PROPOSED SMPTE STANDARD

SMPTE 320M

for Television —

Channel Assignments and Levels on Multichannel Audio Media

1 Introduction

This standard provides specifications for the placing of a 5.1 channel audio program onto multitrack audio media. As specified in ITU-R BS.775-1, the internationally recognized multichannel sound system consists of left, center, right, left surround, right surround, and low-frequency effects (LFE) channels. SMPTE RP 173 specifies the locations and relative level calibration of the loudspeakers intended to reproduce these channels. This standard specifies a mapping between the audio signals intended to feed loudspeakers, and a sequence of audio tracks on multitrack audio storage media. This standard also specifies the relative levels of the audio signals. Media prepared according to this standard will play properly on a loudspeaker system calibrated according to SMPTE RP 173.

In consumer audio systems, the LFE channel is considered optional in reproduction. Media which conform to this standard should be prepared so that they sound satisfactory even if the LFE channel is not reproduced. When an audio program originally produced as a feature film for theatrical release is transferred to consumer media, the LFE channel is often derived from the dedicated theatrical subwoofer channel. In the cinema, the dedicated subwoofer channel is always reproduced, and thus film mixes may use the subwoofer channel to convey important low-frequency program content. When transferring programs originally produced for the cinema over to television media, it may be necessary to remix some of the content of the subwoofer channel into the main full bandwidth channels. It is important that any low-frequency audio which is very significant to the integrity of the program content is not placed into the LFE channel. The LFE channel should be reserved for extreme low frequency and for very high level, <120-Hz, program content which, if not reproduced, will not compromise the artistic integrity of the program.

2 Scope

This standard specifies the audio channel assignment, and the relative levels of the audio channels, for recordings of audio programs containing between three and six audio channels, onto storage media for television sound. This standard may also be applied to other types of media (such as transmission) where a sequence of audio channels is available to carry a multichannel audio program. This standard is not intended for application in the area of film sound.

3 Normative reference

The following standard contains provisions which, through reference in this text, constitute provisions of this standard. At the time of publication, the edition indicated was valid. All standards are subject to revision, and parties to agreements based on this standard are encouraged to investigate the possibility of applying the most recent edition of the standard indicated below.

SMPTE RP 155-1997, Audio Levels for Digital Audio Records on Digital Television Tape Recorders

4 Channel assignments

4.1 Signal definitions

Table 1 specifies the nomenclature used to identify audio channels. The Lt,Rt terms refer to a two-channel matrix surround encoded program, and Lo,Ro to a conventional two-channel stereophonic program.

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Table 1 – Audio channel abbreviations

Audio channel	Abbreviation
Left	L
Center	C
Right	R
Left surround	LS
Right surround	RS
Low-frequency effects	LFE
Mono surround	MS
Mono surround at a –3 dB level	MS (–3 dB)
Left total	Lt
Right total	Rt
Stereo left	Lo
Stereo right	Ro
Monophonic	M
Freely usable	F
Unassigned / unused	U

4.2 Track assignments

Table 2 indicates two track assignments identified as A and B which differ only in the assignment of tracks 7 and 8. The only flexibility allowed in tracks 1-6 is whether or not to include the indicated track. The media shall either include the indicated track, or shall leave the track unused (U). Assignment A requires tracks 7 and 8 to carry a two-channel stereo program intended for reproduction over L and R loudspeakers. Assignment B leaves tracks 7 and 8 freely usable (F), so they may be used for any purpose.

Eight tracks are shown. For media with six tracks, the assignments for tracks 1-6 apply and there is no relevance to these A,B identifiers. Six-channel media shall be identified as following assignment A. Some media have more than eight tracks or a multiple of eight tracks. Media with more than eight tracks shall follow the assignment in table 2 for the first group of eight tracks. When relevant (i.e., additional multi-channel programs are carried), it is recommended that other groupings of 8 tracks (i.e., tracks 9-16, or 17-24 on a 24-track media) also follow the channel assignments shown in table 2.

5 Digital signal characteristics

5.1 Sampling frequency

Digital media conforming to this standard shall employ a nominal sampling frequency of 48 kHz.

NOTE - There is some limited usage of digital audio with a sampling frequency at a multiple of 1000/1001 or 1001/1000 relative to the 48-khz value. these applications are considered within the scope of this standard. storage media labeled as conforming to this standard shall be playable at the 48-khz sample rate. these storage media may also be playable at the off-speed rates, where appropriate, in order to remain in sync with a corresponding source of pictures.

