Synopses of Papers Presented at the 19th Television Conference

Digital Components

1. Digital Component Television Made Simple for Everyone. *By E. Stanley Busby, Jr.*, Ampex Corp., Redwood City, Calif.

This is a pure tutorial, intended to leave a vivid and memorable recollection of the essentials of digitization and the ramifications of the spatial loci of the samples taken. No stand is taken on any particular sampling method. Any arguments will be those made previously by others, and will be couched in the very simplest of analogs. The intent is to create an understanding where none exists, rather than expand upon an incomplete understanding.

2. Digital Production Switchers. *By Jacques Vallee, Max Artigalas, and Michel Favreau*, Thomson-CSF, Gennevilliers, France.

Following the 601 CCIR recommendation on a world-wide digital standard, the company Thomson-CSF has developed a great range of equipment for the French Television Experimental Studio of Rennes. The digital switcher constitutes the heart of the studio.

The authors described:

- The chosen distribution system and the associated switching matrix
- The internal structure of the mixing unit comprising four effects buses and two preset program buses allowing four plane pictures to be made (two backgrounds and two foregrounds)
- Classical accessories like electronic pattern generator, colored backgrounds generator, and subtitling unit
- Some new features only feasible when using digital processing and digital recording, mainly: downstream chroma key, accurate color correction, and automatic phasing between different input signals by the direct use of digital signal words described in the SMPTE/EBU recommendation.

Maintenance procedure has been carefully studied. All the functions' inputs and outputs are easily attainable on a dedicated bus to be displayed. Moreover, the video register pin-out standardization allows video signal visualization on an analog form at every level by the use of a specific tool comprising a compact D/A converter probe.

3. A High Bandwidth Multiport Frame Store System. *By Charles A. Poynton*, Poynton Vector Corp., Ottawa, Ont., Canada.

This frame-store system was developed for NASA as the basis for a new system to convert the field-sequential color television signals from the cameras on board the Space Shuttle to composite NTSC.

A general approach to the system design of this frame store integrates in one unit functions normally ascribed to such diverse items of television equipment as infinite-window time-base correctors, synchronizers, still stores, standards converters, character generators, test signal generators, video analysis equipment, and computer graphics frame buffers. These functions require various configurations of what a computer designer calls "ports" to the central video memory. For instance, a synchronizer requires one full-rate write (input) port and one full-rate read (output) port operating at independent addresses. A computer graphics frame buffer typically requires one slow-speed read/write port for computer access, and one full-rate read port for video refresh.

It is not generally realized that constraints normally associated with frame stores such as subcarrier dependence, sampling structure (samples/line, lines/frame), and interaction between ports are not inherent but are merely conveniences to the designers of particular pieces of equipment. A general approach to the implementation of a memory system can remove these constraints.

The Poynton Vector frame-store system implements five completely independent ports. Each port can be operated at any rate from zero to full video rate, each is dynamically selectable between write and read functions, and the addressing and timing of each port is free from interaction with the other ports. The sample structure of memory can be chosen arbitrarily, and the system can be "populated" in various configurations to implement the storage and bandwidth required for a particular application. Further, much of this configuration can be done dynamically under software control by an external PDP-11/23 computer. The computer also has direct access to the main video memory and access to parameters associated with real-time video processing.

The NASA "color conversion" system consists of three such frame stores (one each of red, green, and blue), digital summing and matrixing modules, digital television input and output modules, and a computer interface module. The requirements of the color conversion demand an aggregate frame-store data rate of about 200 Mbytes/sec. The system incorporates monitoring, fault detection, and packaging features which are necessary for its use in an operational television broadcast environment.

4. Digital TV Tape Recording: A Report of Progress Toward a Standard. *By F. M. Remley*, Chairman, SMPTE WG-DTTR; The University of Michigan Media Center, Ann Arbor, Mich.

As a result of widespread interest in the development of digital television systems, the SMPTE Vice-President for Engineering established a Study Group on Digital Television Tape Recording more than three years ago. For two years, that group energetically examined the many variables involved in defining a video recording system suited for use in the digital television studios that will be placed into service during the late 1980's. Early in the discussions, it became apparent that the future digital broadcast video recorder could not be a device evolving directly from present recorder designs; new technology would be required to make such a machine practical. During this period several laboratories around the world began seriously to develop experimental machines.

In late 1983, the International Radio Consultative Committee (CCIR) approved a carefully described, universally applicable, digital television studio signal specification, now known as CCIR Recommendation 601. It also became apparent that technology was, indeed, advancing at a rapid pace, and that several equipment manufacturers were ready to discuss specific details of possible DTTR formats. After due discussion within SMPTE, it was decided that the Study Group should be disbanded and, in its place, a Working Group for Digital Television Tape Recording (WG-DTTR) be formed, reporting to the Video Recording and Reproducing Technology Committee (VRRT) and to the Vice-President for Engineering. By this means the interests of both equipment designers and potential users of digital video recorders could best be taken into account as actual format proposals were developed.

In 1984, the work of defining an SMPTE digital television tape-recording format began in earnest. The WG-DTTR met at six-week intervals during the year, always with participating experts from North America, Europe, and the Far East. In addition, a joint meeting and equipment demonstration was held in September, 1984, in Winchester, England, with the MAGNUM Specialists Group of the European Broadcasting Union. MAGNUM has been assigned responsibility, by the EBU Technical Committee, of defining a 625-line digital recording specification based on CCIR Recommendation 601. The joint meeting was fruitful, and the pace of progress was accelerated. Specific areas of intense study include video encoding, audio encoding, tape and tape cassette design, transport characteristics, signal processing, and editing provisions.

At the time of writing this synopsis it seems likely that SMPTE and EBU will succeed in completing specifications for a single digital video recording format. If this is the case, the results will be contributed to the CCIR, which meets again in late 1985. This paper will summarize the current state of digital television tape-recorder format specifications up to the time of the 1985 SMPTE Television Conference.

5. The All-Digital Studio Is Here. *By D. Nasse*, TDF/CCETT, Rennes, France; *J. L. Grimaldi*, Thomson-CSF, Paris, France; and *A. Cayet*, TDF, Paris, France.

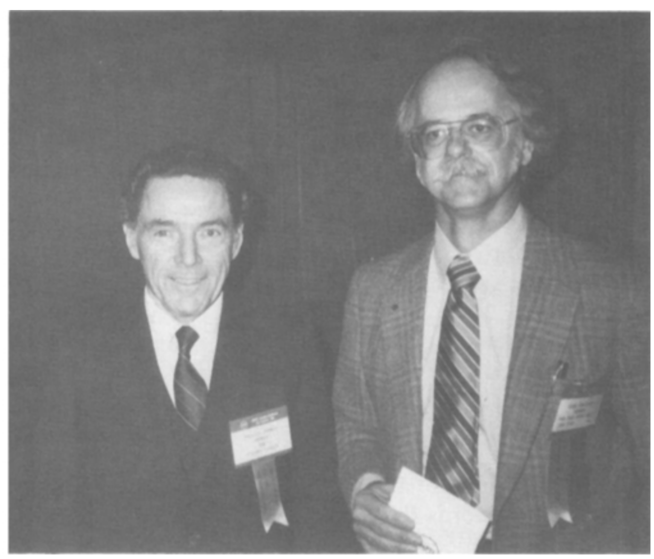
The all-digital television plant has been a dream for the past ten years or so. This is now being realized, as an experimental digital studio is in the process of final implementation in Rennes, France.

The project was initiated in the beginning of 1981, as it was recognized that the relative complexity of digital functions, together with the limited manufacturing runs expected, might prevent desirable equipment from being marketed. It was felt that to make a meaningful assessment of the capabilities of all-digital systems, it was necessary to go farther than the laboratory experiments shown up to that time and to make an experimental but real-size video system on actual marketed products. This project was made possible by the setting of the 4:2:2 digital standard, the progress achieved in the area of digital video recording, and the experience already gained by various French laboratories in digital television and processing.

The all-digital system places emphasis on post-production, but production capabilities include cameras and other picture sources, and an all-digital slide scanner. The most important question is that of the VTRs. By the time the project was initiated, digital video recording was widely demonstrated, but many other requirements were not, although some were subsequently shown on experimental machines (e.g., variable speed). Others are still to appear (e.g., digital audio editing). It was decided to retain the capabilities that are most important, i.e., digital video editing and reliable operation, at the expense of audio (still conventional analog) and non-standard speed operation (no picture in shuttle and no stop motion). This is still thought to permit real-size operation with reasonable feasibility and reliability margins. The most complex piece of equipment is the digital mixer. Initially developed for the project, it features a number of conventional and non-conventional capabilities and is described in a companion paper.

All of the equipment is now ready, and final assembly on location is in progress. The project is supported jointly by the French societies of the public broadcasting service and by Thomson-CSF. Operation is expected to start in mid-1985.

