0.2 dB), integrated DC filters
Output impedance	typ. 19 Ohm
Empty channel noise	typ. –124...–128 dBFS (RMS) typ. –93 dBqp (CCIR 1K) typ. –105 dBu (CCIR 2K RMS)
Unbalance attenuation (output impedance)	60 dB at 20...20.000 Hz typ. 120 dB at 50 Hz typ. 80 dB at 20 kHz
Offset voltage	< 1 mV, typ. 0,1 mV
Unbalance attenuation	40 dB at 20...20.000 Hz 90 dB at 50 Hz 50 dB at 20 kHz
Crosstalk attenuation	100 dB (20...20.000 Hz), typ. > 130 dB
Signal delay	< 230 μs (at 48 kHz sample rate)

Hardware	
Dimensions	222 mm x 40 mm x 222 mm (WxHxD)
Enclosure	Aluminum milled
Connectors	Mic In, Analog Out AES3 In/Out: each D-Sub 25 Tascam pinout GPIO: D-Sub 25 Network: 3- RJ45 (Control, AoIP Primary, AoIP Secondary)

5.4 // GLOSSARY

PTP	Precision Time Protocol - used to synchronize clocks throughout a network- defined in IEEE 1588-2008
QoS	Quality of Service - traffic prioritization and resource reservation control mechanisms. The ability to provide different priorities to different applications, users, or data flows, or to guarantee a certain level of performance to a data flow.
RTP	Real Time Transport Protocol - used for transmission of realtime data
RTCP	Real Time Control Protocol - controls quality of transmissionand negotiates QoS parameters
RTSP	Real Time Streaming Protocol - controls media streamingserver
SDP	Session Description Protocol- describes the configuration of a stream
Session (SDP)	Session describes the stream parameters (audio format, number of channels,...)
Unicast	point to point connection between sender and receiver
URL	Uniform Resource Locator- references to a resource on a network.
Zeroconf	assignment of numeric network addresses for networkeddevices, automatic distribution and resolution of computerhostnames, and automatic location of network services

TABLE OF FIGURES

Fig. 1 TrueMatch converter 8
Fig. 2 NEXUS compact - front panel.... 9
Fig. 3 NEXUS compact - rear panel.... 10
Fig. 4 Web GUI 11
Fig. 5 Dashboard page.... 13
Fig. 6 Mic split panel.... 14
Fig. 7 Digital inputs panel 15
Fig. 8 Processing panel 16
Fig. 9 Outputs panel 17
Fig. 10 MIC IN page.... 18
Fig. 11 INPUTS page 20
Fig. 12 OUTPUTS page 21
Fig. 13 I/O-assignment dialog....22
Fig. 14 PROC/MIX page....23
Fig. 15 Mix bus view 24
Fig. 16 GUI in "copy"-mode....26
Fig. 17 GUI in "paste"-mode 27
Fig. 18 MIX SEND view....28
Fig. 19 Detail view page....29
Fig. 20 Equalizer Detail View 30
Fig. 21 Compressor Detail View 31
Fig. 22 Expander Detail View 32
Fig. 23 Limiter Detail View 33
Fig. 24 Delay Detail View 33
Fig. 25 XY ROUTING page 34
Fig. 26 SYSTEM page / general 35
Fig. 27 Network panel 36
Fig. 28 Audio panel....37
Fig. 29 System panel....37
Fig. 30 Power On State panel....38
Fig. 31 Generator panel....39
Fig. 32 GPIO panel.... 40
Fig. 33 Snapshots page 41
Fig. 34 Headphone panel....42
Fig. 35 Headphone settings dialog....42
Fig. 36 Snapshot menu bar....43
Fig. 37 RAV2SAP SCREEN SHOT....49
Fig. 38 RVSC screen shot: Tx Stream Properties (payload format selection) 52
Fig. 39 RVSC screen shot: Tx Stream Properties (audio channel assignment) 52
Fig. 40 RVSC screen shot: Overview with one TX stream created 53
Fig. 41 RVSC screen shot: TX stream extended properties....53
Fig. 42 RVSC screen shot: TX stream SDP link 54
Fig. 43 Accessing SDP data with browser (by HTTP) 54
Fig. 44 Opening SDP data with editor....54
Fig. 45 RVSC screen shot: RX stream properties (stream source selection) 55
Fig. 46 RVSC screen shot: RX stream properties (stream parameters filled in)....56
Fig. 47 RVSC screen shot: RX stream properties (channel assignment) 57

Fig. 48 RVSC screen shot: overview with one RX stream connected 57
Fig. 49 RVSC SCREEN SHOT: statistics with packet jitter and receiver buffer utilization........ 57
Fig. 50 RVSC screen shot: RX stream properties with pasted SDP data 58
Fig. 51 RVSC screen shot: overview with 2 RX streams connected 58
Fig. 52 Mirror control port 59
Fig. 53 PTP Master or slave status 60
Fig. 54 PTP-Mode........ 60
Fig. 55 PTP extern........ 61
Fig. 56 PTP Profile and Clock Settings 61
Fig. 57 Clock domain........ 61
Fig. 58 Dante controller - unit as PTP translator between PTPv1 and v2 61
Fig. 59 DSCP values PTP and RTP packets AES67........ 62
Fig. 60 DSCP values Dante AES67 62
Fig. 61 Dante Controller AES67 Config 63
Fig. 62 Output Stream Settings Dante compatible AES67 stream 64
Fig. 63 Input Stream Settings - Stream Discovery 64
Fig. 64 Input Stream Settings – RTP Error 65
Fig. 65 Input Stream Settings - no warning 65
Fig. 66 Ember+ viewer........ 66
Fig. 67 Ember+ viewer: establishing a conntection 66
Fig. 68 Ember+ viewer: connection established........ 66
Fig. 69 Ember+ viewer: parameter selected........ 67
Fig. 70 Companion: Startup dialog........ 68
Fig. 71 Companion: Adding generic Ember+ 68
Fig. 72 Companion: Connection parameters........ 68
Fig. 73 Companion: Button parameters 69
Fig. 74 Companion: adding a press action........ 69
Fig. 75 Companion: Press action details 69
Fig. 76 Companion: Web buttons 70
Fig. 77 Companion: Emulator 70

STAGETEC GmbH

Tabbertstraße 10-11 | 12459 Berlin | Germany
T. +49 (0) 30 / 63 99 02 0
sales@stagetec.com
www.stagetec.com