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MAGNETIC DIGITAL AUDIO RECORDING:
A REALISTIC LOOK AT AN EMERGING TECHNOLOGY

By

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ABSTRACT

The general considerations towards the design of Magnetic Digital Audio Recorders are presented and a typical block diagram is explained.

Some areas of hardware implementation which are considered state-of-the art are discussed to provide the potential user with a realistic understanding of what will limit the performance specifications of digital audio recorders.

Certain machine operating requirements which the author feels are necessary for the general acceptance of digital recorders are presented to show their dependence upon the recorders basic signal and tape formats.

INTRODUCTION

The emergence of digital technology into the audio recording studio in the last few years has been primarily through the use of special effects boxes with analog inputs and outputs. This has allowed the recording engineer to continue to think in analog terms and not be seriously confronted with the special terminology and formats of digitized audio.

Without the knowledge of what magic is happening inside these boxes, a misunderstanding of the capability and potential of digital audio has sometimes resulted. There have been descriptions of digital audio recorders which were extravagant and sometimes misleading. Such comments as "total absence of noise and distortion" and "signal to noise at virtual infinity" are comments which the author has read in recent publications. It is with these thoughts in mind that this paper is being written to provide a realistic perspective of the potential performance and use of magnetic digital audio recorders.

GENERAL DESCRIPTION OF DIGITAL SYSTEM

The object of any recording process is to store information and then to faithfully reproduce it. In conventional analog recorders one of the most significant limiting parameters is the inherent non-linear amplitude characteristics of the magnetic tape.

To prevent the amplitude dependent parameters from influencing the amplitude characteristics of the signal to be recorded we must encode the signal into a form which is insensitive to amplitude distortion before recording. One form of this type of encoding is called pulse code modulation (PCM).
A recorder which uses PCM converts the instantaneous amplitude of the signal being recorded into a binary number which represents that amplitude and which is then recorded on tape. It is PCM encoding which is generally regarded as digital recording.

Before delving into the description of the record and reproduce process in a digital recorder it would be worthwhile to review the basic requirements of a digital audio data channel.

Sampling Process

In order to generate the binary numbers which are to be recorded on tape, the audio waveform must be periodically sampled. It is important that the sampling process be often enough to accurately determine the amplitude of the highest frequency of interest. The Nyquist criterion states that the highest frequency to be passed through the system must be sampled at least twice per cycle. If the audio is to have the bandwidth of 20 kHz, then the sampling rate must be at least 40 kHz. Figure 1a shows the resulting waveform when a 6 kHz sine wave is periodically sampled.
Figure 1b shows the same waveform after the sampling frequency and image frequencies have been filtered out as would be done in the output of a digital audio recorder.

If a baseband frequency higher than 20 kHz is sampled by a 40 kHz sampling rate, aliasing products are generated.
To understand the generation of aliasing products let's examined what happens when a 21 kHz sine wave is sampled at a 40 kHz rate. The sampling process multiplies a 40 kHz carrier by the fundamental 21 kHz sine wave. This carrier is amplitude modulated by the 21 kHz sine wave causing 21 kHz side bands. The frequency of the carrier's lower side band is

40 kHz - 21 kHz = 19 kHz.

The frequency of the carrier's upper side band is

40 kHz = 21 kHz = 61 kHz.

When the sampled sine wave is low pass filtered to remove the carrier and upper side bands, it can be seen that the resulting signal is now a combination of the original 21 kHz sine wave and the carrier's lower side band at 19 kHz. This unwanted 19 kHz sine wave is called an aliasing product. Aliasing products will result whenever a frequency above 1/2 of the sampling frequency, called the Nyquist frequency, is sampled.

To prevent aliasing products from being generated, the audio wave form is low pass filtered before it is sampled to eliminate all frequencies higher than the Nyquist frequency. If the Nyquist frequency is equal to the highest frequency to be passed through the system it is impossible to realize a filter to eliminate all alias products. What is required in a practical system is to place the Nyquist frequency well above the highest frequency of interest to allow practical filters to sufficiently attenuate input frequencies that may be above the Nyquist frequency.