Program Chairman John Streets (right) with Program Committee Member Gary Thompson.



Conference Vice-President Maurice L. French (left) and General Arrangements Chairman Glen Pensinger.

The session auditorium was filled to capacity.

6. Digital Video Standards: A Progress Report. *By Stanley Baron*, Thomson-CSF Broadcast, Inc., Stamford, Conn.

This paper provides a snapshot of the current status of SMPTE efforts in the area of digital video component interface standards for use in studio applications. Electrical and mechanical interface requirements and considerations are given. The technical bases and performance level requirements of CCIR 601, SMPTE RP-125, and current efforts towards a serial digital standard are provided. The relationship among the three documents is also described.

CCIR 601

The SMPTE has directed its efforts towards encouraging standards having worldwide compatibility. During 1980 and 1981, the combined efforts of the SMPTE Working Group on Digital Video Standards, chaired at that time by Ken Davies, and the Special Task Force on Digital Video Standards, chaired by Frank Davidoff, developed liaison channels between the SMPTE and like organizations in other administrations, for the purpose of exchanging information relating to worldwide standards.

These efforts, combined with the efforts of similar organizations in other parts of the world, resulted in CCIR Recommendation 601, Encoding Parameters of Digital Television for Studios. Recommendation 601 adopted basic digital component coding standards, defining a sampling frequency at 13.5 MHz and a sampling structure.

The details of CCIR Recommendation 601 are provided.

RP-125 Bit-Parallel Digital Interface for Component Video Signals

The SMPTE Working Group on Digital Video Standards then developed a bit-parallel digital interface based on CCIR 601. The document defining this interface has been published as RP-125.

RP-125 defines an interface for System M (525/60) digital television equipment based on CCIR Recommendation 601. The characteristics of the interface are summarized below:

1. The video signal is transmitted in the form of one luminance and two color-difference components (*Y*, *R-Y*, and *B-Y*).
2. The video signal is transmitted at the 4:2:2 family level of CCIR Recommendation 601, with a luminance sampling frequency of 13.5 MHz.
3. The bits of the digital code words that describe the video signal are transmitted in a parallel arrangement using eight conductor pairs. Each pair carries a multiplexed stream of bits (of the same significance) for each of the component signals, *Y*, *R-Y*, and *B-Y*. A ninth conductor pair carries a clock signal at 27 MHz.
4. The interface consists of one transmitter and one receiver in a point-to-point connection.
5. Parameters of the signal format were chosen to facilitate conversion to and from a serial digital interface format.
6. The interface allows the transmission of the appropriate ancillary signals that may be multiplexed into

SMPTE Executive Director Lynne Robinson with Program Chairman John Streets.

the data stream during video blanking intervals.

The electrical and mechanical details of RP-125 are provided.

RP-125 does not yet completely describe the interface. The Working Group continues to study filtering requirements, particularly the shaping filters for the luminance and color-difference signals, and reconstruction filter requirements. The subject of the encoding of ancillary signals also remains to be defined.

Serial Digital Interface for Component Video Systems

In investigating the studio implementation of digital interface, the desirability of being able to transmit the signal over the already in-place, coaxial cable quickly becomes apparent.

Work has begun on a serial interface based on the following assumptions:

1. The serial signal format was derived from the parallel format as specified in SMPTE RP-125.
2. The interface signal should be compatible with coaxial and optical fiber operation.
3. The format adopted must have a small low-frequency content in order to allow ac coupling to prevent the buildup of dc potential due to earth loops and to simplify low-frequency equalization. An 8/9 bit-mapped block code was selected.
4. Parameters of the bit-mapping system and the signal format were chosen to facilitate conversion to and from the parallel digital interface format. The parallel format timing reference signal was assumed to be always present.
5. The interface must allow the transmission of appropriate ancillary signals that may be multiplexed into the data stream during video blanking intervals.
6. The code selected must exhibit limited overhead in order to restrict the frequency range and clocking rate of the high-speed circuitry.
7. The code must be simple to implement.

The electrical and mechanical details of the Serial Document, as they exist, are provided.

7. The SMPTE-EBU Control Network: A Progress Report. *By Michael Stickler*, British Broadcasting Corp., London, England; and *Thomas R. Meyer*, Dynair Electronics, San Diego, Calif.

Joint efforts by the SMPTE Subcommittee for Remote Control and the European Broadcast Union (EBU) Specialist Group for Remote Control have produced a complete basic architecture for a control network for television. The network is based on the concept of distributed processing, with each item of controlled equipment connected to the network through an intelligent interface or tributary.

Network specifications are divided into logical service "layers" according to principles established by the International Organization for Standardization (ISO). Four layers are specified for the network:

- Electrical/Mechanical. The lowest layer — specifies the electrical and mechanical characteristics of the communications channel.
- Supervisory. Provides access to the communications channel and guarantees error-free delivery of messages.
- System Service. Provides routing of messages between tributaries, identification of the type of equipment connected to a tributary, and blocking/de-blocking of long messages.
- Virtual Machine. Responds to control messages in a defined manner, regardless of the type of physical equipment connected to the tributary. Each type of virtual machine uses a distinct dialect within the overall control language.

The specifications permit use of the lower two layers of the network pending final specifications for the higher layers. The higher layers provide for "user-defined" or nonstandard control messages which will not conflict with standardized control messages. These provisions permit early use of the network, with orderly transition to the full standards as they become available. Several manufacturers are now using the network on this basis.

SMPTE has finished work on two specifications of the basic network architecture: ANSI/SMPTE 207M, Electrical and Mechanical Characteristics; and SMPTE RP-113, Supervisory Protocol. The remaining basic specifications are complete and will be published in the *SMPTE Journal* as proposed recommended practices in the near future: Tributary Interconnect (system service layer); and Control Message Architecture (virtual machine layer). An overview of the network is in preparation and will be published as an Engineering Committee Report (ECR).

The EBU has approved and published a single specification for the control network: Tech. 3245, Specification of a Remote Control System for Broadcasting Equipment. The specification includes an overview of the network and four sections which are functionally identical to the SMPTE specifications.

Specialists from the SMPTE and EBU groups have prepared three proposed sets of standard messages to be used within the network: system service messages, common control messages, and VTR control messages. Public discussion and comment for these documents has started.

Work has been started on prepara-

Session Chairman Gary Thompson (left) and Session Co-Chairman Jess Stone.

tion of control message dialects for other types of equipment. We invite all interested parties to participate in the development of these standards.

The work of the EBU and SMPTE groups is conducted primarily through the CONFER computer conferencing facility. This service allows participation in the work of the groups at minimal cost.

Analog Components

8. In the Beginning, There Were Red, Green, and Blue: A Tutorial on Component Video and a Working Group Progress Report. *By S. Merrill Weiss*, Chairman, SMPTE Working Group on Component Analog Video Standards; Imagex Corp., Berkeley, Calif.

Since their earliest experiments, developers of color television systems have recognized that there are three primary colors. They are red, green, and blue. Among them, they carry all the information that can be obtained about a picture, including brightness, hue, and saturation. Through varying combinations of amplitudes of the three primaries, all levels of brightness, all shades of color, and all intensities of those colors can be created.

Television cameras use optical separation of these primaries to permit them to be separately sensed by pickup devices for sending to distant monitors. Monitors, in turn, use separate phosphors for each of the primaries and combine them optically by recreating the picture out of phosphor dots or stripes too small for the eye to resolve. The eye integrates the small spots of the primary colors into small areas having all the characteristics of the original.

In the early development of color television, the emphasis was to find means to distribute color pictures, in particular to the home. Little or no thought was given to developing a production system, since many of the production capabilities taken for granted today had not yet even been conceptualized.

Monochrome television had been available to the public for a number of years, and there was a moderate number of receivers in the public's homes. Therefore, one constraint placed on the delivery system was that it be "compatible" with existing receivers. This meant that the monochrome portion of the color signal had to work with monochrome receivers just as a monochrome signal would. It also meant that the signal had to fit into the same channel bandwidth as the original monochrome signal.

Some of the earliest attempts at color television actually sent the three primary colors separately, one after another in sequence. This field-sequential color depended on the eye to integrate the primaries over time rather than over space. It worked as long as there was no motion in the picture. Motion produced colored edges. This might be called the first component color system because it kept the components of the picture (in this case, the primaries) separated in time.

The systems which eventually came to be accepted for distribution are the so-called composite techniques. In these, by matrixing the primaries, a luminance signal is developed to carry the brightness information. It then is sent in the same way as the original monochrome signal. The primaries are also matrixed to produce a pair of color-difference signals which carry the hue and saturation information. The color-difference signals are bandwidth-restricted and then modulated onto a subcarrier which is interleaved into the upper spectrum of the luminance signal. These systems are the NTSC, PAL, and SECAM systems currently in use around the world.

