How to establish an AES67 compliant AoIP stream with MONTONE.42 and RAV.IO

Version 1.0
March 2020

DirectOut GmbH

Leipziger Strasse 32
09648 Mittweida
Germany

T: +49-3727-5665-100
F: +49-3727-5665-101
www.directout.eu

Table of Contents

Introduction

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.

For information on how to establish an ST2110-30 or -31 compliant AoIP stream please refer to this guide.

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.

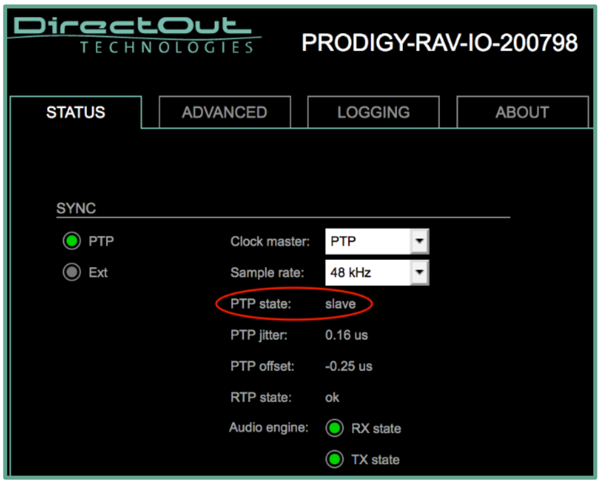

Figure 1: PTP Master or slave status

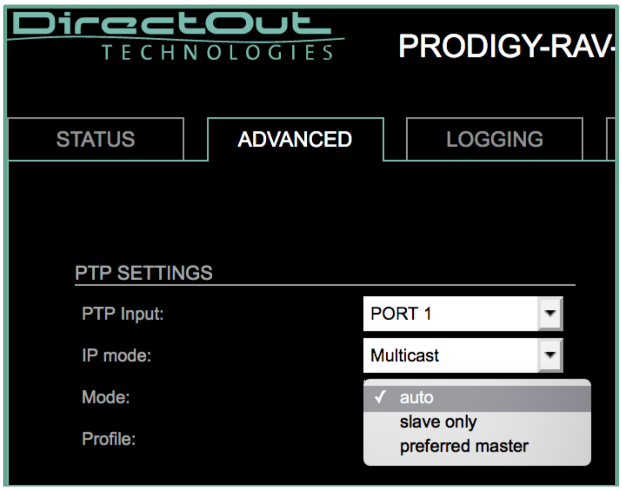

Figure 2: PTP-Mode

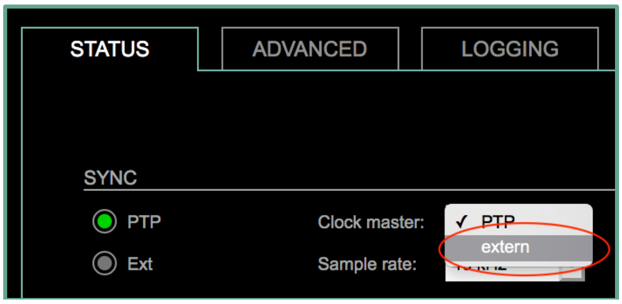

Figure 3: PTP extern

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".

