Digital Audio Technical Committee Report

MINUTES OF THE MEETING OF THE DIGITAL AUDIO TECHNICAL COMMITTEE

Date: 1980 May 4

Time: 915 hours

Place: Los Angeles, California, USA

Present: Chairman, Bart N. Locanthi (Pioneer North America); members, L. Abbagnaro (CBS Technology), R. Blinn (Capitol), B. Blüthgen (Polygram), G. Bogantz (RCA), P. Burkowitz (Polygram), C. Chan (Sony), T. Doi (Sony), E. Engberg (Ampex), R. Feldman (Soundstream), M. Fujimoto (JVC), D. Gravereaux (CBS), B. Hochstrasser (Studer), H. Ipsen (Sonopress), J. Kawada (Victor Co. of Japan, Ltd.), T. Kogure (Matsushita), T. Kohler (Philips Lab), M. Komamura (Pioneer), R. Lagadec (Studer), P. Macdonald (*AES Journal*), C. Mataw (Sea-Saint Mastering Lab), H. Matsushima (Matsushita), J. McKnight (Magnetic Reference Lab), K. Niimi (Nippon Gakki Co.), R. Pryor (Sony), C. Rodgers (Northwestern University, Auditory Research Lab), P. Samson (Systems Concepts), T. Stockham (Soundstream), K. Tanaka (Mitsubishi), H. Tendeloo (Polygram), S. Traiman (RIAA), M. Willcocks (AudioMobile Corp.), T. Yamamoto (Pioneer), T. Yazawa (Sony).

1. J. McKnight discussed the expanded scope of the S4 committee of ANSI from sound recording to audio engineering The details of the changes in scope of S4 are outlined in an AES document prepared by Geoffrey Langdon, available from the AES on request.

Basically, the Acoustical Standards Management Board has given its approval for the AES and EIA to jointly sponsor S4 of ANSI. The scope of S4 will change from sound recording to audio engineering, a much more general case. Various AES and EIA technical committees will come under the management of S4, prepare standards documents when and as they deem appropriate, and submit them to ANSI through S4.

J. McKnight also discussed the AES outline for standards published in the *AES Journal* 1979 Jan/Feb, preprints of which are available from the AES on request.

2. Dr. T. Doi discussed the EIAJ standard for home VTRs, which was published in a previous communication. Dr. Stockham raised the question as to what happens in the performance of a machine satisfying the EIAJ standard for home VTRs if one wishes to use 16-bit samples.

Dr. Doi answered that one would have to sacrifice the Q parity encoding, and the system would then be less robust against the effects of dropouts. Dr. Stockham raised another question as to how the two stereo channels are sampled in the EIAJ system, and Dr. Doi replied that the channels are sampled alternately.

Questions were raised by Dr. Stockham and others about the possible problems caused by alternate sampling of channels, particularly if one were to use a single-point stereo microphone for pickup.

Most felt that the time delay of 12 μs between channels would not produce any audible effects, but there was no evidence of the results of any serious tests.

Dr. Stockham suggested that for professional machines the D/A converters should be strobed synchronously. Dr. Doi felt that 14-bit accuracy for home machines is quite adequate and he did not anticipate trouble from alternate sampling of the channels.

Dr. Stockham repeated his feelings that one should not cast in concrete any digital audio system, a fact that would prevent any future improvements.

Dr. Doi indicated that since the home VTRs are drifting from an initial playing time of 1 to 2 hours to 6 hours or longer, the possibility of dropout problems seems to be increasing, and therefore the feeling in Japan is that 16-bit linear samples should be sacrificed to 14 bits and that very strong error correction encoding should be maintained.

Major discussion continued concerning whether A/D converters are strobed simultaneously or sampled alternately. Mr. Blüthgen felt that alternate sampling presented a problem; Mr. Tendeloo said that he would be satisfied if machines were identified as to whether the A/D converters were strobed simultaneously or sampled alternately.

Mr. Bogantz felt that 14-bit sampling with preemphasis may offset the signal-to-noise problem, but he prefers 16-bit linear.

Mr. Abbagnaro felt that systems should provide for the possibility of 16 bits in the future, even if 14 bits are used initially.

3. D. Yamamoto discussed the recent Digital Audio Disk (DAD) Conference in Japan. There exists a strong feeling that in addition to digital audio disks which may be compatible with video disk machines, a smaller digital audio disk should be considered, perhaps no larger than 135 mm.

