Synopses of Papers Presented at the 20th Television Conference

Television Audio Technology

1. The BTSC Multi-Channel Television Sound System. *By Carl G. Eilers*, Zenith Electronics Corp., Glenview, Ill.

The BTSC multi-channel television sound system (MTS) provides for a full 15-kHz audio response stereo pair with low distortion and signal-to-noise ratio (SNR) equivalent to the monophonic audio component. Also provided is a separate audio program (SAP) with a 10-kHz audio response at reasonably low distortion and an SNR slightly inferior to the monophonic main program. A professional channel is also provided for use by the broadcaster.

Over 150 TV stations have gone on the air with BTSC MTS in the short time since FCC authorization on April 23, 1984. Almost one-half of newly manufactured top-of-the-line color television receivers are MTS-equipped, and two of the major television broadcast networks are providing programs in stereo sound.

2. Audio Companding for the BTSC MTS. *By Leslie B. Tyler*, dbx, Newton, Mass.

BTSC multi-channel television sound includes a sophisticated noise-reduction system (compander) as an integral part of the standard. This paper reviews the rationale for including a compander and the choices made in the BTSC design. The compander's effects on signal-to-noise ratio, dynamic range, distortion, coverage area, and multipath are discussed. Reference is made to the extensive objective and subjective testing done by the EIA in evaluating transmission and compansion systems.

The presentation includes a demonstration comparing MTS performance both with and without companding.

3. Transmission System Characteristics Affecting SAP Performance. *By Robert M. Unetich*, ITS Corp., McMurray, Pa.

The growing industry experience with stereo transmission indicates that in most cases acceptable stereo performance can be achieved with fairly modest changes to a TV station's transmitter and diplexer. Field experience also indicates that the successful addition of a secondary audio program (SAP) will be more difficult because of video component interference and aural system deficiencies. Based on the observation of both lab and field tests of SAP systems, a set of requirements for system performance has been established. Means for achieving this level of performance are also explored in this paper. The use of SAP for other than standard audio transmission is also explored, and field test results of a direct frequency shift-keyed SAP data transmission system are reported.

It is expected that the use of the SAP channel will expand significantly in the next few years. This paper attempts to explore the system considerations necessary to facilitate the use of this channel.

4. Production Facilities for MTS. *By Douglas F. Dickey*, Solid State Logic Ltd., Oxford, England.

Stations which have equipped for multi-channel television sound transmission have discovered a non-technical "bug" in the system. There is a distinct scarcity of quality stereo program material. While devices for stereo synthesis have proved useful as an interim measure, there is significant controversy as to whether they enhance or degrade the artistic content of the broadcast soundtrack. If personal opinions are eliminated from consideration, one point remains clear. Such post-mix processing is beyond the control of a program's producer and director, and offers them none of the creative or dramatic advantages of stereo.

Moreover, the sophisticated high-fidelity capabilities of MTS transmission and the improved audio quality of home receivers have introduced a new element into the competitive equation. While it is still early, the overall sound quality of particular stations and programs will most certainly become part of the viewers' critical opinion. As was the case during the transition to color, deviations in the audio quality of national and local programming and commercials are already apparent in properly-equipped households, and will become more so.

If the U.S. is to complete its transition to nationwide stereo television service, broadcasters who have achieved MTS *transmission* capabilities must also convert their plants for multichannel sound *production*. This is a fundamentally different task from distribution and transmitter conversion, in that it affects the daily hands-on operating environment. It involves changes in technique as well as technology. In addition to equipment upgrades, it may involve acoustic modifications, revisions to control room layouts, and even structural renovation in some cases.

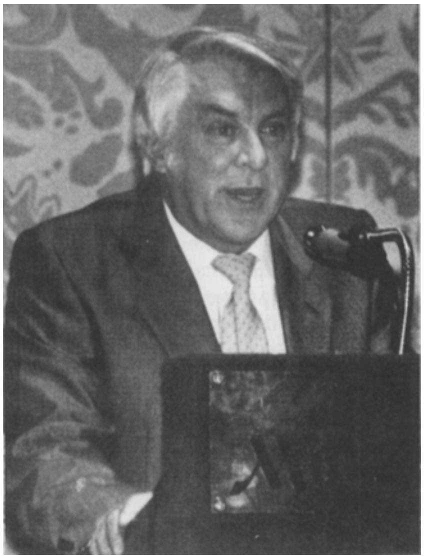
The success of this conversion is measured not only in terms of audio performance, but in terms of creative flexibility and operational efficiency. The challenge is to create a plant that can produce high-quality stereo programming with little or no increase in overall production time. The planning process requires extensive dialogue between the management, production, and technical engineering teams.

As a framework for that dialogue, this paper presents an overview of the improvements which may be required, including sound stage and control room acoustics; stereo monitoring systems; live and post-production audio consoles; communications requirements; the audio tape machine complement; machine control requirements for audio post-production; digital audio storage and effects processing; and the growing role of audio mixing computers. Particular attention is paid to the functional and creative requirements of the audio console with regard to stereo production, mono compatibility, and creation of the secondary audio program (SAP).

Unlike transmitters, production plants can be upgraded over a period of time. Certain steps can be taken immediately with little or no capital expenditure. Others may involve considerable time and money. The paper provides suggestions for establishing the operational requirements in each production and post-production area, and for determining transitional and long-term priorities.

5. Audio Processing for Broadcast Television Transmission of Stereophonic Sound. *By Robert Orban*, Orban Associates, Inc., San Francisco, Calif.

Audio processing combines engineering and esthetic considerations. There are no measurements which adequately describe the "sound" of an audio processor, which is in general a non-linear, time-varying, program-adaptive system. The usual frequency response, noise, harmonic distortion, and inter-modulation distortion measurements are made on *static* test signals. While a good audio processor will exhibit good performance in these instrument tests, the ear must be used to assess other aspects of audio processor design. Some of these aspects include *consistency* (are there annoying variations in audio quality or loudness when switching between sources?), *loudness control* (do certain types of program material produce irritatingly high loudness?), and *freedom from overt processing artifacts*. Some of these artifacts (which should not be present in any well-designed processing



General Arrangements Co-Chairman Michael Bailey at the Get-Together Luncheon.

system) include *noise breathing* (audible "pull-up" of background noise during pauses), *pumping* (unnatural short-term loudness variations which give the subjective impression that "holes" are constantly being punched in the signal), and *dynamic distortion* (where a complex signal will sometimes sound overtly fuzzy or distorted, even if the processor produces clean sinewave measurements).

All of the above considerations apply regardless of whether the transmission channel is mono or stereo. In the case of broadcast TV, many of the older-generation mono processors fell short in at least one of the above areas. Because of the minimal-quality audio sections of most mono TV receivers, these shortcomings were generally unnoticed by most viewers.

The advent of stereo broadcast TV has changed this situation. Many of the new stereo TV receivers are provided with audio sections of substantially higher quality than their mono predecessors. In addition, the rise of the integrated home audio/video entertainment center has resulted in many consumers' connecting the audio outputs of their stereo TV receivers directly to high-powered amplifiers and high-quality, high-fidelity loudspeakers. Under these conditions, processing artifacts which were formerly masked by inadequate sound sections of mono TV receivers become painfully obvious to the ear. However, it must be simultaneously remembered that whatever audio processing is applied to stereo TV must also be compatible with the smaller mono receivers which are still in the vast majority, and which will continue to be so for some years to come.