It is the artifacts of the interleaving process which cause most of the problems with composite signals. The most notable of the artifacts are the cross-color and cross-luminance effects which are quite visible in the picture. Next in importance are the facts that, so far, it has been impossible to build a perfect decoder, at any price, or to build a very good decoder at a reasonable price. Furthermore, even if a perfect decoder were available, the bandwidth-limited color-difference signals do not carry enough information to adequately perform any of the required functions. This is most important from a production standpoint when one considers that many of the techniques used in production today depend upon decoding the signal first. Examples are encoded chroma-keying and any form of picture manipulation which resizes the picture. The problem becomes even more critical when the multiple passes of today's production through the encoding and decoding process are taken into account. The degradations from encoding and decoding can ruin what started out as a very good picture.

Modern technology now permits a solution to the problem of encoding and decoding. The primaries themselves or matrixed signals derived from them, such as luminance and color difference, can be kept separated while they are moved from one piece of equipment to another, and at the same time, the bandwidths of the components can be tailored to what is required in the production process. Since the parts of the picture are kept apart, these are called component systems. The separation can be achieved in one of several ways: separation in space, separation in time, or separation in frequency.

Separation in space requires that the

More than 700 people registered for this technical conference, which resulted in the auditorium being filled to capacity at almost all times.

components be carried over separate pathways from one place to another. This implies an individual connector cable for each component, or a connector with several coaxial circuits and a cable with several coaxial pairs. For small systems such as editing suites or small studios, this parallel component system is the cleanest and most economical technique.

Separation in time requires that the components be placed on a single cable one after another in a time-division multiplex. Since the three components all originate at the same time and take the same time, a way must be found to shorten them so that they will all fit sequentially in the time they originally took together. This is done through a technique called time compression, in which the individual components are squeezed to fit the time available. In this process, the frequency spectra of the components are multiplied by an amount proportionate to the amount of the squeeze, so that the final signal takes up a wider bandwidth. This type of signal is more expensive than parallel interconnections but is useful for connecting together larger systems, including subsystems built with the parallel interface. Such a serial interface has proven to be compatible with most existing facilities built for composite operations. It also has the advantage of being the easiest to translate to and from the digital component form.

Separation in frequency requires that at least two of the components be modulated on a carrier or carriers so that each can have its own spectrum area, in what is called frequency-division multiplex. The components can be the primaries or signals derived from a matrix of the primaries. This technique may be slightly less expensive to implement than time-division multiplex but has the disadvantage of being more susceptible to distortions in the signal path. It is also much more difficult to translate to and from the digital component environment.

The Working Group on Component Analog Video Standards is working to standardize the parallel component interface and the time-division multiplexed form of serial component interface (called S-MAC, multiplexed analog components for the studio), as well as an interface for the direct interconnection of cameras and recorders. Results of these efforts are expected to be published shortly.

9. Component Analog Switcher Design: A General Overview. *By Birney D. Dayton*, The Grass Valley Group, Inc., Grass Valley, Calif.

Not available at time of publication.

10. Component Processing in Time-Base Correctors and Post-Production Switchers. *By David E. Acker*, FOR-A Corp., West Newton, Mass.

Part I — Component Production Switchers

The advent of component recording, coupled with the improvements offered by doing certain types of video processing in component form, has created the need for component production switchers. The lack of standards for component signals, however, raises questions as to the best processing format for such equipment.

Part I of this paper explores some of the advantages and some of the problems of component processing. It deals with the decisions as to the best signal format to use within a production switcher and also discusses some of the other aspects of the design implementation.

Part II — Time-Base Correction

When the digital time-base corrector (TBC) was introduced and used with the ¾-in. color-under format VTRs, electronic news gathering (ENG) was born. The rapid advancement of digital technology that followed has allowed a steady and significant improvement in TBC performance.

Part of the performance improvement was achieved with the use of component signal processing within the TBC. When coupled with component VTRs, the combination provided significantly improved pictures relative to ¾-in. formats. The component TBC is also an important bridge that permits further component signal processing in production switchers, chroma-keyers, and effects equipment.

Part II of this paper will present an overview of the evolution of TBC component processing and define some of the basic terminology associated with this technology. Block diagrams of principal system configurations and some discussion of their advantages and disadvantages will be presented.

11. Component Codec. *By Virgil Lowe*, Fortel Inc., Norcross, Ga.

Not available at time of publication.

12. New and Unique Method for Measuring Video Analog Component Signal Parameters. *By Dan Baker*, Tektronix, Inc., Beaverton, Ore.

Based on the growing need to distribute and process the video signal in analog component form, research has been conducted to provide an *X/Y* display capable of simultaneously monitoring three video component signals. Since relative amplitude and delay inequality are important parameters to adjust and quantify in the component environment, such a display should provide a clear and accurate illustration of these errors. This paper reports on the progress of the evaluation of an *X/Y* display that approaches these goals.

Traditional CRT displays (amplitude versus time) are certainly useful for component measurements. To allow simultaneous display of more than one component, such techniques as a progressive parade or overlay of selected components would be useful for relative amplitude measurements. This type of display, however, may be difficult to use with standard test signals, such as color bars, and is inadequate for determining small delay inequalities in color bar transitions.

A vectorscope provides a two-dimensional display of the two color-difference signals in an *X* versus *Y* Cartesian form. Although relative subcarrier phase would not exist in a component environment, the vector display could still be quite useful for relative chroma amplitudes and delay. The delay inequalities are contained in the transitions between dots when used with a standard color bar test signal. The dot positions provide chrominance amplitude indication. In this way the entire color gamut is displayed and *R-Y* to *B-Y* timing errors are exposed. The luminance information is not displayed, however, and there is no indication of luminance amplitude or delay errors.

The proposed new display would make use of the advantages of an *X* versus *Y* type of format by displaying luminance versus *R-Y* and luminance versus *B-Y*. The *Y* (luminance), *R-Y*, and *B-Y* signals could be fed directly to the monitor; or if the component system is *RGB*, the monitor would encode these signals into *Y*, *R-Y*, and *B-Y*.

First, a line of positive luminance signal is applied to the vertical deflection while, at the same time, a line of the *R-Y* color-difference signal is applied as horizontal deflection. Second, a line of the negative of the luminance signal is applied to vertical deflection and a line of the *B-Y* color-difference signal is applied to the horizontal deflection.

The back porch of the luminance and the zero value of the color-difference signals are clamped to screen center. The resulting display with a standard color bar test signal provides independent indication of luminance, *R-Y* and *B-Y* amplitudes, and encoding errors. The transitions between dots indicate the relative delay inequality of all three components with resolution better than 100 nsec. The display can be used to evaluate relative delay in *RGB* components as well. Using this new display with a sinusoidal test signal (as proposed by the IBA) provides delay inequality measurement with a resolution of a few nsec.

This paper discusses the generation, interpretation, and utility of this new display for component video monitoring and measurement.

13. Improved PAL by a Combination of NTSC, SECAM, and PAL. *By Gerhard Holoch*, Institut fuer Rundfunktechnik, Munich, West Germany.

In improving an existing television system, it is of utmost importance that the new signals are compatible with existing transmission characteristics such as radio frequency (RF), and with cable links and video recorders as well as with existing receivers. More-

Session Chairman Vince Perry (left) and Session Co-Chairman Graeme Little.

over the improved system should be able to work with the signals of the conventional system.

This paper describes an improved PAL (I-PAL) system which meets the requirements mentioned above. Cross-color and cross-luminance are removed and the horizontal resolution of the luminance signal is enhanced. A similar system with a different target had been proposed in the early 70's.

I-PAL can be considered to be a mixture of NTSC, SECAM, and standard PAL. It has the quadrature modulation from NTSC, the line sequential color transmission from SECAM, and the line alternating burst phase from PAL. The basic idea is that at the sending end, during the first line a luminance signal is transmitted with the full bandwidth of the channel, together with the normal PAL burst signal at 4.43 MHz. No color information is transmitted during this line. The second line carries a luminance signal with reduced bandwidth, the normal PAL burst, and a chrominance signal in frequency multiplex format. The chrominance component is quadrature modulated as in NTSC. It is essential that the spectra of the luminance signal and the chrominance signal do not overlap. The signal in the third line corresponds to the signal of the first line. This line-alternate format may be transmitted with or without a frame reset of the count at the start of the frame. Without a frame reset, the information of the even and odd lines is exchanged every frame.

At the receiving end, the first line provides a wide-band luminance signal. In the following line, the narrow-band luminance signal is complemented by the high-frequency luminance components of the previous line by means of a line delay. Due to this, the luminance signal has a full horizontal resolution in all lines. The resolution in the diagonal direction, however, must be restricted by a suitable prefilter in order to avoid alias structures.

