STANDARDS AND INFORMATION DOCUMENTS

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AES standard for audio applications of networks - High-performance streaming audio-over-IP interoperability

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AES standard for audio applications of networks - High-performance streaming audio-over-IP interoperability

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Abstract

High-performance media networks support professional quality audio (16 bit, 44,1 kHz and higher) with low latencies (less than 10 milliseconds) compatible with live sound reinforcement. The level of network performance required to meet these requirements is available on local-area networks and is achievable on enterprise-scale networks. A number of networked audio systems have been developed to support high-performance media networking but until now there were no recommendations for operating these systems in an interoperable manner. This standard provides comprehensive interoperability recommendations in the areas of synchronization, media clock identification, network transport, encoding and streaming, session description and connection management.

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Foreword

This foreword is not part of the AES67-2013 *AES standard for audio applications of networks - High-performance streaming audio-over-IP interoperability*

This document was developed in project AES-X192, in the SC-02-12-H task group on high-performance streaming audio-over-IP interoperability, under the leadership of Kevin Gross.

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Foreword to second edition, 2015

This revision includes minor changes identified during 'plugfest' testing in October 2014 and was developed in task group SC-02-12-M. It includes updated references to RFC 7273, and clarifications in 6.3, 8.1, and 8.5.

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Chair, working group SC-02-12, 2015-07-27

Foreword to third edition, 2018

This revision contains clarifications and minor corrections and adds a Protocol Implementation Conformance Statement (PICS) as Annex G. A new sender keep-alive recommendation, has been added to clause 6.3. Minor clarifications and corrections include a specification of MTU requirements in the presence of allowed (but not recommended) additional information in the RTP header and correcting SDP examples in clause 8.5 to match an erratum issued by the IETF on RFC 7273. Corrected domainNumber range to conform to IEEE 1588-2008 requirements. These revisions were developed in task group SC-02-12-M.

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Note on normative language

In AES standards documents, sentences containing the word “shall” are requirements for compliance with the document. Sentences containing the verb “should” are strong suggestions (recommendations). Sentences giving permission use the verb “may”. Sentences expressing a possibility use the verb “can”.

AES standard for audio applications of networks - High-performance streaming audio-over-IP interoperability

0 Introduction

0.1 General

High-performance media networks support professional quality audio (16 bit, 44,1 kHz and higher) with low latencies (less than 10 ms) compatible with live sound reinforcement. The level of network performance required to meet these requirements is available on local-area networks and is achievable on enterprise-scale networks but is generally not available on wide-area networks or the public internet.

The most recent generation of these media networks use a diversity of proprietary and standard protocols. Despite a common basis in Internet Protocol, the systems do not interoperate.

This standard provides specific recommendations for interoperability. The standard focuses on defining how existing protocols are used to create an interoperable system. No new protocols have been developed to achieve this.

The standard is expected to be useful for commercial audio applications including fixed and touring live sound reinforcement. It is also expected to be useful for distribution within broadcast, music production and post-production facilities.

This standard depends on established network protocols (see clause 2). These protocols may include additional options that are not required by this standard. Implementations of AES67 should tolerate these additional options.

0.2 Patents

The Audio Engineering Society draws attention to the fact that it is claimed that compliance with this AES standard or information document may involve the use of a patent.

The AES holds no position concerning the evidence, validity and scope of this patent right.

The holder of this patent right has assured the AES that it is willing to negotiate licenses under reasonable and non-discriminatory terms and conditions with applicants throughout the world. In this respect, the statement of the holder of this patent right is archived with the AES and listed on the public AES web site.

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Attention is drawn to the possibility that some of the elements of this AES standard or information document may be the subject of patent rights other than those identified above. AES shall not be held responsible for identifying any or all such patent rights.

1 Scope

This standard defines an interoperability mode for transport of high-performance audio over networks based on the Internet Protocol. For the purposes of the standard, high-performance audio refers to audio with full bandwidth and low noise. These requirements imply linear PCM coding with a sampling frequency of 44,1 kHz and higher and resolution of 16 bits and higher. High performance also implies a low-latency capability compatible with live sound applications. The standard considers latency performance of 10 milliseconds or less.

2 Normative references

The following referenced documents are indispensable for the application of this document. For dated references, only the edition cited applies. For undated references, the latest edition of the referenced document (including any amendments) applies.

AES11 - *AES recommended practice for digital audio engineering - Synchronization of digital audio equipment in studio operations*; Audio Engineering Society, New York, NY., US.

IEEE 1588-2008 - *IEEE Standard for a Precision Clock Synchronization Protocol for Networked Measurement and Control Systems*, July 2008, Institute of Electrical and Electronics Engineers (IEEE), US.

RFC 768 – *User Datagram Protocol*", Internet Engineering Task Force

RFC 791 – *Internet Protocol*, Internet Engineering Task Force

RFC 1112 – *Host Extensions for IP Multicasting*, Internet Engineering Task Force

RFC 2236 - *Internet Group Management Protocol, Version 2*, Internet Engineering Task Force

RFC 2474 – *Definition of the Differentiated Services Field (DS Field) in the IPv4 and IPv6 Headers*, Internet Engineering Task Force

RFC 2616 - *Hypertext Transfer Protocol - HTTP/1.1*, Internet Engineering Task Force

RFC 2974 – *Session Announcement Protocol*, Internet Engineering Task Force

RFC 3190 – *RTP Payload Format for 12-bit DAT Audio and 20- and 24-bit Linear Sampled Audio*, Internet Engineering Task Force

RFC 3261 - *SIP: Session Initiation Protocol*, Internet Engineering Task Force

RFC 3264 - *An Offer/Answer Model with the Session Description Protocol (SDP)*, Internet Engineering Task Force

RFC 3376 - *Internet Group Management Protocol, Version 3*, Internet Engineering Task Force

RFC 3550 – *RTP: A Transport Protocol for Real-Time Applications*, Internet Engineering Task Force

RFC 3551 - *RTP Profile for Audio and Video Conferences with Minimal Control*, Internet Engineering Task Force

RFC 4566 – *Session Description Protocol*, Internet Engineering Task Force

RFC 5939 – *Session Description Protocol (SDP) Capability Negotiation*, Internet Engineering Task Force

RFC 7273 – *RTP Clock Source Signalling*, Internet Engineering Task Force