An artful compromise is therefore necessary between the requirements of the low- and high-quality receivers. The low-quality receivers have small loudspeakers with poor response at the frequency extremes. They are usually driven by underpowered amplifiers, and are not capable of reproducing high dynamic range without clipping and distortion. For this reason, a certain amount of compression is required with most program material to keep program material comfortably listenable through such receivers. If the dynamic range is excessive, either low-level program material will be lost in the background noise of the typical domestic listening environment, or high-level material will cause distoriton in the receiver (and/or annoy the viewers' families and neighbors!). The author believes that 10 dB of compression is optimum for typical program material, given the real-world limitations of the small mono receivers and domestic listening environments.

At the same time, this compression must be accomplished subtly to avoid making it obvious on the high-quality receivers. Ideally, the sound should be relatively "open" (i.e., very short-term dynamic range is preserved), yet long-term dynamic range should be controlled. Doing this requires implementation of program-adaptive attack and release times in the audio processor, and also requires a "gating" circuit to "freeze" the gain when the loudness drops below a certain threshold. The details of such circuit implementations separate good-sounding from bad-sounding processors.

Stereophonic processors must be configured so that the gains of the left and right channels track together to avoid shifts in stereo imaging with gain reduction. It has been empirically discovered that only the slow components of the gain reduction need track; the fast components (such as peak limiting or clipping) sound best when uncoupled because they act quickly enough so that the ear does not perceive changes in stereo imaging. High-frequency limiters, which are required in broadcast TV processors to control the effects of the pre-emphasis curve, also sound best when the left and right channels operate without coupling.

In FM stereo using the standard "pilot tone" method of transmission, "inter-leaving" occurs: peak modulation is determined by the higher of the left or right channels. Accordingly, FM stereo limiters are configured so that the amount of gain reduction is determined by that required to reduce the higher of the left or right channels to the 100% modulation point (referred to the processor output).

Because the broadcast TV stereo system incorporates a noise-reduction compressor in the difference ($L-R$) channel, and because the peak modulation constraints are different than in FM stereo, the FM stereo interleaving model is not entirely applicable to TV stereo. The noise-reduciton compressor causes amplitude and phase changes which reduce the amount of interleaving, and the total permitted stereophonic modulation is twice as high as the total permitted $L+R$ modulation.

It can nevertheless be shown that use of FM-stereo-style processing is conservative, in that it will always prevent overmodulation in the TV stereo system (assuming that the noise reduction encoder and/or stereo generator do not overshoot). In addition, several experts have expressed the subjective opinion that FM-stereo-style processing provides more natural loudness balances and long-term stereo perspective than does "matrix" (sum-and-difference) processing. The author agrees, and recommends that FM-stereo-style (greater of left or right) processing be used for broadcast stereo TV.

Similarly, some have advocated that the entire plant be operated in sum-and-difference ($L+R/L-R$) form instead of conventional Left/Right (L/R) form. The author supports the L/R plant, because it has been used successfully in FM stereo for 25 years, and because a given amplitude and phase error between the two transmission channels causes much more severe damage to the stereo image in an $L+R/L-R$ plant than the same error causes to the mono sum in an L/R plant.

Finally, loudness control. Loudness has no physical reality; it is a subjective sensation produced by the ear and brain. Considerable research has been done on measurement and control of loudness in the broadcast environment (most notably by CBS Laboratories and its successor), and effective automatic loudness control techniques have been developed. The author believes that automatic loudness control should be part of audio processing for stereo TV to achieve highest viewer satisfaction, as measured by the lack of complaints received by broadcast stations.

6. Panel Discussion.

Moderator: Edmund A. Williams, *NAB*
Panelists: Douglas F. Dickey, *Solid State Logic Ltd.*
Randy Hoffner, *NBC*
Larry Ocker, *WTTW*
Steven Sarafian, *Sony Broadcast Products Co.*

Type D-1 Digital Television Recorder (Part I)

7. Digital Television Recording — History and Background. *By John L. E. Baldwin*, Independent Broadcasting Authority, United Kingdom.

In the early 1970s there was already much interest in digital standards for 625 lines, but although these studies were interesting, they could only become important when it was shown that the standards being considered could be recorded.

In the summer of 1971, some detailed measurements of TR70 channels convinced me that digital recording was feasible and could provide markedly better performance for the same tape consumption.

Working for a broadcaster (ITA, which later became the IBA) rather than for a manufacturer, it seemed sensible to mention these thoughts, which included a helical format proposal, in a paper entitled "The Digital Future of Television Studio Centres" presented at the International Broadcasting Convention in September 1972, and in further detail at the Video and Data Recording Conference in July 1973. At the IBC, much of the impact of the recording part of the paper was lost, as much of my time was used for a description and demonstration of DICE.

The next significant event was that it became possible in 1966 to obtain a Rank 9000 (IVC 9000) recorder, which had many physical similarities to the format already proposed, as a test bed for digital recording. We were constrained effectively to using the existing record amplifiers, the heads, and the pre-amplifiers of the analog machine, but could change the form of the signal passing through them. We knew that only a fraction of the picture could be recorded, and that fraction chosen was to be in the form of a vertical strip down the picture so that it would be possible to get some subjective impression of the importance of drop-outs and of the quality of the picture. The fraction of the active picture time that we started with was about 21%, but over a period of about six months this was progressively increased to about 54% of the active picture width. Initially we used Miller Code, but this changed to an 8/9 code, and finally to the IBA 8/10 Code. Initially the shuffling of word order was rudimentary, later it was optimum for one-dimensional shuffling.

The recorder was taken to Venice in April 1977 and demonstrated to the Technical Committee of the European Broadcasting Union (EBU). Pictures from the recorder were shown at the Montreux Television Symposium in June of the that year. Between April and June we had proved interchangeability of digital recordings between machines and that half track width heads did not alter the performance. This convinced us that the whole picture could be recorded on a 2-in. wide tape travelling at 8 in./sec. The video applied to the recorder was PAL with only 2 eight-bit words for each cycle of sub-carrier; this corresponds to 71 Mbit/sec.

Soon after Montreux 1977, two manufacturers became associated with our work in the IBA. The first was Bosch Fernseh, soon followed by Sony; others applied later.

On the principle of first come, first served, a Bosch BCN (SMPTE B-Format) machine was progressively modified and developed to provide the ever-increasing bit-rate requirements of proposed digital standards and eventually gave a 160 Mbit/sec capacity in time for the EBU consideration of a 12:4:4 MHz component coding proposal in spring 1980. A Sony recorder based on the SMPTE C-Format but using 6 heads was developed to provide a capability of 228 Mbit/sec used in the important EBU demonstrations at Crawley Court in January 1981.

