Digital Audio Technical Committee Report

MINUTES OF THE MEETING OF THE DIGITAL AUDIO TECHNICAL COMMITTEE

Date: 1980 October 30

Time: 900 hours

Place: Waldorf-Astoria Hotel, New York, NY, USA

Present: K. Bäder (EMT Lahr), B. Blesser (B. Blesser Associates), R. Blinn (Capitol Records), B. Blüthgen (Polygram), P. Burkowitz (Polygram), A. Clegg (Panasonic), A. Conte (SMPTE), Erik DeBenedictus (Consultant, Pioneer), C. Dieterich (RCA Laboratories), T. Doi (Sony), H. Ford (Consultant), M. Fujimoto (JVC), T. Fujino (Yamaha), H. Hadden (Consultant), T. Hidaka (JVC), R. Jaeger (Lexicon), E. Jaffe (Microworks, Consultants), J. Johnston (Bell Labs.), B. Jones (CBS Technology Center), T. Kato (Pioneer), Y. Kimura (Matsushita), T. Kohler (Philips Labs.), M. Komamura (Pioneer), H. Korte (University of Hanover/Sonopress), A. Kurahashi (Matsushita), R. Lagadec (Studer), G. Langdon (AKG Acoustics), B. Locanthi (Pioneer), P. Macdonald (AES Journal), L. Martin (AEG Telefunken), C. Matassa (Consultant), R. McDonough (Harris Corporation), J. McKnight (Magnetic Reference Lab.), T. Mori (JVC), F. Morre (EMI Studios), M. Onishi (Technics/Panasonic), B. Pisha (Audio Magazine), S. Pramanik (Bang & Olufsen), D. Ranada (Stereo Review), P. Rodgers (Rauland-Borg), K. Sadashige (Matsushita), W. Schott (Magnavox), L. Schuweiler (3M), R. Shinokawa (Toshiba), S. Sohma (Totsuco), T. Stockham (DRC-Soundstream), M. Stubbe (IRT, Munich), N. Takahashi (JVC), K. Tanaka (Mitsubishi), H. Tendeloo (Polygram), E. Torick (CBS Technology Center), R. Wartzok (RCA), L. White (Pioneer), B. Whyte (Audio Magazine), R. Youngquist (3M).

After the members had introduced themselves, the Chairman asked J. McKnight and G. Langdon about the Justice Department question raised by the SMPTE relative to official clearance for standards activities for digital audio. The reply was that the legal staffs of the AES and the SMPTE are in communication with each other. Since three U.S. manufacturers are now in the market with digital audio equipment—3M, Soundstream, and MCI—along with Sony and Mitsubishi from Japan, standards activities could probably begin.

Since the SMPTE asked for a representative from the AES Technical Committee, the Chairman asked for a volunteer. R. Youngquist offered his services and was appointed.

The first scheduled item on the agenda was a report from Japan outlining the activities in digital audio in that country. The report was presented by T. Kato of Pioneer and appears on p. 58.

The principal question to this report, raised by T. Stockham, was relative to possible psychoacoustical problems caused by the delay resulting from alternate sampling and also by interfacing problems to other digital audio equipment using parallel sampling. According to T. Kato, critical listening tests show that a delay of about 10 μs caused by alternate sampling cannot be detected. Both T. Stockham and R. Youngquist suggested that T. Kato carry back to the digital audio committee in Japan the feeling that a simple delay circuit be incorporated in the "leading" channel to eliminate the relative delay problem and simplify interfacing to other equipment. It would also eliminate any further discussions about the psychoacoustical matter.

T. Doi presented a brief description of the Sony/Philips format for the compact PCM audio disk, which appears on p. 60.

R. Lagadec presented a brief description of the format for the professional stationary-head recorder jointly supported by Sony and Studer. This report appeared in the *Journal*, vol. 28, no. 9, p. 624 (1980 Sept.).

P. Rodgers presented a short discussion on the progress of her activities in assembling a bibliography (which appears on p. 66) on the matter of maximum perceptible audible bandwidth. She then presented some of her thoughts about the nature of previous measurement methods on bandwidth limits most of which involved monaural sources. It was her feeling that if signals up to 20 kHz were presented to listeners for test purposes, then the 20-kHz levels would have to be higher than the threshold of 100 dB sound pressure level for their presence to be noticed. She further felt that the tests should be done in stereo to preserve the spatial characteristics of natural sounds and that by doing so the thresholds might be more critical. It was also her feeling that one should strive for an upper bandwidth limit of at least 20 kHz.

B. Blüthgen discussed some details of the digital audio Umatic test tape which was sent to 24 laboratories around the world for evaluation of the following:

1) Perceptibility of upper bandwidth limits (15 kHz to 20 kHz).

2) Perceptibility of group-delay distortion (18-kHz cutoff).

3) Perceptibility of passband ripple (0.1 dB versus 0.0001 dB).

4) Perceptibility of aliasing components (10-kHz bandwidth, 140-dB and 60-dB attenuation in stop band).

B. Blüthgen supplied a document explaining the procedure used in making the tapes. Since this report is rather lengthy, we will send copies on request only. It should appear in this *Journal* in the near future.

T. Hidaka presented a brief description of JVC's AHD system which delivers 3-channel stereo plus one channel of video at a rate of about 20 frames per minute. Mr. Hidaka referenced the early work of Snow of the Bell Telephone Laboratories which showed that two-channel stereo with a derived third channel is better than two-channel stereo, and that three-channel stereo is the best. JVC feels that the accompanying slow video adds still another important dimension to home entertainment. A description of the system appears on p. 68.

On other matters we received a document from 3M relative to digital audio signal compatibility for the 3M professional digital recorder (see p. 70).

K. Tanaka of Mitsubishi presented a discussion on the need for data from potential users and manufacturers of digital audio equipment relative to interfacing problems. After Mr. Tanaka's presentation, R. Lagadec indicated that he had a similar set of questions from Europe. An ad hoc committee was formed with Dr. Lagadec as chairman to coordinate the activities of the group on digital interfacing matters. The committee's questionnaire appears on p. 74.

With no further business at hand, the meeting was suspended for lunch at 1215 hours, to be reconvened at 1400 hours at Columbia Records for listening tests to a digital audio tape prepared by Polygram covering the following:

1) Perceptibility of the audible bandwidth limits (15 kHz and 20 kHz).

2) Group delay distortion at 15 kHz (conventional analog filtering versus phase linear filtering).

3) Passband ripple 0.0001-dB reference versus 0.1 dB, 18-kHz cutoff frequency.

4) Aliasing distortion 10-kHz cutoff. Reference signal 140-dB attenuation in passband versus test signal 60-dB attenuation in passband.

The meeting was reopened at 1400 hours at Columbia Records, and the listening tests started at approximately 1500 hours. Since nearly 40 people were located in a small listening area, critical listening tests could not be performed by everyone present.

