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# LONGITUDINAL DIGITAL RECORDING OF AUDIO

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ABSTRACT:

This paper describes a two channel digital recording system intended for highest quality audio use. A criterion for the system has been the achievement of all the potential improvements promised by the application of digital techniques to audio; such as dynamic range expansion, elimination of wow and flutter, eliminating the need for noise reduction systems, and elimination of tape hiss, while avoiding the major difficulties typical of this application. Most of these goals have been met in this system by the application of existing technology and theory.

## INTRODUCTION:

Digital techniques are rapidly finding their way into an increasing number of areas previously thought immune to digital applications. Some of the reasons for this invasion include the ease of manipulation of digital data and the facility of transfer, transmission and storage of digital information. With respect to audio recording, another advantage of digitizing is that the audio information is then moved to the digital domain where some of the commonly encountered problems in audio recording either have no effect at all or can be easily partially or totally eliminated. Digitized audio has already been proven effective in areas ranging from speech and music analysis and synthesis [1] to passenger entertainment systems on Boeing 747 aircraft. It should come as no surprise, therefore, that the use of digitized audio would eventually be extended to the professional recording industry. The potential of achieving a significant improvement in the quality of audio recording on tape by the application of digital techniques makes the development of a digital recorder for audio both desirable and very attractive. Until recently, the required state-of-the-art in several different areas had not developed far enough to allow the realization of all goals necessary to make the application of digital methods acceptable to the audio recording industry. But recent innovations and the proper application of existing technology brings the ultimate goal of a complete and highly automated digital (computerized) studio within sight. This paper describes only one small but significant step towards that goal, the development of a digital tape recording system designed specifically for ultra-high quality recording of audio in a mode and format compatible with the recording industry. Thus in longitudinal digital recording all digital information defining one channel of audio is recorded on one tape track.

## WHY DIGITAL RECORDING?

Typically analog tape recording of audio is hounded by a number of problems which are a function of the medium (magnetic tape) and the mechanics (tape moving mechanisms). These problems include (1) inadequate dynamic range, i.e., low signal to noise ratio, (2) inherent phase distortion, (3) inherent harmonic distortion, (4) insufficient transient response, (5) modulation noise, (6) cross talk, (7) print through, (8) multi-copy degradation, (9) flutter and wow, (10) inherent limitations in noise reduction systems, (11) storage degradation with time, and (12) limited low-end frequencies response. Digital recording provides either an improvement in, or total elimini-tion of each of the above problems.

Of the above mentioned problems, some are eliminated because the problem source exists in the analog domain, and the audio in digital form is largely immune from the effects of the analog world. Other problems can either be greatly reduced or totally eliminated because of the increased facility for handling the signal once it is converted to digital form. The following chart should be helpful in understanding how the above problems are solved.

| TABLE I PROBLEM | Extent of Improvement | | | Cause of Improvement | |
|---|---|---|---|---|---|
| | Totally Eliminated | Red* To Inaud. | Significantly Improved | Digital Domain | Digital Handling |
| Inadequate dynamic range (i.e.,low S/N | | | X | X | |
| Phase distortion | | X | | | X |
| Harmonic distortion | | X | | X | |
| Transient response | | | X | X | X |
| Modulation noise | X | | | X | |
| Cross talk (recording & playback) | | X | | X | |
| Print through | X | | | X | |
| Multi-copy degradation | | | X | | X |
| Flutter and wow | X | | | | X |
| Noise reduction system | X | | | X | |
| Storage degradation | | | X | | X |
| Limitations in low end frequency response | X | | | X | |

* Reduced to inaudibility under worst-case conditions

Thus, of twelve major problems associated with standard analog recording techniques, five can be totally eliminated, three can be reduced to inaudibility and four others can be significantly improved upon. Several examples will now be given.

Modulation noise is an audible distortion in analog recording that increases and decreases with increasing and decreasing amplitude of the recorded waveform. It results from the varying head to tape spacing caused by dirt and dust particles and uneven oxide coating of the tape among other things. This varying tape to head spacing causes a slight time varying modulation of the originally recorded waveform. In digital recording the audio content is in the encoded serial combination of 1's and 0's and although the tape to head spacing may change, that can affect only the amplitude of the ones and zeros and not their encoded audio content. Thus modulation distortion is totally eliminated as a problem in the reproduced audio waveform, simply because the audio is encoded digitally.

Some problems are not solved simply by the transition to the digital domain, but are solved rather by the improved ability to handle or manipulate a signal digitally. Wow and flutter is a problem of this type. Wow and flutter results from the inability to move tape at a perfectly uniform speed. This inability pertains to all recorders, even digital recorders. Unless special precautions were taken, wow and flutter would also be present in a digitally recorded and reproduced waveform. To totally eliminate it, one must re-establish the original timebase for the signal after it has been reproduced from tape. In the analog world this is difficult if not impossible, but when recording digitally it is very simple to dump the encoded 1's and 0's into a memory as they come from the tape, and then strobe them to the DAC for reconstruction under control of a crystal controlled clock, which has the same frequency as the original clock used for sampling in the A/D conversion process. Thus, the timebase is re-established with the precision of a crystal and wow and flutter, due to non-uniform tape movement is totally eliminated, not by the use of the digital domain itself but by the ease with which the signals can be handled digitally. The improvements for the other problems of Table 1 can similarly be easily explained. Some of these will be discussed later. Others are discussed in the references, [2,3,4,5,6].

