Open Standards in Pro Audio: AES67

Conrad Bebbington
Focusrite

Pro Audio

Studio

Live Sound

Theatre

Broadcast

House of Worship

Audio Network Devices

Microphones

Preamplifiers

Mixers

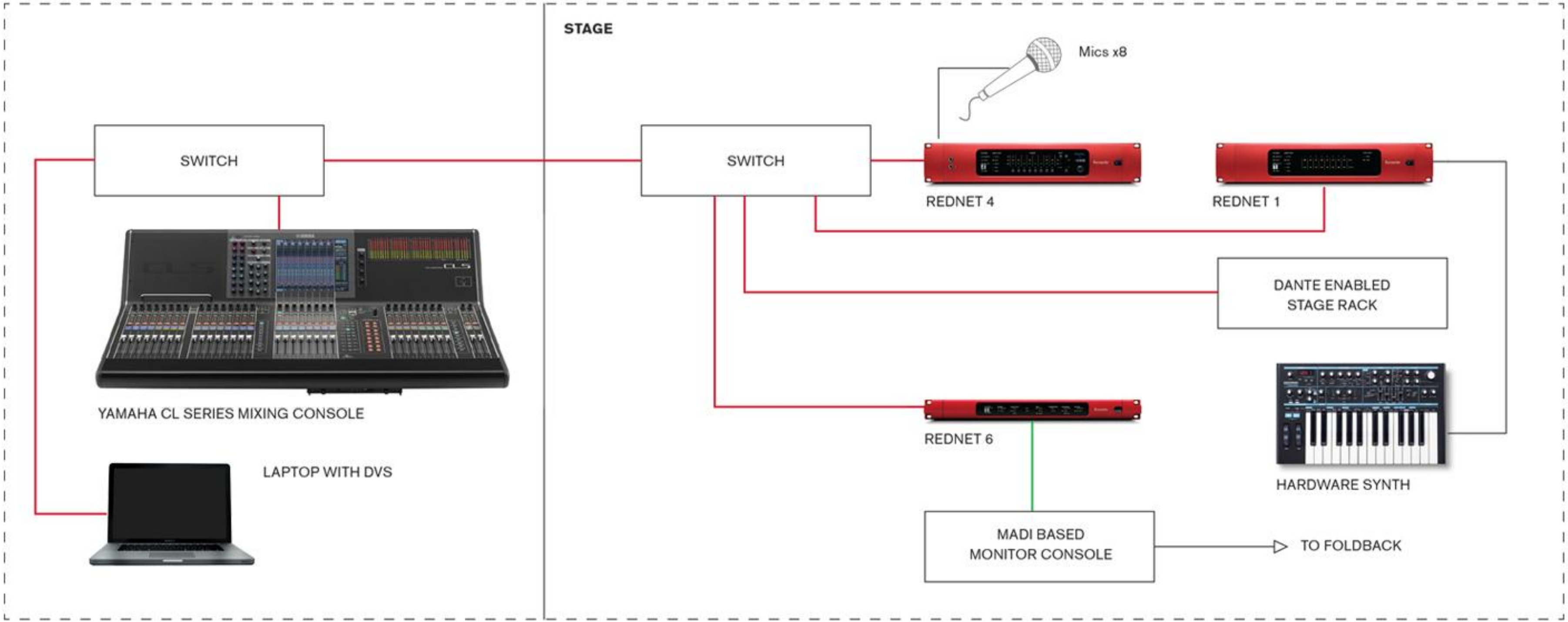
Effects

Interfaces

Amplifiers

Speakers

LIVE SOUND



CABLE KEY

- CAT5e or CAT6
- MADI
- ANALOGUE

Requirements

Local area networks only

Low latency

High sample rate and bit depth

Many channels

No lossy compression

Existing audio networking technologies

Dante

RAVENNA

Livewire

Q-LAN

WheatNet

	Dante	RAVENNA	Q-LAN	Livewire
Control	Proprietary	HTTP, Ember+	TCP, HTTP	HTTP, Proprietary
Discovery	Proprietary	Bonjour	Proprietary	Proprietary
Connection Management	Proprietary	RTSP, SIP, IGMP	Proprietary	Proprietary, HTTP, IGMP
Session Description	Proprietary	SDP	Proprietary	Channel #
Transport	Proprietary, IPv4	RTP, IPv4	RTP, IPv4	RTP, IPv4