The technical specifications may be as follows:

a. Sampling frequency 44.1 kHz to 50.4 kHz,
b. Number of bits 16 bits linear,
c. Encoding overhead less than 60%,
d. Data rate less than 2.688 Mbit/s,
e. Number of channels 2,
f. Recording time 60 min minimum,
g. Minimum data radius 50 mm,
h. Outside diameter less than 135 mm,
i. Constant linear track velocity less than 2.07 m/s,
j. Recording bit density less than 1.3 Mbit/m,
k. Track pitch 1.6 μm maximum.

Future work will consist of the selection of the best modulation scheme, as well as block construction, to be verified by experiments.

4. Dr. Doi then read a prepared paper on the sampling rate for digital audio recorders. The paper follows this report.

Many questions were asked with respect to items 4, 6, and 7 of this paper and were answered reasonably satisfactorily by Dr. Doi.

5. The meeting adjourned for lunch at 1200 hours.

6. The meeting recommenced at 1345 hours, and a general discussion followed.

7. M. Willcocks agreed with Dr. Doi's recommendation of 44.1 kHz or 50.4 kHz for the sampling frequencies.

8. B. Locanthi commented that the maximum number of digitally mastered tape albums in the world is perhaps 1000. With five 32-track machines in Hollywood, why have only 15 albums been mastered in a one-year period? The consensus was that editing with the present mastering machine is very difficult.

9. Dr. Stockham raised the question as to what bandwidth is required for high-quality recording. Some feel that 20 kHz is not adequate. Others feel that 15 kHz is more than adequate.

Mr. Blüthgen estimated that a responsible research study on that point would cost in excess of $2 million. Puddie Rodgers agreed to be the collector and ultimate source on technical published papers on bandwidth perception going back in time at least to the *Bell System Technical Journal* papers of Snow and others of 1934. Anyone knowing of other papers should send the reference data to Miss Rodgers at the Auditory Research Lab, Northwestern University, Evanston, IL.

10. RCA suggested a demonstration of bandwidth perception limit tests at the next AES meeting. Polygram offered to make test tapes available for such a demonstration. Since the room acoustics at most conventions are not suitable for critical listening tests, it was suggested that we contact Mr. Porterfield of CBS to see if one of his studios would be available for next November. Bart Locanthi will handle this.

11. Several discussions about possible prevention of counterfeiting of digital records followed, but no satisfactory proposals were offered.

12. The meeting was adjourned at 1630 hours.

BART LOCANTHI
Chairman

ON THE SAMPLING RATE FOR DIGITAL AUDIO RECORDERS[1]

Toshi T. Doi

SONY Digital Audio Division, Tokyo, Japan

0 The purpose of this paper is to recommend a frequency set of 44.1 kHz and 50.4 kHz as the sampling rates of digital audio systems for professional applications.

1 The sampling rate of 44.056 kHz is widely accepted for digital audio recorders using VTR as a recording medium [1], [2]. This frequency does not have any alternation because it is tightly bound to horizontal or vertical frequencies of NTSC video systems. For the same reason, the sampling rate should be 44.1 kHz for digital recorders based on PAL or SECAM video tape recorders. There is a 0.1% difference between these two sampling rates.

2 The higher sampling rate of 50.4 kHz was proposed for stationary-head recorders [3] which have the integer relationship of 8:7 to 44.1 kHz. The higher rate is said to be necessary for the safety margin as well as to guarantee the upper limit of the audible frequency range of 20 kHz when the tape speed is reduced to 90% of the original for the purpose of pitch control.

The reason for using the integer relationship is to permit the use of digital sampling rate converters for communication between different rates. In this case the size of the ROM that can be reduced will be greater when the least common multiple of the two rates is smaller.

There are several companies which have adopted 50.35 kHz as the sampling rate of stationary-head systems, which have the integer relationship of 8:7 to 44.056 kHz.