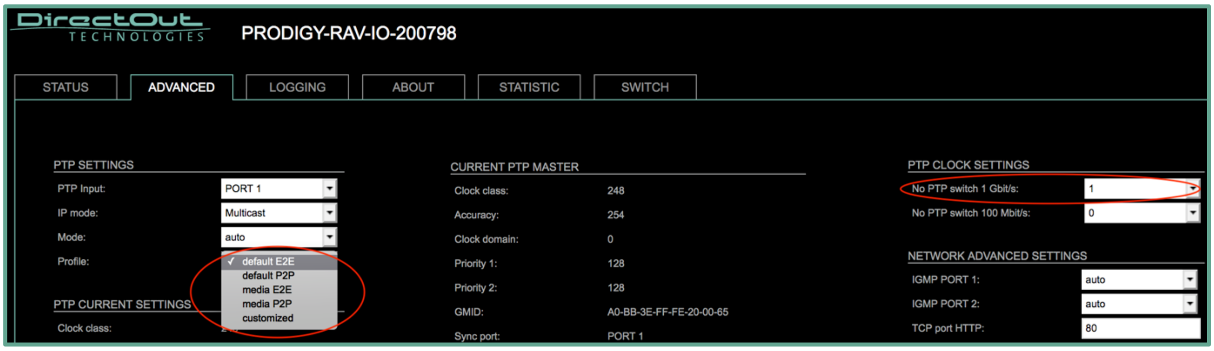

Figure 4: PTP Profile and Clock Settings

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

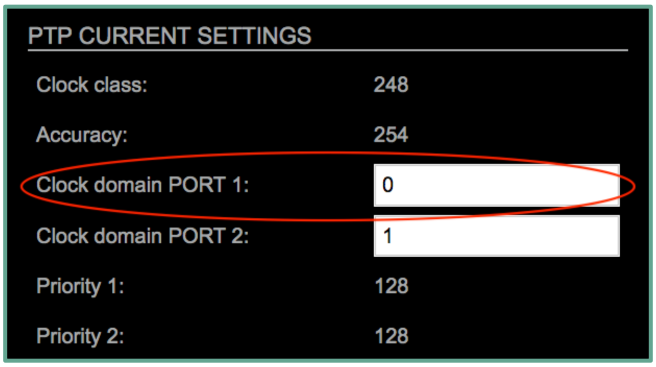

Figure 5: Clock domain

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.

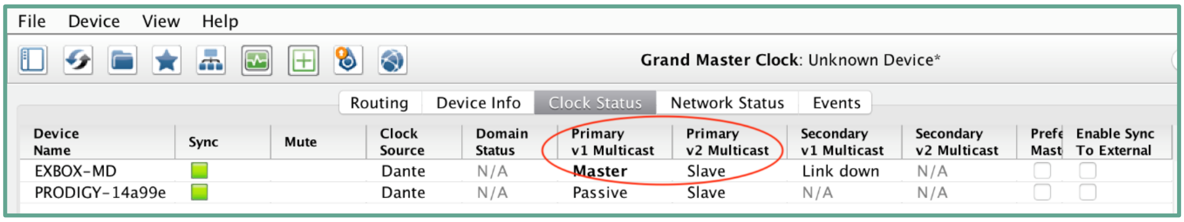

Figure 6: Dante controller - unit as PTP translator between PTPv1 and v2

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.

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

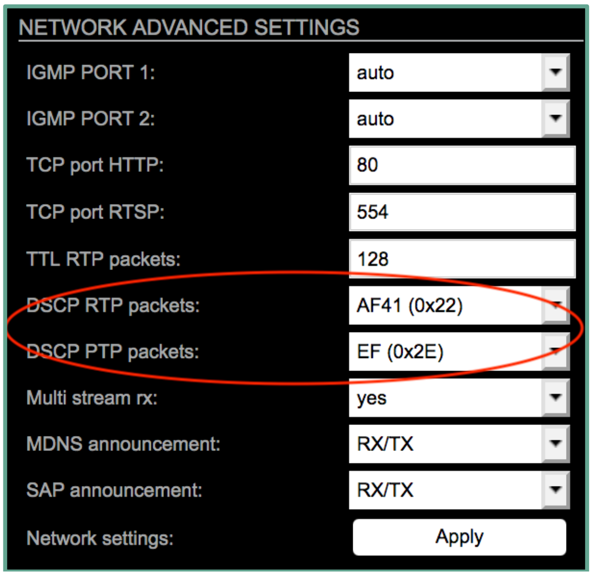

Figure 7: DSCP values PTP and RTP packets AES67

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.

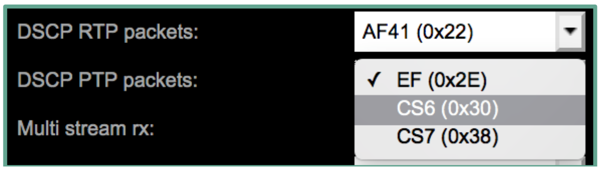

Figure 8: DSCP values Dante AES67

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.

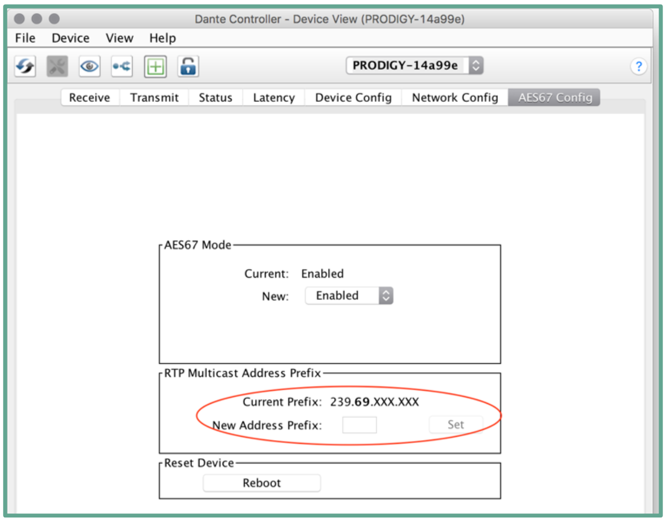

Figure 9: Dante Controller AES67 Config

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 per-stream 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.