## BASICS OF DIGITAL RECORDING:

The Nyquist Sampling Theorem states that if the spectrum of a function $h(t)$ is zero for all frequencies greater than a certain frequency $f_c$, then the continuous function $h(t)$ can be uniquely determined from a knowledge of its sampled values [7]. To have the spectrum zero for all frequencies above $f_c$ means that the function is band limited at $f_c$. The frequency $f_c$ is known as the Nyquist frequency, and the frequency $2f_c$ is the Nyquist sampling rate. Sampling means measuring the amplitude of the waveform at distinct times separated by $1/2f_c$. To reconstruct the original waveform from a knowledge of the sampled values, it is necessary to pass the samples through a system with rectangular frequency response. The rectangular frequency response is that of an ideal low pass filter with cutoff at $f_c$, the Nyquist frequency. In practice, the sampled waveform is a stepped or staircase waveform with each step representing one sample. When this waveform is passed through a low pass filter with a cutoff frequency at or below the Nyquist frequency, the result is the smoothing necessary to reconstruct the original waveform. This process is represented by the block diagram of Fig. 1. This is, of course, an over-simplification. As Stockham indicates [3] there are a number of additional criteria that must be met to obtain highest quality A/D and D/A conversion of audio. For example, it is necessary to employ a precision low-jitter oscillator to control the sampling and reconstruction process. The D/A converter must be de-glitched to prevent the DAC glitches from causing unwanted harmonic distortion and noise. Also, it is desirable to include some equalization for high frequencies because of "minor losses in the high frequency output of the D/A converter" [3]. Note also that a Sample and Hold had already been included in the block diagram of Fig. 1 which is necessary to convert higher frequencies without error, (i.e, during the time the A/D is converting the signal its senses must be held constant).

In longitudinal digital recording the digital equivalent of each sample is first combined with synchronization bits, and then written serially onto a single tape track. In the reconstruction process the serial data stream is received from tape, the synchronization bits extracted, and the original parallel digital word for each sample reconstructed. When all of this is added to the block diagram, a single channel of a longitudinal digital recorder appears as shown in Fig. 2. The synchronizer on the reproduce side of the block diagram is used to extract the clock timing information from the encoded serial data stream, and the memory used on the reproduce side is there to allow re-establishment of the original timebase to eliminate wow and flutter.

## THE LOGITUDINAL DIGITAL RECORDER SYSTEM:

The system to be explained is a two channel, 15 bit, fully self-contained proto-type system, that has successfully been used to make professional recordings. All the basic principals mentioned to this point (with one exception) have been implemented in this system. Now a more detailed analysis of the complete system will be given.

## THE RECORDER:

A goal of the design was to make the system compatible with the professional recording industry. This places many requirements on the recorder mechanism, such as cost of the tape, playing time, tape speed, number of tracks and tape width. Certain limitations of the instrumentation recorder also limit what can be accomplished. It is possible, however, to achieve 14 or 16 tracks, on 1.0 inch instrumentation tape that costs about the same or less than standard 2.0 inch professional recording tape, and move tape at the familiar speed of 30 IPS. (The system described here records only two channels of audio - leaving other tracks on tape free for multiple tape use or for addition of more tracks). It is possible, in fact, to put as many as 40 or 50 tracks on one inch tape, due to the fact that digital recording can be adequately handled with a much lower signal-to-noise ratio (as low as 25 db) than is needed for high quality analog recording. Signal-to-noise ratio is a function of the number of tracks per inch, the particle density of the tape used, and the linear density with which the data is written onto tape. Doubling the number of tracks per inch on tape results in approximately a 3.0 db loss in signal-to-noise ratio, while doubling the linear density involves a loss of approximately 6 db.[8] Theoretically, 40 tracks on 1.0 inch tape with a linear recording density of 25.6 KB/inch will give an SNR of 36.2 db, perfectly acceptable for digital recording.[8] One additional requirement for the recorder not yet mentioned is the need for external servo control of the tape speed during playback. This need arises from the use of the memory in the playback system to eliminate wow and flutter. When a memory is loaded from tape in this fashion, it is necessary to control tape speed to prevent memory underflow or overflow. Thus external control of the tape speed servo of the recorder must be possible during playback.

A tape speed of 30 IPS is the minimum tape speed at which the desired data rate can be easily accomplished, at present, due to the state-of-the-art in instrumentation recording. This is a speed already used in professional recording and will give recording times of 30 min. on a standard 4600 foot roll of instrumentation tape. Many commonly available instrumentation recorders meet the requirements of tape speed and bandwidth, and have excellent operating characteristics, and relatively compact size. Instrumentation recorders are extremely expensive, and must be replacedin the near future with a more economical dedicated recorder designed for this express purpose.

