Digital Audio Technical Committee Report

MINUTES OF THE MEETING OF THE DIGITAL AUDIO TECHNICAL COMMITTEE

Date: 1981 May 10

Time: 0930 hours

Place: Los Angeles Hilton Hotel

Present: H. Arisaka (JVC), R. Blinn (Capitol), B. Blüthgen (Polygram), K. Dauphinee (ABC-TV), K. Davies (SMPTE), T. Doi (Sony), J. Gibson (RCA & SMPTE), D. Gravereaux (CBS), T. Griffiths (Decca), A. Heaslett (Ampex), J. Kawada (JVC), G. Keller (Harris Corp.), T. Kogure (Matsushita), T. Kohler (Philips), D. McEwen (Ampex), C. Matassa (Consultant), F. Morrison (Ampex), K. Sadashige (Matsushita), P. Samson (Systems Concepts), L. Schuweiler (3M), T. Stockham, Jr. (Digital Recording Corp.-Soundstream), A. Weisser (Tele-Diffusion de France), M. Windram (Independent Broadcasting Authority, UK), R. Youngquist (3M).

In order to interface with two other organizations which will be large users of professional digital audio recording equipment, official representatives from the EBU (European Broadcast Union) and the SMPTE (Society of Motion Picture and Television Engineers) attended this meeting.

1. R. Youngquist reviewed the U.S. proposal to the IEC WG16 group meeting in Prague, March 2–6 (his report is Appendix 1).

2. B. Blüthgen reviewed the AES Technical Committee Meeting in Hamburg, March 16, and the associated ad hoc meetings which dealt with input/output interfacing and sampling frequency considerations. Reports of these meetings appeared in the *Journal*, vol. 29, no. 6, p. 444 (1981 June).

3. B. Blüthgen reviewed the Los Angeles ad hoc meeting of May 9th, which was a continuation of the Hamburg ad hoc meetings.

4. B. Locanthi read a report from Japan concerning the status of the DAD committee recommendations and other digital matters from that country (see Appendix 2).

5. B. Locanthi reviewed digital audio machines now in the marketplace, and some soon to be in the marketplace, discussing items of compatibility and noncompatibility (see Appendix 3).

The meeting was adjourned for lunch at 1215 hours.

6. At the request of the Chairman, Mr. Dan Gravereaux of CBS Laboratories presented a brief description of their compatible companding (CX) system proposed for new phonograph recordings. This system makes it possible for one to play back a CX encoded phonograph record without a decoder in a manner that has been acceptable to many listeners. For some classical recordings the decoder is necessary.

Clearly, this is a reaction of a record company to the development of digital audio. The CX system using a decoder is said to have a signal-to-noise ratios of 80–85 dB. Digital audio may be responsible for improvements in analog recording, and such improvements may be responsible for maintaining digital audio standards at a high level.

The remainder of this meeting commencing at 1430 hours was an extension of the ad hoc meeting of May 9, and the following is an attempt to put the two Los Angeles meetings in perspective.

With no presently agreed upon common sampling frequency for digital audio recording machines it will be necessary:

1) To provide the digital audio data in a common format so that it will be possible for one machine to communicate with another (i.e., one recorder to another), one recorder to a digital recording console, a digital audio tape through a digital recorder to a digital broadcast network, and so on. This is part of the input/output interface problem.

2) Since several sampling frequencies may be in use, it will be necessary to use transcoding devices to convert to digital audio data sampled at a different sampling frequency. These transcoding operations are not without penalty, the loss in signal-to-noise ratio being 4–6 dB. Furthermore, if the conversion is from a high sampling frequency to a lower one, a digital antialiasing filter must be used to take care of the new lower sampling rate.

3) Since it is clear to everyone that there are a large number of problems concerned with keeping track of

which recordings were sampled at what frequency and the attendant loss in signal data at each transcoding operation, discussions are continuing with an attempt to arrive at a common sampling frequency for professional and consumer machines with a single transcoding operation for the European Broadcast Union and satellite communication networks (32 kHz). Furthermore, at this juncture, recommendations from the Digital Video Study Group of the SMPTE for integer samples of digital audio data per frame and a possible sampling frequency of 60 kHz are cause for further study.

Two principal features emerged from the input/output interface meetings. The first related to the audio data word size. Many believed that 20 bits should be allocated for audio data with the possibility of extending the word size to 24 bits by the use of user bits.

Second, future proposals for the channel encoding of the input/output interface should be accompanied by information on the circuit in today's technology, the performance at a distance up to 500 m with a transmission rate in the region of 1.6 Mbit. Characteristics of any proposal should also indicate whether the signal has a dc component, whether it is polarity free, and what its synch and noise behavior is. All measurements should be made on real lines. EIA-RS 422 should be used for guidance in the above measurements.