5.2 Emphasis

The audio signals carried by digital media which conform to this standard shall not contain any frequency emphasis.

6 Level alignment

6.1 Level calibration

With the exception of the LFE channel, all audio channels belonging to a common program are recorded on the storage media at levels appropriate for reproduction over a set of loudspeaker channels which produce the same acoustic sound pressure level for a common stimulus. This means that the program would play correctly over a reproduction system in which each of the individual speaker channels in the 3/2 configuration (L, C, R. Ls, Rs) has the same relative acoustic output when presented with equal level signals from the media.

The LFE channel is stored with a level offset of –10 dB. This offset is compensated for in the reproduction system, where the LFE loudspeaker has an acoustic output (within its low-frequency passband) of +10 dB with respect to the other channels.

6.2 Level calibration test signals

6.2.1 Tones

Level calibration tones are the subject of SMPTE RP 155.

Table 2 – Track assignments for media with 6 or more channels

Track	Stanard Assignment A			Standard Assignment B
1	L			L
2	R			R
3	C			C
4	LFE			LFE
5	LS[1]			LS[1]
6	RS[1]			RS[1]
7	Lt	or	Lo	F[2]
8	Rt		Ro	F[2]

NOTES

1 In the case of programs with a monophonic surround channel, the MS (–3dB) monophonic surround signal can be placed on both tracks 5 and 6. This allows a program with a single surround channel to be treated as a program with two surround channels. The MS (–3dB) signal will be reproduced out of both the LS and RS loudspeakers, with a relative level of –3dB with respect to the front channels. The combined power into the room will be the correct relative level of 0 dB.

2 In assignment B, tracks 7 and 8 are freely usable for any purpose.

3 If the main audio program requires less than six tracks, the use of any of the unneeded tracks 1–6 for any other purpose is not compliant with this standard. While nonconforming assignments can be expected to occur in practice, such usage is outside the scope and specification of this standard. It is recommended that track assignments which are not according to this standard remain as consistent as possible with table 2, so as to minimize the need to repatch studio configurations. For example, if a 2/2 program (L, R, Ls, Rs) and a 2/0 program (Lt,Rt) are all placed in the first 6 tracks, the 2/2 program could occupy tracks 1, 2, 5, and 6 respectively, and the 2/0 program could occupy tracks 3 and 4.

6.2.2 Pink noise

Pink noise test signals, if present, shall be at equal levels on all channels. Excepting the LFE channel, if each channel of pink noise is reproduced at the same acoustic sound pressure level, then the relative balance of the program audio channels should be correct.

In the case of the LFE channel, the pink noise test signal is intended to be reproduced at an acoustic sound pressure level (within the LFE channel <120-Hz passband) of +10 dB relative to any of the other individual channels. Note that due to the limited bandwidth of the LFE channel, if the acoustic level produced by the LFE pink noise is measured with a wideband sound pressure level meter, the reading will not (and should not) measure +10 dB with respect to the other channels. The acoustic level of the LFE channel should measure + 10 dB within its <120-Hz bandwidth when measured with a frequency selective meter.

7 Media labeling

Media which conform to this standard shall have a label which clearly documents the contents of each individual track, and any test or alignment signals which are present. The preferred form of the label is a standardized electronic label. At the time of issuance of this standard, a standardized method of electronic labeling is not available.

For recorded media, the preferred form of labeling is to store the label in electronic form on the media along with the audio information. For transmission media, the electronic label should be included in the transmission.

For removable media, until a standardized form of electronic label is available, and for those removable media which cannot carry an electronic label, a human readable textual label shall be closely associated with the media.

The label shall indicate compliance with either the A or B columns in table 2 by using the notation: SMPTE 320M-A or SMPTE 320M-B.

Examples of labeling are shown in annex A.

Annex A (informative)
Tape Label

Examples of a tape label for an eight-channel tape recorded according to this standard shall be as follows (see table A.1):

Tape number: 123

Title: An illustrative example

Program details: A set of examples to illustrate labeling and usage of this standard.