## ENCODING:

Restrictions imposed by the magnetic recorder require the proper code selection for successful digital recording. Instrumentation recorders will give an upper bandwidth of 500 Khz at 30 IPS, and a low frequency response down to 400 Hz. This may seem more than adequate, but these limits are pushed and possibly even exceeded on both ends by PCM requirements for audio. To achieve a bandwidth of 15 KHz with 16 bit conversion capability and allowing three bits for synchronization and 1 bit for parity checking, requires a bit rate in excess of 600 K Bits/second. Of the known codes for recording, only three make logical choices; the NRZ code, (Non-Return to Zero); Miller code, sometimes known as narrow band phase (NBØ) or delay modulation; and the Biphase (BiØ) codes. The characteristics of each of these codes have been tediously explained [9,6,10] relative to their power spectra, and their required bandwidth for a given bit rate. NRZ codes require DC response, but also require only a bandwidth of 1/2 the bit rate. The BiØ codes eliminate the need for DC response since there is a transition every bit period regardless of data word construction, but they require an upper frequency response twice that of NRZ and equal to the bit rate. NBØ, known as delay modulation or Miller code, requires about 25% more bandwidth than NRZ, but does not require the same DC response as NRZ codes. Unfortunately, it has other drawbacks, such as the requirement for a clock at twice the data rate and the need for frequent 101 sequences for proper decoding.[9] BIØ and NBØ codes require only about 1 decade of bandwidth while NRZ codes require 10 octaves of bandwidth. Considering all these factors either NRZ codes or NBØ (Miller) codes are the best choice. The simplicity of NRZ and the particular character of digitized audio signals permit it to be used about equally well as NBØ.

Selection of the NRZ code allows standard "true" TTL levels (above 2.4 volts) to represent an encoded "1", and the false TTL level (less than 0.8 volts) to represent a "0". For example, after a start bit, sixteen bit data can be transmitted followed by a parity bit and two stop bits for synchronization purposes.

## SAMPLE AND HOLD, AND A/D CONVERTER SELECTION:

The key to superb quality digitizing of audio lies in the selection of the sample and hold module and the analog to digital converter. These two components together are not only responsible for the resultant quality of the digitized audio, but they are the active circuits within the analog record path of the signal that have the most serious degrading effects on the analog signal prior to digitization. Once digitized, the audio content is immune from further degradation, so the S & H and the A/D must be of exceptionally high and proven analog quality. In addition, they must exhibit other vital characteristics. For the Sample and Hold, these include an accuracy of better than .005%, full audio bandwidth, fast settling time (about 300 ns to .005%), a very low hold decay rate, aperture uncertainty time of less than 5 ns, low feed through characteristics, excellent linerity and stability and very small switching transients. At this time, an S/H that meets all of these specifications has not been found. Circuits with slightly relaxed specifications are available. The Sample and Hold, it turns out, can become the weak link in the chain and some improvement in this area is needed.

The A/D converter must also meet some stringent requirements. These include moderately high conversion rates (10 us/conversion), a true 16 bit accuracy of near .0015%, small size, full scale range of $\pm$ 10 volts and sufficient digital status information. Again, units meeting all of the specifications have not been found, but some with slightly relaxed performance are available. It should be mentioned that specifications alone do not make the product. It is of highest importance that those units selected be critically tested in actual operating environments and the selection be made on the basis of experience, rather than just on specifications.

RECORD ELECTRONICS:

The record electronics must synchronize and control the Sample and Hold function with the A/D conversion of each channel, and the parallel to serial conversion and transmission to tape of the digitized data. These functions must be performed in an interleaved fashion under control of a local low jitter crystal oscillator. The interleaving allows the parallel to serial conversion and serial transmission to tape for the previous digital word to occur while the A/D converters are performing the conversion on the next sample. Fig. 4 depicts this interleaved timing, which is generated in TTL sequential and combinatorial circuitry. It should be mentioned that digital systems are hard limiting systems, which do not exhibit increasing distortion with increasing amplitude but rather the distortion characteristics remain both excellent and constant at all amplitudes up to overrange, at which time clipping type distortion occurs. Experience has shown that operation with a standard OVU reading set 12 db below clipping level adequately avoids clipping without wasting significant available dynamic range.

Amplifiers and filters in the analog signal path of the record electronics must, of course, be of the highest quality. The amplifiers used should exhibit total audio band harmonic distortion and noise better than 90 db down from full signal amplitude.

The recording electronics does not need to contain any special electronics for noise reduction, companding, pre-emphasis and post-emphasis as all high quality analog recorders require. The quality of digitized audio is impeccable without these aids, and the avoidance of them greatly simplifies its use.