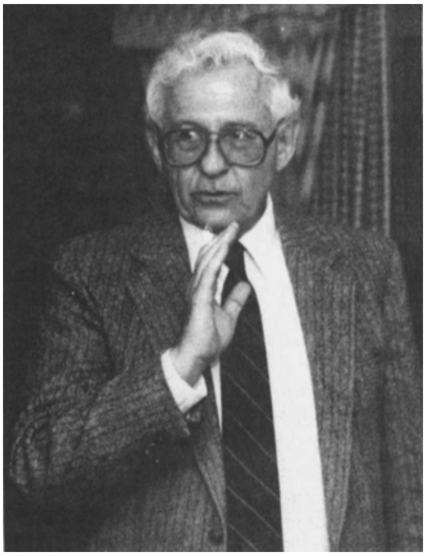
Sony independently provided the 8-headed modified C-Format machines for the corresponding SMPTE demonstrations and tests of Digital Standards in San Francisco in February 1981. Many other important demonstrations have been made by different organizations, including the Ampex composite transverse scan demonstrations in the U.S. and Montreux in 1979.

The Digital Television Tape Recording Group of the SMPTE was set up in 1979 a few months before the MAGNUM Group of the EBU was formed. Initially progress was slow, as the signals to be recorded had not been defined; progressively, as time passed, the rate of progress has steadily increased, and in the last year it seemed at times that there was a risk of its becoming too fast.

In theory, two groups considering the same subject could have led to two different standards or to excessive duplication of work. In practice there was sufficient common membership that liaison was excellent, and these risks were avoided. On a number of occasions the totally different structure of the two groups made it possible for decisions to be made in one group when deadlock threatened in the other.

There were many controversial issues that needed to be resolved; to make matters worse, a number of issues were often interrelated. It is not the intention to give a blow-by blow account of these controversies, but to give the relevant history when this improves the understanding of the decisions made or of the concepts. Probably the most important concept, without which agreement would have been impossible, was that we should not define the digital recorder format, but should define the format of the digital recording laid on the tape.

The other concepts to be introduced will include: the self-contained segment, segments in a field, the 2/4 channel approach, the gap-in-the-middle, distribution of words between sectors, distribution of words within sectors, channel coding and randomization, video and audio error correction and concealment, and overlap editing. For each concept or group of interrelated concepts, the general approach will be to define its purpose, to give the background thoughts behind the concept, to introduce the specific terms or jargon, to explain how different realizations of the concept may occur depending on the structure of the recorder, to mention the history if this might be expected to improve understanding of the concept, but not to go into details of how the concept is realized.



Charles J. Lipow, Chairman of the SMPTE Public Relations Advisory Committee, before the press at the press briefing.

8. The User Requirements for the 4:2:2 Component Digital VTR. *By William C. Nicholls*, CBS Operations and Engineering, New York, N.Y.

The agreement on the 4:2:2 Component Digital VTR standard by the SMPTE and the EBU represents an outstanding example of international cooperation involving both broadcasters and manufacturers. A number of ad hoc groups were formed in the SMPTE to work on the various aspects of the standard. One of these ad hoc groups was composed entirely of users, and had as its charge the generation of a report detailing user requirements for the 4:2:2 machine. The report accurately described the users' primary requirements, but did not specify an excessive amount of detail which would stifle design and manufacturing creativity.

Deliberate use is made of the words "shall," should," and "may" to convey the relative urgency of each requirement. Many items are listed without specifying their exact implementation, thus allowing manufacturers to develop innovative methods.

The main sections of the User's Report are: General Requirements; Tape Characteristics; Cassettes; Performance Parameters in Normal Play Mode; Operational Requirements; and Maintainability.

This paper will address specific user requirements from the above list and explain the thought processes and discussions which formed the background of each item. They cover configuration, functionality, operation, performance, and maintenance. The delicate balance between user requirement, manufacturer preferences, and market considerations will also be addressed.

9. Type D-1 Digital Television Tape Recorders: An Overview *By Bernard L. Dickens*, CBS Engineering and Development, New York, N.Y.; Chairman, SMPTE Video Recording and Reproduction Technology Committee.

The D-1 format was planned to meet the needs of TV broadcasters and program producers in the all-digital studio environment. However, consideration must be given to utilization of the D-1 format recorder in the interim period when televison studios will use both digital and analog equipment. In this paper, the author will discuss the work of the new ad hoc group created to study the application of D-1 recorders in current and future studios. One possible configuration of the D-1 studio recorder to meet the interim hybrid studio requirements will be presented.

The author will also provide an overview of the D-1 format and recording system.

10. Magnetic Media for DTTR. *By Arthur R. Moore and Michael P. Sharrock*, 3M Co., St. Paul, Minn.

A general discussion of the industry trends for magnetic media will be given, with comments on new materials. The digital television tape recorder (DTTR) requires the use of today's advanced particulate tapes to satisfy the system needs. The tape requirements imposed by these needs will be discussed. The tape industry appears to be capable of supplying media having the signal-to-noise ratios and the error rates, and will be addressing the reliability factors as the DTTR enters the marketplace.

The DTTR 4:2:2 is designed to use 16 or 13μ-thick, 19mm-wide tape packaged in cassettes. The system is capable of making 20 generations of copies, while retaining broadcast quality. Details of the testing results obtained with the tape samples submitted to the SMPTE committee will be summarized.

11. The SMPTE D-1 Cassette Family. *By K. Ike/P. Dare*, Sony Corp., Teaneck, N.J.

Unlike previous videotape formats using a cassette implementation, the SMPTE D-1 format consists of a family of cassettes specifically designed to meet user-defined needs. As a starting point, an examination of broadcasters' needs was made, and a reference to recent improvements in basic cassette technology provided a substantial starting point for an intensive study which led to a final approved design.

The SMPTE D-1 cassette family consists of three sizes of cassettes, allowing for play time varying from 11 min to 94 min. In addition, a series of identification holes allows for a variety of tape thickness and tape surface properties to be sensed by the recorder player, along with four plugs for end-user application. The SMPTE D-1 cassette design has been a cooperative effort between the EBU, SMPTE user groups, and manufacturers. This paper will describe the characteristics of the cassette in detail.

12. Introduction to the SMPTE D-1 Format and to the Related Mechanical Configurations. *By Takeo Eguchi*, Sony Corp., Atsugi Plant, Atsugi-shi, Kanagawa-ken, Japan.

After a number of committee discussions and technical demonstrations, the DTTR Working Group of the SMPTE and the EBU MAGNUM Committee finally achieved the first worldwide digital video recording format now designated as the SMPTE D-1 format.

This paper will introduce the tape pattern and some of the associated discussions which led to the D-1 format, and will describe three difference mechanical configurations which create the same footprint on the tape.

It is expected that there will be some supporting practical results for the D-1 format shortly, such as the achievement of initial tape interchange between manufacturers.

Type D-1 Digital Television Recorder (Part II)

13. The D-1 DTTR: The Design of the Electrical Part of the Standard. *By Jürgen K. R. Heitmann, Robert Bosch GmbH*, Darmstadt, W. Germany.

The mechanical part of the D-1 standard defines the magnetic pattern on tape. The basic mechanical work was done prior to the electrical work but not without looking at the electrical problems. The D-1 track pattern allows the recording of 227 Mbit/sec of data. The standard defines how this data rate is split between video according to CCIR 601, and four digital audio signals according to AES/48 kHz standardization. Major work was done to find the best solution for synchronization, error protection, and video and audio sample distribution, i.e., to define where on tape each sample should be recorded.

The mechanical standard leads to a segmented recording, using more than one scan to record a field. Out of 525 lines forming a frame, 500 lines are recorded by removing the vertical blanking interval. Ten helical tracks are needed for a field of 250 lines. It was agreed within the SMPTE Working Group on DTTR to distribute the incoming video signal over four adjacent tracks, insuring a minimum sensitivity of the recorded signal against tape defects such as drop-outs.