Tone alignment level: –20 dBFS

Pink noise level: –30 dBFS rms

Table A.1 – Labels for eight-channel tape

	Example 1	Example 2	Example 3	Example 4
Track	SMPTE 320M-A	SMPTE 320M-A	SMPTE 320M-A	SMPTE 320M-B
1	L	L	L	L
2	R	R	R	R
3	C	C	C	C
4	LFE	U	U	U
5	LS	U	MS (–3 dB)	MS (–3 dB)
6	RS	U	MS (–3 dB)	MS (–3 dB)
7	Lt	Lo	Lt	M (Japanese mono downmix)
8	Rt	Ro	Rt	M (Spanish mono downmix)

Annex B (informative)
Bibliography

SMPTE 323M, Motion-Picture Film — Channel Assignments and Levels on Multichannel Audio Media

SMPTE RP 173, Loudspeaker Placements for Audio Monitoring in High-Definition Electronic Production

ITU-R BR.1384, Parameters for International Exchange of Multi-channel Sound Recordings

ITU-R BS.775-1, Multichannel Stereophonic Sound System With and Without Accompanying Picture

PROPOSED SMPTE STANDARD

SMPTE 323M

for Film — Channel Assignments and Levels on Multichannel Audio Media

1 Introduction

This standard provides specifications for the placing of a 5.1 channel audio program onto multitrack audio media. The multichannel audio system consists of left, center, right, left surround, right surround, and subwoofer (SW) channels. This standard specifies a mapping between the audio signals intended to feed loudspeakers, and a sequence of audio tracks on multitrack audio storage media. This standard also specifies the relative levels of the audio signals. Media prepared according to this standard will play properly on a loudspeaker system calibrated according to the theatrical exhibition standard, SMPTE RP 200.

In theatrical exhibition systems, the SW channel is always reproduced. Domestic consumer multichannel audio employs a low-frequency effects (LFE) channel which can carry the information included on the cinema SW channel. However, in consumer audio systems, the LFE channel is considered optional in reproduction, and is intended to carry extreme high-level and low-frequency effects. When an audio program originally produced as a feature film for theatrical release is transferred to consumer media, the LFE channel is often derived from the dedicated theatrical subwoofer channel. When transferring programs originally produced for the theatrical reproduction over to media intended for domestic consumer reproduction, it may be necessary to remix some of the content of the subwoofer channel into the main full bandwidth channels so as to maintain the artistic integrity of the program content in case the LFE channel is not reproduced

2 Scope

This standard specifies the audio channel assignment, and the relative levels of the audio channels, for recordings of audio programs containing six audio channels, onto storage media for film sound. Programs which are placed on media according to this standard are intended for reproduction in the cinema where the SW channel is always reproduced. This standard is not intended for application in the area of sound intended for reproduction in the domestic consumer environment.

3 Normative references

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

SMPTE RP 127-1999, Specifications for Type U Audio Level and Multifrequency Test Film for 35-mm Studio Audio Reproducers, Magnetic Full-Coat Type

SMPTE RP 155-1997, Audio Levels for Digital Audio Records on Digital Television Tape Recorders

SMPTE RP 200, Relative and Absolute Sound Pressure Levels for Motion-Picture Multichannel Sound Systems

4 Channel assignments

4.1 Signal definitions

Table 1 specifies the nomenclature used to identify audio channels. The Lt,Rt terms refer to a two-channel matrix surround encoded program.

Table 1 – Audio channel abbreviations

Audio channel	Abbreviation
Left	L
Center	C
Right	R
Left surround	LS
Right surround	RS
Subwoofer	SW
Mono surround	MS
Mono surround at a –3 dB level	MS (–3 dB)
Left total	Lt
Right total	Rt
Unassigned/unused	U

4.2 Track assignments

Table 2 indicates three allowed track assignments denoted A, B, and C. Eight tracks are shown. For media with six tracks, the assignments for tracks 1-6 apply. Some media have more than eight tracks or a multiple of eight tracks. Media with more than eight tracks shall follow the assignment in table 2 for the first group of eight tracks. When relevant (i.e., additional multichannel programs are carried), it is recommended that other groupings of 8 tracks (i.e., tracks 9-16 or 17-24 on a 24-track media) also follow the channel assignments shown in table 2.

Table 2 – Track assignments for media with 6 or more channels

Track	Standard assignment A	Standard assignment B	Standard assignment C
1	L	L	L
2	R	LS (see note)	C
3	C	C	R
4	SW	RS (see note)	LS
5	LS (see note)	R	RS
6	RS (see note)	SW	SW
7	Lt or U	Lt or U	Lt or U
8	Rt or U	Rt or U	Rt or U

NOTE – In the case of programs with a monophonic surround channel, the MS (–3 dB) monophonic surround signal can be placed on both tracks 5 and 6. This allows a program with a single surround channel to be treated as a program with two surround channels. The MS (–3 dB) signal will be reproduced out of both the LS and RS loudspeakers, with a relative level of –3 dB with respect to the front channels. The combined power into the room will be the correct relative level of 0 dB.