3 Consequently there are two sets of integer-related sampling rates:

Set I 44.056 kHz – 50.35 kHz
Set II 44.1 kHz – 50.4 kHz

The following are the choices for selecting the sampling rate:

a. 44.056 kHz
b. 44.1 kHz
c. 50.35 kHz
d. 50.4 kHz
e. Set I
f. Set II
g. 44.056 kHz and 44.1 kHz
h. 50.35 kHz and 50.4 kHz
i. Sets I and II
j. The other rates

4 The choices of a, b, and g might be possible under the following conditions:

i. A frequency bandwidth over 20 kHz has been proved

[1] Presented at AES Digital Audio Technical Committee Meeting, Los Angeles, 1980 May 4.

to be unnecessary for human ears.

ii. The cost of the filter has proved not to be extremely high compared with that of the system of higher rate.

iii. The variable pitch of -10% is not to be used very often, or the aliasing is considered not serious, or some method is introduced such as to switch the filter when the tape speed is reduced.

These points are not very clear today.

5 The choice of c, d, or h might be very difficult because digital audio systems based on VTR are too important nowadays to be neglected.

6 The choice of i might be a good compromise. The difference of 0.1% between the two sets is not too large compared with the speed tolerance of analog machines.

However, if possible it should be avoided because the accuracy of speed is one of the features of digital recorders, in which the stretch of the tape is always compensated for by referring to a quartz oscillator. And it is far better to select one set of sampling rates among which the digital rate conversion can be easily adopted.

7 This paper will not deal with choice j, therefore only choices e and f will be considered.

Following are the reasons to recommend f, namely, 44.1 kHz and 50.4 kHz, rather than e, that is, 44.056 kHz and 50.35 kHz.

i. The sampling rate for disk systems might be chosen to be either 44.1 kHz or 50.4 kHz.

ii. It is easier to design a sampling rate converter between 32 kHz (transmission standard) and set II, rather than set I.

iii. The only drawback of set II is that the easiest system-lock method using composite synchronization signals is lost for NTSC broadcasters. The same thing will happen with the choice of set I at PAL/SECAM broadcasters. However, it is not a serious problem because there are many methods for system locking. For instance, if the machine is designed to lock by sampling rate frequency, it is very easy to design a PLL circuit to bind the sampling clock to the frequency of vertical synchronization frequencies.

iv. It is true that many machines have been marketed at the frequency of 44.056 kHz, but one should consider the future of digital audio instead of thinking of today's problem in changing all the quartzes on the market.

8 REFERENCES

[1] T. T. Doi, Y. Tsuchiya, and A. Iga, "On Several Standards for Converting PCM Signals into Video Signals," *J. Audio Eng. Soc.*, vol. 26, pp. 641–649 (1978 Sept.).

[2] EIAJ Technical File STC-007, 1979 June.

[3] T. T. Doi, "On the Selection of Higher Sampling Frequency," presented at the AES Digital Audio Technical Committee Meeting, 1978 April 29.

[4] T. T. Doi, "On the Synchronization Between Digital Audio Recorders and NTSC, PAL/SECAM, and Film Systems," presented at the AES Digital Audio Technical Committee Meeting, 1978 July 21.

PROFESSIONAL STATIONARY-HEAD RECORDER QUESTIONNAIRE ON USER REQUIREMENTS[2]

Björn Blüthgen

PolyGram GmbH, Hannover, Germany

1 GENERAL

The questions listed below refer to such user requirements in digital audio recorders as recorded tape interchangeability, tape consumption, tape-editing functions, recorder alignment and maintenance, peripheral equipment interfacing, and operational data storage. Due to the complexity of the issues involved, simple answers obviously cannot be expected, and trade-offs must be made between conflicting requirements. Answers to the questionnaire should, whenever possible, indicate which features are essential or secondary with respect to recorder performance, usability, and exchangeability of recorded tapes.

2 QUESTIONS

2.1 Which numbers of audio data tracks should be harmonized?

2.2 To which track widths should they correspond?

2.3 Which additional tracks should be provided? Where should they be located? Which information should they carry? (For example, should they carry digital or analog audio, time code, recorder control, audio signal control, user-definable bits.)

2.4 As already demonstrated by prototype recorders, the ¼-inch tape format can be realized with different track numbers and different recorder features (such as editing, error correction, and play time). Could the parameters be chosen in such a way that the user would be offered the option of 1, 2, 4, or 8 data tracks per audio signal?

2.5 Could this also be achieved in multitrack recorders?

2.6 Which channel code (modulation code) is to be preferred? For what reasons?

2.7 Is more than a single channel (modulation) code to be provided in digital recorders? How would more than a single code influence the recorder's performance and the exchangeability of recorded tapes?