Several persons near the front of the group were able to hear differences between some reference signals and the corresponding test signals.

Polygram sent digital audio test tapes of the above program to 24 different groups around the world for comment. As soon as the results are available, they will be distributed to this committee. The initial mailing of the test tapes took place in July of this year, with clarifications received toward the end of September. We expect to receive data resulting from critical listening tests by the end of this year.

The meeting was adjourned at approximately 1700 hours.

The Chairman was invited to attend an RIAA meeting on November 3 to present information about digital audio activities in Japan. At that meeting he was presented with a policy statement endorsed by the RIAA engineering committee, a resolution proposed by P. Burkowitz, President of the AES, to wit:

"In view of the increasing evidence of digital technology becoming the successor of existing analog techniques in professional recording and mass media, and in view of the thus ultimately resulting vital need of exchangeability of recorded means, and in view of the foreseeable failure of such means if not applicable and exchangeable worldwide, and in view of the foreseeable application by radio and television stations, it is hereby

Resolved that the Recording Industry Association of America through its technical experts consider the formulation of mutually acceptable user requirements for recorded professional grade and mass media, with due emphasis on the most essential qualities which are:

1) unrestricted interchangeability and
2) technical quality parameters within the confines of economical optimization.

Resolved further that RIAA invite other concerned industry associations to join in this undertaking."

BART LOCANTHI
Chairman

DIGITAL AUDIO ACTIVITIES IN JAPAN

Presented by T. Kato

1 PCM ENCODER-DECODER UTILIZING A VTR

1.1 Problems discussed at the last meeting in Los Angeles

1.1.1 Standpoint for the technical file STC-007.

1.1.1.1 It is originally designed for consumer use.

1.1.1.2 It takes into account the average capability of a home VTR.

1.1.1.3 It considers the reliability and economical efficiency rather than pursuing top-quality performance.

1.1.1.4 From the preceding factors the system adopts 14-bit plus preemphasis and a powerful error-correction scheme.

1.1.1.5 We shall make such a proposal to other organizations concerned, such as the IEC.

1.1.2 Alternate sampling.

1.1.2.1 Alternate sampling is the most attractive for a consumer system using a VCR or a prototype digital audio disk player for economic reasons.

1.1.2.2 It generates no problem if a machine em-

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ploying the same alternate sampling system is used for recording and playback.

1.1.2.3 A master recorder uses simultaneous sampling due to the need for multiple tracks.

Consequently problems may arise if digital copy from the master recorder to another machine is carried out.

However, since it is considered necessary to convert the sampling rates, we see less of a problem if the time (or phase) differences are aligned at the same time.

1.1.2.4 According to our listening test one cannot detect a phase difference of about 10μs.

1.2 Subsequent activities in EIA-J

1.2.1 Examination of a signal format for a 625-line video system.

1.2.1.1 A test format is generated (refer to documents).

1.2.1.2 Accumulation of experimental data using a prototype machine based on this format.

1.2.1.3 It will be documented in the technical file when basic requirements are overcome.

1.2.2 Address code.

1.2.3 Definition of terms in the technical file.

1.2.4 How to present the specifications.

1.2.5 Measurement methods (on signal-to-noise ratio, dynamic range, distortion, etc.).

1.2.6 Possible problems in interfacing home VTRs are under examination.

1.2.6.1 We are planning to improve the STC-007 some time next spring, after analyzing the above results.

2 DIGITAL AUDIO DISK

2.1 Recent activities in DAD conference

2.1.1 The conference was held three times in 1980; the study meeting, WG-1, and WG-4 convened twice.

2.1.2 In WG-4 a test format was generated and was proposed to the study meeting which tried to examine it and at the same time opened the door to further eventual proposals from members.

2.1.3 It resulted in three different proposals from AEG-Telefunken, JVC, and Philips. These three were submitted to the conference meeting.

2.1.4 A steering committee has narrowed down its candidates for a DAD format to these three issues.

2.1.5 The study meeting is now responsible for their examination.

2.2 Activities in study meeting and its schedule

2.2.1 The study meeting, having been requested by the steering committee, started to examine the following items:

2.2.1.1 Clarification of a DAD concept.

2.2.1.2 Reorganization of the working groups and steering committees.

2.2.2 Concerning 2.2.1.1, after examination of the proposed formats, it is confirmed that there exist two different ideas for DAD: one for a compact (small) and audio-oriented system and another for a compatible system with a videodisk player. For the time being we will first look at an audio-oriented system in the DAD conference.

2.2.3 Concerning 2.2.1.2, as WG-4 had accomplished its task, we reorganized three working groups as below.

2.2.3.1 WG-1: Extraction of problems from the software side in DAD industries.

2.2.3.2 WG-2: Examination of productivity and eventual requirements, etc., in a DAD industrialization (for both hardware and software).

2.2.3.3 WG-3: Evaluation of DAD signal formats.

2.2.4 In the study meeting we plan to push these three working groups energetically forward and to make clearer the conditions necessary for a DAD system to grow up as a future industry.

GENERAL INFORMATION ON A COMPACT DIGITAL AUDIO DISK

Presented by T. T. Doi

Sony Corporation/N.V. Philips

1 DISK (Fig. 1)

Playing time, single side, 2 channels	Approximately 60 min
Scanning velocity (2 channels)	1.2–1.4 m/s
Sense of rotation seen from reading side	Counterclockwise
Track pitch	1.6 μm
Disk diameter	120 mm
Disk thickness	1.2 mm[1]
Diameter of center hole	15 mm
Starting diameter of program area	50 mm

2 SIGNAL FORMAT

Number of channels	2 and/or 4
Quantization, per channel	16 bits linear
Encoding	2's complement
Sampling frequency	44.1 kHz
Error-correction code	CIRC[2]
Channel modulation code	EFM[3]
Channel bit rate	4.3218 Mb/s

[1] Double-sided disk optional.
[2] *C*ross *I*nterleave *R*eed Solomon *C*ode.
[3] *E*ight-to-*F*ourteen *M*odulation.

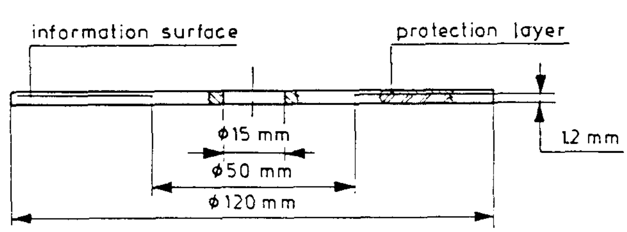

Fig. 1. Dimensions of disk.