The first discussion dealt with the desirability of a stereo-pair or mono frame structure of the data stream.

Ampex presented information on a measured practical interface coding method using RS 422 type circuitry in response to the conclusion of the previous meeting.

Sony presented a paper pointing out the distance recommended by RS 422 and expressed some concern about the distance limitations. The practical measurements of Ampex were on the lines of greater length, and it was clear to everyone that much depends on the noise environment, of which little is known today.

The discussion then turned to the subject of word synchronization. Two methods were extensively discussed; however, no final conclusion could be made because again the noise performance is a crucial factor in the design of the synchronization system. The simplest method was by violation of the rules and a modulation code. The other method was to use a synchronizing word within the data stream.

There was also discussion on whether the input/output system would extend to the PTT. If such length would be envisaged, a synch-word system would perhaps be applicable.

The discussion also ranged over the subject of errors on the line. It was generally felt that the line should be considered perfect, thus not using error correction methods.

Further input/output interface suggestions were presented by Matsushita and added to the existing table of answers to the questionnaire.

It seemed clear to all concerned that the adoption of an internationally accepted single sampling frequency would make life a good deal more pleasant for everyone concerned and would result in a better product for the ultimate user—the home listener.

Specific documents on hand presented at the ad hoc and regular meetings are available on request to B. Locanthi [(213) 795-3785]. These documents are:

1) "Tests of the Ampex Serial Digital Audio Communications Format," John Brennan, Ampex Corp.

2) "Consideration of Digital I/O Interface," Timiura et al., Matsushita Electric Industrial Co. Ltd.

3) "System Lock for Simultaneous Video/Audio Production," T. Doi, Sony.

4) "Recommendation of Audio Sampling Rate for Digital VTR," T. Doi, Sony.

5) SMPTE letter on video/audio study group activities, William B. Connolly.

6) Chart showing derivation of proposed sampling frequencies from a crystal clock at 157 500 Hz.

7) Chart showing derivation of various proposed sampling frequencies derived from 13.5-MHz and 36-MHz crystal clocks and also the number of samples per video frame for various field rates and frame rates, E. Gibson, RCA.

8) Matsushita's thoughts on the sampling frequency for professional digital audio use, T. Miura, H. Matsushima, and T. Kogure.

9) Sampling frequencies and integer fractional factors, T. Stockham.

10) Chart showing derivation of master oscillator frequency for integral values of k from 1 through 10 and some higher values for an integral number of audio samples for video/motion picture frame rates of 30, 30/1.001, 25, and 24 frames per second, A. Heaslett, Ampex.

11) DVTR editing considerations, etc., K. Clunis, 3M.

12) "A Format for Digital TV Tape Recording," J. Baldwin, IBA, United Kingdom.

13) "A Digital I/O Interface Suitable for Broadcasting Use," A. Weisser, TeleDiffusion de France.

BART LOCANTHI
Chairman

APPENDIX I

REVIEW OF PROPOSAL TO IEC WG 16

R. Youngquist, T. Stockham, and A. Heaslett

1 Scope

The proposal covers the area of sample rates and source encoding protocols for professional digital audio systems.

2 Sampling Rate

The sampling rate for professional digital audio signal interchange should be 50.00 kHz ± 50 ppm. This proposal was based upon the following:

2.1 There already is a substantial body of commercial precedent usage of this rate.

2.2 50 kHz is one of a small and restricted set of sampling frequencies which provide, without any compromise of timing or pitch, the following benefits:

2.2.1 Absolute synchronization with 525/NTSC, 525/Monochrome, 625/PAL/SECAM, and film chain frame rates.

2.2.2 Integer samples per frame or per small number of frames for each of the systems.

2.2.3 Absolute synchronization of digital audio systems to video production environments, utilizing normal composite video sync signals.

2.2.4 Because of 2.2.1 and 2.2.3, transcoding to domestic VCR-based PCM systems may be effected synchronously to sample rates which are related to video horizontal line rates with comparative ease and with no compromise of compatibility with audio plus video applications.

Note. Some of these specific relationships are detailed in the Appendix.

3 Source Encoding

3.1 The source encoding of audio analog signals should be in the form of a minimum of a 16-bit linear 2's complement representation.

3.2 The positive maximum analog input shall be represented by a digital word of:

0111 1111 1111 1111 [7FFF]

that is, a zero MSB (most significant bit) followed by 15 1's, where a 1 is a logic high state.