2.8 How does your preferred channel code influence alignment and mechanical maintenance?

2.9 On the basis of your answers to the preceding questions, do you think that some formats and/or parameters can now be harmonized?

2.10 What tape specification would ensure proper program recording and playback on any harmonized type of equipment?

2.11 The nominal tape speed depends on such parameters as number of tracks per channel, channel code, word length, redundancy, sampling rate, etc. Can a standardized packing density be defined which would allow tape han-

[2] Received 1980 January 14; revised 1980 March 27.

dling such as is common practice today? How should this packing density be defined?

2.12 The SMPTE time code is a worldwide standard for synchronization, and should presumably be the reference for a tape transport control. How should the SMPTE code be used in a digital audio recorder? What would be the interface for external synchronization? Can other signals (a MHz-range master clock, or a kHz-range sampling clock) be used for external synchronization? What are the advantages and disadvantages of other synchronization signals? How should the interfaces be designed?

2.13 What technical parameters would ensure or would not prevent the choice of one or several sampling rates in the digital recorder?

2.14 If recorder parameters are controlled by code information recorded on a dedicated data track, which uniform code format should be selected?

2.15 Which head configuration (for example, record/playback/record or syncplay/record/playback) is preferable? For what reasons?

2.16 Head spacing may be influenced by tape format, data format, and processing time. What value for the processing time required in updating might be suitable for harmonization?

2.17 How should the mechanical and electrical interconnection interface be designed for stereo and multitrack recorders?

2.18 Which word length should be proposed for the digital audio samples? Are several word lengths required?

2.19 Which error-protection code and method do you support?

2.20 Which data format for error protection (interleaving, sequence, etc.) do you support? What operational features does it offer to the user?

2.21 Have essential features relating to the use of digital recorders been left out in this questionnaire? What are they?

ANSWERS TO QUESTIONNAIRE ON PROFESSIONAL STATIONARY-HEAD RECORDER USER REQUIREMENTS[3]

Toshi T. Doi

Sony Corporation, Tokyo, Japan

1 GENERAL

Sony will propose a format.[4] Meanwhile, we are open to all discussions for the purpose of harmonization.

[3] Received 1980 February 18; revised 1980 July 3.

[4] It appears on p. 624 of this issue.

2 ANSWERS

2.1 Number of data tracks recommended are as follows:

a) ¼ inch   8 tracks
b) ½ inch   24 tracks
c) 1 inch   48 tracks

2.2 Track width should be decided according to today's technology of heads and tapes, which satisfy the requirements of interchangeability and electromagnetic characteristics.

2.3 Each tape-width format should allow for the following to be included:

a) digital audio tracks shown in (2.1),
b) two channels of analog tracks,
c) a time-code track, and
d) a control track.
e) Audio signal control data should replace any of the digital audio tracks with the same error-correcting schemes, in which case 800 kb/s is possible for one track, and error concealment is also effective for mixdown data.

2.4 The switchable format between 1, 2, or 4 data tracks/audio signal is possible.

2.5 Yes.

2.6 The modulation scheme should be chosen which has the highest efficiency in keeping with reliability.

2.7 It might be too difficult to design a machine which covers more than one modulation scheme.

2.8 The differences in circuit alignment and maintenance due to a modulation scheme might be negligibly small.

2.9

a) Considering the future of the digital audio disk, the playing time of the master recorder should be more than one hour in order to cover one side of the disk.

b) The number of audio channels for a certain tape width will depend on the number of tracks per audio channel, which should, therefore, be reduced to a minimum number for multichannel machines. The playing time of multichannel recorders is requested as being long enough for one piece of music, which is not always as much as (a).

c) The best compromise is to settle on both a 1-track/channel and 2-track/channel format, so that the user can select channel number or playing time. The two formats should be switchable so that the interchangeability might not cause any problems, just as in today's formats of 7.5 in/s, 15 in/s, and 30 in/s in analog machines.

2.10 In order to start the digital audio era quickly, the tape for the type C format video is recommended. If we consider metal or vaporized tapes, a significant improvement is not expected in the digital audio field.

2.11 A packing density which would not be affected by fingerprints and dust would be too low. These affects can be solved by appropriate error-correcting schemes. Therefore, the handling problem should be solved by error correction, not by low packing density.

