STANDARDS AND INFORMATION DOCUMENTS

AES47-2018
(stabilized 2018)

STANDARDS

AES standard for digital audio — Digital input-output interfacing — Transmission of digital audio over asynchronous transfer mode (ATM) networks

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AES47-2018
incorporating AES47-Am1-2008
(s2018)

AES standard for digital audio — Digital input-output interfacing — Transmission of digital audio over asynchronous transfer mode (ATM) networks

Published by
Audio Engineering Society, Inc.
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Abstract

This document specifies the method of carrying multiple channels of audio in linear PCM or AES3 format in calls across an asynchronous transfer mode (ATM) network to ensure interoperability. The specification includes the method of conveying information concerning the format and sampling frequency of the digital audio signal when setting up the calls.

This revision adds code points in the ATM Adaptation Layer Parameters Information Element to signal that the time to which each audio sample relates can be identified as specified in AES53.

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Foreword

[This foreword is not a part of *AES standard for digital audio — Digital input-output interfacing — Transmission of digital audio over asynchronous transfer mode (ATM) networks*, AES47-2002.]

This document was developed under project AES-X92, Digital Audio in Asynchronous Transfer Mode (ATM). It is derived from draft X92-PWD-JG-010617, prepared by J. Grant for task group SC-02-02-E, incorporating the result of the task group meeting on 2001-06-12.

The membership of the task group was K. Brown, C. Chambers, J. Dunn, P. Eastty, C. Gaunt, J. Grant, U. Horbach, H. Lemcke, G. Mills, M. Schindele, J. M. Strawn, T. Thompson, and M. Yonge.

The task group has prepared a report for AESSC SC-02-02 of guidelines for *AES47-2002, AES standard for digital audio - Digital input-output interfacing — Transmission of digital audio over asynchronous transfer mode (ATM) networks* which has now been published as AES-R4-2002, *Guidelines for AES standard for digital audio — Digital input-output interfacing — Transmission of digital audio over asynchronous transfer mode (ATM) networks, AES47*.

Julian Dunn, chair
Robert A. Finger, vice-chair
SC-02-02

Foreword to 2006 revision

This revision of AES47 was prepared under project AES47-R by task group SC-02-02-E. It specifies use of the Organisationally Unique Identifier 00-0B-5E, recently acquired by AESSC, in 5.2.2.2 and 8.1.2.2. It also simplifies the minimum requirements to ensure interworking, in clause 4.3.

The membership of the task group at the time of writing the second edition was K. Brown, R. Caine, C. Chambers, P. Eastty, C. Gaunt, J. Grant, U. Horbach, H. Lemcke, M. Page, M. Schindele, J. M. Strawn, T. Thompson, J. Waas, and M. Yonge.

John Grant, chair
Robert A. Finger, vice-chair
SC-02-02

Addendum 2010-02-19

A new multi-part revision of AES3 was published in 2009. Its technical content is intended to be identical to the relevant parts of the 2003 edition as amended by Amendment 5 (2008) and Amendment 6 (2008). Where this document refers to clauses of earlier editions of AES3, equivalent references to AES3-2009 are also offered, *[identified by italic text in square brackets]*.

Foreword to 2018 revision

This revision of AES47 is an editorial combination of AES47-2006 and AES47-Am1-2008.

Richard C. Cabot, secretary
AESSC

Note on normative language

NOTE 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."

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AES standard for digital audio — Digital input-output interfacing — Transmission of digital audio over asynchronous transfer mode (ATM) networks

0 Introduction

0.1 Background

This document describes means for the transmission of professional audio across digital networks, including metropolitan- and wide-area networks, to provide the best performance with regard to latency, jitter, and other relevant factors.

Current-generation wide-area communications are based on two very similar systems, synchronous optical network (SONET) and synchronous digital hierarchy (SDH), SONET being used in the United States and SDH in Europe. On top of them are run integrated services digital network (ISDN), asynchronous transfer mode (ATM), and Internet protocol (IP).

ISDN provides telephone call connections of a fixed capacity which carry one 8-bit value per 125 µs; when a call is set up, its route through the system is chosen, and the switches that route the data are configured accordingly. Each link, between switches or between switch and end equipment, is formatted into frames that take 125 µs to transmit, and each data byte is identified by its position in the frame.

ATM, also called broadband ISDN, provides a service similar to ISDN, but with the capacity of each call being specified by the caller. Links are formatted into cells, which consist of a header and 48 data bytes; the header is typically 5 bytes long, and most of it is taken up with the `virtual channel identifier` (VCI) that shows to which call the cell belongs. Call setup, routing, and switching are done in the same way as in ISDN, but with calls not being restricted to 1 byte every 125 µs.

IP provides a very different service, not designed for continuous media such as audio and video. There is no call set-up, and each packet contains enough information within itself to allow it to be routed to its destination. This means that the header is much larger than in the case of ATM, typically 74 bytes, and packets will also typically be much larger, if only because otherwise the overheads would be excessive. Each packet is liable to be routed separately, so two packets that are part of the same flow may well take different routes. This can mean that the one that was sent first does not arrive first.