Unfortunately, it is impossible to divide the number of helical tracks/field (10) by 4. Therefore it was decided to divide each helical track into two, one upper half and one lower half. This results in 20 half-tracks/field, a number which can be divided by 4. Each half-track contains one video sector and two audio sectors. Four video sectors form a segment of 50 lines.

Following the explanation of video segments and video sectors, the paper will deal with the distribution of four digital audio signals into the video recording. Audio and video are not only sharing the same heads, but to a large extent, the same electronics. A basic block diagram of a D-1 standard DVTR will be described.

In general, a data signal is not suitable for the direct recording onto a magnetic medium. The reason is the dc-content of the signal, which cannot be recorded due to the coupling of the rotating heads to the outside world with rotary transformers. Another point is that dc magnetization cannot be reproduced by ferrite heads. It is the task of the channel coding to reduce the disturbing dc-content of the digital video and audio signals prior to recording. In the D-1 format, this is done by a randomization or scrambling of the signals. The randomization leads to a reduction of the correlation in a serial bit sequence so that it statistically approximates a random sequence.The occurrence of long sequences of logical 1s or logical 0s is reduced to a level which can be handled by the built-in error-correction scheme.

Finally, the tracking control record will be explained.

14. Specifying Error Correction and Detection for a DVTR. *By John P. Watney*, Ampex Corp., Redwood City, Calif.

This paper will look at some of the experimental results and engineering judgment calls that went into estimating the bit error rate statistics that might be expected from tapes to be used in the DVTR, and the error concealment and misdetection rate that would be acceptable to the user. Knowing the gap between what is available and what is required allows the correct error-corrections system to be specified to bridge the gap.

15. Formatting and Coding the Audio in the DTTR. *By Kenneth P. Davies*, Canadian Broadcasting Corp., Montreal, Quebec, Canada.

The recording of digital audio in the D-1 format makes use of small sections of the helical tracks and associated channels which are optimized for recording video data. Appropriate processing and pre-

Leonard Coleman (left) and Jack Spring, at the Kodak-sponsored Friday evening reception.

coding must be used to adapt this channel to the special needs of the audio data for secure, low error-rate reproduction. In addition, the processing must consider the need to interface with digital signal sources, the synchronization requirements between the data streams in the DTTR, and the special functions required for audio editing and other operational needs.

The paper discusses the conclusions reached for the D-1 recording format in detail, their implications, and the potential applications of this format. In addition, a summary is included of the essential elements concerning the longitudinal tracks provided for audio and time code in the D-1 format.

16. The SMPTE Type D-1 Digital Television Recorder — Error Control. *By J. H. Wilkinson*, Sony Broadcast Ltd., Basingstoke, U.K.

The objective of the paper will be to explain the method of error control employed in the SMPTE D-1 digital television recorder. Starting with an introduction, which will outline fundamental aspects of information engineering, the paper will then move on to outline the objectives of the error-control scheme. The nature of redundancy against error-correcting power will be explained, but in conjunction with the constraints set by the operational requirements of slow-motion and high-speed shuttle replay.

In the main body of the paper, the first part will deal with the concept of the concatenated error-coding structure, with its inner and outer code blocks. The use of a combined error-correcting, error-detecting code for the inner code will then define the optimum decoding procedure for all operational conditions, with full error correction possible in normal play and slow motion, and with inner correction only for replay at high shuttle speeds.

The paper will then move on to detail the size of the concatenated array and the levels of redundancy in both inner and outer code blocks, and this will ultimately define the dropout correction capability. Two key aspects of the error-control performance will then be calculated in detail.

First, a tape error model will highlight the nature of errors which must be corrected in normal play. From the tape model, and from the defined decoding procedure, the rate of uncorrected error samples can be calculated. This computation involves some statistical calculations, the results of which can be used to indicate whether the visual impairments at the 20th generation will be acceptable to broadcast users.

The second aspect of error-control performance is the likelihood of failing to indicate the location of uncorrected errors. The visual impairment due to undetected errors is much greater than that for concealed errors. The rate of detection failure is therefore an important aspect of the error control code and is calculated for the most critical operation: slow-motion replay. The results of both computations are positive, and indicate that the performance of the error-control scheme should exceed broadcasters' expectations.

No error-control scheme can be decoded without some means to define the location of the block start position. The paper will explain the principles of the sync block format, which is used both to define the location of the block start position and further to identify the block position in a four-field window. Since the details of sync block construction are available in the D-1 format documents, the paper will highlight the objectives of the sync block format and a method of decoding. The decoding of valid blocks and the rejection of invalid data both exceed the capacity of the error control coding.

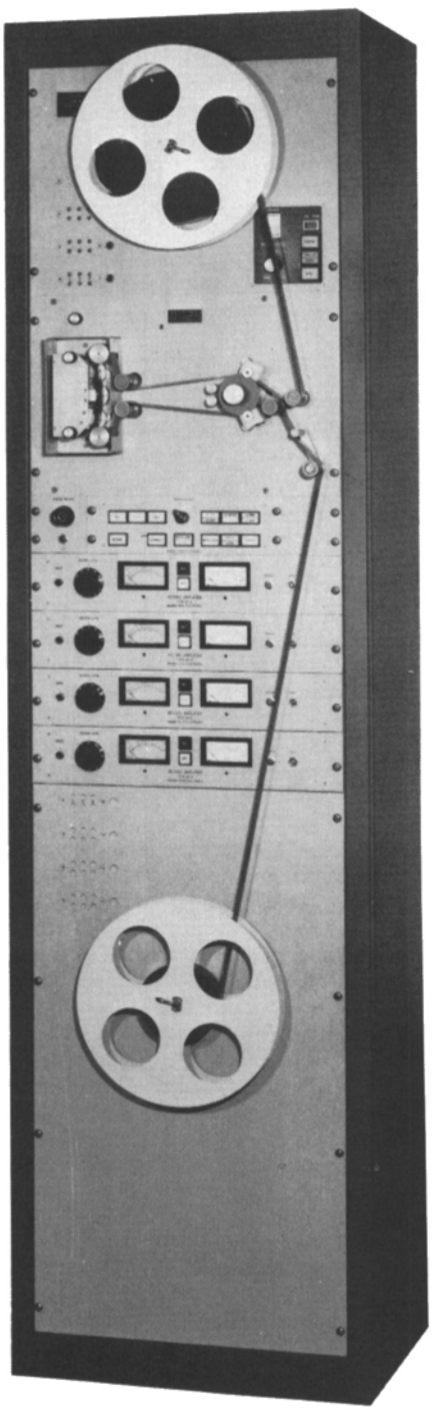
Finally, the paper will conclude with the observation that the detailed aspects of error-control coding, including both the mechanics of error-control coding and the computations involved in the decoding processes, are the realms of the specialized equipment designer. To understand the error-control scheme of the D-1 format does not require a detailed understanding of the precise mechanics, but does require an understanding of coding objectives and their reponse to a given tape error model. The paper will concentrate on this latter part, as it represents the interests of a wider group of the expected audience.