3 FRAME FORMAT

	Data Bits	Channel Bits
Synchronization		24
Control and display	8	14
24 data symbols	192	336
8 error-correction symbols	64	112
Merging and low-frequency suppression		102
Total frame		588

4 OPTICAL STYLUS (Fig. 2)

The wavelength λ and the numerical aperture NA have to fulfill the requirement

$$\frac{\lambda}{\text{NA}} \leq 1.75\ \mu\text{m}.$$

The stylus should be diffraction limited, and the information is viewed through a transparent plane parallel plate of 1.2-mm thickness (refractive index $\approx$ 1.5).

The system is optimized for a wavelength of 0.78 μm (e.g., laser wavelength of AlGaAs). The depth of focus of the optical stylus is $\pm 2\ \mu$m.

The method of radial tracking is differential, and the method of high-frequency detection is integral.

5 MODULATION SYSTEM

The NRZ signals from the A/D converter and the error-correction parity generator may have a high dc content and are not self-clocking (the run length is not limited).[4] Therefore they cannot be used on the disk. The signals have to be converted into another code which should meet some special requirements.

5.1 Requirements

5.1.1 Clock Content. The bit clock must be regenerated from the signal after readout. Therefore the signal must have a sufficient number of transients and the maximum run length must be as small as possible.

5.1.2 Correct Readout at High Information Densities. The light spot with which the disk is read out has finite dimensions. These dimensions give rise to intersymbol interference. This effect can be minimized by making the minimum run length as large as possible. However, too large a value has a negative influence on the clock content of the signal.

5.1.3 Servo. The modulation code must be dc free, because the low frequencies of the spectrum give rise to noise in the servo systems.

5.1.4 Error Propagation. The error propagation of the modulation system must be as small as possible.

5.2 Eight-to-Fourteen Modulation Code (EFM)

5.2.1 Each block of 8 data bits is mapped onto 14 channel bits. To each block of 14 channel bits 3 extra bits are added, 2 bits for merging the blocks and 1 redundancy bit for LF suppression.

5.2.2 The information is contained in the positions of the transients. For mapping 8 data bits 256 combinations of channel bits are needed.

5.2.3 The code is generated in such a way that the minimum distance between 2 transients is 3 channel bits ($\approx$ 1.5 data bits), and the sampling window or eye pattern is 1 channel bit ($\approx$ 0.5 data bit). This yields a good compromise between intersymbol interference and clock accuracy (phase jitter). The maximum run length within the blocks is 11 channel bits ($\approx$ 5.5 data bits). An example is shown in Fig. 3.

5.2.4 Since the extra 3 bits do not contain any information, an extra transient may be inserted in these bits. In this way the maximum run length $T_{\max}$ in two successive blocks and the dc content of the frequency spectrum can be controlled.

5.2.5 The modulator and demodulator can be realized with a lookup table in a ROM.

5.2.6 Because of the block structure this modulation code is extremely suitable for use in conjunction with the error-correction system, whose operation is based on 8-bit blocks.

5.3 Frame Format

Because the system must be self-clocking, synchronization is necessary. Therefore the data stream is split up into frames. Each frame contains:

1) A synchronization pattern of 24 bits
2) 12 data words of 16 bits each
3) 4 error-correction parity words of 16 bits each
4) A control and display symbol of 8 bits.

The data and error-correction words are each split up into two 8-bit blocks, which are fed into the modulator circuit. After modulation each block is converted into 3 + 14 channel bits.

[4] NRZ—*N*on-*R*eturn to *Z*ero coding; A/D—*A*nalog to *D*igital; run length—distance between transients in the signal.

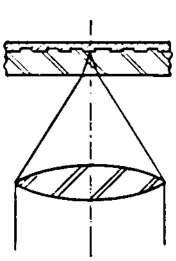

Fig. 2. Optical stylus.

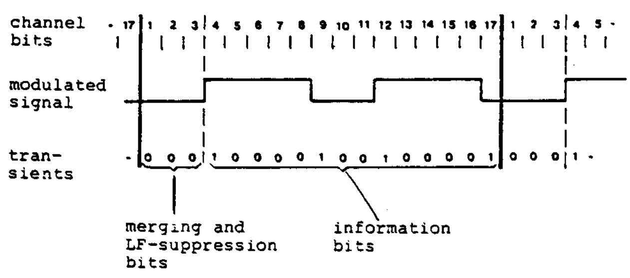

Fig. 3. Modulation code.

The total number of channel bits per frame is:

Sync pattern	24	channel bits
Control and display	1 × 14	channel bits
Data	12 × 2 × 14	channel bits
Error correction	4 × 2 × 14	channel bits
Merging and LF suppression	34 × 3	channel bits
Total	588	channel bits

6 ERROR-CORRECTION SYSTEM

An efficient error-correcting system, named CIRC, has been developed with different decoder strategy possibilities. A simple 4-frame correction to a more complex 16-frame correction is possible, keeping full compatibility.

6.1 Requirements

1) High random error correctability
2) Long burst error correctability
3) In case burst correction is exceeded, a graceful degradation
4) Simple decoder strategy possibility with reasonably sized external random access memory
5) Redundancy as low as possible (not much parity should have to be added)
6) Possibility for future introduction of four audio channels, without changes in the decoder chip.

6.2 Cross Interleave Reed Solomon Code (CIRC)

6.2.1 The code corrects most errors that occur on the disk. However, some error patterns are not correctable.

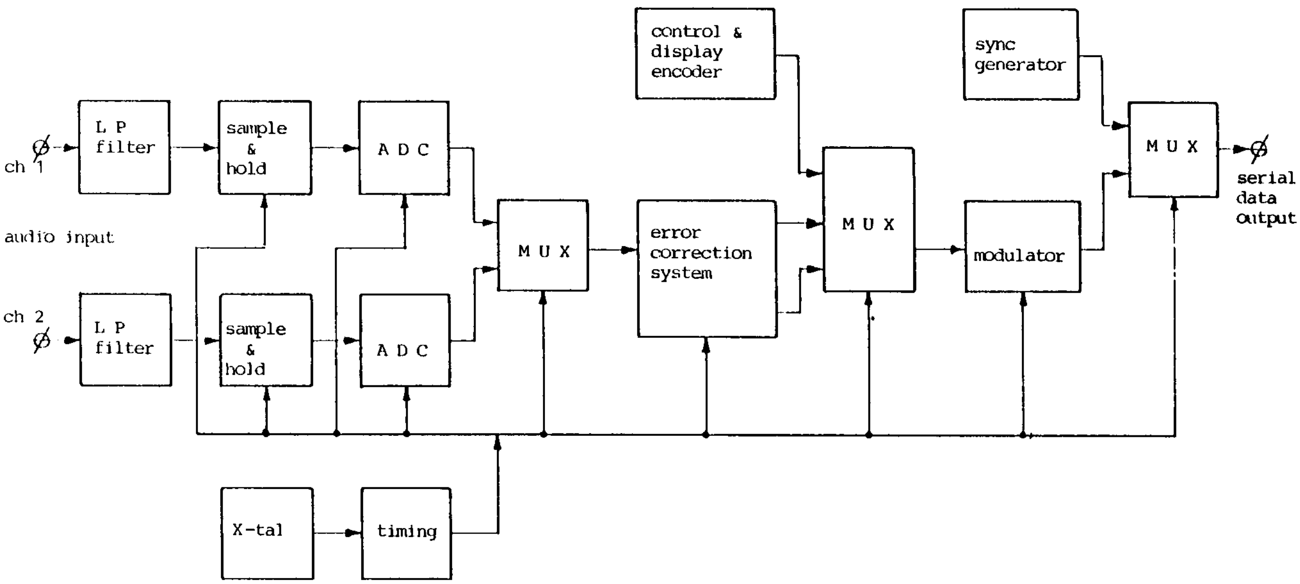

Fig. 4. Encoding system. MUX—time multiplexer; ADC—analog-to-digital converter.

