Open Standards in Pro Audio: AES67

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Focusrite
Pro Audio

Studio
Live Sound
Theatre
Broadcast
House of Worship
Audio Network Devices

- Microphones
- Preamplifiers
- Mixers
- Effects
- Interfaces
- Amplifiers
- Speakers
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
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Why standardize?

Poor interop limits adoption

Future proofing

Broadcast industry - external feeds and outside broadcasts
Scope of AES67

Bridge between existing systems

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

Excludes discovery and control

Uses existing networking standards
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.


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 = \( \frac{(T1' - T1 - T2 + T2')}{2} \)
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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a=mediaclick: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)
Connection Management

Multicast uses IGMP

No direct communication between sender and receiver

Senders transmit 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.
Focusrite Audio Engineering
Lawo Ag
The Telos Alliance

Bosch Security Systems, Inc.
Harman Professional
QSC, LLC
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