2.12

a) A time-code track separate from the normal control track is essential, because the time code might be rewritten according to the editing of the source. Besides, the control track should have a special relationship to the block format of the digital audio coding.

b) If the time-code track is separated from the main control, the synchronization would be easy because it is the same as in today's application.

c) The sample rate or master clock can easily be used for synchronization, but they are not sufficient for editing purposes because of the lack of the absolute address, and the miscounting of pulses cannot be overcome.

2.13 We have tried to make a multisampling-rate machine but, because of its complexity, we would like to reduce the number of rates to only those which have a simple integer relationship.

2.14 In the distant future, an ultrahigh-speed microcomputer might be able to handle all the signal processing. At that time a universal machine might be possible.

2.15 There is no difference for standardization.

2.16 This distance depends greatly upon the block format, error correction, and the method of editing. It might be extremely difficult to find a specific distance which is common to different formats.

2.17 We proposed an electrical-code format of digital I/O at the 65th Convention of the Audio Engineering Society.

2.18

a) 16-bit

b) If the 16-bit slot is harmonized, the lower bit system (15, 14, 13, ······) should be used without changing tape consumption. 16-bit is appropriate because a 17-bit word would require almost double the amount of hardware for easy control. All digital devices are designed to handle 2^n bits. The requirement for the higher bit resolution, if it arises, should be solved by emphasis/deemphasis or DPCM.

2.19 Detailed information will be proposed.

2.20 Detailed information will be proposed.

2.21 Detailed information will be proposed.

ANSWERS TO QUESTIONNAIRE ON PROFESSIONAL STATIONARY-HEAD RECORDER USER REQUIREMENTS[5]

Roger Lagadec

Willi Studer, Regensdorf, Switzerland

1 GENERAL

Studer supports all efforts toward the harmonization of digital recorder parameters, with free machine interconnection and tape interchangeability as final goals. Studer and Sony have recently agreed on a common format for a wide range of applications in digital audio recording, an outline of which is added to this answer as an appendix.

[5] Received 1980 May 9; revised 1980 July 22.

2 QUESTIONS

2.1 We propose 8 audio data tracks on ¼-inch tape, 24 on ½-inch tape, and 48 on 1-inch tape.

2.2 The approximate track widths can be derived from 2.1. Precise widths and tolerances should be discussed in the technical community.

2.3 We propose 2 analog tracks on the tape edges, and 2 additional data tracks for time code, control and user data.

2.4 We support 1, 2, and 4 tracks per audio channel.

2.5 1, 2, and 4 tracks per channel should refer to both stereo and multichannel machines.

2.6 Code requirements for high-density digital magnetic recording are well understood. A new code will be proposed, and described in the near future.

2.7 A single modulation code is sufficient.

2.8 The choice of a code does not affect the requirements for alignment or mechanical maintenance.

2.9 Yes, many essential parameters of a format can, and should, be harmonized very soon.

2.10 Conventional video-grade tape should be adequate. It is essential to discuss in the near future which mechanical tolerances can ensure tape interchangeability.

2.11 Any recording density not greatly in excess of 40 kbits/inch (referring to data bits prior to modulation coding) ensures that today's tape handling practice can be maintained, provided that machines are designed properly.

2.12 Digital recorders should have a time-code track, and interface according to the rules of the SMPTE time code.

2.13 The sampling rate itself is the wrong issue. A digital recorder should have a few nominal sampling rates, dictated by applications such as high-quality professional recording, digital audio disk, broadcasting, and consumer applications. As variable pitch is an essential feature anyway, *ranges* of sampling rates should be defined.

2.14 This point will be answered in the near future.

2.15 Both head configurations are basically acceptable.

2.16 The distance between the heads in relation to processing delay is not a critical design constraint.

2.17 An input/output interface is essential if recorders are to be interconnected freely. We will propose an I/O format in the near future.

2.18 At present, a single-word length of 16 bits, with uniform (''linear'') quantizing, in two's complement code, is appropriate for recording.

2.19 Error protection is a complex matter, and should be dealt with in the form of a detailed technical proposal. We shall do so.

2.20 See 2.19.

2.21 Some comments:

Digital recorders must interface without any degree of mutual interference. Stringent rules on electromagnetic compatibility (EMC), such as presently under study in telecommunications, industrial electronics and computers, will be essential for tomorrow's digital studios.