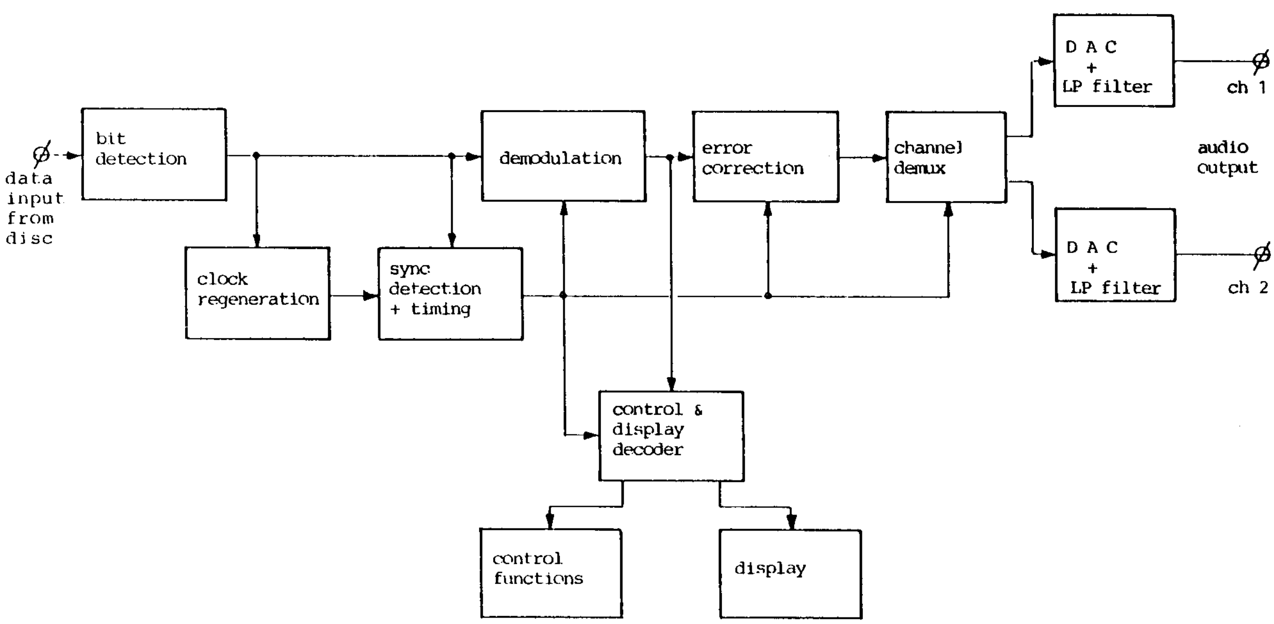

Fig. 5. Decoding system.

In this situation the error is detected and the decoder reconstructs the sample value by means of interpolation.

6.2.2 The performance of the CIRC is such that 1000 samples per minute (out of 2.6 million samples per minute) will have to be interpolated at 10^{-3} BER. If the BER is 10^{-4}, only 1 sample per 10 hours will have to be interpolated. However, an average BER of 10^{-5} is typical.

6.2.3 Since the probability that an uncorrectable error is not detected is nonzero, which may lead to a click, the detection capability of the code was designed to ensure less than 1 click per month at 10^{-3} BER.

6.2.4 A disk that is handled very roughly might have scratches. Because of that the code should be capable of dealing with long burst errors. CIRC can fully correct burst errors up to 4000 bits (2.5 mm).

6.2.5 The decoder complexity of the CIRC has been reduced considerably by splitting up the decoder into two main parts:

1) A special-purpose decoder LSI
2) Standard 2k words of 8 bits.

6.2.6 CIRC has an efficiency of 3/4, which means that 3 data bits will result in 4 bits after encoding.

6.2.7 The signal format has been designed in such a way that 4 channels are possible in the future, without changes in the decoder chip.

6.3 CIRC Encoder and Decoder (Figs. 6 and 7)

The CIRC consists of a C1 and a C2 Reed Solomon code as follows: C1 is a (32, 28) Reed Solomon code over GF[5] (2^8), and C2 is a (28, 24) Reed Solomon code over GF (2^8). The horizontal blocks between C1 and C2 represent 8-bit-wide delay lines of unequal lengths (interleaving). Before the C2 encoder a delay of one symbol is inserted in the even words to facilitate concealments in simplified decoder versions. After the C1 encoder a delay of one symbol (8 bits) is inserted in the even symbols (scrambling).

The decoder operates as follows: The C1 decoder accepts 32 symbols of 8 bits each from which 4 parity symbols are used for C1 decoding. The parity is generated according to the rules of Reed Solomon coding, and because of that the C1 decoder is able to correct a symbol error in every word of 32 symbols. If there is more than one erroneous symbol, then regardless of the number of errors, the C1 decoder detects that it has received an uncorrectable word. If this is the case, it will let all 28 symbols pass through uncorrected, but an erasure flag is set for each symbol to mark that all symbols from C1 are unreliable at that moment.

Because the delay lines between the C1 and the C2 decoders are of unequal lengths, the symbols marked with an erasure flag at one instant arrive at different moments at the C2 decoder input. Thus the C2 decoder has for every symbol an indication whether it is in error or not. If a symbol does not carry an erasure flag it is error free. If no more than 4 symbols carry an erasure flag, then the C2 decoder can correct a maximum of 16 frames.

In cases that even the C2 decoder cannot correct, it will let the 24 data symbols pass through uncorrected, but marked with the erasure flags that had originally been given out by the C1 decoder.

[5] Galois field.