5 Digital signal characteristics

5.1 Sampling frequency

Digital media conforming to this standard shall employ a nominal sampling frequency of 48 kHz.

NOTE – There is some limited usage of digital audio with a sampling frequency of 1000/1001 or 1001/1000 relative to the 48-kHz value. These applications are considered within the scope of this standard. Storage media labeled as conforming to this standard shall be playable at the 48-kHz sample rate. These storage media may also be playable at the off-speed rates, where appropriate, in order to remain in sync with a corresponding source of pictures.

5.2 Emphasis

The audio signals carried by digital media which conform to this standard shall not contain any frequency emphasis.

6 Level alignment

6.1 Level calibration

The audio channels are recorded onto storage media with levels relative to the intended acoustic level for reproduction as indicated in table 3.

Table 3 – Relative levels of recorded channels

Channel	Relative level
L	0 dB
R	0 dB
C	0 dB
SW	–10 dB
LS	+3 dB
RS	+3 dB

The three front channels are recorded at equal level. The surround channels are recorded at a level offset of +3 dB, which is suitable for reproduction in a cinema which has left and right surround speaker systems, each of which has an acoustic output of –3 dB relative to the frontal channels (given a common stimulus).

The SW channel is stored with a level offset of –10 dB. This offset is compensated for in the reproduction system, where the subwoofer channel has an acoustic output (within its low-frequency passband) of + 10 dB with respect to an individual front (L, C, or R) channel.

6.2 Level calibration test signals

6.2.1 Tones

Level calibration tones are the subject of SMPTE RP 127 and SMPTE RP 155.

6.2.2 Pink noise

Pink noise test signals, if present, shall be at equal levels on all channels.

7 Media labeling

Media which conform to this standard shall have a label that clearly documents the contents of each individual track, and any test or alignment signals that are present. The preferred form of the label is a standardized electronic label. At the time of issuance of this standard, a standardized method of electronic labeling is not available.

For recorded media, the preferred form of labeling is to store the label in electronic form on the media along with the audio information. For transmission media, the electronic label should be included in the transmission.

For removable media, until a standardized form of electronic label is available, and for those removable media which cannot carry an electronic label, a human readable textual label shall be closely associated with the media.

The label shall indicate compliance with either the A, B, or C columns in table 2 by using the notation: SMPTE 323M-A, SMPTE 323M-B, or SMPTE 323M-C.

Examples of labeling are shown in annex A.

Annex A (informative)
Tape label

Examples of a tape label for an eight-channel tape recorded according to this standard are shown in table A.1.

Tape number: 123

Title: An Example Film

Program details: A set of examples to illustrate labeling and usage of this standard.

Tone alignment level: −20 dBFS

Pink noise alignment level: −30 dBFS rms

Table A.1 – Examples of tape label

	(Example 1)	(Example 2)	(Example 3)	(Example 4)
Track	SMPTE 323M-A	SMPTE 323M-A	SMPTE 323M-B	SMPTE 323M-C
1	L	L	L	L
2	R	R	LS	C
3	C	C	C	R
4	SW	SW	RS	LS
5	LS	LS	R	RS
6	RS	RS	SW	SW
7	Lt	U	Lt	Lt
8	Rt	U	Rt	Rt

Annex B (informative)
Bibliography

SMPTE 320M, Television — Channel Assignments and Levels on Multichannel Audio Media

ITU-R BS.775-1 (1994), Multi-channel Stereophonic Sound System With and Without Accompanying Picture

PROPOSED SMPTE STANDARD

SMPTE 324M

for Television — 12-Channel Serial Interface for Digital Audio and Auxiliary Data

Table of contents

1 Scope

1.1 This standard defines a synchronous, self-clocking serial interface for up to 12 channels of linearly encoded audio and auxiliary data. The interface is designed to allow multiplexing of six two-channel streams compliant with AES3.

1.2 Audio sampled at 48 kHz and clock locked to video is the preferred implementation for studio applications. However, this interface supports any frequency of operation supported by AES3, provided that all the audio channels are sampled by a common clock. Ideally all the channels should be audio synchronous (see 3.5) for guaranteed audio phase coherence.

1.3 This standard is intended to provide a reliable method of distributing multiple cophased channels of digital audio around the studio without losing the initial relative sample-phase relationship. A mechanism is provided to allow more than one 12-channel stream to be realigned after a relative misalignment of up to ± 8 samples.