As to the chrominance signal, a line repeat post-filter in the receiver interpolates the missing lines. This can be accomplished by the conventional PAL delay line circuit and an additional electronic switch. In line-sequential color transmission, aliasing components are produced within the original spectrum which can cause disturbing effects on vertical highly saturated color transitions. By prefiltering the color-difference signals prior to alternate line omission, these alias defects must be reduced to an insignificant level. This procedure limits the vertical resolution of the color information. With a line repeat post-filter and a pre-filter of the same degree, the vertical resolution is restricted to an equivalent horizontal resolution of 1.85 MHz, −6 dB.

An additional drawback of this system is its sensitivity to phase errors as the modulated chrominance signal no longer changes its phase in alternate lines. However, this problem (if this is a problem at all) can be overcome by using a frame reset mode. In this way a phase alternation frame (PAF) signal is transmitted, the phase errors of which might be compensated at the receiving end by means of a frame repeat post-filter. The use of a frame delay in the receiver also offers the possibility to avoid the loss of diagonal resolution for the luminance signal mentioned above.

There is evidence from ample subjective assessment tests that I-PAL really guarantees an improved picture quality with I-PAL receivers. The downward compatible reception (I-PAL signal on a standard PAL receiver) suffers from a small loss in signal-to-noise ratio and slightly increased crosstalk. It should be mentioned, however, that with a modified version of I-PAL full compatibility can be retained. In this case, color information is transmitted in all lines but in such a way that in two consecutive lines the information is absolutely the same. With this modified principle, the NTSC system might be improved in a compatible way.

Session Chairman Joe Roizen (left) and Session Co-Chairman John Carlson.

14. B-MAC: A Standard Transmission for Pay DBS. *By Dr. Keith Lucas*, Digital Video Systems, Toronto, Ont., Canada

The signals in use today for the transmission of broadcast TV services are direct descendants of the original monochrome transmissions of the 1940's. The method adopted for the addition of color (namely the color subcarrier) was dictated by the requirement for compatibility with the early monochrome receivers, and by the technology then available. Since that time, there have been significant advances in technology, due in particular to the advent of large-scale integration of electronic components. Moreover, the economic environment in which television services must operate has been transformed through the introduction of cable TV, VCRs, home computers, etc., which compete for the screen. (It is foreseeable that the home TV receiver may evolve into a monitor, fed by a series of standalone input devices, each providing a separate service.)

To be successful, any new medium must offer high-quality programming, with enhancements to the range and technical quality of the services provided. The cost of providing such programs will be high. At the same time, the increasing number of program channels available in the home will fragment the audience and decrease the value of associated advertising.

Consequently, there is a requirement to reach a very large audience and to augment advertising revenue through novel payment schemes related to program quality and cost.

It is considered that a viable new service may be based on the following attributes:

1. High quality programs
2. Direct-to-home satellite broadcasting (compatible with subsequent cable distribution)
3. Significant improvements to video quality
4. Multiple-channel, high-quality digital audio
5. A range of useful data and text services
6. Upward compatibility to extended-definition TV (1.77:1 aspect ratio); downward compatibilty to existing receivers
7. A simple-to-use impulse pay-per-program feature
8. Low-cost equipment for the consumer.

Distribution by high-power satellite will ensure availability to a very large audience, and the above list of features will attract a significant percentage to accept the service.

The introduction of direct TV broadcast by satellite (DBS) offers a unique opportunity to provide a quantum improvement in the quality and range of services provided and to pave the way to higher-definition transmissions in the future. This may be achieved by employing a transmission standard which is optimized for the DBS medium, thereby providing high-quality services appropriate to the video, audio, and data equipment available to the consumer now and in the foreseeable future. Compatibility with conventional TV receivers can be provided in the decoder by transcoding to the current transmission standards using LSI techniques.

This paper describes a DBS transmission format which has been designed to satisfy the criteria just described and the range of services provided.

15. The D2-MAC Packet System for All Transmission Channels. *By J. Sabatier, D. Pommier, and M. Mathieu,* CCETT, Cesson-Sevigne, France.

The need to define new standards for television became apparent with the initial projects for direct broadcasting by satellite. This need reflected a wish to improve the services transmitted, both qualitatively and quantitatively, with a view to optimizing the use of the spectrum.

Although all the systems proposed up to 1984 checked out the constraints imposed by planning, it is more difficult to assert that the economical and operational aspects were always recognized for what they were. Indeed, only large-volume production permits the price of individual receivers to drop to a level that makes them an acceptable consumer item. This fundamental condition can be met only if a single system operates on all available transmission networks and answers to the various service requirements.

Consequently, the major aims for such a system are:

- Availability of at least four high-quality sound channels to be broadcast simultaneously
- Improvement in picture quality over current broadcasting standards
- Capability to transmit in various cable distribution networks all the service components conveyed by a satellite channel
- Use of the same standard for VHF-UHF terrestrial broadcasting
- Flexibility of structure to allow the later introduction of new services and improvement in quality
- Suitable implementation of control access procedures with this system.

After almost five years of research carried out within the European Broadcasting Union (EBU), the awareness of such new features has led the CCETT to design the so-called D2-MAC packet system, as a member of the MAC-packet family promoted by the EBU. The D2-MAC packet system consists of:

- Baseband time multiplex between the digital and analog components (TDM)
- Duobinary coding for the digital signal
- Line and frame synchronization
- Digital service multiplexing using a packet procedure
- Picture transmission by companded and time-multiplexed analog components (MAC)
- Sound coding methods, using either near-instantaneous companding law or linear law and a Hamming code for sample protection
- Control of the TDM
- Conditional access procedures
- Service identification by means of a dedicated digital channel.

One of the noticeable properties of MAC transmission is clearly the lack of any absolute limitation on the bandwidth reduction. This interesting property is lost as soon as the MAC signal component is multiplexed with a digital signal having a bandwidth greater than that of the MAC signal. The D2-MAC packet system has been designed in such a way that a bandwidth reduction down to 4.5 MHz can be allowed. Experiments on satellite and terrestrial networks have validated the good performance of the system either with wide or narrow-channel bandwidth.

After a brief chronological summary of the various stages of the work, the paper describes all these characteristics of the D2-MAC packet system and gives practical results and measurements obtained in DBS, VHF, MATV, or CATV. Whatever the reactions to these subjects around the world, the CCETT's laboratories have, like others elsewhere in Europe, played an active part in standardization and will continue to do so.

16. Panel Discussion

Merrill Weiss, Moderator
Birney Dayton
Stanley Baron
Larry Thorpe
Charles Poynton
Dominique Nasse
David Griffin

Future Technology

17. Technical History of Home VTR Development. *By Yuma Shiraishi,* Victor Co. of Japan, Ltd., Yokohama, Japan.

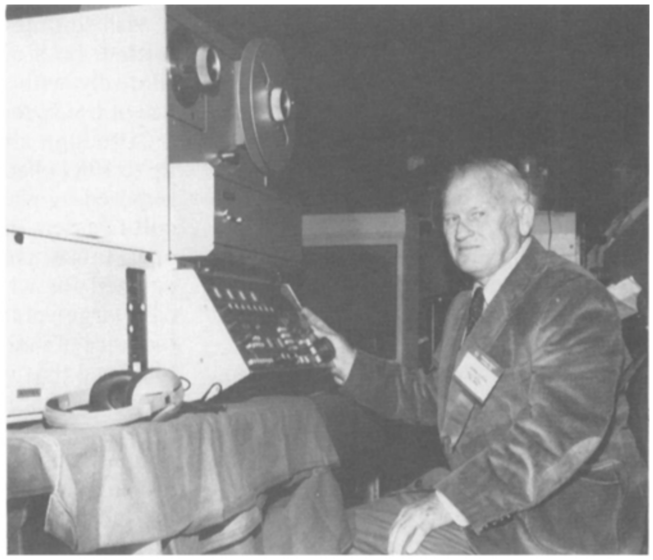
The penetration of home VCRs into the consumer market is proceeding rapidly. Video technology is increasingly seen as a necessity, not only as a television-signal recorder, but also as an information-transfer medium through prerecorded cassettes.

Now is a good time to look back on the early days of home video development to see how the original conditions required for the popularization of this machine were met by specific technical developments.

Birth of the VTR

Although many years of research

Registration Chairman Donna Foster-Roizen (center) with Larry Filby (left) and John Corso (right).



Audio/Visual Chairman Vernon Kipping seated on the projection platform.

preceded it, the appearance in 1956 of the quadruplex VTR marks the beginning of video as we know it, because, remarkably, technology introduced in this system, such as FM recording, is still in use today.

Simplified Mechanism

The relative simplicity of today's home video recorders owes much to the two-head helical scanning system, whose development it is interesting to trace.

Magnetic Tape and Head

The high performance of today's oxide tapes and heads is the result of step-by-step progress over many years. The technical history leading to today's improved reliability will be examined.

Chrominance Signal Recording

Because a home recorder requires far less tape consumption than a professional machine, many steps were taken to arrive at an appropriate color signal recording system.