THE REPRODUCE ELECTRONICS:

Quality reproduction of digitally recorded audio is by far the most difficult problem. Although commonly encountered analog recording problems are eliminated, other problems concerned with the recording of digital information at extremely high bit rates become worrisome. The major tasks that must be performed are (1) signal enhancement and extraction of the clock or timing information from the serial bit stream, (2) serial to parallel reconstruction of the digital words, (3) memory buffering of the digital words and re-establishment of the original timebase, (4) reconstruction of the analog signal through proper selection of a digital to analog converter and low pass filtering, (5) analog buffering of the reconstructed signal, and (6) generation of the sequential, combinatorial and timing circuitry to handle all the above tasks. A discussion of each of these tasks and their requirements will be given.

BIT STREAM AND CLOCK EXTRACTION:

High density digitally recorded data returns from tape more as a "quasi sinusoidal wave form" [11] than as a square wave representing one's and zero's. High density bit packing supresses harmonics necessary for square wave reproduction, and causes shifts in bit transition locations due to phase distortions near band edge. In addition to this, tape noise, track to track cross talk, tape vibrations, wow and flutter, and problems with the tape composition itself all complicate the detection process [4]. A standard detection process consists of passing the signal through a limiter with a DC restorer, then through a phase-locked loop and finally a decoding circuit. The limiter and DC restorer derive a TTL compatible bit stream from the quasi-sinusoidal waveform. The phase-locked loop then tracks the timing variations due to non-uniform tape motion and establises a clock relative to the bit stream that can be used for decoding. The decoder then presents the resulting data stream and a synchronous clock to the following circuitry. A block diagram of this process is shown in Fig. 5. This combination of circuits is called a bit synchronizer. The outputs of the bit synchronizer are a TTL compatible square wave data stream, and a synchronous TTL compatible symmetrical clock.

SERIAL TO PARALLEL CONVERSION AND WORD SYNCH.:

To successfully reconstruct the exact digital words originally encoded, it is necessary to decipher word divisions in the serial data stream received from the bit synchronizer. The recording electronics adds synch pulses to the bit stream for this express purpose, a START bit which is always zero (false) at the beginning of each word and two stop bits, which are always true at the end of each word. When words are transmitted serially, this forms a 110 code which is always present regardless of the status of other bits, much like serial ASCII encoding used on a standard teletype. The decoder demands that 110 codes be present and spaced the correct number of bits apart before word synch is obtained. When word synch occurs, bits between the 110 codes are assumed to form one parallel data word plus the parity bit and are transmitted in parallel to a holding register.

While in this holding register error-checking, using parity is accomplished. The correct digital word with all synch bits, and the error-checking bit stripped from it, is then strobed into a FIFO memory. When errors are detected, the previously correct word can be substituted with minor audible effect. [5]

MEMORY BUFFERING AND RE-ESTABLISHING THE TIMEBASE:

The only degradation that the tape drive itself is able to impress on the audio contained in the digital code, is that of timebase error caused by the tape speed error, (i.e., wow and flutter, vibration, speed drift, etc.). This is possible because the rate at which the digital words arrive for reconstruction will vary from the original rate of digitization by the difference between the crystal clock and the instantaneous playback tape speed and its associated variations. A slower reconstruction rate would reduce the audio pitch - a higher rate would increase it - while wow and flutter of the reconstruction rate would sound just like wow and flutter from a conventional machine. In the digital domain, this problem could be totally eliminated by storing the digital words as they return from tape in an infinite length memory, and then strobing them out of memory for reconstruction at a rate based on the original crystal clock. The memory would have to be of infinite length because the average long-term error of tape speed during playback relative to the tape speed during recording will not be zero, and thus any size finite length memory would eventually overflow. Of course, infinite length memories do not exist and the solution to this problem is to servo control the tape speed as a function of the number of words in memory. (i.e., when the number of words in memory is being depleted, increase the tape speed slightly, and when the number of words in memory is increasing, slow the tape down slightly). This can be handled by counting the number of words in memory and then using the binary output of the counter to control a DAC which, in turn, controls a VCO (voltage controlled oscillator), which, in turn, regulates the tape drive servo. This gives continuous and somewhat linear speed control of the digital recorder resulting in a need for fewer memory storage words than may be anticipated.

When servo control is implemented, the number of memory words needed is a function of the recorders servo speed-error specifications, and the error coefficient of the servo control circuit. If the circuitry has been designed to have a zero error coefficient and if the transient response of the servo system is good, the number of words of memory storage can be held to very few, possibly less than ten. Otherwise, as many as 32 or 64 words of memory may be needed.

SELECTION OF THE DAC, FILTERS, AND OUTPUT AMPLIFIERS:

The DAC, filters, and output amplifiers are again the only components in the reproduce path with the opportunity to degrade the analog signal directly. Thus, they must be of exceptionally high quality. The digital to analog converter should have 16 bits, be totally monotonic, with linearity of .0015% and a slew rate as high as possible-preferably above 200 volts/microsec...A DAC with high output voltage swing is preferred to gain maximum signal-to-noise ratio. An output of ±10 volts is easily obtained and some with higher outputs are available. With a 16 bit system having ±10 volts output, the weighting of the LSB is 152 micro-volts. Therefore, if the LSB is to be at all significant all noise, hum, RFI and EMI must be significantly less than 152 microvolts, preferably less than 100 micro-volts. This is not easily accomplished.