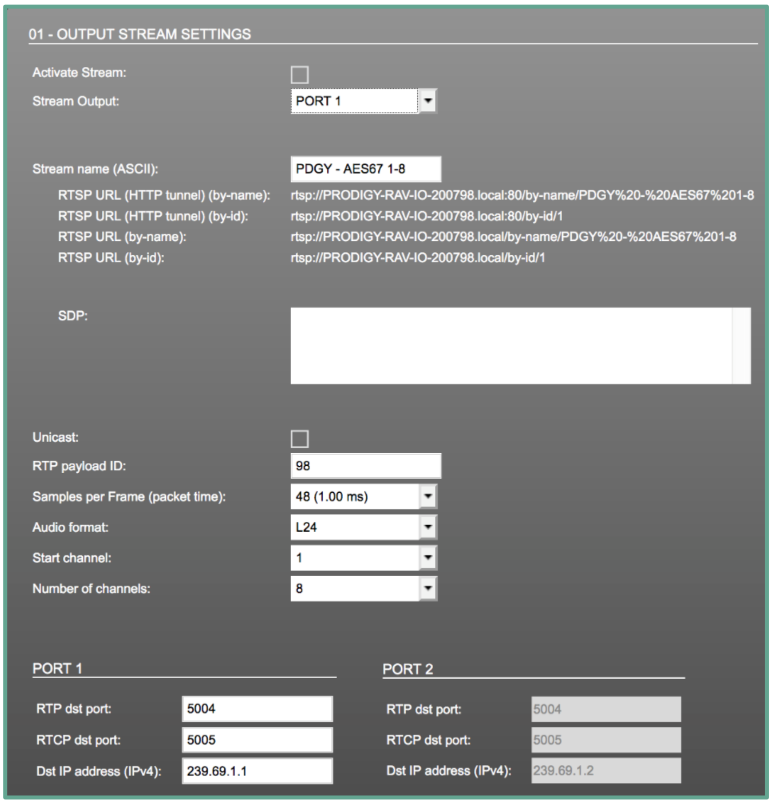

Figure 10: Output Stream Settings Dante compatible AES67 stream

Activate the stream and continue at the receiving end.

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.

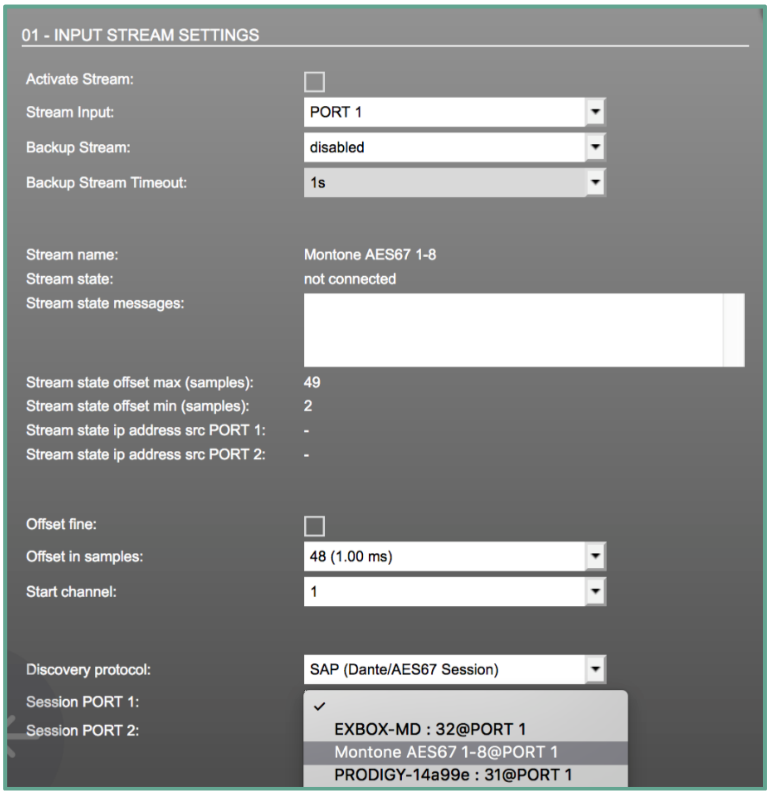

Figure 11: Input Stream Settings - Stream Discovery

Stream Delay/Offset

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.