Sampling Rate

The selection of the proper sampling rate must take into account many considerations, some of which are not obvious. The higher the sampling rate, the easier it is to build the anti-aliasing filter. But it is also true that the higher the sampling rate the more samples and therefore the more binary digits are generated per unit time. This increases the bit packing densitiy on the magnetic tape at a given tape speed. The higher the bit packing density the less reliable the recording will be, since bit error rate is highly dependent on the bits stored per inch of track. Therefore it is equally important to keep the sampling rate as low as possible. The balance between these two conflicting requirements is the first guideline towards the selection of the proper rate.
Heaslett (1) in his report to the AES Digital Standards Committee has described in detail the criteria for the selection of a digital audio sampling rate which will be compatable with worldwide television and film standards. Of the sampling rates available between 45 kHz and 60 kHz it appears that for professional recording applications the most desirable sampling rate is 50.000 kHz.

Several Japanese manufacturers have designed digital audio recorders based upon consumer helical video tape recorders. Because of the need to format digital audio into the NTSC television standard and because of the nature of helical video recorders, a special sampling rate is involved. When a PAL television standard helical video recorder is used a different sampling rate is required. The sampling rate which has been chosen for NTSC is 44.0559 kHz and for PAL is 44.100 kHz. Because neither meets the criteria for standards compatibility it is expected that neither frequency will find general acceptance for professional recording applications which require international interchange.

Data Rate

To describe the data rate required to record digital audio, two assumptions will be made. The first is that each sample amplitude will be described by a 16 bit linear binary word. The second is that the sample rate will be 50 kHz. Both these assumptions are consistent with the direction being taken by the Digital Standards Committee of the Audio Engineering Society.

There are 50,000 16 bit binary numbers generated per second. When these are converted into serial form the basic serial bit rate becomes 800 kilobits/sec (kb/s). To provide the capability of error detection and correction and to provide synchronizing signals, an additional amount of data equal to 25 to 50% of the basic data must be added. When this additional data is added to the basic data the recorded data rate becomes 1 to 1.2 Mb/s. A conventional professional analog recorder has a bandwidth of about 25 kHz. Because one cycle can ideally convey two bits of information this is equivalent to 50 kb/s. The ratio between the digital requirements and the analog capability is 20 to 1.

One important characteristic of a digital data channel is that it requires only a 25 to 30 dB signal to noise ratio to achieve an acceptable reproduce data error rate. A typical professional analog audio recorder will provide a signal to noise ratio of more than 70 dB. It is the ability of trading the excess 40 dB signal to noise ratio for a bandwidth increase of 20 to 1, which allows designing a practical digital audio recorder which operates at a conventional tape speed. An additional tool which the designer has is to provide more tracks in the same tape width. If one wide track is split into two, the result is twice the record data rate with only a 3 to 4 dB signal to noise ratio loss in each track.

Record Wavelength

The shortest wavelength recorded on professional analog tape recorders is between .5 and 1 mil. (12.7 and 25 uM). For digital applications the wavelength which results from a 1.2 mB/s data stream which has been encoded into a typical channel code such as Miller and recorded at 30 in/s (762 mm/s , is 50 micro-inches (1.27 uM). This represents a recorded density of 40 kb/in (1.57 kb/mm).

The reproduce signal amplitude decreases at the rate of 55 dB per wavelength of separation between the tape surface and the reproduce head. It is obvious that for recorded wavelengths of 50 microinches (1.27 uM) surface imperfections or dirt on the tape which have a dimension approaching 10 microinches (.254 uM) can have a significant effect upon the output signal from the head. At these wavelengths even a fingerprint will cause separation loss and thus cause errors to occur. Even without contamination of the tape it is very difficult to maintain head to tape contact of less than 10 u inches (0.254 uM) without requiring much higher than usual head to tape pressure which may result in tape and/or head damage.
It is the effectiveness of the recorder's error detection and correction mechanism which determines whether any audible effect exists upon playback.

The alternative to recording at high densities is to either increase the tape speed, which decreases total recording time, or to provide more than 1 tape track per channel, which requires more electronics. In either case the user of digital audio recorders should expect to take reasonable care in keeping the recorder's transport area clean and protecting the tape from contamination such as wrinkles or fingerprints.

Error Detection and Correction

Playback will occasionally contain errors that will be objectionable unless corrected or concealed. The key to doing either correction or concealment is knowing when an error has occured. A first level of error indication is provided by looking at the envelope of the playback signal. This is however much too coarse an indication to totally rely upon.

If errors are to be reliably detected there must be a sound basis of judgement. This basis is provided by recording additional data along with the normal signal data. This data, called overhead, can be in the form of parity bits or, more elegantly, in the form of special error checking characters. Whichever way it is done the end result should be that all errors wich can occur can be detected.