17. Shuffling Algorithm for the 4:2:2 DVTR. *By Richard Brush*, Ampex Corp., Redwood City, Calif.

The author's paper will be divided into three parts. The first part will be introductory and will discuss shuffling in general terms, as it might apply to any digital video tape recorder (DVTR). The second part will discuss the requirements and nature of shuffling as it specifically applies to the 4:2:2 DVTR. The relationship between shuffling and other system level considerations, especially the error-correction coding (ECC) method, will be emphasized. The third part will discuss the particular shuffling algorithm proposed by Ampex, and adopted by SMPTE in the 4:2:2 DVTR standard. The superior error-concealment properties, picture-in-shuttle quality, and ease and simplicity of implementations will be emphasized.

During the first part of the talk, the author will answer the questions: "What is shuffling and why it is necessary or desirable for a DVTR?" The importance of shuffling to error concealment and picture-in-shuttle quality will be explained. A criterion will be presented for quantitatively evaluating a shuffling algorithm with respect to concealment, namely the largest tape defect that can be concealed using specified concealment filters. Criteria will also be presented for evaluating a shuffling algorithm with respect to picture-in-shuttle quality. These include uniformity of pixel distribution on the TV screen, freedom from obvious geometric patterns or artifacts, and updating cosited pixel components simultaneously.

The second part of the talk discusses shuffling as it applies to the 4:2:2 DVTR. There is an intimate relationship between shuffling, the segmented recording format on tape, and the ECC method, which



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uses a two-dimensional Reed-Solomon (R-S) code.

The "sector memory" plays a central role in this relationship. It is a byte-wide memory organized as a matrix of 600 columns by 32 rows, holding the equivalent of 50 lines of video data. During record, outer ECC code blocks (30 video data plus 2 outer check bytes) are written into the memory array column-wise. Inner code blocks (60 data bytes, including video plus outer check bytes) are read from the array row-wise. Mathematically, a shuffling algorithm can be regarded as a permutation of the column and row addresses while writing the outer blocks into the sector memory.

Cosited CB and CR bytes on the TV screen are in the same inner code block on tape, ensuring they will always be updated from the same field during shuttle mode. To achieve this, it is necessary to perform a simple recording of the bytes within a line before they are sent through the outer coder. This is referred to as the intraline shuffle, whereas the permutation of the sector memory addresses is referred to as the intrasector shuffle, which is done in such a manner as to preserve the cositing achieved by the intraline shuffle.

There is a compromise between too little and too much randomness in the permutation functions. Too much randomness gives good picture-in-shuttle quality at the expense of concealment, whereas too little randomness gives the opposite. Numerous permutation functions based on arithmetic congruence formulas were investigated, using two interactive computer programs written at Ampex. One program analyzed the concealment properties of a shuffling algorithm, while the other gave an indication of picture-in-shuttle quality.

A shuffling algorithm was found which has excellent concealment properties and good picture-in-shuttle quality. To improve shuttle quality even more, a 4-field sequence variation was introduced, which amounts to a reversal and/or offset of the sector memory row addresses when reading out data to the inner coder. The shuffling algorithm was proposed by Ampex, and accepted by SMPTE for the 4:2:2 DVTR standard. A block diagram of the shuffling algorithm shows that it may be implemented using three small PROMS for the permutation functions.

18. The Optimization of the D-1 DTTR Standard by Simulation Techniques. *By Roland Mester, Robert Bosch GmbH,* Darmstadt, W. Germany.

The D-1 DTTR standard was agreed upon without the hardware realization of a D-1 DTTR. This could only be done through sufficient practical experience with experimental digital recording systems. Measurements made with these systems were basics in the standardization process. Experimental experience led to realistic mathematical models, with which new approaches could be simulated and optimized. The D-1 standard reflects the theoretical developments based on realized experimental systems.

A realistic model founded on measurements is basic for every optimization by simulation. The model's complexity depends mainly on the achieved improvements. Building up a model requires measurements of essential hardware independent of system parameters, allowing an abstraction of the equipment used.

A complex model itself should be an assembly of several submodels, each describing a special part of the system. Modification of one of these submodels must not change the behavior of others. Thus the whole system can be optimized by independent modifications of the subsystems. Each modification step shows the effects on the complete system caused by a subsystem.

Experimental measurements of error visibility in digital TV signals made by Bosch and the BBC were the starting point in finding the best error-protection scheme. A derived mathematical model permitted the development and optimization of a redundancy-free mapping code, reducing the visibility of single and double-bit errors compared to binary coding.

The operating range of a DTTR error-protection system is determined by the off-tape error rate and the desired residual error rate. The errors requiring concealment should be below 200/sec according to measurements for a perfect picture quality. The off-tape errors are a combination of single-bit and burst errors. This error rate was measured by several manufacturers of experimental models with a pseudorandom data sequence, representing a digital video signal. An average error statistic was agreed upon by the SMPTE working group as the basis for further work. Tape-error statistics gave the possibility of calculating the performance of various error-correction codes for normal play and stunt modes.

A model was created concerning data recovery in case of cross-tracking in fast and slow-motion modes. In most stunt modes only one part of the error-correction scheme, the inner code, is working. With the model, it could be verified that the error-detection capability of the inner code is sufficient in stunt modes, also.

The analysis of cross-tracking also allowed a realistic estimation of data readability at all tape transport speeds. Combined with a model of several scanner types, an analysis of data pickup in stunt mode was possible.

A simulation program developed by the author is able to calculate the process of data collection at every tape speed. It handles any scanner type and head arrangement as well as any track pattern and width. Data distribution within the tracks (shuffling) can be treated independently, and every possible shuffling scheme could be simulated. A graphic output of data collection while crossing the tracks gives a realistic impression of the way scanner types and shuffling schemes work together at different tape transport speeds.

The simulation was verified by a shuffling scheme in an experimental system. As seen in earlier attempts to optimize shuffling, no hardware system was as flexible as the computer simulation. This was the tool for the experts of the SMPTE Working Group to use in optimizing the shuffling scheme independent of any given electronic or mechanical hardware.

19. Measurements and Diagnostics in a digital Videotape Recorder. *By Rolf Hedtke*, Robert Bosch GmbH, Darmstadt, W. Germany.

The introduction of the digital videotape recorder (DVTR) will change all test and measurement procedures dramatically. Today a service engineer can monitor the analog waveforms with a simple oscilloscope, and can decide which part of the system is faulty. This is not possible with the DVTR. The video and audio signals are digital data streams which are processed in many ways.

The user is expecting a new generation of products to simplify all the service and maintenance work. Therefore, the engineers will have to develop not only a DVTR but, in addition, new methods which simplify the testing procedure to achieve better maintenance and higher availability of the system.

The digital videotape recorder can be divided into three parts:

1. The "head-to-tape" electronics is the most critical part. Here the digital data stream is converted into analog current for recording, and vice-versa for playback. The parameters of this channel must be adjusted very precisely because this part mainly determines the quality of the DVTR. An insufficient adjustment results in an increased error rate which can lead to an overload of the error protection scheme. Therefore it is necessary to measure the basic parameters of the channel: signal-to-noise ratio, frequency characteristics, and group delay behavior. This can be done by a special "missing pulse" test signal. A spectrum analyzer is needed for the interpretation of the channel's response. The built-in error counter can help to adjust the parameters because misadjustments result in increased error rates.