Slew rate, as well as "glitching", can become another problem with existing DAC's. [3,16] Slew rate distortion is caused by errors between the ideal stepped output of the DAC and the actual output which requires a finite time for new steps to be reached. This type of distortion is most prevelant at higher frequencies and large amplitudes and may cause either odd harmonic distortions or, in the worst case, distortions that are not harmonically related to the output. The ear is very sensitive to this distortion and, therefore, it cannot be ignored. The best solution is to follow the DAC with a high speed Sample and Hold prior to filtering. This makes the signal dependent only on the slew rate of the Sample and Hold which can be much faster than the DAC. This same Sample and Hold may also be used as the DAC deglitcher if it is of sufficient quality.

When a DAC of sufficiently high quality has been selected, it is then necessary to carefully design it into the rest of the system. Errors in grounding, timing of control pulses, the placement of wires, power distribution, and the proximity of other digital and analog circuits can all cause loss of resolution by overshadowing the LSB's of the DAC, with noise. Shielding between the digital electronics and the high quality analog electronics, including the DAC, must also be properly handled and the rise and fall times of control signals going to the DAC or near it can have a detectable effect on the quality of output.

OPERATING SPECIFICATIONS AND TEST RESULTS:

The following information is given as a summary of the overall full loop operating characteristics of an experimental system. (full loop means that a signal to be recorded was digitized, recorded onto tape as a serial bit stream, played back from tape, and reconstructed.) The parameters discussed here pertain to the quality of the reproduced analog signal directly and not just to the mechanics of the recording mechanisms. The method of testing each parameter will also be explained.

DYNAMIC RANGE or SIGNAL-TO-NOISE RATIO: When a short circuit is applied to the analog input, the total noise at the output of the system is 90 db below the maximum signal level. If a signal is applied at 0 vu, which will cause most bits to be changing, total harmonic distortion and noise measures -86 db. Some of this is harmonic distortion, but most is noise.

HARMONIC DISTORTION: Total harmonic distortion for the system across the audio bandwidth has been measured and plotted as shown in Figs. 6 & 7. Fig. 6 is a plot of distortion VS input amplitude and Fig. 7 is a plot of distortion VS input frequency run at 0 vu level. Note that in most cases total harmonic distortion and noise is 86 db below max. signal level. Exceptions occur at frequencies near the upper band edge, where slew rate limiting occurs and at input levels which push the amplifiers and filters to near saturation.

PHASE DISTORTION: Phase distortion, which is synonomous with group delay, or a non-linear phase response versus frequency, may create a subjective haziness or dullness of the reproduced signal. In conventional recording, phase distortion occurs in the magnetic recording mechanism itself as well as in the amplifiers and equilizers that precede and follow the tape. In digital recording, no phase distortion exists in the digitizing of the audio, and once in digital form magnetic recording cannot affect the phase. From digitization through reconstruction, phase is perfectly maintained. The analog paths prior to digitizing and following reconstruction, which contain filters and amplifiers, may add some phase distortion of their own, but that can be controlled and kept to a minimum. The important fact is that the audio in digital recording is totally immune from recorder-induced phase distortions. Total phase distortion in this system is almost unmeasurable.

TIMEBASE ERROR: In this context, timebase error refers to any discrepancy between the timebase of the original signal, and that of the reconstructed signal. In a standard tape recorder this would be dominated by tape speed error components such as wow and flutter, but in this system, because of the reconstruction method, the timebase is reestablished with the precision of a crystal clock, and the error is unmeasurable. Crystal stability is the only factor which could contribute to timebase error.

CROSS TALK: Cross talk is not possible while the audio signal is digitally encoded, and may, therefore, occur only in the amplifiers, filters, and D/A components. Care has been taken to hold cross talk in these areas to better than -85 db relative to $\pm$ 6VU signal amplitude.

ERROR RATE: In digital recording, this is an important specification. Drops in digital signal amplitude from the recorder during playback, commonly referred to as drop outs, may cause catastrophic failures of the reconstruction process resulting in clicks or momentary pops that are very audible in the reproduced signal. These drop outs are caused by imperfections in the oxide coating of the instrumentation tape used and cannot be avoided, although high quality tape can reduce the frequency of their occurrance.

Error correction schemes are available which can reduce audible errors due to drop outs to as few as 1 or 2 in 5 minutes of listening. This would give an error rate of 1 in $10^8$.

MODULATION NOISE: Modulation noise is totally eliminated when recording audio digitally.

