# STANDARDS AND INFORMATION DOCUMENTS

**STANDARDS**

**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

**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 needed to meet these requirements is typically available on wired 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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# Contents

## 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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This document was edited by Kevin Gross.

Richard Foss
Chair, working group SC-02-12, 2013-07-18

### 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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The task group was led by Kevin Gross.

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

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## 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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The task group was led by Kevin Gross.

The PICS (annex G) was edited by Gints Linis

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Chair, working group SC-02-12, 2017-12-11

**Foreword to fourth edition, 2023**

This revision contains multiple updates and corrections, categorized as indicated later in this foreword.

Authors of this revision have intentionally minimized the amount of text changes in the body of the standard. Those engaged in implementation of the standard are encouraged to also consult the PICS (Annex G) as this presents testable behavioral requirements that, in many cases, provide clarification of the normative statements found in the body of the standard.

The changes have resulted in deprecating or moving content of existing PICS statements, as well as inserting one new PICS section and a number of new PICS statements.

The existing PICS statements have retained their statement numbers within sections. New statements received new, formerly non-existing statement numbers. Where the contents of a PICS statement were moved between sections, the previous statement number is marked as deprecated, and a new statement number is created in the new position.

Some of the PICS sections have been renumbered.

Detailed list of changes by categories follows.

**Changes directly affecting interoperability:**

- 4.0, 8.2: Defined handling of PTP version IEEE 1588-2019. This is a new requirement added in this revision of the standard.

- 5: Added a discussion of using zero RTP offset to achieve interoperability with ST 2110-30:2017.

- 6.1.3 Multicast address range: Previous versions of this standard required use of the administratively scoped multicast address range, which puts all other multicast addresses outside the scope of this standard. It narrowed usability of AES67 for SSM applications and limited interoperability with ST 2110. To resolve this issue, the requirement to use the administratively scoped multicast addresses is replaced with a requirement to support them. Thus, using other addresses, although not necessarily supported by all conformant devices, becomes a valid AES67 application. Informative details are provided about usage of multicast addresses under ST 2110.

- 6.2: In previous versions, the requirement for senders to use the default DSCP settings in absence of a management interface was not required by the language of the standard. This has been clarified as a requirement.

- 6.3: Defined an exact range of transport port numbers to be supported by senders and receivers. This requirement was largely undefined before, only talking about "other or additional ports", without any further details.

- 6.3, 7.2.0, G.3.4.3 (PICS): Clarified RFC 3551 requirements – explicit overrides are defined for silence suppression, channel numbering, ordering, content mapping, and packetization requirements. In particular, a new explicit requirement is added in this revision of the standard, which disallows use of silence suppression by senders.

- 7., Annex G: Reworked streaming interoperability criteria. The criteria specification method is changed from individual stream attributes to attribute vectors called *stream modes*. The concept of stream mode is introduced in 7.0, but most of the essential changes are applied to Annex G - PICS. Section 7 has been left mostly unchanged in this respect – it still provides a general description. Annex G specifies conformance criteria.

  As a result, a number of the previously existing PICS entries are deprecated, and new entries are created in G.3.5, to follow the new qualification criteria as defined in G.4. An instruction is provided in G.2 for documenting stream mode capability declarations in the PICS proforma.

These changes are aimed at resolving ambiguities and filling requirement gaps in previous versions of the standard. They are not aimed at changing existing interoperability requirements. The new specification method can, however, reveal possible misinterpretations of this standard's intentions in existing implementations.

- 7.5: Interoperability exception explained – when a sender of the lower timing accuracy class is connected with a receiver implementing only the minimally required buffering capacity, reliable reception of all streams cannot be guaranteed. An interoperability matrix summarizing the issue is provided.

- 8.2: Added guidance for handling of traceable time references. This option is defined in RFC 7273, but was not addressed in previous versions of AES67. ST 2110-30:2017 requires that traceable clock references are labeled so, therefore lack of this mechanism in AES67 devices can create a potential for interoperability problems. Changes applied:

  o Added an example of SDP signaling of the "traceable" property of a clock reference.

  o Stream connection setup rules are updated to reflect cases involving traceable references.