2. The signal processing part of the DVTR is a pure digital system. Therefore one can use the well-known techniques of digital trouble-shooting: With signature analysis, for example, a special digital test pattern is used as a stimulus, and one can trace the footprints of the signal by comparing the signature with the precalculated pattern at fixed test points. This procedure can be done automatically if a

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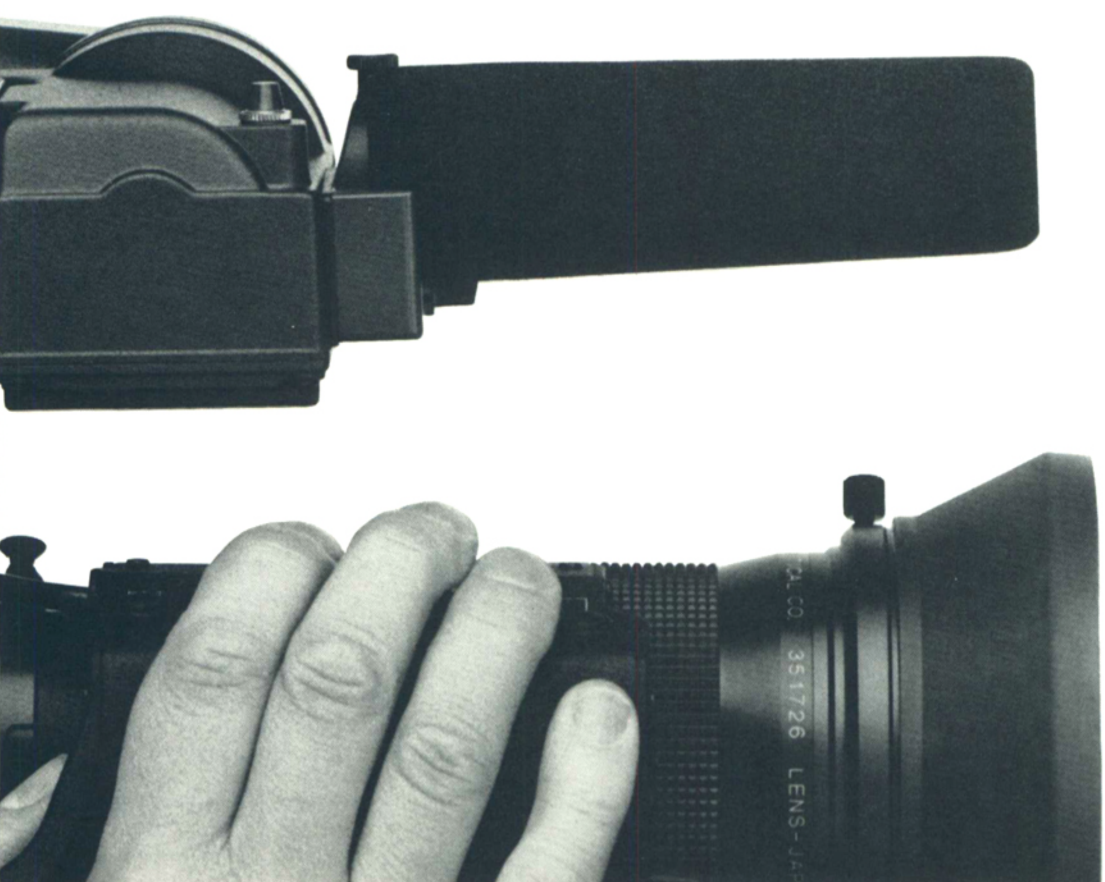
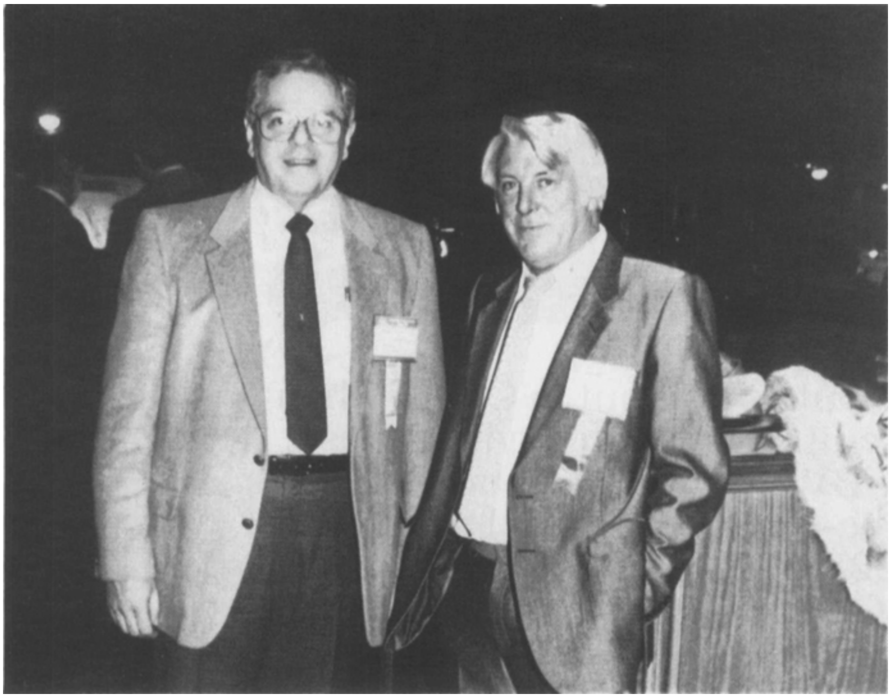




General Arrangements Co-Chairmen Norman Thelen (left) and Michael Bailey.

microcomputer is used where all the necessary information of the signatures is stored in its internal memory.

3. The control systems of modern videotape recorders normally include a microcomputer with integrated self-diagnostic capabilities.

A digital videotape recorder is a very complex machine. Special test and diagnostic instruments are needed. Built-in diagnostics will reduce the time for service and maintenance and increase the availability of the system.

20. Panel Discussion.

Moderator: Federick M. Remley, *University of Michigan*
Panelists: John L. E. Baldwin, *Independent Broadcasting Authority*
Kenneth P. Davies, *Canadian Broadcasting Corp.*
Bernard L. Dickens, *CBS Engineering & Development*
Takeo Eguchi, *Sony Corp.*
John P. Watney, *Ampex Corp.*

Television Post-Production Techniques

21. "Harry" and the SMPTE Standard in the Edit Suite. *By Howard Shephard*, Quantel Ltd., Newbury, England.

Harry is a real-time digital video recorder which can replay frames in random order in real time. Harry can therefore edit material on a frame-by-frame basis without the delays normally associated with the mechanics of editing, i.e., tape shuttling, recueing and synchronizing. All the prerequisites for editing short segments exist within Harry. Clips may be cut, keyed, or mixed together without using any external equipment.

Video editing has taken on a new dimension in recent years since the advent of high-quality graphics systems such as the Quantel Paintbox. The Paintbox has the capability of retouching video frames with such realism that an expert could not tell where the alteration has taken place. In addition, production techniques such as cartooning and rotoscoping, previously the domain of the optical animation laboratory, can be done electronically using a Paintbox. However, the Paintbox is limited in that it is designed to work with still graphics, i.e., single frames of video. Video retouching, cartoon animation, and rotoscoping all, by definition, involve stringing together frames of moving video.

Harry has full random access capability to record and replay frames, which can be instantly and digitally transferred between the Paintbox and Harry, using the SMPTE 4:2:2 digital standard. The paper makes the case for using the digital component standard in the edit suite and discusses the advantages to the creative user and to the engineer of taking this approach.