Operability

In retrospect, it is clear that the open-reel system had to be replaced by a cassette system for home use. What is more interesting is that the cassette format was subsequently adopted for professional use.

Long Playing Time

A key requirement for home video recorders is a recording time of at least two continuous hours. Several different techniques were combined to reach this objective.

Continuing Development

Once penetration into homes began, video encountered new demands. New techniques such as the miniaturization of Video Movie and the addition of Hi-Fi audio continue to be developed to meet these demands, thus widening the applications of home video.

Conclusion

Intrinsic in the whole process of home video development is the element of cost reduction. The presentation closes with a discussion of the cost-reducing process.

18. Editing with the DRAW Videodisk. *By Gary Mantz*, Spectra Image Inc., Burbank, Calif.; and *Bill Justus*, Ampex Corp., Redwood City, Calif.

Ampex and Spectra Image Inc. demonstrated videodisk editing at the 1984 NAB. The first shows have now been edited using videodisks. Experiences and insights from the ongoing development of these high-speed access editing systems and how they may change our editing habits will be presented.

The paper will demonstrate differences in quality and speed of the current ¾-in. videotape cassette standard and the DRAW laser videodisk standard. Observations from the first interactions of the Hollywood editing and production staffs with this medium will be given.

A means of carrying time code, frame numbers, and other picture information on the DRAW videodisk will be presented. Needed standards for the use of DRAW videodisk in the editing marketplace will also be discussed.

19. Resolution Considerations in Using CCD Imagers in Broadcast-Quality Cameras. *By Thomas M. Gurley and Carl J. Haslett*, RCA Corp., Gibbsboro, N.J.

A previous paper, given at the 126th SMPTE Technical Conference, presented considerations for using CCD imagers in broadcast-quality cameras, with emphasis on those CCD advantages that benefit ENG camera design. New ideas concerning static and dynamic resolution were offered, emphasizing the need for measurement techniques that correlate well with subjective assessment. It was shown that the superior performance of frame-transfer CCDs over pickup tubes — in the areas of sensitivity, SNR, lag, and highlight handling — can be exploited to portray motion more clearly. This is an important advantage, since the vast majority of televised scenes involve motion. This paper will provide further analysis of dynamic resolution and consider the effect of temporal resolution on motion reproduction.

Both film and television systems create a set of stationary records of the scenes they image. The quality of motion reproduction depends on the sharpness of detail preserved in each record, the continuity of action from one record to the next, the even weighting of records during playback, and the repetition rate. These factors are all embodied in the concepts of dynamic and temporal resolution.

Dynamic resolution is a measure of the faithful reproduction of fine detail in moving subjects. In real-time pictures, good dynamic resolution adds a feeling of presence. More importantly, blur-free stop action requires better dynamic resolution than tube cameras have been able to provide. Stop action converts real-time dynamic resolution to static resolution, so the presence or absence of fine detail is much more noticeable.

When a camera captures a scene, detail in any moving subject is blurred proportionally to the amount of relative movement that occurs while the image is being recorded. For a frame-transfer CCD camera, as for a film camera, this is the only factor affecting dynamic resolution. Both cameras use shuttering to control the exposure time, with faster shutters and shorter exposure times producing the highest dynamic resolution. For both, a simple mathematical expression relates dynamic resolution to the subject's velocity and the camera's shutter speed. Measurements taken using a variable-shutter-speed CCD camera have verified this expression. Good agreement between these measurements and subjective picture assessments have been demonstrated previously. For tube cameras, the simple expression does not apply. It must be expanded to consider lag and field-to-field image retention, which are more difficult to characterize mathematically. Because of these additional considerations, shuttering a tube camera, at the expense of sensitivity or SNR, provides limited improvement in dynamic resolution.

As the dynamic resolution of a camera is increased, effects related to its temporal resolution become more apparent. Temporal resolution is the ability to distinguish events that are closely spaced in time. It is determined by the number of images recorded per second. Since the records constitute a sampling of the scene in the time domain, the imaging is time-discrete. Thus, aliasing can occur when a temporal Nyquist rate, related to the system's imaging rate, is exceeded. For a moving repetitive pattern, this temporal aliasing appears as an incorrect velocity of the image. A well-known example of this effect is the apparently backward-turning wagon wheels seen in western movies. Video from a tube camera shows less temporal aliasing than film. This is due, in part, to the faster imaging rate of the television system. Primarily, though, it results from the tube's poor dynamic resolution. The higher dynamic response of the CCD camera — a major contributor to its praised "film look" — renders it susceptible to temporal aliasing, although to a lesser degree than film.

In designing a camera for stop-action or slow-motion applications, both temporal and dynamic resolution must be considered. An ENG camera based on frame-transfer CCDs has superior dynamic resolution to a tube camera. Its dynamic performance can be further enhanced for stop-action applications in sports productions, with no increase in system complexity, by merely increasing the camera's shutter speed. However, this increase in dynamic resolution can lead to strobing or judder effects in some slow-motion applications. Optimum slow motion requires greater temporal resolution, which, in turn, implies a faster imaging rate. The benefit must be traded off against the increase in system complexity involving special time-base-conversion hardware. Moreover, an implementation that provides adequate temporal resolution for smooth slow motion may not simultaneously provide sufficient dynamic resolution for blur-free stop action.

20. Super Motion System. *By Larry Thorpe*, Sony Broadcast Products Co., Teaneck, N.J.

Sony has developed a radically new television field-acquisition system specifically designed to enhance the capture of motion. The Super Motion system is comprised of a new genre of color television camera and a new VTR system. The Super Motion system can be considered a variation on a theme of high-definition television. A wideband camera system is employed to increase the temporal resolution of the camera, rather than the spatial resolution we are accustomed to associating with HDTV. The Super Motion system is specifically designed to overcome the motion judder and image blur associated with conventional TV pictures when played on a standard slow-motion recorder. This is accomplished by shooting the scene with a camera whose frame rate is three times higher than that of a normal 525/60 camera. In a given time period, therefore, three times as many TV frames are captured and thus enhanced capture of motion is allowed.

The novelty of the system lies within the specific technique employed to record this high-speed TV picture. The basic stipulation levied upon the system was that the final output should be a standard Type-C 1-in. videotape capable of replay on any standard Type-C VTR while retaining the enhanced motion resolution. The output of the system is of course a standard encoded 525-line NTSC video signal, off tape. This NTSC signal has been enhanced from the viewpoint of temporal resolution, however. This perhaps casts the Super Motion system as a significant contributor to the improved NTSC genre of signal, even though HDTV techniques are employed to achieve this.

The problem became, therefore, one of converting the high-speed frames to standard 525/60 NTSC, and then finding appropriate means to record all of these frames. An additional interesting system requirement was the provision of a real-time standard 525/60 NTSC output from the high-speed camera system. This was important in that it provided simultaneous on-air capability from the camera while recording the super motion signal.

The BVP-3000 Super Motion camera scans 525 TV lines at a 90-Hz frame rate (180 fields interlaced 2:1). In order to record this signal, it must first be standards-converted. This conversion is accomplished in the remote camera CCU, using contemporary digital technology. The final output of the camera CCU consists of four standard 525/60 NTSC signals with a spcific relative time delay between them. Three of these are individual representations of the high-speed frames originally captured by the camera. These three individual NTSC signals are fed to the BVH-2700 VTR.

The BVH-2700 is a specially modified 1-in. VTR. The drum contains three recording heads, spaced 120° apart. The tape, in the record mode, moves at three times normal Type-C speed. This combination produces a perfect standard Type-C recording on tape. Specifically, each of the NTSC signals is laid down on the tape, according to Type-C specifications, and each field of these separate signals is laid down in consecutive order. This tape can be played back on any standard Type-C machine, at the standard rate, but with three times normal temporal resolution.

The new technologies included in the Super Motion system comprise mixed-field pickup tube technology, fiber-optic transmission, digital processing and standards conversion, and high-speed C-MOS digital CSI chips to encode *RGB* to NTSC. This paper will describe each of these technologies.

The Equipment Exhibit.

21. Enhanced Television: A Progressive Experience. *By John L. E. Baldwin*, Independent Broadcasting Authority, Winchester, Hants, U.K.

Before adopting an incompatible higher-definition television standard, which cannot be broadcast in an acceptable bandwidth without the use of several field stores in every receiver, it is sensible to investigate the possibility of using field stores in receivers to provide noticeably better subjective vertical resolution by progressively scanning the display. The transmission would continue to use conventional interlaced 525 or 625-line standards. If aliasing due to the sampling inherent in the number of lines in a displayed field is the important limitation, then the use of progressive scanning in the display could result in a doubling of vertical resolution. This paper reports on the progress of this investigation at the Independent Broadcasting Authority (IBA).