- 8.3: Added a discussion of an ST 2110-30:2017 requirement to specify mediaclk always at the media level of SDP. Not doing so could affect interoperability with ST 2110-30:2017.

- 10.1: Relaxed the strong requirement ("shall") to support SIP for unicast connections – changed to a recommendation ("should").

- 10.2 (was in 6.1): IGMPv3 support level elevated from "may" to "should"; behavior related to IGMPv3 is defined.

- 10.2 (was in 6.1): removed requirement for senders to use IGMP to request their own streams. See the new Annex F for a discussion of the motivation for this change.

**Normative language improvements:**

These editions include clarification of ambiguous or implied requirements, filling omissions, correcting unclear or misused terms or normative expressions. No changes of requirements are intended by this category of editions. In particular:

- Corrected improper or inconsistent use of media streaming terms, to achieve better alignment with the referenced RFC documents. Specifically:

  o The term "payload format" was sometimes used interchangeably with "encoding format" and "payload type". Use of these terms is corrected in this version.

  o The terms "sampling rate", "sampling frequency, and "sample frequency" are interchangeable, and they all have been used variously throughout the standard. Those cases are brought to uniformity now, using a single term "sampling rate".

- 0.1: Corrected improper use of normative language – verb "should" was used to express a generally valid but practically untestable goal. The "should" is replaced to clarify this sentence is informative.

- 4.0 The previous revisions of this standard used a term "common clock", which is inconsistent with PTP, where common time is shared between devices. "Common time" terminology is used in this revision.

- 4.1, 4.2: "IP network" and "1588 network" terms are replaced with "ordinary IP network" and "IP network with IEEE 1588-2008" respectively; "standard IP network" replaced with "ordinary IP network"; "use" (a user's act) replaced with "support" (device's property); "achieved" replaced with "achievable".

- 6.1.1: Added clause clarifying that senders are allowed to choose whether they support only multicast, only unicast, or both.

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- 6.1.2: Corrected the terms describing ICMP signaling related to packet fragmentation and MTU size.

- 6.2 Table1: Clarified mapping of IEEE1588-2008 traffic types to QoS classes.

- 6.2 Corrected an unclear expression "shall make no assumptions ...". It is replaced with "shall not depend on", a more actionable statement.

- 7.0: Added clause to clarify devices may support receiving of audio streams, or sending, or both.

- 7.1 Clarified the scope of this standard with respect to combinations of sampling rates and payload formats.

- 7.2.2 Details added to the table listing the required and recommended packet times.

- 7.3 Clarified channel count requirements.

- 7.4 Clarified requirement for the link offset to be retrievable from the device. Previous versions used an expression referring to communication of the link offset within the device.

- 8.1 Term "milliseconds decimal" replaced with "milliseconds fractional part".

- 8.1 Clarified the requirement to support alternative packet time signaling beyond the examples given in Table 4. Specifically, besides other values, valid alternative representations must be supported too, such as containing unnecessary precision digits or a decimal point not followed by the fractional part.

- 8.3: Added a discussion of specifying mediaclk at the media level of SDP as a precondition for interoperability with ST 2110-30:2017.

- 8.4 Reference redirected to section 7.1. In previous versions it pointed to table 2, which does not provide a complete specification.

- 10.1 Added explicit language allowing the use of other protocols or management interface for unicast connection management.

- Annex G (PICS): n/a option is now provided consistently, where relevant. In the previous version it was often missing.

- G.3.5 (PICS) – multiple statements: Streaming interoperability requirements reworked from individual attributes to attribute vectors called "stream modes".

- G.3.6.1 (PICS) - 8.0-1: SDP support

  o Clarified applicability of SDP output generation and input interpretation requirements to senders and receivers, respectively.

  o Added a discussion of using off-the-shelf test suites as a tool for verification of SDP document handling.

  o Added a table row for information about the test tools used for verification of SDP document handling and the test configuration.