The choice of the SMPTE 4:2:2 recording standard for Harry allowed the design of a flexible production tool which will maintain first-generation quality by digitally interfacing to other equipment such as the Paintbox and in the future other devices that use the SMPTE standard.

22. Panel Discussion.

Moderator: Robert J. Vavra, *Video Corp. of America/Technicolor*
Panelists: Anthony Izzo, *Edit Chicago*
Mickey Mitidero, *Lake Shore Post Production*
Lenard E. Pearlman, *Editel/Div. Columbia Pictures*
Jimmy Smyth, *Optimus*
David Triunfal, *Swell Pictures*
Jack A. Weinberg, *Post Pro Video, Ltd.*

New Developments in Video Recording

23. Introduction of Small Format Videotape Recorders in NHK. *By Iwao Obata*, NHK (Japan Broadcasting Corp.), Tokyo, Japan.

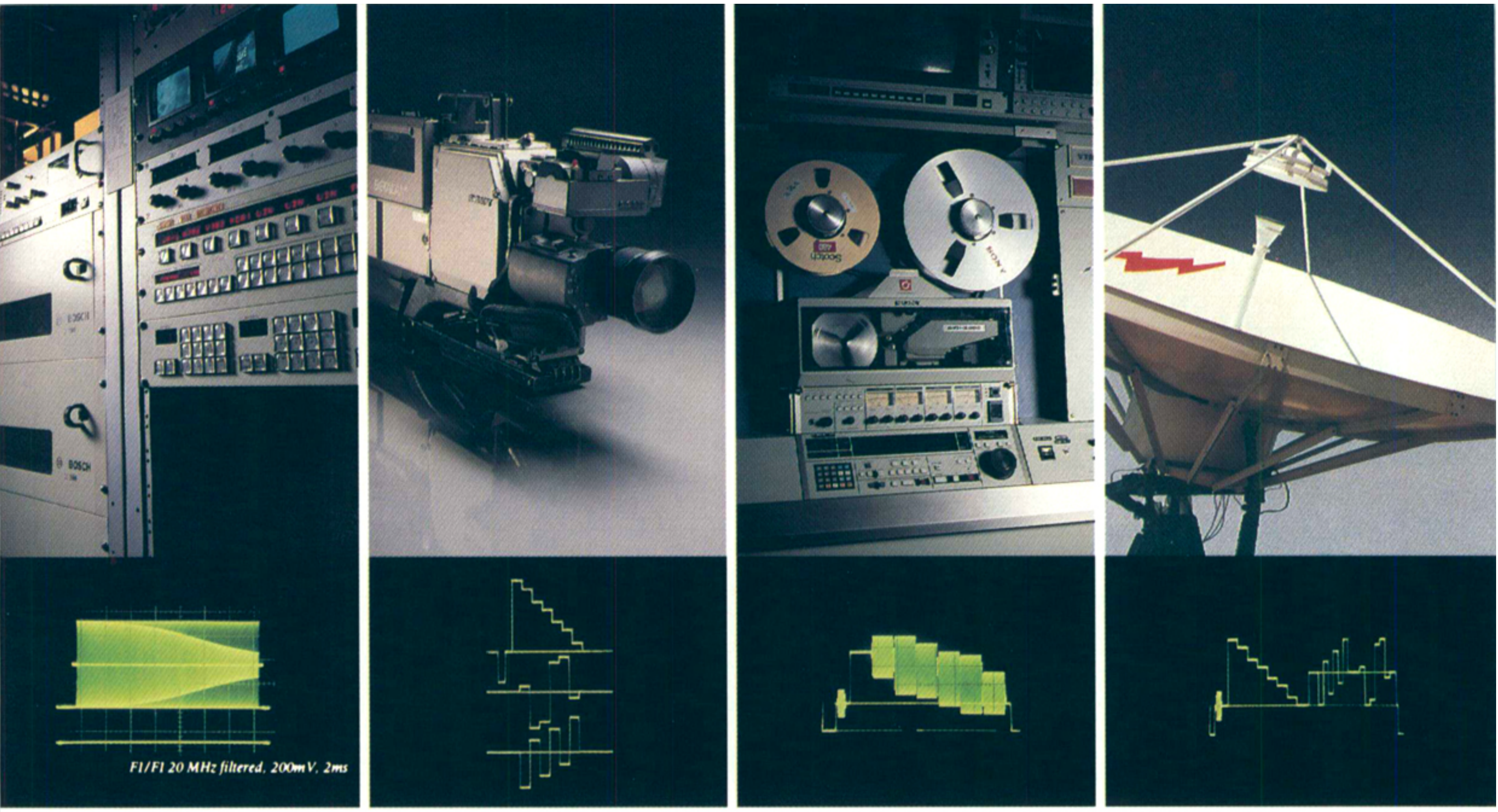
At NHK, there are many requirements of VTRs for the production of programs and ENG. There is no one format to meet all of these needs. We employ 1-in. VTRs for studio use, and recently we have developed a new small-format VTR (MII) with the Matsushita Electric Co. in order to replace the old 2-inch VTRs at our local stations. We also use Sony Betacams for our extensive ENG operations. Ideally, we would like a single format for all applications, but we do not anticipate finding such a format for at least several years.

24. PCM Audio Recording on An MII Format VTR. *By Shiro Tsuji, Masamitsu Ohtsu, and Mobuyoshi Kihara*, Matsushita Electric Industrial Co., Moriguchi, Osaka, Japan.

An MII-format VTR for television broadcast use has been recently developed. The first generation of this VTR adopts an analog audio recording method using longitudinal tracks, but a PCM audio recording method using rotary heads has been taken into consideration to get excellent audio quality.

A new MII-format VTR, with a PCM audio recording method, has been successfully developed, diverting one of the longitudinal audio tracks to PCM audio recording using rotary heads. In this system, two channels of PCM audio signals are separately time compressed and recorded by rotary heads on the extended luminance and chrominance video signal recording tracks (about 22° tape wrapping angle for PCM audio).

For realizing high-quality PCM audio, the AES recommends a standard in which the sampling frequency of 48 kHz and quantization of 16 bits is adopted. The