For normal composite television systems there are three causes of moving artifacts on stationary and nearly stationary pictures: cross color, cross luminance, and interlace effects. While the removal of the two cross-effects achieved by adopting component systems has a number of benefits, it leaves the eye with more time to spend on noticing the moving artifacts associated with interlace.

If looked at in one way, there is no doubt that interlace works, since with 525 lines in $^1/_{30}$ sec, the large area flicker frequency is 60 Hz using interlace, rather than the 30 Hz it would be for progressive scan. On the other hand, looking at an interlaced monitor at normal brightness, the pitch of the lines appears to be double that which would be expected from calculation, for example 1 mm compared to 0.5 mm. Interlace can only work with static pictures, since the rate of movement, in the vertical direction, of a picture height in 8 sec completely neutralizes the effect of interlace.

Investigation has shown that it is not the use of interlace for transmission that is the main problem, but the use of interlace of the display. Fortunately the cost of field stores is rapidly decreasing, and it seems likely that it will become practical to use these in domestic receivers with larger screens before the end of this decade.

An experimental converter which accepts a 625-line 25-Hz 2:1 interlaced input and provides a 625-line 50-Hz progressive scan output is operating and providing very satisfactory results with conventional camera inputs. It is adaptive, depending on the amount of movement, and works well for normal pictures. On computer-generated moving graphics there has been a problem with the movement adaption, but today this has been substantially overcome. However, it is necessary for tests with a wider range of such material to take place.

Provision has been made in a proposal for a satellite transmission system to increase the aspect ratio to about 5:3 and the horizontal resolution to the equivalent of that achievable with a luminance bandwidth of 20 MHz on a 1125-line 30-Hz 2:1 interlaced standard. The transmitted signal is interlaced and would simultaneously provide a service for normal reception on 4:3 aspect ratio interlaced displays as well as enhanced reception with improved resolution on 5:3 aspect ratio progressively scanned displays.

22. High-Definition Optical Video Disc by MUSE. *By Tateo Toyama, Yoshihiro Morita, Osamu Ohta, Toshiaki Hioki, and Yasuhiro Ishii,* Sanyo Electric Co., Development Center, Gifu, Japan; *and Yuichi Ninomiya, Yoshimichi Ohtsuka, Yoshinori Izumi, and Seiichi Goushii,* NHK Science and Technical Research Laboratories, Tokyo, Japan.

A high-definition television system having 1125 horizontal lines has been developed, mainly by NHK in Japan. This system has about five times the volume of data capacity compared to the existing television system, and is able to create a marvelous high-quality picture for projection onto the wide screen.

In order to realize this high-definition television system, the bandwidth must be over 20 MHz. NHK developed a new method called MUSE (multiple sub-Nyquist sampling encoding) in January 1984. By this MUSE system, the wide bandwidth of the HDTV signal is compressed into approximately 8 MHz, or twice the frequency of the existing television format such as NTSC.

In cooperation with NHK, Sanyo has developed an optical videodisc system for HDTV.

MUSE

The analog input signals of *R*, *G*, and *B* channels are first converted into digital signals. They are then combined into a time-division multiplexed signal, where a simple version of time-compressed integration (TCI) signal is used. Line-sequential transmission is applied for the process of two color-difference signals.

At the subsampling stage, intra-field and inter-frame offset subsampling are adopted. The sampling is made in a four-field sequence. The subsampling signals are transmitted in analog form, with additional signals which are necessary for reconstruction of the picture at the receiving end. The receiver should be equipped with a frame memory in which four fields of the subsampled signal are memorized. Still pictures can be reconstructed with the four-field data in the memory.

In moving pictures, the reconstruction is carried out by using only intra-field data, and may exhibit some blurriness in comparison with still pictures. This blurriness is considered acceptable for partial movement of an object contained in the picture. It is, however, annoying when the picture moves slowly and uniformly in one direction, for example, in panning or tilting the camera.

In order to prevent this type of degradation in picture quality, a motion-compensation technique is applied. At the sending end, motion vector information is detected based on the inter-frame correlation, and multiplexed into the output signal. At the receiving end, motion compensation is carried out by using the motion vector received, in such a way that the readout address of the frame memory is shifted in accordance with the motion vector and the data can be handled in still picture mode. Thus, conspicuous blurring caused by the uniform movement can be avoided as with still pictures.

These two modes of interpolation, with inter-frame processing for still pictures and intra-field processing for moving pictures, are switched by detecting the motion portion at the detector.

Disc Format

The MUSE signal is recorded on the optical disc as an analog signal by the frequency modulation method, in which deviation is chosen from 12 MHz to 16 MHz (from black to white level).

A pilot signal frequency is selected to 67.5 horizontal line frequency (fH) to reduce the moiré interference between the video and pilot signal. The pilot signal is also used to control the disc's spinning and to drive the tangential mirror for jitter compensation. Dropouts have been compensated by means of stopping the input of data into the memory of the MUSE encoder when the carrier faults are detected.

As a result, high-definition TV pictures are obtained for 17 min from the CAV disc, which has a diameter of 300 mm, and recorded radius of 90 mm–145 mm; the rate of disc revolution is 1800 rpm. By using a CLV disc, 30-min playback is achieved. The disc's linear velocity is 18 m/sec, recorded radius is 55 to 145 mm, and the track pitch is 1.65 m.

Y. Ninomiya, Y. Otsuka, Y. Izumi, "Single channel HDTV broadcast system — the MUSE," NHK Laboratory's Note No. 304, September 1984.

23. A Possible Digital VTR for HDTV. *By Yoshizumi Eto, Seiichi Mita, Masuo Umemoto, and Shusaku Nagahara*, Central Research Laboratory, Hitachi, Ltd., Tokyo, Japan.

As the interest in high-definition television has widely increased, a few HDTV systems including VTRs have been successfully demonstrated. Since all of them were implemented with analog techniques, one of the problems with the HDTV VTRs is the S/N ratio in the playback video signal.

It is well known that one of the powerful techniques for improving the S/N ratio is digital recording. However, it was believed to be very difficult to apply conventional digital recording techniques to the HDTV VTRs when the required bit rate is considered. The results of our study show that the following parameters are possible examples for the digital VTR, which records HDTV pictures of 1125 lines/30 frames:

1. 46 and 11.5-MHz sampling frequency for luminance and chrominance signals, respectively; 8-bit quantization, line-sequential processing for two chrominance signals, and 5-channel recording.
2. Type-C tape transportation parameters, except that the drum rotation speed and the linear tape speed are twice as high as those of the C-format.

Parameters in (1) result in a channel bit rate of 92 Mbit/sec which can be processed by either ECL serial or TTL paralell circuits. Those in (2) result in the shortest wavelength of 1.1 μm and the track pitch of 36 μm, which is not a difficult recording density to realize with the latest and the most advanced digital recording techniques. With these parameters, an S/N ratio can be attained of about 56 dB under the same tape consumption as the current analog HDTV VTRs.

Based on these parameters, our HDTV digital VTR laboratory model has been developed. With the improvement of the dc restoration and the NRZ-type channel coding methods, the average block (consisting of 168 bits) error rate in the 460 Mbit/sec video data stream shows a figure of around $(3 \sim 4) \times 10^{-4}$. The error-protection method, considering human visual sensitivity, efficiently corrects and conceals all of these errors.

This is one of the world's first experiments on an HDTV digital VTR. More details of our findings will be shown at the conference.

24. Major Parameters of HDTV. *By Tetsuo Mitsuhashi*, NHK Science and Technical Research Laboratories, Tokyo, Japan.

The number of scanning lines of HDTV must satisfy suitable vertical resolution and invisible line structure disturbance. The desirable number of lines for a person with a visual acuity of 1.0 to see the picture at 3H, is about 1100.

According to the results of subjective tests with a progressive scanning TV system of from 375 to 1125 lines, the 1125-line system is about half a grade superior to the 825-line system in vertical resolution, line structure disturbance, and total picture quality. Therefore, 1125 lines are required for HDTV.

25. HDTV Production Standards — Interlace or Progressive? *By Kerns H. Powers*, RCA Laboratories, Princeton, N.J.

The objective and subjective factors comparing progressive with interlaced scanning in a high-definition electronic production system are examined to establish preferred choices for the parameters in a single, worldwide HDEP standard. The approach first taken is to establish the preferred scanning system for the consumer display, then translate the vertical, horizontal, and temporal resolution requirements into the preferred source-camera scan, assuming a transparent transmission channel. The advantages of progressive scan in picture manipulation, spatial and temporal filtering, aperture correction, bandwidth compression, frame-rate conversions, and other post-production processes are compared against the bandwidth penalty over interlaced scan. Several myths about that penalty in the conventional wisdom are exposed. It is shown that the true penalty is much lower than is generally recognized. A specific set of scanning parameters is proposed as the preferred HDEP standard.

Stereo Audio in Television

26. Stereo Audio in TV Tutorial. *By Thomas B. Keller*, National Association of Broadcasters, Washington, D.C.

Not available at time of publication.