FREQUENCY RESPONSE: The frequency response of this system extends from DC to 15 KHz as shown in Fig. 8. The DC response is real in that a DC level of arbitrary selection can be digitized, recorded on tape as a serial bit stream, played back and faithfully reconstructed as the same DC level plus or minus the quantization error. All input and output amplifiers and filters have DC response. The high frequency response is limited by the use of the 15 KHz low pass filters needed for anti-aliasing and smoothing functions. These filters could easily have been selected at 20 KHz with only a slight broadening of the needed recorder bandwidth. Since both filters are down 3 db at 15 KHz the total full loop response at 15 KHz, due to the filters, is down 6 db. In addition, the DAC's roll off slightly at the high end. All this ends up to slightly more attenuation at 15 KHz than desirable. The frequency equalization necessary to compensate for most of this loss is being installed in the experimental system in the near future. In spite of the limited upper frequency response of the system as it presently exists, none of those who have heard it have complained about the lack of the high end, but rather speak highly of its crispness, and realism. This is almost certainly due, in part, to the lack of other types of distortion. Excellent amplitude response across the audio band in a system that exhibits phase distortion will not necessarily give excellent transient response (or attack). Both phase and frequency response must be present and distorrion free to obtain good impulse or transient response. Even when both are present, if other signal dependent distortions such as modulation noise or transient intermodulation distortion exist, the reproduced audio may be subjectively poor.

REMAINING PROBLEMS:

The system described is an experimental system intended to demonstrate the feasibility of the principals and theory. However, there are still problems to be overcome. The effect of tape dropouts can be further reduced, cost and size reduction is necessary, standardization of tape formats will be desired, and reliability needs to receive critical testing. These problems are all presently receiving highest priority and solutions are immediately forthcoming.

THE FUTURE:

Digital recording promises an upgrading of the sound of music, that will eventually be realized. Industry compatible multi-channel digital recording systems available for the professional recording industry are becoming a reality. The improvement this will bring and the extra facility digital techniques offer will eventually lead to digital mix-down facilities, digital mastering and, perhaps in the future, digitally recorded music for the consumer. The world will soon hear higher quality, crisper, more realistic recorded sound than it has ever heard before.

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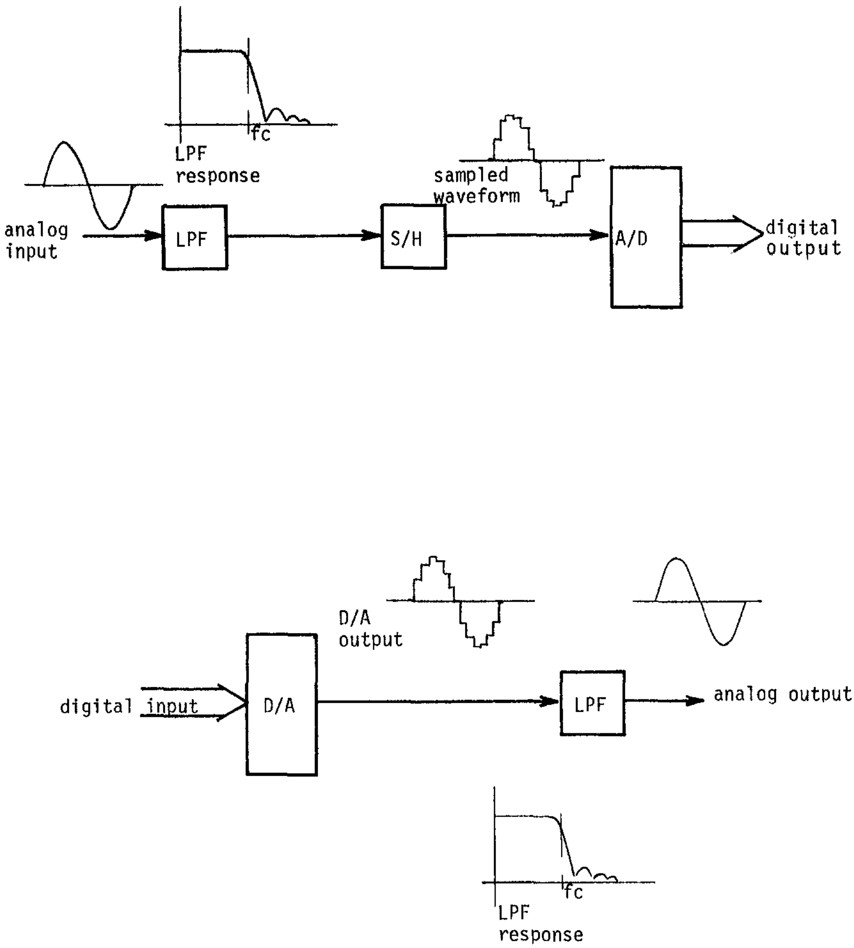
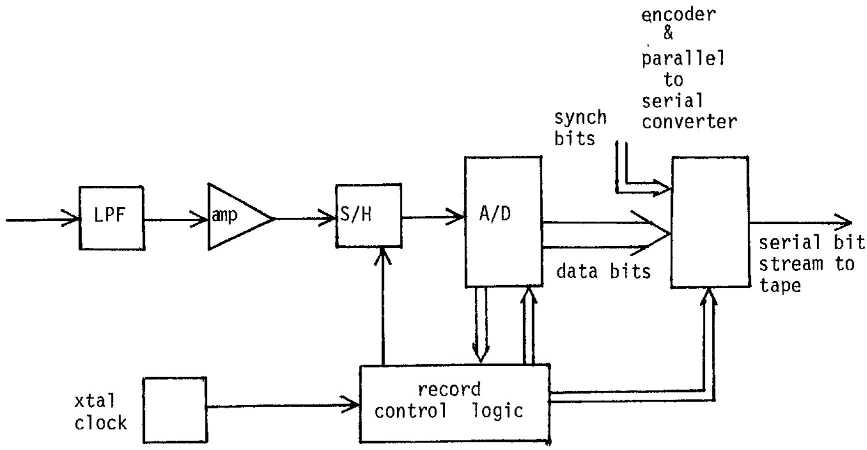
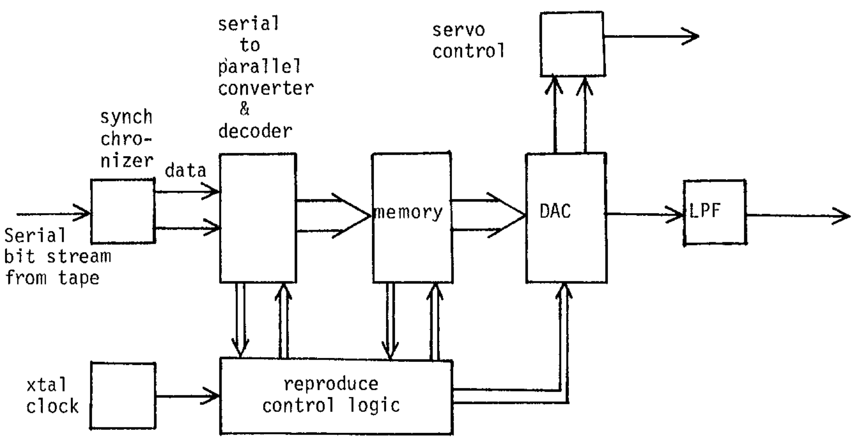