1.4 This interface is intended to be compatible with the complete range of digital television scanning standards and standard film rates.

1.5 This interface may be used for distribution of multiple channels of audio in either a pre-mix or post-mix situation. In the post-mix case, a channel assignment is defined in SMPTE 320M to facilitate interchange.

2 Normative references

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

AES3-1992, Digital Audio Engineering — Serial Transmission Format for Two-Channel Linearly Represented Digital Audio Data

AES3-id-1995, AES Information Document for Digital Audio Engineering — Transmission of AES3 Formatted Data by Unbalanced Coaxial Cable

ANSI S4.44-1997, Digital Audio Engineering — Synchronization of Digital Audio Equipment In Studio Operations (AES11)

ANSI/SMPTE 276M-1995, Television — Transmission of AES-EBU Digital Audio Signals Over Coaxial Cable

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3 Definition of terms

3.1 AES audio: All the data, audio and auxiliary, associated with one AES digital stream as defined in AES3.

3.2 AES frame: Two AES subframes, one with audio data for channel 1 followed by one with audio data for channel 2.

3.3 AES subframe: All data associated with one AES audio sample for one channel in a channel pair.

3.4 audio data: 24 bits: 20 bits of AES audio associated with one audio sample, not including AES auxiliary data, plus the following 4 bits: sample validity (V bit), channel status (C bit), user data (U bit), and parity (P bit).

3.5 audio synchronous: One audio channel is defined as being synchronous with another when the two channels are running from the same clock and the analog inputs are concurrently sampled as described in ANSI S4.44 (AES11).

3.6 auxiliary data: Four bits of AES audio associated with one sample defined as auxiliary data by AES3. The four bits may be used to extend the resolution of the audio sample.

3.7 carrier data: Available data bits in the header data that may be used for metadata transmission or other user applications.

3.8 channel pair: Two digital audio channels derived from the same AES audio source.

3.9 header data: All the data in the multiplex header.

3.10 multichannel phasing flag (MC bit): A dedicated bit that identifies the start of a 16-frame period with a 1 state. This bit immediately precedes the Z 1-2 bit and has a 1 state in the first frame following the frame containing the Z preamble in the AES11 timing reference.

3.11 multiplex frame: All the data associated with each sample of audio in each AES frame for each of six AES inputs and the multiplex header associated with that sample period.

3.12 multiplex frame sync: A code violation at the beginning of the multiplex header that identifies the start of each multiplex frame.

3.13 multiplex header: 48-bit data packet at the start of each frame that includes a code violation for synchronization, 32 bits of carrier data, an optional carrier block flag (CB bit), a parity bit for carrier data (CP bit), a multichannel phasing flag (MC bit), two reserved bits (R1 and R2 bits), six Z bits, and a parity bit for the MC, R, and Z bits (HP bit).

3.14 video synchronous: Audio is defined as being clock synchronous with video if the sampling rate of the audio is such that the number of audio samples occurring within an integer number of video frames is itself a constant integer number.

4 Overview

Audio data derived from six synchronous AES frames are multiplexed together in a single, biphase-mark-encoded serial data stream at exactly six times the bit rate of each incoming AES signal. Each AES subframe is organized in time sequence in the multiplexed frame. All the data in each AES subframe is included with the exception of the preamble. The four bit times from each preamble are moved to the front of the multiplexed word to produce a 48-bit header. Four of the header bits are used for a code violation that provides frame synchronization. One bit is used for a 16-frame flag that allows multiple streams that have been subjected to differential delay to be rephased with minimum latency. Six bits are assigned as Z bits to identify the block start frame for each of the six AES channel pairs. Two bits are reserved. One bit is assigned to a parity bit that covers the MC bit and the Z bits. Thirty-two bits are allocated to carrier data. One bit is reserved for an optional block start reference for carrier data, and the last bit is used to ensure even parity across carrier data bits, the carrier block bit, and the reserved bits.

5 Multiplex carrier channel coding

The data stream in the multiplex carrier is coded biphase mark. Data zeroes are identified by a transition only at the beginning of the bit period. Data ones are identified by transitions at the beginning and at the middle of the bit period as shown in figure 1. Note that the receiver shall be able to detect the biphase mark data in either normal or reverse polarity so that it can operate normally if the data has passed through an odd number of inversions.