7 AUDIO PERFORMANCE

Frequency response	20–20 000 Hz
Quantization, per channel	16 bits linear
Signal-to-noise ratio	>90 dB
Dynamic range	>90 dB
Channel separation	>90 dB
Harmonic distortion	<0.05%
Wow and flutter	equal to crystal oscillator accuracy

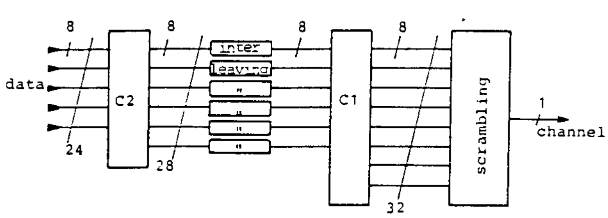

Fig. 6. Encoder.

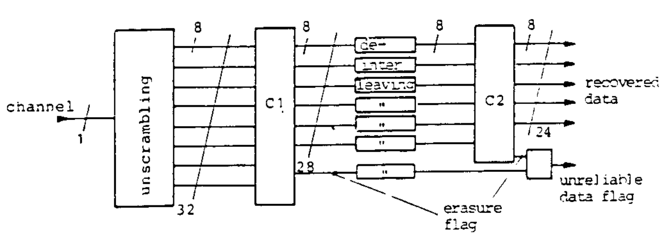

Fig. 7. Decoder.

BIBLIOGRAPHY

Puddie Rodgers

1 THRESHOLD STUDIES

1.1 Early Studies

L. J. Sivian and S. D. White, "On Minimum Audible Sound Fields," *J. Acoust. Soc. Am.*, pp. 288–321 (1933). (*Note:* this article provides an excellent bibliography for all of the earlier threshold studies.)

1.2 Current Studies

S. A. Fausti, R. H. Frey, D. A. Erickson, B. Z. Rappaport, and E. J. Cleary, "A System for Evaluating Auditory Function from 8000–20 000 Hz," *J. Acoust.*

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Soc. Am., vol. 66, pp. 1713–1718 (1979).

J. D. Harris and C. K. Meyers, "Tentative Audiometric Threshold Level Standards from 8 to 18 kHz," *J. Acoust. Soc. Am.*, vol. 49, pp. 600–608 (1971).

J. L. Northern, M. P. Downs, W. Rudmose, A. Glorig, and J. L. Fletcher, "Recommended High-Frequency Audiometric Threshold Levels (8000–18 000 Hz)," *J. Acoust. Soc. Am.*, vol. 52, pp. 585–595 (1972).

2 BANDWIDTH ARTICLES

2.1 Early Studies

D. K. Gannett and I. Kerney, "The Discernibility of Changes in Program Band Width," *Bell Sys. Tech. J.*, pp. 1–10 (1944 Jan.).

W. B. Snow, "Audible Frequency Ranges of Music, Speech, and Noise," *J. Acoust. Soc. Am.*, pp. 157–166 (1931).

2.2 Current Studies

T. Muraoka, M. Iwahara, and Y. Yamada, "Examination of Audio-Bandwidth Requirements for Optimum Sound Signal Transmission," presented at the 64th Convention of the Audio Engineering Society, New York, 1979 November 2–5, preprint 1545.

G. H. Plenge, H. Jakubowski, and P. Schöne, "Which Bandwidth Is Necessary for Optimal Sound Transmission?" presented at the 62nd Convention of the Audio Engineering Society, Brussels, 1979 March 13–16.

H. Tanabe and T. Fujita, "Upper Limit Frequency For Program Transmission," pp. 231–232; ASJ, (1979 June).

3 RELATED PHYSIOLOGY ARTICLES

G. A. Manley, "Some Aspects of the Evolution of Hearing in Vertebrates," *Nature*, vol. 230, pp. 506–509 (1971 Apr.).

B. Masterton, H. Heffner, and R. Ravizza, "The Evolution of Human Hearing," *J. Acoust. Soc. Am.*, vol. 45, pp. 966–985 (1969).

V. Bruns, "Function Anatomy as an Approach to Frequency Analysis in the Mammalian Cochlea," *Verh. Dtsch. Zool. Ges.*, pp. 1–73 (1979).

A. M. Brown and J. D. Pye, "Auditory Sensitivity at High Frequencies in Mammals," in *Advances in Comparative Physiology and BioChemistry*, vol. VI (Academic Press, New York, 1975), pp. 1–73.

P. Dallos, *The Auditory Periphery—Biophysics and Physiology* (Academic Press, New York, 1973); pp. 448–460 discuss subharmonics and fractional harmonics.

4 OTHER RELATED ARTICLES OF INTEREST

R. J. Pumphrey, "Upper Limit of Frequency for Human Hearing," *Nature*, p. 571 (1950).

B. H. Deatherage, L. A. Jeffress, and H. C. Blodgett, "A Note on the Audibility of Intense Ultrasonic Sound," *J. Acoust. Soc. Am.*, vol. 26, p. 582 (1954).

J. F. Corso, "Pitch Discrimination in Air Conduction and by Bone Conduction for Sonic and Ultrasonic Frequencies," *J. Acoust. Soc. Am.*, vol. 35, p. 804 (1963).

J. F. Corso, "Bone Conduction Thresholds for Sonic and Ultrasonic Frequencies," *J. Acoust. Soc. Am.*, vol. 35, pp. 1738–1743 (1963).

JVC AHD DIGITAL AUDIO DISK SYSTEM

Presented by T. Hidaka

0 INTRODUCTION

JVC has been engaged in the research and development of the AHD (audio high-density) digital audio disk system, compatible with the VHD (video high-density) video disk player. This newly developed AHD system has 3-channel stereo capabilities along with a still-picture playback function.

In June of this year, at the Digital Audio Disk (DAD) Conference where various systems are studied, JVC proposed its AHD system as the system upon which a standard could be based.

1 OBJECTIVES OF DEVELOPMENT

The digital audio disk, which employs the latest advances in video disk and digital audio technology is attracting the attention of the industry as an entirely new hi-fi stereo disk which is known for its wide dynamic range and freedom from noise and distortion.

To improve the total quality of sound reproduction through signal digitalization and to provide various ways to enjoy both audio and video programs, JVC engaged in research and development of the AHD digital audio disk system, which derives its standard programs from 3-channel stereo music and still-picture playback.

2 FEATURES OF JVC'S AHD DIGITAL DISK SYSTEM

1) This 3-channel stereo system with a wide dynamic range, free of distortion and noise, features super hi-fi reproduction over an optimum hearing range and gives the user the aural perception of the precise direction and movement of each sound source.

2) The AHD digital disk system is capable of playing back high-quality still pictures or visual information on the TV screen through digital transmission.

3) The AHD digital disk system is capable of playing either 3-channel or 2-channel stereo music in addition to the option of playing back still video pictures simultaneously. This system therefore provides a wide selection of entertainment programs through both audio and video playback functions.

4) Quick and exact search of recorded music and also the location of the beginning of any specified music program are made possible by electronic tracking.

5) The manufacturing of the AHD disk can be ac-

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complished by applying VHD disk manufacturing processes. At home the user can also apply the same player for both AHD and VHD disks.

