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Digital audio – Digital input-output interfacing – Transmission of digital audio over asynchronous transfer mode (ATM) networks

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International Standard IEC 62365 has been prepared by technical area 4: Digital systems interfaces and protocols, of IEC technical committee 100: Audio, video and multimedia systems and equipment.

The text of this standard is based on the following documents:

FDIS	Report on voting
100/1517/FDIS	100/1550/RVD

Full information on the voting for the approval of this standard can be found in the report on voting indicated in the above table.

This publication has been drafted in accordance with the ISO/IEC Directives, Part 2.

The main changes with respect to the previous edition (2004) are listed below.

– Second, third, and fourth required formats in 4.3 removed.
– 4.3 reformatted, eliminating Table 2, and subsequent Tables renumbered.

The committee has decided that the contents of this publication will remain unchanged until the maintenance result date indicated on the IEC web site under "http://webstore.iec.ch" in the data related to the specific publication. At this date, the publication will be

- reconfirmed;
- withdrawn;
- replaced by a revised edition, or
- amended.

A bilingual version of this publication may be issued at a later date.

INTRODUCTION

This International Standard 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 that 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 set-up, routeing, 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 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 standard to carry dual-channel IEC 60958-4 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 IEC 60958-4 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 standard 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 the Bibliography.

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1 Scope

This International Standard specifies a means to carry multiple channels of audio in linear PCM or IEC 60958-4 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.

IEC 60958-1, *Digital audio interface – Part 1: General*

IEC 60958-4, *Digital audio interface – Part 4: Professional applications (TA4)*

ITU-T Recommendation I.150: *B-ISDN asynchronous transfer mode functional characteristics*

ITU-T Recommendation I.363.5, *B-ISDN ATM Adaptation Layer specification: Type 5 AAL*

ITU-T Recommendation Q.2931: *Digital Subscriber Signalling System No. 2 – User-Network Interface (UNI) layer 3 specification for basic call/connection control*