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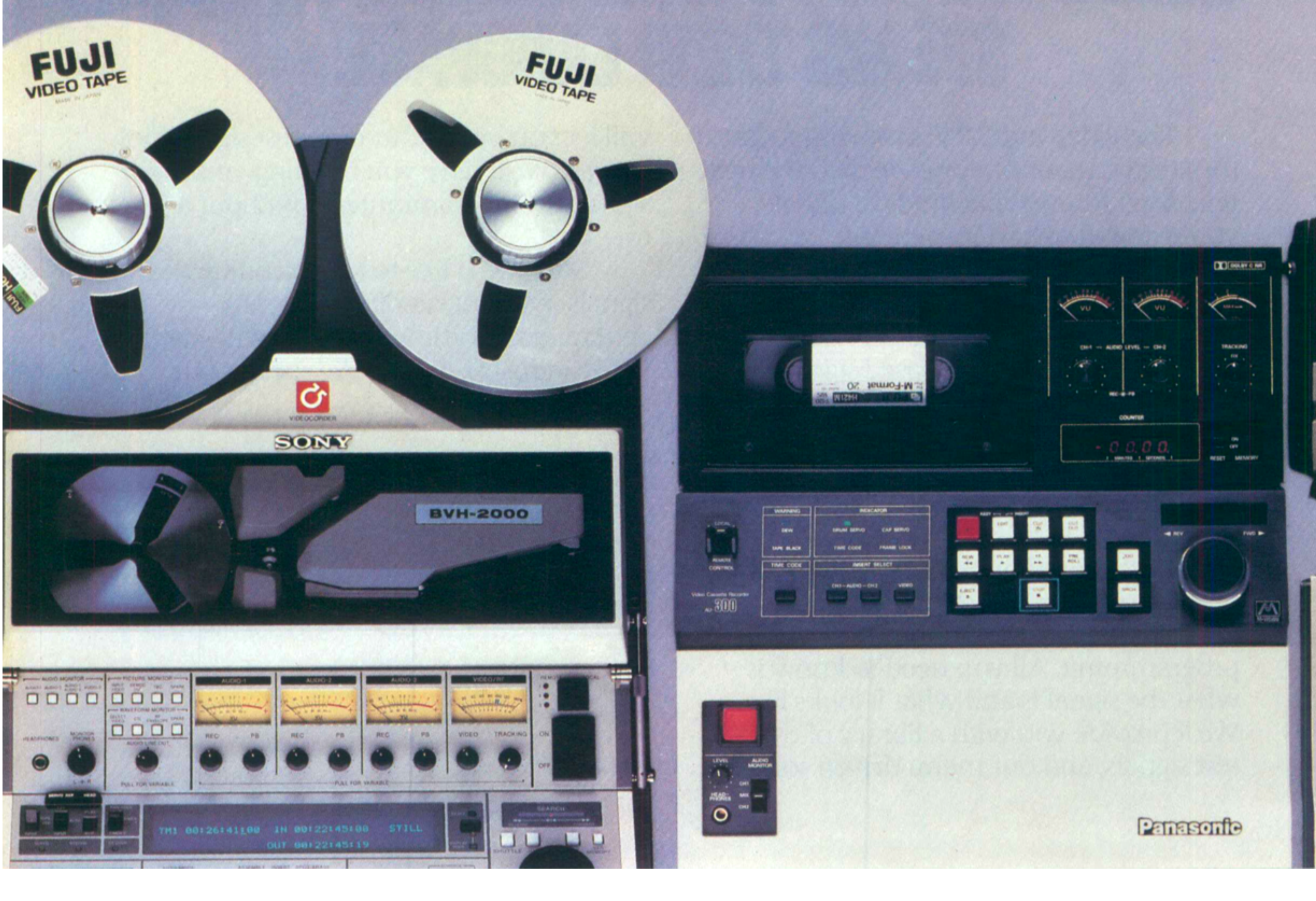

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PCM signal of one TV field consists of 67 blocks/channel, and each block consists of sychronous signal, address data, CRCC for address, audio data of 24 symbols (8 bits/symbol), and error correction code.

For the error-correction code, the combination of two Reed-Solomon codes ([32, 28, 5], [28, 24, 5]) over Galois Field (2^8) is adopted, after investigating the most suitable method for MII VTR PCM audio. These Reed-Solomon codes are generated in each field audio data. Using this error-correction method, powerful error correction and low probability of miscorrection occurrence are realized by means of 4 times of error flags.

The recording data rate of this time-base-compressed PCM signal is 11 Mbits/sec, which is a relatively high data rate for the MII VTR. Therefore, a new modulation method improving recording density needed to be developed.

It was decided to revise the 8–14 conversion code with greater T min (larger density ratio), smaller T max, and less dc energy, considering sync signal code and better clock-regeneration. This 8–14 conversion code is dc-free and has a respectively large density ratio of DR = 1.14.

The typical symbol error rate in MII VTR PCM audio using this 8–14 conversion code is about 10^{-4}, and in this case, the uncorrected error probability is about 10^{-23}.

Almost all the editing functions required for broadcasting-use VTR are realized, such as one-field-period accuracy editing, separate channel editing, digital fade-in/fade-out at editing point, and digital dubbing. A leap-field problem caused by a different number of samples/field in the NTSC system is solved by recording an index signal mixed with PCM audio data, and field-accurate editing can be done by processing the leap-field using this index signal.

In using time-compressed PCM audio recording in a VTR, a reproduced audio signal is delayed relative to the reproduced video signal. This problem was solved by adopting relative reproducing heads which trace PCM audio tracks to video head tracing relative video tracks. (In the recording mode, PCM audio is recorded by the same heads as the video recording heads.)

As described above, a broadcasting VTR with high-quality PCM audio has been developed, which satisfies almost all of the functions required for broadcasting use.

25. New Small-Format VTR Using 8mm Cassette. *By Toshiaki Kawamura, Susumu Kasai, Tamotsu Tominaga, Hideo Sato, and Minoru Inatsu,* Hitachi Denshi, Ltd., Tokyo, Japan.

Hitachi Denshi, Ltd. has developed a new small-format videotape recorder using an 8mm metal tape cassette. The system using this new small-format VTR is a compact, lightweight, high-performance camera/VTR for ENG/EFP.

The luminance signal and TDM chrominance signal are recorded separately on two video tracks to achieve a high-density recording. Thus, a high resolution and a high signal-to-noise ratio are obtained. (Luminance channel band of 4.5 MHz, chrominance channel band of 1.5 MHz, SNR better than 48 dB.)

A family of products has been developed that presently consists of: a combined Camera/VTR; a field playback adaptor (when the adaptor is combined with VTR detached from camera, broadcastable color TV signal is reproduced); and a studio player/recorder that connects directly to existing editors or controllers for use with another present VTR system.

The paper describes the recording techniques and equipment, with technical descriptions of both.

26. Considerations on the Improvement of an HDTV Digital VTR. *By Yoshizumi Eto, Masuo Umemoto,* Central Research Laboratory, Hitachi, Kokubunji, Tokyo, Japan; and *Toshiaki Kawamura,* Hitachi Denshi, Ltd., Kodaira, Tokyo, Japan.

The results of tests on our experimental HDTV digital VTR were presented at the 19th Annual SMPTE TV Conference, and were also demonstrated at the NAB '85 Show. This five-head, 1-in.-tape machine operates at a bit rate of 460 Mbits/sec, which had been confirmed to be the minimum rate needed to obtain acceptable HDTV picture quality, and which appeared to be the maximum available bit rate needed to implement a practical digital VTR at that time. However, with the latest developments in recording technology, even higher bit rates and resultant expanded applications of HDTV digital VTRs may be expected.

There are two approaches to consider as the coding parameters for an improved HDTV digital VTR. One is to follow the sampling rate being discussed for adoption as the HDTV studio standard or its lower hierarchy level. The other is to regard a digital VTR as a high-quality VTR with an analog interface, which has a sampling rate high enough for analog HDTV signals. In either approach, at least a 30–40% bit rate increase over our 460 Mbits/sec machine may be required.

According to our basic research, about 1 hr of recording time at the above bit rate will be possible, assuming the use of 1-in. or 19mm tape with reasonable tape consumption. Electronic commonality between 4:2:2 digital VTRs and HDTV digital VTRs is desirable. Some examples of coding parameters for an improved HDTV digital VTR have been studied.

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