QoS	DiffServ	DiffServ	DiffServ	DiffServ/802.1pq
Encoding	L16-L32	L16-L32	Floating Point	L24 stereo/surround
Synchronization	PTP1588-2002	PTP1588-2008	PTP1588-2008	Proprietary
Sample Rates	44.1kHz - 192kHz	44.1kHz - 384kHz	48kHz	48kHz

Why standardize?

Poor interop limits adoption

Future proofing

Broadcast industry - external feeds and outside broadcasts

Image: Richard Yeowart / Wikipedia (Creative Commons Attribution-Share Alike 3.0 Unported)

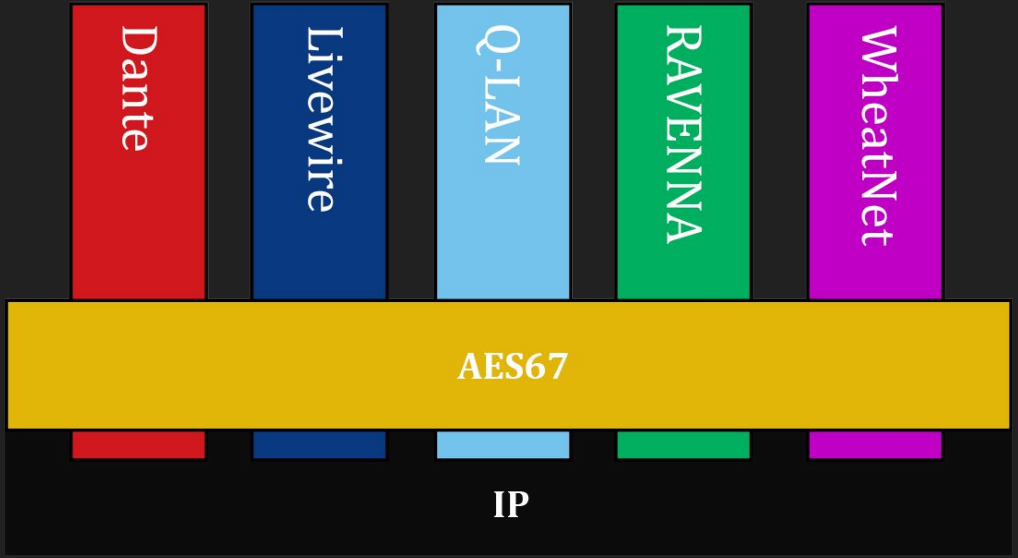
Scope of AES67

Bridge between existing systems

Specifies encoding, transport, synchronization, session description, connection management

Excludes discovery and control

Uses existing networking standards



Dante
Livewire
Q-LAN
RAVENNA
WheatNet
AES67
IP

Overview

Audio format: PCM

Packetization: RTP

Synchronization: Precision Time Protocol

Session description: SDP

Connection Management: SIP (unicast), IGMP (multicast)