3 AHD DIGITAL DISK SYSTEM SPECIFICATIONS

Pickup method	Grooveless electro-tracking capacitance pickup system
Disk size	260 mm (10 in)
Revolutions	900 r/min, the same in PAL and SECAM countries
Playing time	2 hours (1 hour per side)
Number of channels	4 (3 audio channels and 1 still-picture channel)
Quantization	16 bit linear
Sampling rate	47.25 kHz
Picture transmission method	Digital

INFORMATION FOR DIGITAL AUDIO SIGNAL COMPATIBILITY

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and

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0 INTRODUCTION

The following information is being made public in an effort to facilitate signal compatibility between present digital audio equipment and that which will become available to recording studios in the future.

3M herein provides the necessary technical interfacing data in hopes that by doing so, potential manufacturers of multitrack recorders and digital consoles, when designing their equipment, will be able to meet the realistic need of studios to be able to transfer signals in the digital domain between audio recorders and consoles of different brands.

While adoption of a universal machine format standard remains a more distant possibility, establishment of a signal standard at an early date would represent a major step forward and one which could head off potential future problems for studios around the world.

1 STANDARD SIGNAL INTERFACE FOR DIGITAL AUDIO SYSTEMS

In order to allow digital audio systems to interface with one another, it is necessary to standardize the systems in several areas. These areas are as follows:

1) Sampling rate.
2) Digital code to represent data.
3) Serial or parallel format.
4) System to allow for small variations and time delays in the transmission system.
5) Provisions for transmitting extra bits for other than audio data. These can be such things as transmission parity bits, good/bad data indicators, optional format indicators, or user assigned bits.
6) Means of locking the systems together with a common clock.
7) Driver–receiver configuration and impedance.

The following standards are recommended to accomplish the above:

1) Sampling rate to be 50 kHz.
2) Data code to be 16-bit 2's complement.
3) Serial data transmission with MSB first and LSB last.
4) Provide quadrature clocks for outputting and inputting data to prevent time delays causing erroneous data reception.
5) Provide space in the serial data stream for up to 9 extra data bits.
6) Two standard clocks to synchronize systems:
   a) 2.5 MHz synchronizing clock which can be divided by 2 for transmission and reception clock.
   b) 50 kHz clock to synchronize word rates.
7) Use 50-Ω line drivers and receivers.

Fig. 1 illustrates how these clocks are used and the phase relationships between them.

A—2.5 MHz clock supplied to systems for synchronization.
B—50 kHz [2.5 MHz (A) ÷ 50]
C—Data Cells

Cell Number	Use	Notes
1–16	D_{15}–D_0	D_{15}–D_0 is 16-bit 2's complement data; D_{15} = MSB, D_0 = LSB
17–25	Optional	
Suggested Options:		
17–20	Extra data bits (20-bit code possible)	
21	Signal data parity bit	
22	Good/bad data	
23–25	Format or user option	

Note: Data valid for entire cell

D—1.25 MHz [A ÷ 2] data transfer clock.
E—Not taking data.
F—Clock for data transmission.
G—Clock for data reception.

2 50-kHz COMPATIBILITY AND SIGNAL GENERATION

A sampling frequency of 50 kHz is recommended as the best overall compromise to allow compatibility with television, film, and other currently used digital audio transmission systems.

In the case of PAL and NTSC television systems, the lowest common denominator is 2.25 MHz. Dividing this by 144 gives the PAL horizontal sweep rate, by 143 it gives the NSTC horizontal sweep rate, and by 45 it gives us a 50-kHz audio sampling rate. In the proposed interface standard it is recommended that a phase-lock loop be used to convert 2.25 MHz to 2.5 MHz to be used for master synchronization. A block diagram of this loop is shown in Fig. 2. The VCO is run at 22.5 MHz and divided by 9 to generate 2.5 MHz for the master recorder clock, and divided by 10 to give 2.25 MHz. This 2.25 MHz is phase compared to the television system 2.25 MHz, and the resulting error signal controls the 22.5-MHz VCO.

In the case of the EBU 32-kHz transmission system it is necessary to go through a sampling rate converter. The relationship of 50 kHz to 32 kHz is 25 to 16, and it is relatively easy to make a phase-lock loop to generate a sampling rate clock. The conversion of the digital data, although requiring quite a few digital multiplications, is felt to certainly be acceptable.

In the case of motion picture film, at 24 frames per second, there are 6250 words per three picture frames, and again synchronization is possible.

Fig. 3 is a block diagram of the system to derive the necessary machine synchronizing and data transfer clocks. The 2.5 MHz is supplied to the recorder as a master reference clock. The word synchronizing signal is obtained by dividing the master clock by 50.

Data transmission and reception clocks are obtained by first dividing 2.5 MHz by 2 to give 1.25 MHz. The 50-kHz signal is made to be exactly sixteen 1.25-MHz clock cycles long by the 16-cycle timer, and this signal is then not taking data. By appropriately gating this signal with the 1.25-MHz clock, data transmission and data reception clocks are generated. If optional data are also to be transferred, the timer is expanded to include the total number of bits to be transferred.