3.3 The digital value in 3.2 shall produce a maximum amplitude positive analog output.

3.4 No pre- or postemphasis of the analog or digital signal shall be used.

3.5 Aperture correction (sin x/x correction), if applied, shall only be implemented at the time of digital-to-analog conversion.

3.6 This proposal is based upon the following:

3.6.1 Preliminary accords developed during meetings of the AES technical committee (Atlanta), which have subsequently established an almost universal precedent.

3.6.2 In the case of 3.4, the establishment of an audio channel with unrestricted high-level signal handling capability to the upper system band edge is considered a necessary attribute in the face of current trends in power spectral density distribution of modern music.

3.6.3 In the case of 3.5, digital signal-processing techniques may require the lack of any spectral modification of a digital signal. More specifically, the precise and exact compensation for aperture loss is determined by the digital-to-analog conversion process only.

Appendix

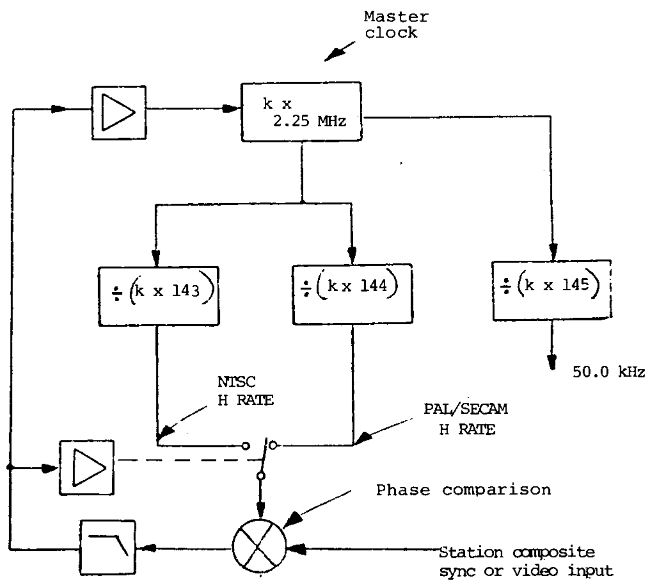
1) Lowest common multiple of NTSC and PAL/SECAM horizontal rates is 2.25×10^6 Hz.

2) Division of 2.25 MHz by 45 yields 50 kHz.

3) Division of 2.25 MHz by 143 yields 15 734.26573 Hz, the NTSC horizontal rate. Further division by 525 yields 29.97002996 Hz, the NTSC vertical rate.

4) Division of 2.25 MHz by 144 yields 15 625 Hz, the PAL/SECAM horizontal rate. Further division by 625 yields 25 Hz, the PAL/SECAM vertical rate.

5) Division of 2.25 MHz by 93 750 yields 24 Hz, the standard film frame rate.



Typical system synchronization

APPENDIX 2

REPORT ON SUBSEQUENT DAD CONFERENCE ACTIVITIES IN JAPAN

1) A DAD study meeting was held for the 14th time on 1981 March 27. Results from activities of each working group (WG) were reported:

WG 1: Report dictating future prospect of a DAD industrialization and its eventual problems.

WG 2: A table was submitted listing a series of concepts of three different proposed systems (CD, MD, and AHD) and their characteristics to be specified later.

WG 3: A table was submitted listing results from subsequent discussions on formatting in three sub-WGs formed to examine each of the three different proposals.

2) At this meeting the nature of each of the three systems was made clear and characteristics to be specified later were also formulated.

As for the system concept, there seem to exist two different concepts. It is taken for granted that the CD and MD systems are audio oriented, in contrast with the AHD system which rather takes care of the common use of a video system. It was also reported that at the examination stage in a sub-WG, a great deal of interest was manifested for both the CD and the AHD systems and that nobody participated in the MD sub-WG. After all activities had been carried out in the meeting it ended with a closing statement.

3) Following this meeting, a DAD conference was held on April 10. It acknowledged the report of the study meeting. The conference played an important role: It minimized the number of possible systems as a DAD to three, and many different types of conceptions to two. It also made each of the three systems clear to all members

concerned, as well as problems summarized at both WG 1 and WG 2. However, a decision as to what system should be adopted was beyond the scope of the conference, and individual companies are requested to decide upon their own system or systems. The conference closed all activities which required more than three years.

4) A report on the DAD conference activities, its minutes, decisions made during the study (such as tentative specifications), and other valuable material shall be published.

APPENDIX 3

REVIEW OF DIGITAL AUDIO MACHINES CURRENTLY AVAILABLE

Table 1. Professional PCM Digital Audio Recorders, 16-bit linear quantization.