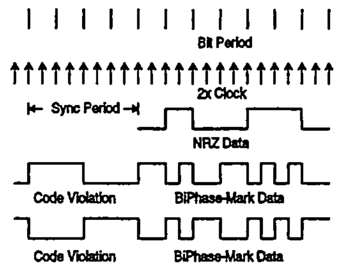

Figure 1 – Channel coding

6 Synchronization and audio phasing

Multiplex frame synchronization is achieved by use of a code violation at the start of each multiplex frame. The code violation consists of a transition at the beginning of the first 2-bit period and a second transition at the beginning of the second 2-bit period. This produces the maximum violation time achievable in a 4-bit period for an even-parity violation. A multichannel flag bit (MC bit) is provided to allow rephasing of multiplex streams that have suffered differential delay up to ± 8 samples. The MC bit is a 1 in the sample frame following the frame that contains the Z preamble in the AES11 system sync reference. Subsequently, the MC bit is a 1 every 16 frames, and has a value of 0 in all other frames. This bit must always be passed with the identical delay as that for the audio data.

7 AES data mapping

AES frame data from AES inputs are rigidly organized in the multiplex carrier. AES input 1 follows the multiplex header with subframe 1 preceding subframe 2. AES inputs 2 through 6 follow in the same fashion. With the exception of the preamble, all data in each AES subframe is replicated in order in the multiplex carrier. Each subframe preamble is replaced by a single bit which is zero at all times except during a Z preamble at which time it is a one. These single Z bits immediately follow the MC bit in the multiplex header. They are arranged in the same order as the AES inputs. The subframes are numbered 1 through 12 in the multiplex according to the above mapping scheme. All multiplex and demultiplex equipment shall have AES input and output connectors accurately labeled with channel numbers in the form channel 1-2, channel 3-4, channel 5-6, channel 7-8, channel 9-10, and channel 11-12. Figure 2 shows the bit assignments.

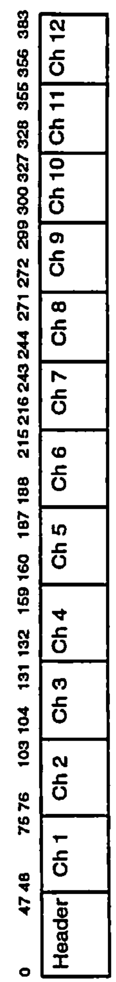

Figure 2 – AES data mapping

8 Multiplex header

Each incoming AES subframe includes a 4-bit preamble from which only the block start information (Z bit) is necessary in a rigid multiplex. The four bit-periods from each AES preamble are assigned to the multiplex header. The header is thus 48 bits long.

Bits 0 through 3 in the header are assigned to multiplex frame sync (a code violation sequence).

Bits 4 through 35 are assigned to carrier data bits and organized as four bytes with MSB first.

Bit 36 is assigned to an optional carrier data block start bit (CB bit).

Bits 37 and 38 (R1 and R2 bits) are reserved and should be set to zero until defined.

Bit 39 is assigned to a parity bit (CP) which ensures even parity across the carrier data bits, the R bits, and the CB bit.

Bit 40 is assigned to the multichannel flag bit (MC).

Bit 41 is assigned to the block start (Z 1-2 bit) for the first AES pair (channels 1 and 2).

Bit 42 is assigned to the block start (Z 3-4 bit) for the second AES pair (channels 3 and 4).

Bit 43 is assigned to the block start (Z 5-6 bit) for the third AES pair (channels 5 and 6).

Bit 44 is assigned to the block start (Z 7-8 bit) for the fourth AES pair (channels 7 and 8).

Bit 45 is assigned to the block start (Z 9-10 bit) for the fifth AES pair (channels 9 and 10).

Bit 46 is assigned to the block start (Z 11-12 bit) for the sixth AES pair (channels 11 and 12).

Each Z bit is set to one in the frame coinciding with the Z preamble in the corresponding AES input. At all other times, the Z bits are set to zero.

Bit 47 is assigned to a parity bit (HP bit) that ensures even parity across the MC bit and Z bits. Figure 3 shows the header detail.

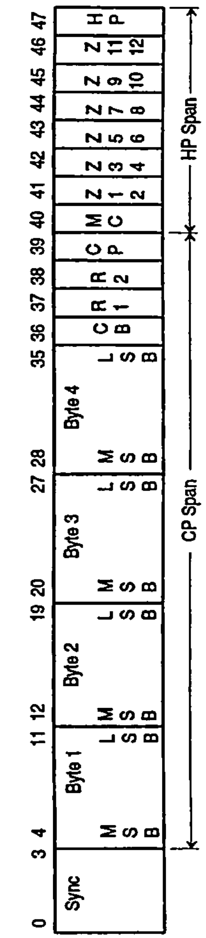

Figure 3 – Header detail

9 Electrical characteristics

9.1 The output interface of the multiplexer is a 1-volt peak-to-peak, 75-ohm, coaxial, source-terminated signal, centered on ground.