Discussions on formats should be conducted openly, and refer to real user considerations and concerns, such as: can machines be interconnected? do machines accommodate the same user data? do they provide tape interchangeability? how are errors diagnosed and repaired? and how are the machines maintained?

APPENDIX

AN OUTLINE OF THE FORMAT JOINTLY SUPPORTED BY SONY AND STUDER FOR STATIONARY-HEAD DIGITAL AUDIO RECORDING

Toshi T. Doi

Sony Corporation, Tokyo, Japan

and

Roger Lagadec

Willi Studer, Regensdorf, Switzerland

At the 66th Convention of the Audio Engineering Society in Los Angeles, Sony and Studer announced that they had agreed on a new format for stationary-head digital audio recording. This is an outline of the new format's essential features, as seen from a user's viewpoint. A detailed document will be submitted to the appropriate technical committees.

THE FORMAT'S SCOPE

At present, a large number of formats for stationary-head digital audio recording have been proposed, and some are no doubt excellent. A main reason in submitting yet another format was to cover a much wider range of application than hitherto possible, in order to make future machine design and interfacing easier. The newly developed format more than covers the range of channel numbers in use today, while based on the same nominal tape speed as today's machines. It applies, with no basic change in electronic boards, to professional studio recording, to broadcasting, and even to consumer applications.

THE SAMPLING RATES

Regrettably, sampling rates have been a central issue in discussing digital audio formats. All practical studio machines should include pitch control, which is achieved by varying the tape speed from its nominal value. Thus, digital audio recorders inherently accommodate at least moderate ranges of sampling rates. The jointly supported format has three nominal sampling rates, corresponding to different applications: 50.4 kHz for highest quality studio recording, 44.1 kHz for other studio applications and digital audio disks, and 32 kHz for broadcasting and other applications, in accordance with the EBU standard.

THE WORD FORMAT

In accordance with the recommendation presented to the AES Digital Audio Technical Committee, a 16-bit uniform quantization word format is used.

THE TAPE FORMAT

Today's tape widths and nominal tape speeds are used in the format, which accommodates ¼-inch, ½-inch, and 1-inch tape at 30, 15, and 7½ in/s. Nominal tape speeds refer to a 50.4-kHz sampling rate, and are accordingly decreased for 44.1 kHz and 32 kHz. Video-quality tape is used, with mechanical tolerances in width and track geometry ensuring tape interchangeability. One-quarter-inch tape has 8 digital audio tracks, ½-inch tape has 24, and 1-inch tape has 48. In addition, all tapes have 2 analog tracks, 1 on each edge, and 2 auxiliary digital tracks for control and time code. The time code is SMPTE compatible and the control track allows storage of an adequate quantity of user data.

TRACKS AND CHANNELS

At full speed, 30 in/s nominal, a single track is assigned to each channel. Thus, ¼-inch tape accommodates 8 channels, while ½-inch and 1-inch tape accommodate 24 and 48, respectively. At medium speed (15 in/s), 2 tracks are used per channel, the electronics remaining basically unchanged. At low speeds, finally, 4 tracks are assigned to each channel, so that ¼-inch tape is used for stereo recording.

DROPOUTS AND EDITING

A strong error-correcting method, called cross-interleave code, with 33.3% redundancy ensures a high security against data dropouts. Error protection ensures that punch-in and punch-out, and electronic editing can be performed without signal degradation. The format also offers protection against fingerprints and allows tape-cut editing. The level of fingerprint protection and tape-cut capability is better at high speeds than at low speeds, but should prove adequate in all appropriate applications.

SUMMARY

The jointly supported format is based on today's tape material, width, and speeds. It covers the full range of stereo to multichannel recording, with up to 48 channels at full speed on 1-inch tapes. It also extends to broadcasting and even consumer applications. The design is essentially modular, and identical boards are used for all tracks regardless of tape speed and track-to-channel assignment.

Sony and Studer hope that their joint proposal will generate interest in defining a final format which covers the full range of possible applications of digital audio. They would appreciate response from the technical community, and are ready for an open discussion.

ANSWERS TO QUESTIONNAIRE ON PROFESSIONAL STATIONARY-HEAD RECORDER USER REQUIREMENTS[6]

Tetsuya Yamaguchi

Mitsubishi Electric Corporation, Amagasaki, Hyogo, Japan

[6] Received 1980 February 25; revised 1980 July 24.