Fig. 1 A/D and D/A Conversion Process




RECORD




REPRODUCE

Fig. 2 Block Diagram of a Single Channel Logitudinal Digital Recorder

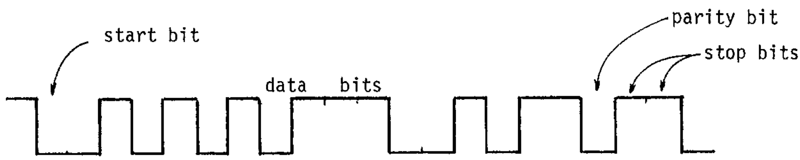



Fig. 3 Serial Bit Stream

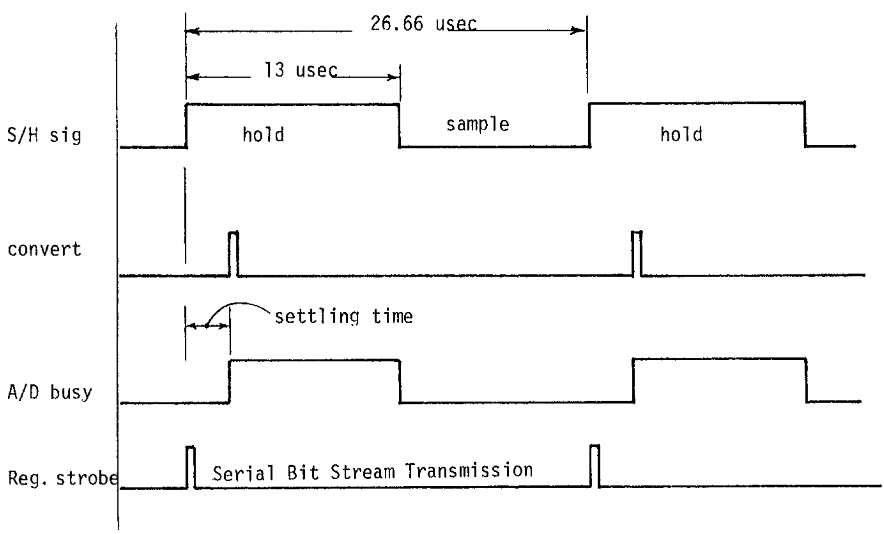



Fig. 4 Record Circuitry Timing

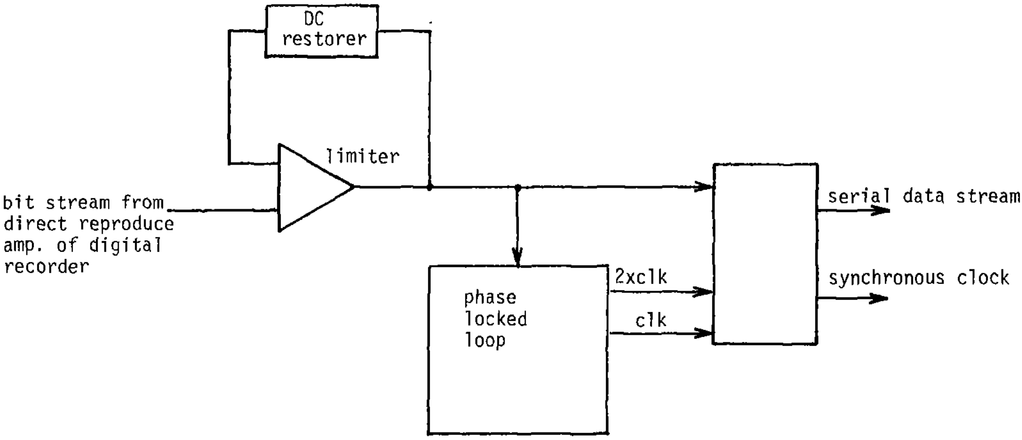