9.1.1 The output level tolerance shall be ± 10%, and the output return loss shall be better than 25 dB from 200 kHz to 40 MHz and better than 15 dB from 40 MHz to 75 MHz.

9.1.2 The rise and fall times, determined between the 10% and 90% amplitude points and measured across a 75-ohm resistive load through 1 meter of high-quality coaxial cable, shall be between 5% and 30% of the baud interval.

NOTE – For example, a 2.7-ns rise time is equal to 10% of the shortest symbol for 48-kHz operation. This results in 7% at 32 kHz and 20% at 96 kHz. Therefore, equipment with an output rise time between 2 ns and 4 ns will meet this specification for operation from 32 kHz to 96 kHz.

9.1.3 Data jitter shall be less than 0.2 unit interval peak to peak.

NOTE – For example, at 48-kHz operation, the shortest symbol period is 27 ns. At this operating frequency, the jitter shall not exceed 5.4 ns peak to peak. As the operating frequency is increased, the allowable jitter is proportionately less.

9.2 The receiver shall have a nominal input impedance of 75 ohms.

9.2.1 The receiver shall have a return loss better than 25 dB from 200 kHz to 40 MHz and better than 15 dB from 40 MHz to 75 MHz.

9.2.2 The receiver shall correctly interpret the data with an input level from 200 mV to 1.2 V peak to peak with a data jitter up to 0.5 unit interval peak to peak.

10 Equalization

Equalization for transmission cable is not required for typical distances. When equalization is used, it shall correct link cable losses over the range from 100 kHz to 50 MHz.

NOTE – Typical coaxial cables exhibit loss, but very little differential group delay from 5 MHz to 75 MHz and equalizers, if used, must have complementary response.

11 Cable

The interconnecting coaxial cable shall have a nominal characteristic impedance of 75 ohms.

12 Connector

The connector shall have mechanical characteristics conforming to the 50-ohm BNC type. Mechanical dimensions of the connector may produce either a nominal 50-ohm or nominal 75-ohm impedance. However, the electrical characteristics of the connector and its associated interface circuitry shall provide a resistive impedance of 75 ohms. Where a 75-ohm connector is used, its mechanical characteristics must reliably interface with the nominal 50-ohm BNC type defined by IEC 60169-8.

Annex A (informative)
Additional data

A.1 Abstract

This standard describes a method for multiplexing up to 12 channels of audio (six AES3 signals) into one data stream. The input and output signal formats of the multiplexer and demultiplexer meet the AES3, AES3-id, or ANSI/SMPTE 276M specification. The aggregate transmitted data rate is six times that of a single AES3 input. Inputs must be mutually frequency locked (isochronous) and ideally phase aligned (synchronous).

A.2 Background

The advent of ATV, be it HDTV, resolution-enhanced SDTV, multicast, or multilingual, has generated an increased desire to improve audio quality. In fact, results from a number of ATV user tests indicate that improving audio quality provides at least as much enhancement to the viewing experience as increasing picture aspect ratio and resolution.

The surround sound or theater sound environment can be implemented and coded in many forms. AC-3, DTS, SDDS, and MPEG audio are some examples. Each of these multichannel coding methods requires original audio source material that is image accurate both during production and for production master distribution in order to maintain the original spatial perception of the program. With multiple diverse audio paths, it is difficult to provide precisely matched path delays for multiple channels of audio in a typical video production environment due to the multiple opportunities for sample slips from the beginning to the end of a production path.

Several proposals have been made for distribution of audio using compression. Although compression is very useful for

recording, long-haul transmission, and final delivery, it creates some problems as a distribution method for production. Production requires that multiple sources be available to a common processing point, and that these sources be in a common editable format. Since there are a number of compression formats available, the point of production must be 'multilingual' and must also be able to manage the different time delays associated with each format, and the variable time delay with a given format. This is a complex process to distribute throughout a production facility.