27. Stereo/Multichannel Audio in Production and Broadcasting: Expectations, Experiments, and Future Trends. *By C. Robert Paulson*, AVP Communication, Westborough, Mass.

On March 29, 1984, the FCC backed into an endorsement of multichannel sound television broadcasting. They issued a ruling reported "to protect the portion of the baseband where the EIA-recommended Zenith/dbx pilot carrier is to be placed." According to a press release, "The Commission's position does not rule out other systems for TV stereo sound transmission, but tacitly gives a vote of confidence to the work of the EIA."

Known both as the MTS (multichannel television sound) and the BTSC (EIA's Broadcast Television Systems Committee) system, the ruling permits simultaneous full bandwidth transmission of both stereo channels and a separate audio program (SAP) channel. The latter is intended for second-language programming, supplementary audio for the visually impaired, or audio unrelated to the program video signal (such as "radio" programming and paging services). Non-commercial (PBS) stations may use the extra channel(s) commercially.

This benign, loosely specified permission for television broadcasters to begin multichannel sound transmissions on their VHF and UHF channels has been received with widely varying degrees of enthusiasm across the broadcasting industry. Station engineers' views range between responses of indifference to "Let's hope that management doesn't get enthusiastic about it." A very few pioneers are on the air with regular schedules of MTS broadcasts. Among cablecasters and broadcasters, concern over "must carry" implications of the rule has created strong feelings of hostility. Production house engineer attitudes are on the side of indifference: "We're already equipped for stereo sound production and editing, for anybody who wants to pay for it."

This paper examines as a systems problem the technical, creative, and practical considerations faced by the broadcasting industry in expanding from monophonic to full MTS operation. The examination includes all links in the chain from the viewer all the way back to the creative activity of planning a video/MTS live or recorded production or commercial. Variables include evolution of audio signal processing from analog to digital, viewer attitude changes manifested or prompted by videocassette movie rentals, VCR sales, combo VD/CD player introductions, CD record purchases, large screen/stereo receiver availability, linear versus interactive television, movie theater competition, etc., all cast in total against the immutability of the 168-hr week.

The creative payoffs, cutover costs, and profit motivations for MTS television broadcasting are then evaluated from both broadcasters' and advertisers' standpoints. A scenario of possible video/HDTV/MTS broadcasting, narrowcasting, and indicasting developments and effects will be presented to stimulate continuing dialogue among broadcasters, producers, hardware manufacturers, and viewers.

28. The Digital Television Tape Recorder: Audio and Data Recording Aspects. *By Kenneth P. Davies*, Canadian Broadcasting Corp., Engineering Headquarters, Montreal, Que., Canada.

The digital television tape recorder (DTTR) is expected to achieve very high video performance, and it is clear that the audio performance must be comparable while offering far greater operational facilities and flexibility. The designer of the audio systems for the DTTR is constrained in many directions: users' needs, specified performance, relationships to the video channel, and interconnection to a completely digital production facility, while having to be very conservative in terms of circuit complexity and reliability. There are few precedents to rely on, and the specifications of the DTTR audio and data formats, to meet these divergent needs, have produced a number of novel ideas and concepts heretofore unknown in television technology.

This paper will outline the various requirements and constraints, propose a format that meets them, and also offer a number of new features not currently found in analog recording.

29. Digital Stereo Sound with Terrestrial Television. *By A. H. Jones*, British Broadcasting Corp., Tadworth, Surrey, U.K.

The BBC is developing a digital system that will carry two high-quality sound signals with existing System I UHF television transmissions. The system uses a second sound carrier that is spaced at about 6½ MHz above the vision carrier and four-phase differential phase-shift keying (DPSK) modulated at a bit rate of about 700 Kbit/sec.

When an additional signal such as this is envisaged within an existing service, it is necessary to make sure that the twin constraints of ruggedness and compatibility are adequately met. On the one hand, the new signal should reliably be received on sets built for the purpose, even in areas where reception is already difficult. On the other hand, existing receivers must not exhibit interference to sound or pictures when the new signal is broadcast. Much of the experimental work on the digital stereo system has been aimed at ensuring that both of these requirements are met.

The ruggedness of the system was established in over-air tests conducted out of normal program hours from a transmitter at Wenvoe in South Wales. Wenvoe was chosen because the area fed by this transmitter consists of hills and valleys in which multipath and low field strength are common. The tests indicated that the area of satisfactory reception of the digital signals will be equal to or greater than the service area limits for satisfactory color television reception. It was also found that the signal will travel satisfactorily through a chain of up to five transposing relay transmitters in tandem.

Compatibility was tested in off-hours tests from the Crystal Palace transmitter that feeds greater London and the southeast of England. These tests used picture and sound material that was particularly susceptible to the effects of interference, and confirmed the compatibility of the system with the widest range of domestic receivers and reception conditions. Further evidence of satisfactory compatibility was provided during tests carried out during normal program hours from the Rowridge transmitter on the Isle of Wight, during demonstrations given at IBC '84.

Other tests were made to assess co-channel and adjacent-channel effects. These were conducted in the laboratory, and it was found that, at the protection levels adopted for planning purposes, reception of the new carrier is not adversely affected by the presence of interfering signals, and the new carrier does not increase the amount of interference caused by the transmissions to which it is added.

Having thus established the basic feasibility of the proposed system, a detailed specification was drawn up. Discussions are now under way with the IBA and the receiver industry, and it is hoped that a U.K. standard based on this specification will be achieved in the spring of 1985. Baseband audio coding is a matter receiving particular attention since several options exist. Plans are also being made for the provision of the necessary modulation equipment at transmitters and for a stereo-capable signal distribution system. Meanwhile, experiments in stereo program production are going on in anticipation of full broadcast stereo service.

30. Implementing the BTSC Companding System for Multichannel TV Sound. *By Leslie B. Tyler and David E. Bates*, dbx, Inc. Newton, Mass.

The BTSC system includes a sophisticated compander for audio-noise reduction. The compander encodes the stereophonic difference (*L-R*) signal before transmission and decodes it after reception. The FCC has specified that the transmission system be capable of 30 dB of separation in the mid-band. To meet this specification, the amplitude response of the *L-R* channel must match that of the *L+R* channel within 0.39 dB and the phase must match within 2.56°. More amplitude error is allowed if the phase is held tighter, and vice versa. This places severe performance requirements on the implementation of the compressor used in broadcast and the expander used in monitoring.

Detailed theory of the system design has been covered elsewhere in the literature.[1] To aid the design process, a computer model was constructed that predicted the response of the system to sine-wave inputs of arbitrary frequencies and levels. Since the expander is a feedforward system (the input signal directly determines transfer function between input and output), the model provided a closed-form solution to the expander response. However, the compressor is a feedback system (the output signal directly determines the transfer function between input and output), so a Newtonian iteration scheme was used to predict its response.

To prevent out-of-band frequencies from interfering with the pilot and the other audio channels, the BTSC system includes a steep low-pass filter within the feedback loop of the compressor. This filter will not change the amplitude response of the ideal encoder if its passband response is perfectly flat, but it will introduce phase shift, which must be compensated in the *L+R* channel. If the filter has any amplitude errors (the amplitude requirements make a Cauer alignment, with ripple in both the stopband and the passband, a likely choice), these must also be compensated in the *L+R* channel.

The modeled performance of the ideal system does not include many of the real-world constraints that will limit performance. A major factor is compressor input noise. When the input signal contains very little high-frequency content (e.g., with a low-frequency sine-wave input), the gain of the compressor becomes very high at high frequencies: +80 dB at 15 kHz is quite possible. This gain causes the slightest input noise to be amplified and interferes with broadband measurements of level at the compressor output. Furthermore, the noise is sensed by the level detectors in the compressor, appreciably changing the actual response to the sine wave.

Other real limitations, such as component tolerances, amplifier-bandwidth constraints, detector accuracies, gain-control-element accuracies, temperature variations, and so on, make it impossible to achieve the theoretical response in practice. The computer model was modified to include the effects of these limitations and to allow comparison of the ideal with the actual response. A proprietary optimization routine (Module) was used to evaluate the impact of the constraints and to provide a design that would minimize it.

The instantaneous compression ratio of the system varies with frequency from 2:1 at low frequencies to 3:1 at high frequencies. This places a frequency-response requirement on the compressor of ±0.195 to ±0.13 dB from low to high frequencies (this is because for sine waves the expander will increase any errors by the inverse of the compression ratio). The response must also vary with level precisely according to theory. This requires good tracking between the rms-level detectors used and the gain-control elements.