2.1 a) ¼ inch 8
b) ½ inch 20
c) 1 inch 40
d) 2 inch 80

2.2 Approximately 30 μm

2.3 Analog track and time-code track. Linear recording density of the data track is usually higher than that of the analog track or time-code track. Therefore it is advisable to allocate those auxiliary tracks at the outer edge portion of the tape.

2.4 Operational features required for 2-channel recorders and multichannel recorders are different. The tape formats of each are different. For simplicity, it is preferable to have one format for each than a multiformat.

2.5 Same as 2.4.

2.6 MFM for 2-channel recorders, because it is simpler. Run-length-limited code for multichannel recorders, because multichannel recorders need high-density recording.

2.7 A tape recorded on a recorder with one modulation code cannot usually be played on a recorder with another modulation code, unless some switch is operated by the user. This makes the recording system complicated. One modulation is definitely better than multimodulation for a recorder.

2.8 In order to evaluate circuitry size of modulation, total consideration of write amplifier, head amplifier, read amplifier, write compensator, read compensator, modulator, and demodulator is necessary. Usually, MFM might require less circuitry than a run-length-limited code. Mechanical maintenance for each code would be the same.

2.9 In addition to those aspects listed, editing method (tape cut or electronic, or both), over-dubbing, and total number of channels should be considered to justify a certain format.

2.10 Hc = 600 ~ 700 Oe, Br = 1300 gauss.

2.11 PCM recordings are generally of very high density because the quantity of information to be recorded is much larger than that of analog. Consequently rough tape handling, such as heavy fingerprints, will cause a major problem to any kind of format. However, our experience shows that even tape-cut editing is possible when recording density is kept lower than approximately 20 kb/in. Higher packing density might be acceptable for multichannel recorders because the tape surface is not usually touched.

2.12 a) In order to allow the recorder to follow an externally generated SMPTE code, such as that from a VTR, the frequency of the master clock should be a multiple of 2.4 kHz and a multiple of the sampling frequency. It means that

$$f_s = \frac{n}{m} \times 2.4\ \text{kHz}$$

where m and n are simple integers.

b) Current drive photo coupler.

c) and d) When a sample clock is chosen as a reference signal, synchronization with a VTR becomes difficult unless the SMPTE code is used in addition to the sample clock. For digital dubbing purposes, a sample clock should be provided in PCM tape recorders. This means that synchronization with a sample clock might be useful for synchronization between recorders which do not have an SMPTE code track.

e) Current drive photo coupler.

2.13 We prefer one sampling frequency, 50.4 kHz.

2.14 Users bits of SMPTE code and flag bits on data tracks could be used for this purpose.

2.15 We think that these aspects should be left to be decided as part of the recorder design, unless it becomes clear that there are certain reasons to choose a tape format.

2.16 More than 35 mm.

2.17 Multiconnector.

2.18 16 bits

2.19 RSC code

2.20 We support formats as follows:

Type of recorder	Format	Operation features
Multi-channel	As presented in "On a Tape Format for Reliable PCM Multichannel Tape Recorders," by K. Tanaka et al., presented at the 66th Convention of the Audio Engineering Society, Los Angeles, 1980 May 6–9, preprint 1669.	Over-dubbing Ping-ponging Electronic editing Read after write Recorder works correctly even if one track fails.
2-channel	As presented in "Improved Two-Channel PCM Tape Recorder for Professional Use," by K. Tanaka et al., presented at the 64th Convention of the Audio Engineering Society, New York, 1979 November 2–5, preprint 1533.	Tape cut editing Electronic editing Read after write Recorder works correctly even if one track fails.

ANSWERS TO QUESTIONNAIRE ON PROFESSIONAL STATIONARY-HEAD RECORDER USER REQUIREMENTS[7]

Tatsuo Yamamoto

Nippon Columbia Co., Ltd., Tokyo, Japan

2.1 2 channels, 8 data tracks. In multichannel, 1 or 2 data tracks per one channel on 6.30 mm (+0, −10 μm) metric tape width.

2.2 Less than 0.35 mm

2.3 Analog data track and time-code track. They should be located at both tape edges. An analog data track should be recorded as a mixed signal on the left and right channels. Time-code track should be recorded in SMPTE time code.