The audio channels of current DVTRs, disk recorders, and other AES3 paths through the typical plant are not transparent data channels in spite of the fact that digital audio is PCM data. There are several audio signal impairment issues including phase uncertainty between AES3 audio and video sync. Error concealment algorithms, sample rate conversions, and audio level processing are operations that may be found in many machines. If compressed data are stored on the machine, the effects of these processing steps, if not successfully bypassed, will typically render the data and subsequent uncompressed audio output useless.

Different machines may well require different coding algorithms to ensure transparency within the recorder. As a minimum, a common, or standard, compression algorithm would need to allow for full error correction given the statistics of every transport, fixed compression time, across all equipment in the plant, and a coding scheme which would allow a cut edit. It is most likely that future machines will employ proprietary compression to accomplish the feature sets desired by individual machine manufacturers. Furthermore, it is probable that any commercially successful DVTR (or disk recorder) will necessarily have full bandwidth outputs for both video and audio, since this is the level at which the signals can be edited.

This standard represents a straightforward intraplant distribution solution based on multiplexed 12-channel full-bandwidth distribution and switching between equipment with CODECs (encoder-decoder pairs), such as recorders or STL interface points, where compression may be required for bandwidth preservation. This topology ensures that digital audio signals are coherent and that image accurate audio may be moved throughout the facility with confidence. It also allows for optimal CODEC design to match the bandwidth and error behavior of specific storage and transmission channels.

Six AES3 signals may be easily transported in a single path for only a small increase in complexity over a single AES3 signal. The complexity of the multiplexer and demultiplexer is minimal when compared with separate distribution CODECs. Given the proliferation of 8-track audio recorders, the presence of 8-, 10-, and 12-channel HDTV DVTRs, and the economies associated with multichannel transport, a 12-channel interface standard will dramatically simplify in-plant distribution of multichannel audio.

A.3 Implementation

A.3.1 Multiplexer inputs and format

There are six AES3 inputs that shall be referred to as AES3 1, 2, 3, 4, 5, and 6, each of which has two channels of audio, A and B, as is consistent with the AES3 standard. Each channel of audio contains 24 bits of data, 4 bits of overhead information (C, V, U, and P), and 4 bits of preamble. The bit usage and position in the frame, as well as the preamble codes, are all identical to AES3 specifications.

A.3.2 Demultiplexer outputs and format

There are six AES3 outputs that shall be referred to as AES3 1, 2, 3, 4, 5, and 6, each of which has two channels of audio, A and B, consistent with the AES3 standard. Each channel of audio contains 24 bits of data, 4 bits of overhead information (C, V, U, and P), and 4 bits of preamble. The bit usage and position in the frame, as well as the preamble codes, are all identical to AES3 specifications.

A.3.3 Input/output electrical specifications

The signals should be electrically compatible with AES3, AES-3id, and/or ANSI/SMPTE 276M for both multiplexer inputs and demultiplexer outputs.

A.3.4 Multiplexer transmission format

The multiplexer serial data format calls for transmitting consecutive data frames composed of 12 AES3 data packets and one header packer in the rigid order shown (see figure 2). Note that the AES3 inputs or outputs of a multiplexer or demultiplexer have a rigid position in the transmission frame format. This is a critical economic and operational advantage for manufacturers and users.

Each AES3 packet is a truncated version of the 32-bit AES3 subframe as shown. This packet is 28 bits long, with complete preservation of the entire AES3 subframe data payload. The block start information is moved to the multiplex header for data efficiency. The header packet contains 48 bits (see figure 3). A four-bit preamble is used for multiplexer framing. A single MC bit is used for alignment of multiple data streams ensuring synchronous performance and preservation of audio image across parallel machines with differential latencies of several AES3 frames.

The header contains 4 bytes of channel data, an optional channel block bit, and two reserved bits. A parity bit (CP) sets this group to even parity. The last byte of the header contains the MC bit, the Z bits corresponding to each of the AES3 inputs, and a second parity bit which sets the last byte to even parity.

The Z preamble from each pair of AES subframes is saved as a Z bit for decoding channel status. The Z bit will be 1 at the start of the AES3 block, and 0 for the remaining 191 subframes in the standard AES3 block. This allows accurate recovery of all the channel status information. The Z bit is coincident in time for the AES3 A/B pair since this is a requirement of the AES3 specification. Equipment is required to pass channel status transparently. Also, equipment that processes audio and reinitializes the channel status bits must restripe the Z framing bit in accordance with maintaining the channel pair correlation.

The complete multiplexed frame of data is 384 bits long generating a bit rate of 384 × FS (the audio sampling frequency). For example, 48-kHz audio is transmitted at 18.432 MHz.