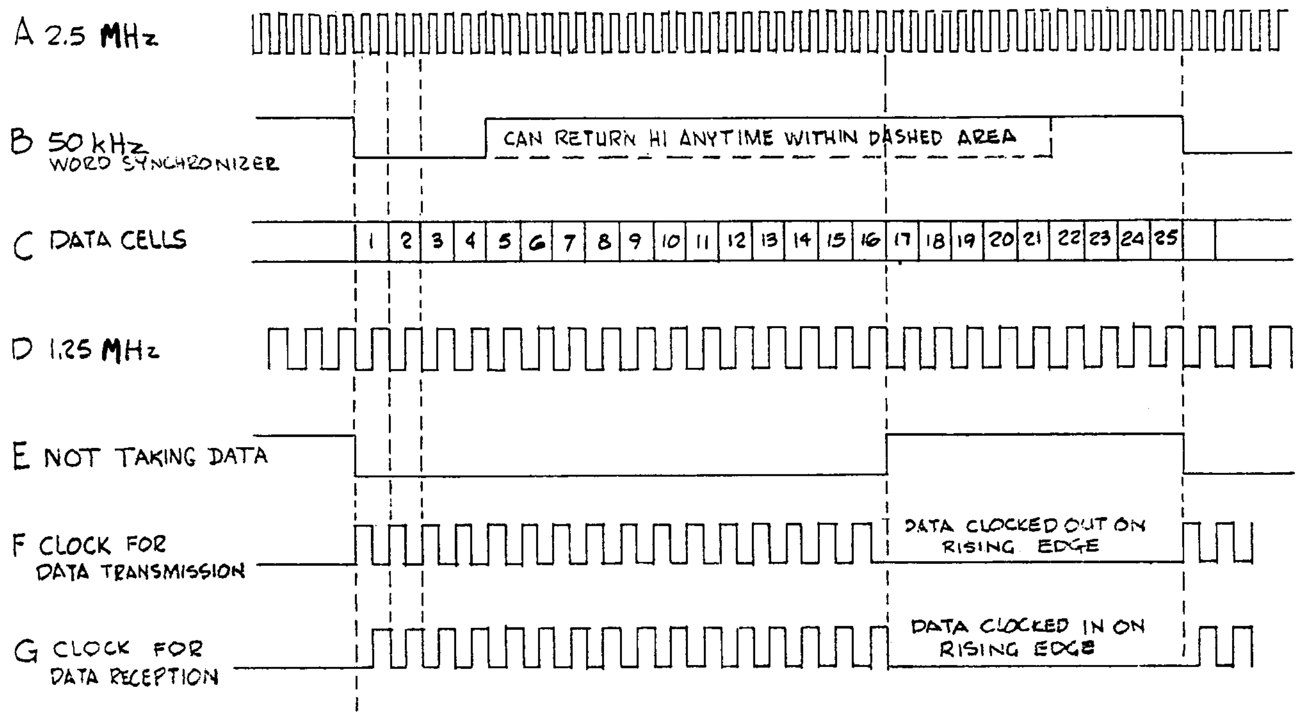

Fig. 1. Timing diagram for digital transfer.

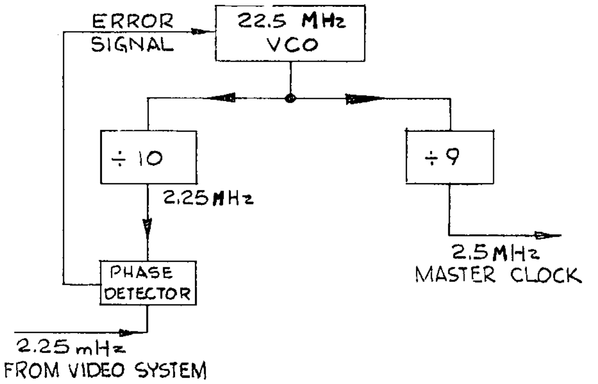

Fig. 2. Block diagram of phase-lock loop used to obtain synchronization of sampling frequency.

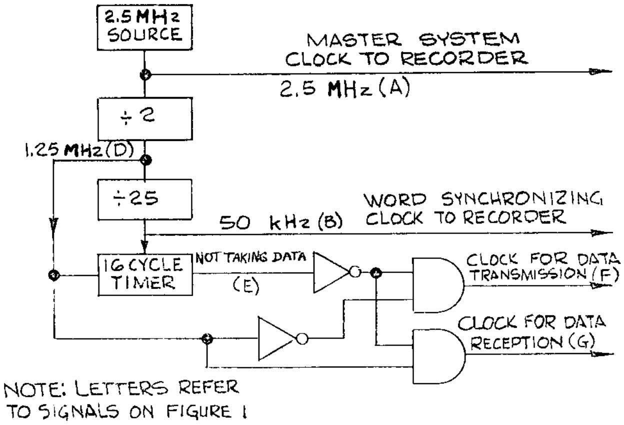

Fig. 3. Block diagram of system for machine synchronizing and data transfer clocks.

QUESTIONNAIRE ON USER REQUIREMENTS FOR A DIGITAL INTERFACE FOR THE LOCAL INTERCONNECTION OF DIGITAL AUDIO EQUIPMENT

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AES Technical Committee on Digital Audio/ad hoc Group on the Digital Interface

0 INTRODUCTION

At the meeting of the Digital Audio Technical Committee of the Audio Engineering Society on the occasion of the 67th AES Convention in New York, a number of participants submitted questions relating to the requirements to be met by a digital interface for the interconnection of audio PCM equipment. It was decided that a questionnaire would be jointly elaborated by an ad hoc working group and submitted to the AES *Journal* for publication. The aim of the questionnaire is to ensure an early discussion of the problems relating to the digital interface, and to help pave the way for a possible future universal digital audio interface.

In addition to its publication in the AES *Journal*, this questionnaire will also be directly mailed to all the participants in the meetings of the Technical Committee. It is hoped that some answers will be returned in time to serve as a basis for further discussions at the next meeting of the Technical Committee in Hamburg in 1981 March. In the very important emerging area of digital audio, where many of the concepts have not yet been finalized, the broadest possible cooperation and dialogue between users and manufacturers is essential.

The digital audio community is already confronted with the lack of compatibility between many digital audio products and with the impossibility of interconnecting them freely in the digital domain. The problem of tape interchangeability among digital audio recorders is being studied separately, and this questionnaire refers exclusively to the problems of digitally interconnecting audio PCM equipment. Although long-distance connections can be envisaged in the future, it is felt that today's emphasis should be on the local interconnection of equipment, for instance, within a recording studio or a production center. Thus the questionnaire refers to the requirements for a digital interface for local interconnections, and to the possible implementation of these functions.

The following questionnaire, by necessity, covers a wide range of both technical and operational aspects of the digital interface. Some questions will be thought too technical, or too vague, or even biased. We hope that those answering the questionnaire will provide not only direct answers, but also add their observations, comments, and criticisms. Answers to a few of the questions is far more useful than no answer at all. Requests for clarification will help us ask better questions in the future. Please feel free to answer the questionnaire in your own way, and to the extent you think necessary.

0.1 Terminology. There is at present no precise definition of the functions of a digital interface for digital audio. Obviously, an essential function is to allow the transfer of *digital audio samples*. For a number of reasons (synchronization, flag bits, possibly error protection, user-definable data, etc.), additional data will have to be transmitted along with the digital audio samples. In this questionnaire, these additional data transmitted along with the digital audio samples will be called *auxiliary audio data*. Further, and as with today's analog equipment, information relating to equipment status and control will have to be exchanged. In this questionnaire, this information relating to equipment status and control will be called *control data*.

Finally, in digital audio, the data streams must be synchronized, and the digital audio samples, auxiliary audio data, and control data may have to refer to the same fixed time reference. In this questionnaire, these functions will be called *synchronization functions*.

0.2 Objectives. It may be useful to list a number of goals which we hope may be achieved in the future by a digital audio interface. We need a simple, single-channel digital interface for digital audio, but also a hierarchy of interfaces allowing multichannel interconnection over a small number of cables. We need bidirectional exchange of audio data and control data. Synchronization functions are necessary, including fast and reliable recovery in case of errors. The digital interface should presumably accommodate different sampling frequencies. Finally, it should permit (with additional circuitry, if required) the interconnection of equipment operating synchronously, semisynchronously, and even with different sampling rates.

1 PRELIMINARY QUESTIONS

1.1 Do you consider it necessary or unnecessary to try to harmonize user requirements relating to a digital audio interface?