Maker	Type	Number of Channels	Tape Speed	Sampling Frequency	Dynamic Range	Total Harmonic Distortion	Frequency Range	Recording Time/Reel Size
Sony	Rotary head/ Umatic	2	0.665 m/s	44.056/44.1	90 dB	0.05%	0–20 kHz	>1 h, ¾ in
Sony	Fixed head	24	30 in/s (0.76 m/s)	44.1/50.4	90 dB	0.05%	20 Hz–20 kHz	60 min (50.5), ½ in
			26.25 in/s (0.72 m/s)				(+0.5–1.0)	69 min (44.1)
JVC	Rotary head/ Umatic	2		44.056/44.1	>90 dB	0.02%	0–20 kHz (± 0.5)	>1 h, ¾ in
Mitsubishi	Fixed head	32	30 in/s (0.76 m/s)	50.4	>90 dB	<0.05%	20 Hz–20 kHz	60 min, 14 in, 1 in
	Fixed head	2	15 in/s (0.38 m/s)	50.4	90 dB		(+0.5–1.0)	60 min, 10½ in, ¼ in
Soundstream	Fixed head	Up to 8	30 in/s (0.76 m/s)	50.0	> 90 dB	<0.03%	0–22 kHz (+0.0–1.0)	30 min, 10½ in, 1 in
3M	Fixed head	32	45 in/s (1.14 m/s)	50.0	>90 dB	<0.03%	20 Hz–18 kHz (± 0.3)	45 min, 14 in, 1 in
		4					10 Hz–20 kHz (+0.5–3.0)	45 min, 14 in, ½ in
Decca	Rotary head	2 [1-in (25-mm) tape]		48.0	>90 dB		20 Hz–20 kHz	>1 h, 1 in
Matsushita	Fixed head	4/2	15 in/s (0.38 m/s)	50.4	>90 dB	<0.05%	20 Hz–20 kHz	1 h, 10½ in, ¼ in

Table 2. Digital Audio PCM Adapters (EIAJ format—14-bit linear or equivalent) for PCM recording on home type VCRs, 44.056 kHz (alternate channel sampling).

Maker	Quantization	Dynamic Range	Total Harmonic Distortion	Frequency Range	Price (Japan) (US$)
Toshiba	12 bits/ 5 slopes	85 dB	0.03%	0–20 kHz ± 1 dB	3750
Sharp	14-bit (equivalent)	90 dB	0.03%	0–20 kHz	2950
Sanyo	14-bit (equivalent)	85 dB		2 Hz–20 kHz ± 1 dB	2900
Sony PCM10	12 bits/ 3 slopes	85 dB	0.03%	0–20 kHz ± 1 dB	3500
Sony PCM100	14-bit linear	85 dB	0.03%	0–20 kHz ± 1 dB	7500*
Technics	14-bit linear	85 dB	0.03%	2 Hz–20 kHz + 0.5 − 1.0	4000
JVC	14-bit linear	85 dB	0.03%	0–20 kHz + 0.5 − 1.0	7500

*Card available for 44.1 kHz.

Table 3. Integrated PCM Recorders using EIAJ format (internal VCR deck), 44.056 kHz (alternate channel sampling).

Maker	Quantization	Dynamic Range	Total Harmonic Distortion	Frequency Range	Price (Japan) (US$)
Hitachi PCMV100*	14-bit linear	85 dB	0.01% 1 kHz	2 Hz–20 kHz ± 0.5 dB	2500
Matsushita SVP100	14-bit linear	>85 dB 44.056/44.1	<0.01%	2 Hz–20 kHz + 0 − 2.5 dB	

*Hitachi PCMV100 is also a video cassette recorder, that is, if a video signal is provided, a video signal may be recorded and played back.

Table 4. Digital audio record players (proposed for end of 1982).

System	Playback Pickup	Record Size	Quantization	Number of Channels	Sampling Frequency	Dynamic Range	Total Harmonic Distortion	Frequency Response	Playing Time
Sony/ Philips CAD	Optical laser	0.12-m disk	16-bit linear	2/4 audio	44.1 kHz	90 dB	0.05%	0–20 kHz	1 h/side
JVC AHD	Capacitance	0.26-m disk	16-bit linear	4/2 audio 3 audio 1 video frame/3 s	47.25 kHz	90 dB		0–20 kHz	1 h/side
Telefunken MD	Mechanical stylus	0.135-m disk	14-bit linear	2 audio	48.0 kHz	>80 dB			1 h/side
DRC	Optical laser	3 × 5-in (76 × 127-mm) plate (rectangular)	16-bit linear	2/4 audio	50 kHz*	>90 dB	0.004%	0–22 kHz	1 h/layer

*System adaptable to any future standard frequency.