Audio Format

All audio is linear PCM

Required formats: 16bit 48kHz, 24bit 48kHz

Optional formats: 16bit 44.1kHz, 24bit 96kHz

Other formats are permitted

Packetization

Audio is transmitted as RTP

No CSRCs or header extensions

Up to 8 channels per stream

Allowed packet times are 125µs to 4ms

Optional multicast

Synchronization

Sample clocks in the network must have the same rate

Precision Time Protocol (IEEE 1588-2008) defines a network clock

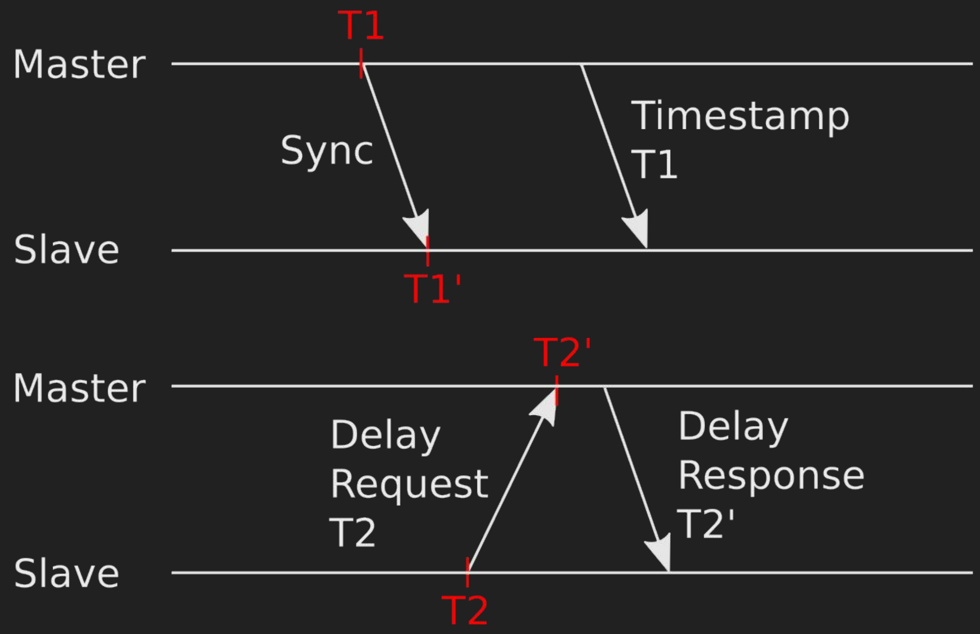
A master clock is selected

Master periodically broadcasts sync

Slaves periodically contact master to measure network delay

Sample clock is derived from network clock

Synchronization



Slave offset = (T1’ - T1 - T2 + T2’) / 2

Session Description

Mostly standard SDP

Format `L16` or `L24`

`ptime` and `maxptime` attributes for packet time

`ts-refclk` attribute for network clock source

`mediaclk` attribute for media clock

Payload mapping using `rtpmap`

Example SDP

```
v=0
o=audio 1311738121 1311738121 IN IP4 192.168.1.1
s=Stage left I/O
c=IN IP4 192.168.1.1
t=0 0
m=audio 5004 RTP/AVP 96
i=Channels 1-8
a=rtpmap:96 L24/48000/8
a=sendonly
a=ptime:0.250
a=ts-refclk:ptp=IEEE1588-2008:39-A7-94-FF-FE-07-CB-D0:domain-nmbr=0
a=mediaclk:direct=2216659908
```

Example SDP

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```

8 channels of 24bit/48kHz PCM

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```

250μs packet time

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```

PTP clock domain 0

Example SDP

```
v=0
o=audio 1311738121 1311738121 IN IP4 192.168.1.1
s=Stage left I/O
c=IN IP4 192.168.1.1
t=0 0
m=audio 5004 RTP/AVP 96
i=Channels 1-8
a=rtpmap:96 L24/48000/8
a=sendonly
a=ptime:0.250
a=ts-refclk:ptp=IEEE1588-2008:39-A7-94-FF-FE-07-CB-D0:domain-nmbr=0
a=mediaclk:direct=2216659908
```

