# AES recommended practice for digital audio engineering — Format for the user data channel of the AES digital audio interface

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#### Abstract

This document describes a method of formatting the user data channels provided within the digital audio serial interface format (AES3). The transmission format is an adaptation of the packet-based high-level data link control (HDLC) communications protocol and provides for the transmission of ancillary data that may or may not be time related to the audio signal. The data rate is constant within a range of  $\pm$  12.5 percent of a sampling frequency of 48 kHz. The standard also provides a data priority and management strategy to ensure that adequate capacity is available for downstream data insertion.

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## -2-

# Contents

Foreword	
1 Scope	
2 Normative references	
3 Definitions	
4 General description	
4.1 Messages	
5 Data formatting	
5.1 Application level.	
6 Channel management	
6.1 Channel formatting136.2 Channel description	
7 Addressing of applications	
7.1 Introduction197.2 Classes of application197.3 Applications197.4 Message exchange207.5 Addressing207.6 Assignment of reserved bits21	
Annex A (Informative)	
New packets	
Annex B (Informative)	
Code tables	
Annex C (Informative)	
Informative References	

#### -3-

#### Foreword

[This foreword is not a part of AES Recommended practice for digital audio engineering – Format for the user data channel of the AES digital audio interface, AES18-1996.]

This document is a revision of AES18-1992. It is a companion document to the AES digital audio interface specification AES3, AES Recommended practice for digital audio engineering — Serial transmission format for two-channel linearly represented digital audio data and its revisions. It is the result of a desire by users of the interface to have a recommended format for the user data channel provided by the user bit.

In this edition, an amendment has been added to provide recommendations for coding to be used in complying with AES18 and to complete the subclauses therein. The coding is presented in an added clause 7 and an added annex B. An added annex C provides informative references for the coding. The coding has been developed in cooperation with the European Broadcasting Union.

The requirement was for a system that is flexible and independent of the user application, and that can carry message data related in time to the audio data, as well as information, such as text, that may be unrelated to the audio data. A further requirement was for a format that treats the user data channel as a transparent carrier at sampling frequencies in the range of 48 kHz  $\pm$  12.5 percent such that a constant data rate is achieved. The system specified is based on a widely used packet communication protocol, high-level data link control (HDLC) (ISO Publication 3309), which is standardized in the information technology industry. The HDLC protocol has been adapted for unidirectional transmission and to permit the accurate transmission of time-dependent information, but the HDLC integrated circuits, which are readily available from several manufacturers, are still able to be used. It is therefore expected that this transmission format will be easy to implement without recourse to special hardware and will, as a result, be included as a matter of routine in commercial interface equipment.

This document results from the work of the Working Group SC-02-04 Signal Labeling and Ancillary Data, established by SC-02 Subcommittee on Digital Audio of the AES Standards Committee, in close collaboration with the European Broadcasting Union (EBU).

Experimental and prototype equipment was designed and built by A. Komly and A. Viallevieille to prove and test the system during development of the specification. Input documents were provided by R. Lagadec, G. McNally, J. Wilkinson, A. Komly, A. Weisser, A. Viallevieille, and J. P. Nunn.

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J. P. NUNN, *Chairman* SC-02-04 Working Group Signal Labeling and Ancillary Data 1996-07

### Corrigendum 2005-05-12

In table 7, message address extension byte 2, units previously shown as "ft/s" have been corrected to read "frames/s"

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-4-

# AES recommended practice for digital audio engineering — Format for the user data channel of the AES digital audio interface

### 1 Scope

This document specifies a recommended method of formatting the user bit of each channel of the AES digital audio interface (AES3, AES Recommended practice for digital audio engineering – Serial transmission format for two-channel linearly represented digital audio data). Each user data channel so formed is independent of the user application and is primarily intended for the transmission of data associated with the audio signal, although data unrelated to the audio signal may also be transmitted. There is no restriction on the length of messages that may be transmitted in the user data channel. The format treats the user data channel as a transparent carrier at sampling frequencies in the range of 42 kHz to 54 kHz, that is, 48 kHz  $\pm$  12.5 percent. There is no theoretical upper limit, but if sample rate conversion to a sampling frequency below this range is performed, data management is to be used to avoid loss of vital data.

The document describes the method of formatting user information into packets, together with the rules for data insertion into the multiplex and for data management. The purpose and the content of the user information for particular applications are outside the scope of this document.

This document describes a practice for professional applications and is not intended to encompass applications related to consumer versions of the digital audio interface.

### 2 Normative references

The following standards contain provisions that, through reference in this text, constitute provisions of this document. At the time of publication, the editions indicated were valid. All standards are subject to revision, and parties to agreements based on this document are encouraged to investigate the possibility of applying the most recent editions of the indicated standards.

1) AES3-1992, AES Recommended practice for digital audio engineering – Serial transmission format for two-channel linearly represented digital audio data. New York: Audio Engineering Society, 1992.

2) ISO 3309, Information processing systems–Data communications–High-level data link control procedures– Frame structure. Geneva, Switzerland: International Organization for Standardization, 1984.

### 3 Definitions

**3.1 transport system**: Method by which messages are carried between source and destination of the application.

**3.2 address**: Identification of either the destination hardware or the application. It can be combined with an address extension.

**3.3 address extension**: Part of the address that extends the range of destination hardware or applications that may be addressed.

**3.4 block**: Repetitive structure chosen for the transport system by the user.

**3.5 continuity index**: Count of the messages or packets with a given address. In this system the count is modulo 8, allowing up to seven missing messages or packets to be detected. This system uses a message continuity index and a packet continuity index.