For many professional audio applications, a round-trip time from the microphone through the mixing desk and back to the headphones of no more than 3 ms is required. Allowing 0,5 ms each for conversion from analog to digital and back again, it follows that the network connections to and from the mixing desk must have a latency of less than 1 ms each. For distances of more than about 200 km, the transmission delay alone will exceed 1 ms, but within a

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metropolitan area the transmission delay should be no more than 0,25 ms (equivalent to about 50 km), leaving 0,75 ms for packetization, queuing within switches, and resynchronization within the receiving equipment.

Packetization delays are proportional to the size of the transmission unit (frame, cell, or packet), and resynchronization delays depend on how evenly spaced the transmission units are when they arrive at their destination. Both classes of delay are thus small for ISDN and large for IP. Using the format specified in this document to carry dual-channel AES3 audio with a 48 kHz sampling frequency over ATM results in an inter-cell time of 125 µs, at which ATM will have similar delays to ISDN. A higher sampling frequency or a larger number of channels would reduce the inter-cell time and hence also the delays.

The queuing time within each ISDN switch is likely to be around one frame time, or 125 µs. The ATM documents limit the queuing time in an ATM switch to approximately the inter-cell time for the call, which, as with the other delays, translates into performance similar to that of ISDN for dual-channel 48 kHz AES3 audio and better for higher sampling frequencies or larger numbers of channels.

The queuing time within an IP router for normal, best effort, Internet traffic is unbounded, and if the router is congested, packets may simply be thrown away. Resource reservation protocol (RSVP) (see annex A) allows capacity to be reserved for a particular traffic flow, but it does not guarantee that the packets will actually be routed over the links on which the capacity has been reserved; if the flow is re-routed, it will only get a best effort service until a reservation has been made on the new route, and it may not even be possible to make a reservation on the new route at all.

ATM has therefore been chosen as providing a more convenient service than ISDN and significantly better performance than IP, even when RSVP is used.

This document does not specify a physical interface to the network because one of the features of ATM is its ability to make a seamless connection between interfaces operating at a wide variety of data rates and with different ways of encoding the ATM cells. Commonly used interfaces provide 25,6 Mbit/s over category 3 structured wiring and 155,52 Mbit/s over category-5 structured wiring or fibre-optic cable.

The physical layer section description and unique ATM abbreviations can be found in ATM forum approved specifications. See annex A.

0.2 Patents

Attention is drawn to the possibility that some of the elements of this AES document may be the subject of patent rights not identified herein. The AES shall not be held responsible for identifying any or all such patent rights.

0.3 Conventions used in this document

0.3.1 Decimal points

According to International Electrotechnical Commission (IEC) directives, the comma is used in all text to indicate the decimal point. However, in specified coding, including the examples shown, the full stop is used, as in IEC programming language standards.

0.3.2 Data representation

In this document, all coding and data representations are printed in an equally spaced font.

0.3.3 Non-printing characters

Non-printing characters are delimited by angle brackets, as in <CR> for carriage return.

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0.4 Reserved bits

Unless otherwise indicated, bit assignments shown as reserved are reserved for future standardization by the AES, only by means of amendment or revision of this document.

1 Scope

This document specifies a means to carry multiple channels of audio in linear PCM or AES3 format over an ATM layer service conforming to ITU-T Recommendation I.150. It includes a means to convey, between parties, information concerning the digital audio signal when setting up audio calls across the ATM network.

It does not specify the physical interface to the network.

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.

AES3-2003 - AES standard for digital audio engineering — Serial transmission format for two-channel linearly represented digital audio data. Audio Engineering Society, New York, NY., US.

AES10-2003 - *AES recommended practice for digital audio engineering — Serial multi-channel audio digital interface (MADI).* Audio Engineering Society, New York, NY., US.

AES11-2003 - *AES recommended practice for digital audio engineering — Synchronization of digital audio equipment in studio operations.* Audio Engineering Society, New York, NY., US.

AES53-2006 - *AES Standard for digital audio - Digital input-output interfacing - Sample-accurate timing in AES47*. Audio Engineering Society, New York, NY., US.

ITU-T Recommendation I.150 (02/99) *B-ISDN asynchronous transfer mode functional characteristics.* Geneva, CH: International Telecommunications Union.

ITU-T Recommendation I.363.5 *B-ISDN ATM adaptation layer specification: Type 5 AAL.* Geneva, CH: International Telecommunications Union.

ITU-T Recommendation Q.2931 (02/95) - *Digital Subscriber Signaling System No. 2 (DSS 2) – User-Network Interface (UNI) layer 3 specification for basic call/connection control.* Geneva, CH: International Telecommunications Union.

ITU-T Recommendation Q.2971 (10/95) - *Broadband integrated services digital network (B-ISDN) – Digital subscriber signaling system No. 2 (DSS 2) – User-network interface layer 3 specification for point-to-multipoint call/connection control.* Geneva, CH: International Telecommunications Union.

OUI and company_id assignments See the resource locator on the databases page of http://www.aes.org/standards/.

3 Definitions and abbreviations

3.1
asynchronous transfer mode
ATM
networking technology in which data are carried in 48-o cells

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