1.2 Which time frame do you consider realistic for the introduction of a harmonized digital audio interface?

2 APPLICATIONS

The types of applications envisaged for digital audio may have a direct impact on the features considered necessary for the digital audio interface.

2.1 What application(s) do you have in mind or do you consider important?

2.2 Which items of digital equipment will be digitally interconnected in your applications, and do you have any particular priorities?

2.3 Which maximal distance for the local interconnection of digital audio equipment do you consider acceptable in your application without digital repeaters?

2.4 Which configuration of equipment interconnections do you envisage? Can you provide a tentative

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block diagram?

3 DIGITAL AUDIO SAMPLES

3.1 Which word length(s) do you recommend for the digital audio interface?

3.2 Can a fixed number of bits be assigned to the digital samples?

3.3 Are there any reasons for moving away from PCM samples in a binary 2's complement code?

4 AUXILIARY AUDIO DATA

4.1 What auxiliary audio data should be accommodated in the digital audio interface? Instances of auxiliary audio data functions might include emphasis, companding, word length indicators, synchronization words, error flags indicating erroneous samples, CRC or parity bits for protection against transmission errors, freely definable user data, etc.

4.2 Which applications do you have in mind for the freely definable user data?

4.3 Which auxiliary audio data bits should accompany every single audio sample?

4.4 Which auxiliary audio data bits can occur less frequently, and what would be the appropriate data rates?

4.5 How many bits do you recommend for the auxiliary audio data functions?

4.6 In which position should the auxiliary audio data be transmitted with respect to the digital audio samples?

5 CONTROL DATA

5.1 What control information do you consider essential to your applications?

5.2 Do you recommend separate or common transmission of audio data on the one hand and control data on the other hand?

5.3 If a separate transmission of control data is preferred, do you know an existing system for the exchange of control information that might be adequate to your application and to the needs of digital audio?

6 SYNCHRONIZATION

Synchronization will be an essential aspect of interconnecting digital audio equipment. In order to ensure the reliable exchange of information, the data streams must be synchronized to some local or external reference.

6.1 Do you recommend a separate transmission of synchronization signals?

6.2 Instead of a separate transmission of synchronization signals, do you recommend extracting the synchronization information from the data stream of the digital interface?

6.3 Do you recommend synchronism between digital audio data and control data, and to what specifications?

7 MODE OF TRANSMISSION

Depending on the requirements of data rate, transmission distance, simplicity of interconnection, type of synchronization, number of cables, etc., digital data can be transmitted in either serial or parallel mode.

7.1 Do you recommend a serial or parallel transmission of digital audio data?

7.2 Do you recommend a serial or parallel transmission of control data?

8 SERIAL TRANSMISSION OF AUDIO DATA

Under the assumption that a serial transmission of audio data is chosen:

8.1 Serial data transmission can take place in a variety of ways: individual words or blocks may be transmitted with guard spaces for separation; data transmission can be continuous; synchronization bits can indicate the beginning of either words or blocks; etc. Which transmission format do you recommend?

8.2 Which sequence of bits do you recommend for each word or block?

8.3 Do you recommend the use of a modulation (channel) code? If yes, which one?

8.4 Do you recommend protection against transmission errors? Should automatic correction of some transmission errors be possible?

8.5 What might be the requirements for error detection and correction?

8.6 Which type of cable would you recommend for the digital audio interface, and up to what maximum transmission length (coaxial cable, twisted pairs, standard audio frequency balanced pairs, optical fibers, etc.)?

8.7 Do you think several different types of cables could be used for the same applications?

8.8 Which type of connector(s) do you recommend?

8.9 What are your requirements relating to electromagnetic interference and impedance mismatching?

8.10 What are your recommendations concerning data timing, timing tolerances, maximal timing jitter, guard spaces if any, of the signals transmitted over the digital interface?

9 PARALLEL TRANSMISSION OF AUDIO DATA

Under the assumption that parallel transmission of audio data is chosen:

9.1 What organization do you recommend for the parallel data transmission?

9.2 What ordering of bit lines do you recommend?

9.3 Do you recommend the use of a modulation (channel) code? If yes, which one?

9.4 Do you recommend protection against transmission errors? Should automatic correction of some transmission errors be possible?

9.5 What might be the requirements for error detection and correction?

9.6 Which type of cable do you recommend for the parallel data transmission, and up to what maximum transmission length?

9.7 What are your requirements concerning electromagnetic interference and impedance mismatching?

9.8 What are your recommendations concerning data timing, timing tolerances, maximum timing jitter, etc.?

9.9 Which type of connector(s) do you recommend?

10 SERIAL TRANSMISSION OF CONTROL DATA

Under the assumption that serial transmission of control data is chosen:

10.1 Which transmission format do you recommend (separated words or blocks, continuous transmission, etc.)?

10.2 What data rate do you consider adequate?

10.3 Do you have recommendations relating to equipment addressing, control instruction format, synchronization, protection against transmission errors, and timing requirements?

10.4 Do you have any recommendation concerning the protocol governing the interchange of control data?

10.5 Do you have recommendations concerning the connectors and cables suitable for interchanging control data serially?

10.6 Do you know an existing system for the serial interchange of control data which might be adequate for digital audio?

11 PARALLEL TRANSMISSION OF CONTROL DATA

Under the assumption that parallel transmission of control data is chosen:

11.1 What data rate would you consider adequate?

11.2 Do you have recommendations concerning equipment addressing, control instruction format, synchronization, protection against transmission errors, and timing requirements?

11.3 Do you have any recommendation concerning the protocol governing the interchange of control data?

11.4 Do you have recommendations concerning the connectors and cables suitable for the parallel interchange of control data?

11.5 Do you know an existing system for the parallel interchange of control data which might be adequate for digital audio?

12 INTERCONNECTION HIERARCHY

12.1 Do you recommend a hierarchy of digital audio interfaces making it possible to transmit several digital audio channels over a single cable?

12.2 What numbers of channels should be accommodated in this hierarchy?

12.3 Which types of cables and connectors would you recommend for the different numbers of digital audio channels?

13 SIZE AND COMPLEXITY

13.1 What size and complexity do you consider acceptable for a single-channel digital interface?

14 OMISSIONS

14.1 Apart from the economic aspects, which cannot be the subject of questions in a questionnaire submitted to the AES, have important aspects of the digital interface been omitted from this document?

15 DISTRIBUTION OF ANSWERS TO THIS QUESTIONNAIRE

15.1 Do you agree that your answers to this questionnaire be distributed by and published in the AES *Journal*?

15.2 If yes, do you prefer to withhold the name of your company or organization?

16 CONTACT ADDRESSES FOR CLARIFICATIONS RELATING TO THIS QUESTIONNAIRE

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17 ANSWERS TO THIS QUESTIONNAIRE SHOULD BE SENT TO:

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