Fig. 5 Bit Stream Enhancement & Clock Extraction

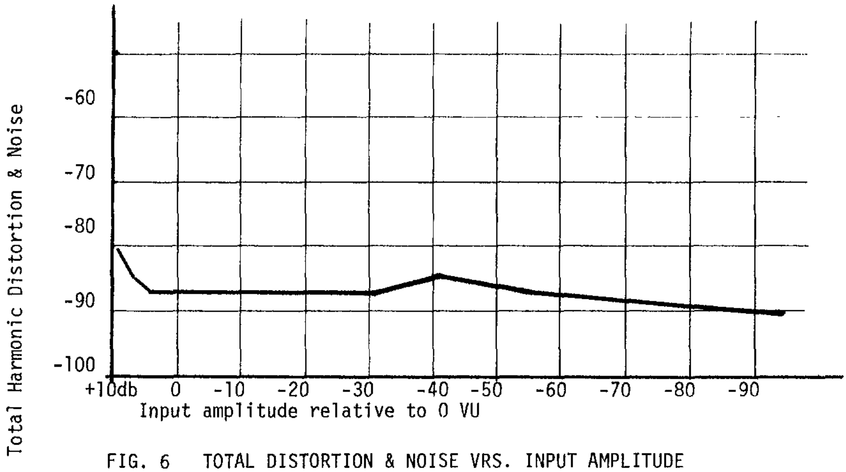



FIG. 6 TOTAL DISTORTION & NOISE VRS. INPUT AMPLITUDE

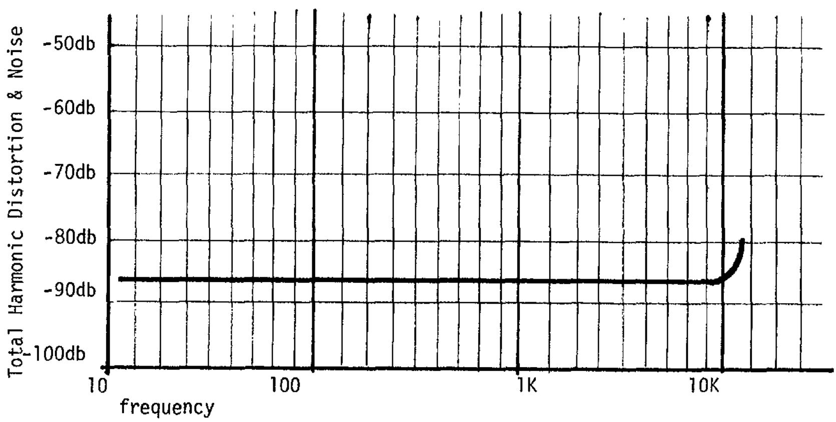



Fig. 7 TOTAL DISTORTION & NOISE VRS. FREQUENCY WITH 0 VU input level

Full loop input at 0 V.U.

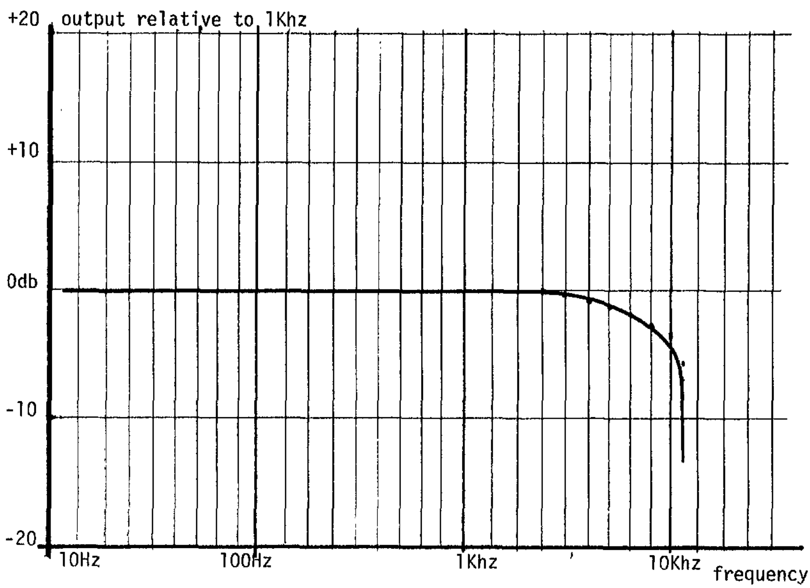



FIG. 8 FREQUENCY RESPONSE