2.4 Although the tape transport could be utilized by

[7] Received 1980 March 24; revised 1980 July 17.

changing only the speed, an electronics processor must be provided for each data track configuration, requiring different demodulation parameters in accordance with different longitudinal packing densities.

2.5 Same as 2.4

2.6 MFM is preferable, because the error rate of MFM is less influenced by level fluctuation than are those of the other channel codes.

2.7 More than a single-channel code would make exchangeability of recorded tapes very difficult.

2.8 Not much difference among any of the channel codes.

2.9 Same answer to 2.19 and 2.20.

2.10 Frequency response, mechanical characteristics, output level, Hc, Br, dimensions, etc.

2.11 20 kb/in or less (by MFM recording) packing density is preferable. A packing density should be defined to ensure an error rate to the extent that at least there are no significant error data which cannot be corrected.

2.12 The SMPTE time code can be used for the address data for the electronic editor. In the case of a PCM recorder, a high-accuracy clock is built in and a synchronizing signal is usually recorded on the tape as well. Therefore, we are reluctant to utilize the SMPTE time code for speed control. Synchronization accuracy would improve by synchronizing with the sampling or with the master clocks. There would be no problem if the sampling frequency is harmonized. If it is not harmonized, a sampling frequency converter would be needed. The sampling frequency converter would also be needed for data dubbing in the case of utilizing the SMPTE time code, even if it can synchronize the speed. If there is a need for providing an interface, a device having a floated signal ground could be used, for instance, a photo coupler.

2.13 Tape speed, longitudinal packing density, tape characteristics, etc., would.

2.14 The users' area of the SMPTE code on the time-code track and the users' bits, which are provided in data tracks, can be used for this purpose. Therefore, there is no need for a specific data track other than the time-code track.

2.15 The latter (sync/record/playback) would be more advantageous for operation, and the former (record/playback/record) would be more practical.

2.16 The distance between the sync-play head and the recording head should not be determined for computation. Data can be processed in approximately 10 ms. The leakage of the recording signal to the playback signal would be a more important consideration by which to determine the position. Data can be easily delayed by a memory. Although the data processing time depends on tape speed and tape format, it will not be so significant as to cause problems.

2.17 For stereo- or 4-channel recorders, every specification (including the wow and flutter of the mechanism) should be harmonized or standardized. However, we wonder if harmonization is necessary for those systems which are mainly used in "in-house" conditions as with multichannel recorders.

2.18 16-bit word length is preferable. Several word lengths are not required.

2.19 Example (2 channel)

Ecc_1	Ecc_1'
DATA 5	
3	
1	
6	
4	
2	
Ecc_2	Ecc_2'
8 bit	
16 bit	

⇨

a_7	b_7
a_5	b_5
a_3	b_3
a_1	b_1
a_6	b_6
a_4	b_4
a_2	b_2
a_8	b_8
8 bit	
16 bit	

Ecc is error correction code. Ecc_1 and Ecc_2 are generated as follows:

$$a_7 = a_1 \oplus a_2 \oplus \text{----} \oplus a_6$$

$$a_8 = T^6 a_1 \oplus T^5 a_2 \oplus \text{----} \oplus T^1 a_6$$

$\oplus$ modulo 2 summation

($b_1 \sim b_8$ as same)

T is companion matrix, shown as follows:

$$T^1 = \begin{pmatrix} 0&0&0&0&0&0&0&1 \\ 1&0&0&0&0&0&0&0 \\ 0&1&0&0&0&0&0&0 \\ 0&0&1&0&0&0&0&1 \\ 0&0&0&1&0&0&0&1 \\ 0&0&0&0&1&0&0&1 \\ 0&0&0&0&0&1&0&0 \\ 0&0&0&0&0&0&1&0 \end{pmatrix}$$

2.20 Example (2 channel). Block format of data

Sync	Ecc_1^1	Ecc_1^2	Ecc_1^3	Ecc_1^4	------		------ Ecc_1^{28}	CRCC
Sync	5L		5R					CRCC
⋮	3L		3R					⋮
	1L		1R		7L		37R	
	6L		6R				42R	
	4L		4R					
⋮	2L		2R		8L			⋮
Sync	Ecc_2^1	Ecc_2^2	Ecc_2^3	Ecc_2^4	-----		------ Ecc_2^{28}	CRCC
16 bit	16 bit							16 bit
256 bit								

There is no interleaving over blocks, considering tape cut editing.

2.21 No special comment.