**Informative and readability improvements:**

These editions include partial restructuring of some sections, correcting inaccurate terminology and descriptions, general editorial corrections. In particular:

- 3. All definitions formerly divided between section 3 "Definitions" and Annex F "Glossary" are consolidated in section 3. New terms are added, and some definitions are reworked for better clarity.

- 5. Clauses refactored and language revised to improve general readability and provide better definitions of the related terms.

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- 6.1, 7.6, 10.2 – content partially reordered and moved between sections to improve readability and consolidate sets of closely related requirements. Specifically:

  o Section 6.1 "Network layer" is now additionally structured by adding 3$^{rd}$-level headings.

  o Discussion of multicast transport was formerly split between section 6.1 "Network layer" and section 7 "Encoding and streaming". It is now all consolidated in section 6.1 "Network layer".

  o All discussion of using IGMP for multicast connection management is moved from 6.1 "Network layer" and consolidated in 10.2 "Multicast connections".

- E.6. Added a brief summary of RAVENNA discovery.

- New Annex F explains the context of using IGMP by senders to request their own streams. Such requirement was originally included in this standard, and it is removed in this revision.

- "Master" and "slave" terminology replaced by "timeTransmitter" and "timeReceiver" respectively, according to recommendations given in IEEE 1588g-2022.

- Multiple smaller editorial changes throughout the standard.

**Maintenance of external document references:**

- RFC 4566 has been revised and declared obsolete by IETF. Reference updated to RFC 8866.

- In previous versions, a number of documents have been misplaced between the lists of normative references and bibliography. Such misplacements have been corrected in this version as follows:

  o RFC 792 and RFC 4028 are moved from bibliography to normative references.

  o RFC 2974 is moved from normative references to bibliography.

- In previous versions, a number of documents appearing on the list of normative references or bibliography were not properly mentioned in the main text of this standard. Such omissions have been corrected in this version.

- A number of new document entries are added to the "Bibliography" section. Some were referenced in the text, and some are entirely new references. Newly added are a number of RFC documents by IETF, ST 2110-10 and ST 2110-30 by SMPTE, and an amendment to IEEE1588.

- Reference to IEEE 802.1Q-2011 is dated and is presented as such consistently in this version of AES67. Other references to IEEE 802.1 series documents are effectively undated, although release dates of these documents occasionally appear indicated in the text of this standard. They remain relevant and reflect the status at the moment of publishing the original version of AES67.

This revision was developed in task group SC-02-12-M.

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The task group was led by Kevin Gross.

This revision was edited by Gints Linis.

Morten Lave
Chair, working group SC-02-12, 2023-12-04

## Note on normative language

In AES standards documents, sentences containing the verb "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 -
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## 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 needed to meet these requirements is typically available on wired 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 can include additional options that are not required by this standard. Robust implementations of AES67 will tolerate these additional options.

Any behavior details not described in the main part of this standard are in some cases clarified in Annex G (PICS), by means of the respective evaluation criteria.

### 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 can 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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# 1 Scope

This standard defines an interoperability mode for synchronization, encoding, transport, and connection management of high-performance audio over networks based on the Internet Protocol. For the purposes of this standard, high-performance audio refers to audio with full bandwidth and low noise. These requirements imply linear PCM coding with a sampling rate 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. This standard considers latency performance of 10 milliseconds or less.

# 2 Normative references

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

In case of a conflict between clauses of this standard and a referenced document, the clauses of this standard take precedence.

The document identification and versioning approach adopted by IETF for RFCs, where each update or addendum is identified as a new document, can produce multiple secondary references related to a single root document. This standard lists only the root documents in force at the moment of writing. For secondary references, readers are encouraged to visit the IETF document library and follow the links indicated in the respective RFC document headers. Secondary references to RFC documents, which are published after this revision of AES67, should be used judiciously. In case of conflicts, the status at the moment of publishing this revision of AES67 takes precedence.

**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 792**, *Internet Control Message 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 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 4028**, *Session Timers in the Session Initiation Protocol (SIP)*, 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

**RFC 8866**, *SDP: Session Description Protocol*, Internet Engineering Task Force