Laying out the compressor on one printed circuit board to provide 80 dB of gain at 15 kHz without oscillation was a difficult task. (The theoretical compressor response specifies more than this amount, from 15 kHz on up with no input or low-frequency energy only; this makes oscillations very hard to control.) A 2-pole low-pass filter was added to the PCB to limit the card's bandwidth to slightly over 15 kHz, and the sum-channel compensation was modified to include the effects of this filter. Another location that picked up stray signals and occasionally led to oscillation was the gain-control port of the VCAs used. This problem was solved by bypassing the gain-control ports directly at the voltage-controlled amplifier integrated circuits (VCA ICs) themselves.

Monte Carlo analysis of the effects of component value tolerances led to the choice of 0.1% resistors and capacitors in many critical locations. This, in turn, required that selection procedures be developed to provide the needed tolerances at reasonable costs and with reasonable lead times.

To ensure consistent performance with temperature, the components used were chosen for low-temperature

Kerns Powers (left) and John Baldwin chatting between sessions at the TV Conference.

coefficients except in the case of the detectors and gain-control elements. These parts are integrated rms detectors and VCAs that have matching temperature coefficients, thereby providing consistent response independent of temperature. Allowance for varying scale factors due to semiconductor-geometry variations was made by providing trims on the board.

Other design considerations were addressed by employing on-card regulators for supply-sensitive sections of the circuitry, offset trims for the VCAs to minimize control-voltage (CV) feedthrough, and low-drift op amps throughout the CV paths.

To align compressors, a procedure was developed that would eliminate the effects of input noise on calibration. This entails making band-limited measurements wherein the band-pass filter is included within the compressor feedback loop. This way, high-frequency noise is not sensed by either the detectors or the measuring instruments. The expander does not need such precautions, since its input noise is not exaggerated by pre-emphasis and gain. NBS-traceable meters, with 0.05% accuracy, are employed.

Finally, sum-channel-compensator networks were designed to mimic the known differences in performance between the theoretical standard and the practical realization. For the expander, the compensation required is simply the difference between the response of the actual expander and the model. For the compressor, the errors at the output must be referred back as equivalent errors at the input; this becomes the compensation required. The actual networks were designed with the assistance of Module.

A typical pair of compressor/expander cards provides 35 dB of separation to 10 kHz and 30 dB to 14 kHz, performance that exceeds FCC requirements.[2]

31. An Audio Broadcast System Using Delta Modulation. *By Kenneth Gundry*, Dolby Laboratories, Inc., San Francisco, Calif.

Digital systems are technically attractive for the delivery of high-quality audio to the home because they can be substantially transparent even under conditions of impaired reception, and permit scrambling without degradation in quality. Conventional multilevel pulse-code modulation (PCM), with its precise digital/analog (D/A) converters (at least 13-bit), elaborate output filters, and complex error correction, can perform very well, but its cost is much higher than that normally associated with consumer audio circuitry. This paper describes a digital transmission technique based on the features outlined below, whose decoder cost is a small fraction of that of a PCM system.

Over-Sampling and Noise Shaping

By using a sampling frequency in the range 200 to 350 kHz instead of 30 to 50 kHz, much of the noise power can be transferred out of the audio band into the otherwise unused part of the spectrum between the highest audio frequency and half the sampling frequency. Furthermore, a high sampling frequency greatly simplifies output filters.

Differential Coding in Conjunction with Companding

In the context of companded systems, differential coding is preferable to nondifferential because it matches more closely the properties of human hearing. It is more important to maximize the signal-to-noise ratio (SNR) for low-frequency signals than for high, since audible noise modulation is much more probable with low frequencies. At practical bit rates, two-level adaptive differential coding (adaptive delta modulation) is similar in audible performance to nondifferential digitally companded PCM.

Adaptive Emphasis

Practical companded systems employ pre- and post-emphasis. Placed around a companded codec, high-frequency emphasis improves the subjective SNR for low-frequency signals but degrades it in the presence of predominant high frequencies. However, by using a suitable program-adapting emphasis whereby predominant high-frequency signals are cut selectively rather than generally boosted, the SNR will always be improved instead of degraded.

Advance Information for Control of Companding and Emphasis

Analysis of the audio signal in advance permits the use of slow adapting response times without transient distortion, leading to a robustness in the face of transmission errors and of wide component tolerance. Control information is conveyed from encoder to decoder by low rate (e.g., 8 Kbit/sec) auxiliary bit streams, eliminating the need for complex analysis of audio waveforms and for delay lines in the decoder.

Low Cost

The decoder requires no components (inside or outside integrated circuits) with tolerances tighter than 1%, and a stereo circuit can be built from off-the-shelf parts costing \$6 or \$7; custom ICs are in development (first samples due early in 1985) and will reduce the cost further. Note that this is less than one output filter for multi-level PCM.

The Dolby digital audio system offers a performance comparable with the near-instantaneously companded PCM systems proposed elsewhere and

1. L. B. Tyler, M. F. Davis, and W. A. Allen, "A Companding System for Multichannel TV Sound," *IEEE Trans. on Consumer Electronics*, vol. CE-30, No. 4, Nov. 1984.
2. As set forth in OST-60.

Part of the Equipment Exhibit.

permits the economical distribution of high-quality audio over any medium which can accommodate 220 Kbit/sec or more per channel with a random error rate up to about 10^{-3}. It is therefore particularly well suited to DBS and cable television.

32. A Digital Audio Time-Base Corrector for Linear Magnetic Recording. *By Thomas J. Rosback*, Harris Corp., Quincy, Ill.

Helical scanning VTRs with linear audio tracks suffer from head azimuth alignment errors and wow-and-flutter, which can significantly affect audio quality. Audio time-base correction is an encode/decode process which reduces these undesirable effects to sub-audible levels.

Wow-and-flutter is tape speed fluctuation caused by imperfect capstan, pinch roller, and idler wheel roundness; uneven reel clutch pressure; and uneven tape pull from the video heads. Wow-and-flutter frequency modulates the audio program.

Azimuth errors are caused by tape guide and head misalignment. Improper azimuth causes an inter-channel time delay in stereo recorders. When this delay approaches one-half wavelength at some audio frequency, and the two tracks are summed to mono, a frequency response notch will be created. The sum of two equal frequency sinusoids with a phase offset of ϕ degrees is: $S = A\ (1 + \cos\phi)$. When ϕ is 0 degrees, S = 2A. When ϕ is 180 degrees, S = 0. Phase offset and time delay are intimately related. $\lambda = x/f$, where x = tape speed, f = audio frequency, and λ = wavelength on tape. When ϕ is 180 degrees, $d = \lambda/2$. Assume that a tape speed of 7.5 in./sec is used, and track separation is 0.1 in. A notch at 10,000 Hz, which is quite audible, corresponds to a delay length of 3.75×10^{-4} in. The azimuth error in degrees is then 0.2. Even a minute azimuth error angle causes significant mono sum losses.

Sync pulses provide a convenient reference signal for video time-base correction. Audio signals are not so predictable, so an inaudible pilot reference signal must be recorded along with the program material. A 19-kHz sinusoidal pilot signal is recorded at a low enough level (27 dB below normal operating level) to prevent audible signal-to-noise degradation due to tape modulation noise. The phase error at 19 kHz often exceeds 360°, so the pilot is 60% amplitude modulated with a 300-Hz sinusoid, preventing phase ambiguity during decoding. Once encoded, the tape may be copied without degradation.

The audio time-base corrector decoder utilizes two voltage-controlled 16-bit-wide digital audio delay lines driven by hybrid signal processing electronics. A series of 19-kHz band-pass filters is used to extract the pilot signal from the output of each digital delay line. The pilot is then envelope detected and band-pass filtered, leaving only the 300-Hz modulation. Any phase difference between the two channels is detected with a sensitive phase detector.

The phase detector output is a dc error voltage which is used to control the frequency of a voltage-controlled oscillator (VCO). The VCO frequency determines the clock rate, and thus the delay, of one of the digital audio delay lines. Since the control electronics look at the output of the delay lines, a feedback loop is created which forces time and phase between both channels. Frequency response notches due to azimuth errors are thus eliminated.

Wow-and-flutter correction is also accomplished through the use of a feedback loop. The 19-kHz pilot signal is FM-demodulated with a phase locked loop. The resulting ac error voltage is used to simultaneously control the delay of both delay lines. This action effectively cancels recorded wow-and-flutter components.

Only an encode/decode audio time-base correction process can provide significant audible reduction of wow-and-flutter and azimuth errors resulting from imperfect tape transport design.

33. Forging an HDTV Tool for Production and Post-Production: A Working Group Report. *By Richard J. Stumpf*, Universal City Studios, Universal City, Calif., and *Birney D. Dayton*, Grass Valley Group, Grass Valley, Calif.

The SMPTE Working Group on Electronic Production was organized in early 1984 to study requirements and to recommend HDEP standards. This report traces progress of the Working Group, whose membership is drawn from both the motion picture and television fields. The work of the group has concentrated upon frame-rate conversion, aspect ratio, bandwidth considerations, and description of film performance in electronic terms. The need for future developmental work will be presented.