Media clock offset from PTP

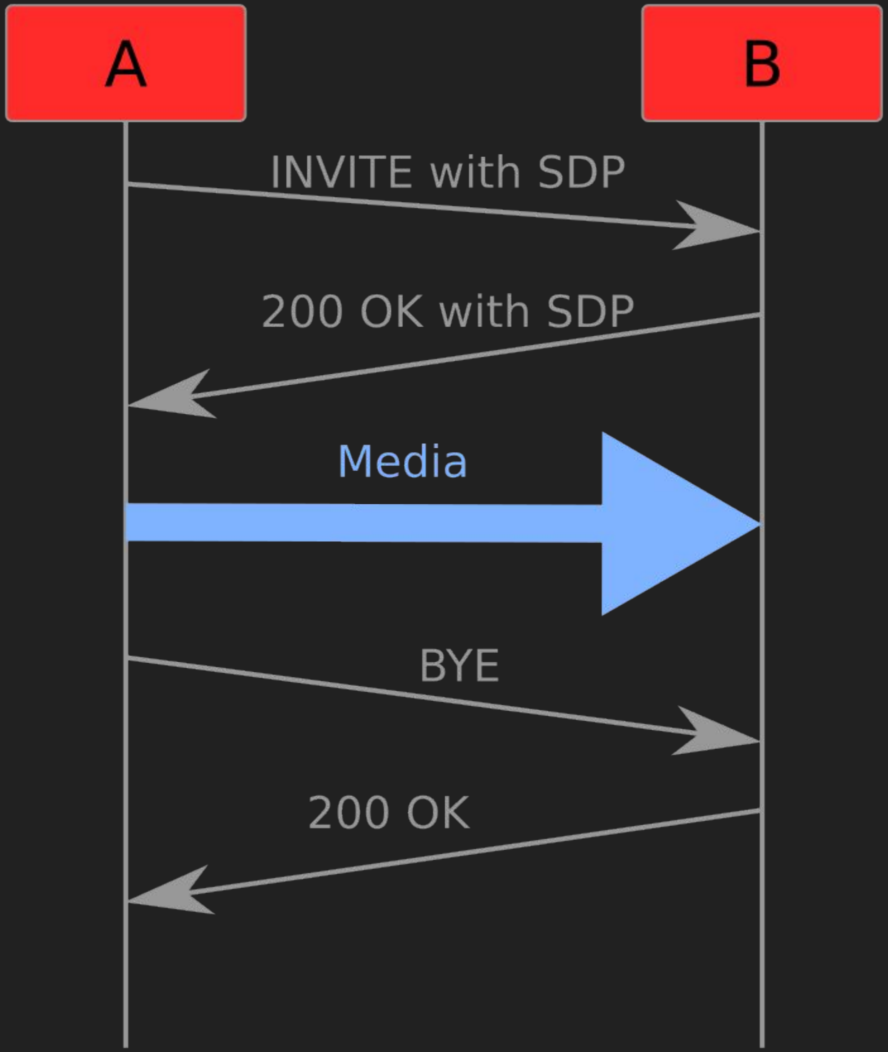
Connection Management

Unicast connections use SIP

Originally designed for VoIP and conferencing

Uses URIs: username@hostname

Direct connections are preferred (serverless mode)


A
B
INVITE with SDP
200 OK with SDP
Media
BYE
200 OK

Connection Management

Multicast uses IGMP

No direct communication between sender and receiver

Senders transmits on multicast IP

Receivers indicate desired streams to network routers

Organisations

AES handles standardisation and technical discussion

Media Networking Alliance promotes adoption and handles discussion of practical implementation

Media Networking Alliance Members

Archwave Technologies, B.V.

Bosch Security Systems, Inc.

Focusrite Audio Engineering

Harman Professional

Lawo Ag

QSC, LLC

The Telos Alliance

Yamaha

Plus 25 Associate Members

Conclusions

Standardisation improves interop

Adoption of general purpose technology for audio use case

More interaction between pro-audio and tech industries

More Information

http://medianetworkingalliance.com/

http://www.aes.org/publications/standards