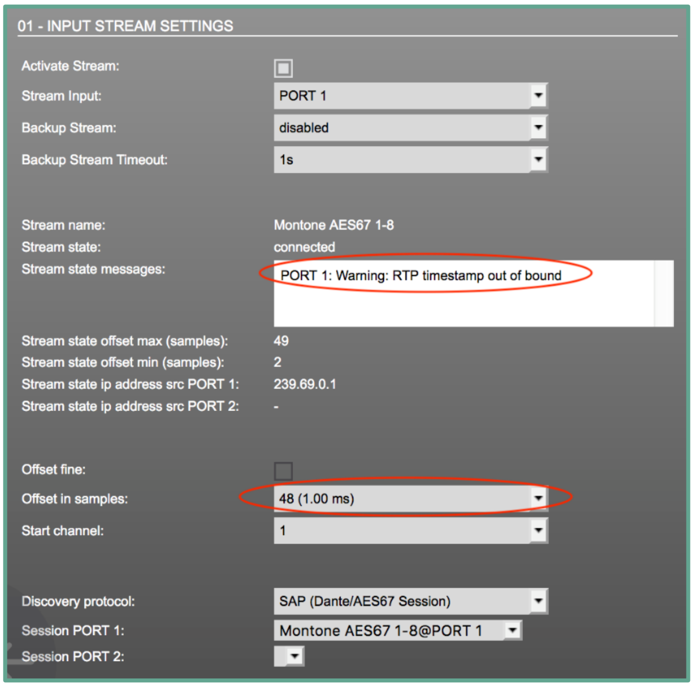

Figure 12: Input Stream Settings – RTP Error

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

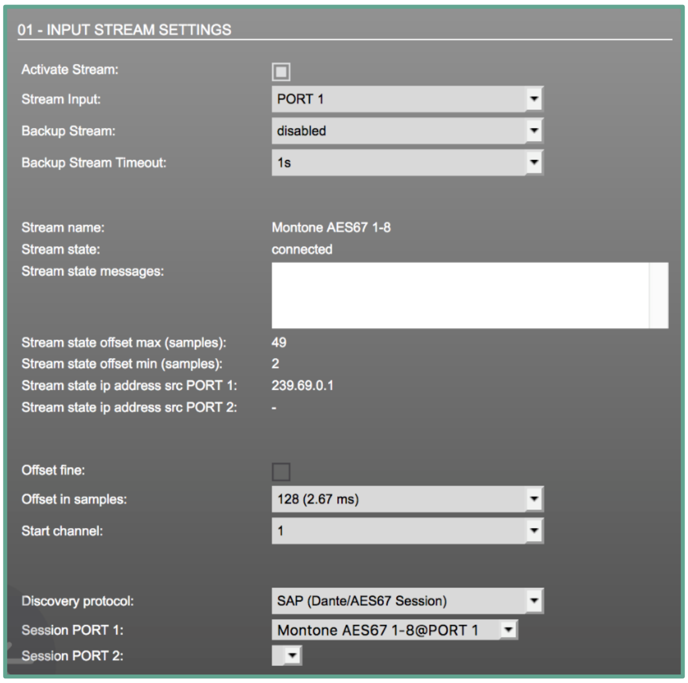

Figure 13: Input Stream Settings - no warning

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 (part of PRODIGY.MC or PRODIGY.MP manual).

Glossary

Audio Format	Payload format of audio data - also known as 'encoding'
Bonjour	Apple ´s implementation of zeroconf.
DiffServ	Differentiated Services- mechanism for classifying and managing network traffic, prioritization of services (e.g. low-latency traffic)
DSCP	The differentiated services code point (DSCP) is a 6-bit field in the IP packet header that is used for classification purposes. DSCP is part of the differentiated services architecture.
IGMP	Internet Group Management Protocol (IGMP) is a communications protocol used by hosts to report their multicast group memberships to IPv4 routers.
IP	Internet Protocol - used to build logical units (subnets) in a network
HTTP	Hyper Text Transfer Protocol - data transmission for application layer, e.g. websites
Latency	delay introduced by packetizing or buffering - number of samples per frame divided by sample rate - also known as 'frame size'
mDNS	Multicast DNS - resolves host names to IP addresses, part of zeroconf
Multicast	one sender to many receivers
Packet	formatted unit of data - consists of control information and user data (payload)
Packet Time	The real-time duration of the media data contained in a media packet. For example, a packet containing 24 samples of 48 kHz audio has a packet time of 24 ÷ 48 kHz = 500 microseconds. Short packet times allow for lower latency but introduce overhead and high packet rates that may overtax some devices or networks. Long packet times imply higher latency and require additional buffering which may not be available on memory - constrained devices.

PTP	Precision Time Protocol - used to synchronize clocks throughout a network- defined in IEEE 1588-2008
QoS	Quality of Service - overall performance of a network
RTP	Real Time Transport Protocol - used for transmission of real time data
RTCP	Real Time Control Protocol - controls quality of transmission and negotiates QoS parameters
RTSP	Real Time Streaming Protocol - controls media streaming server
SDP	Session Description Protocol- describes the configuration of a stream
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 networked devices, automatic distribution and resolution of computer hostnames, and automatic location of network services

List of figures