Once an error has been detected either of two mechanisms can be evoked. The first mechanism is error masking. This may take the form of muting during the error or it may be to hold or interpolate the wave form during the error, based upon remaining samples. Listening tests on different types of program material have demonstrated that some forms of error masking can be very effective. Simpler forms of masking however can be detected and are sometimes objectionable.

The most desirable way to eliminate errors is to correct them. This requires knowledge of the data recorded during the time that the error occured. The means for correcting errors therefore is to provide additional special coding or redundancy of the data. In magnetic tape recorders errors generally are not randomly scattered but come in bursts lasting from 10 to several hundred bits. Therefore the error correcting information must be dispersed on the tape to prevent burst type errors from destroying a recorder's error correcting capability. The rate and duration of tape dropouts must be a prime consideration in the design of the format through which the basic data, error checking data and error correcting data are combined on tape.

It can be seen that the more effectively and reliably we want to conceal and correct data during playback, the more information must be added as overhead to the original recording. This additional information increases the data storage requirements of the recorder and either increases the packing density on tape or causes a corresponding increase in tape usage and speed.

With automatic error correction, gradual deterioration of the recorded bit stream will be concealed until the point is reached where the correction mechanism fails. This is typical of digital systems where performance is normal until they abruptly fail. Such deterioration may come from repeatedly performing an imperfect process (perhaps a punch-in recording sometimes produces a transient error) or from gradual deterioration of the recorder itself. It is important that the format used in professional digital audio recorders have a significant margin between normal operation and failure. This margin will come from a conservative design approach whereby bit packing density, head to tape interface and encoders are not pushed close to their performance limits. For the user it will be very difficult to evaluate the performance margin of different recorders. As in most other areas of recorder evaluation the final determining factor must be the demonstration of reliability over varied operating conditions.

DIGITAL RECORD PROCESS

With an understanding of the basic digital audio channel requirement let's look at a general block diagram of the signal flow for the digital record process, as shown in figure 2. The necessary control and clocking interconnections are omitted for the sake of clarity.

The analog audio signal is applied to the recorder at the line receiver amplifier. This amplifier can be of conventional design except for the requirements of exceptional dynamic range.

The output of the line receiver is applied to the input filter. This is a sharp cut-off low pass filter with the cut-off at the highest frequency to be recorded. The purpose of this filter is to prevent the generation of aliasing products when the audio signal is sampled.

A critical part of the record process is the sample and hold circuitry which samples the analog waveform at the input filter. This circuitry holds the sampled value of the audio waveform constant while the analog to digital (A-D) converter is converting that value to a digital word. Any noise or glitch which becomes part of the held value decrease the accuracy of A-D conversion causing a reduction in the recorder's signal to noise ratio or an increase in it's non-linear distortion. In a 16 bit system, the accuracy must be better than 1 in 2^{16} or 1 in 65,536.

The output of the A-D converter is typically in parallel form where each of the bits which make up the binary word is available simultaneously. After each sample interval this parallel binary number will change to represent the value of the next sample.

One purpose of the "formatter" which follows the A-D converter, is to take this parallel binary word and convert it into a serial NRZ data stream for recording on magnetic tape. An additional purpose is to add other types of data to the serial bit stream. This additional data called overhead consists of bit patterns for synchronization, error checking characters and error correcting codes. Sometimes parity information and redundant data are also added to facilitate error correction. The way in which the basic data and overhead data are combined determines the recorder's basic data format. Different formats will use different combinations of the previously mentioned types of overhead information. The resulting combination of data and overhead is connected to the channel encoder.

The specific nature of a magnetic tape channel is that of a band pass filter. Very specifically there is no dc response. The data which comes out of the A-D converter is in the form of a non-return-to-zero (NRZ) code. The spectral content of NRZ data contains significant amount of energy at dc. If NRZ data were directly recorded on tape the lack of dc response in the channel would severely distort the data wave form making accurate data detection during reproduction difficult. To compensate for the lack of dc response the NRZ data from the data formatter is usually encoded into a form which minimizes or has no dc response and has a spectral content which matches the record channel. The output of the channel encoder is directly connected to the record amplifier.

The record amplifier in a digital recorder can be very much like the record amplifier in a conventional analog recorder when the digital recorder uses biased recording.

However when certain channel codes are used and when the recording density is 15 Kbits per inch and higher, sometimes no bias is used and the channel encoded data itself drives the tape to near saturation. It is important to know that when recording with bias it is necessary to have a separate erase head when recording over previously recorded magnetic tape.

DIGITAL REPRODUCE PROCESS

A general block diagram of the signal flow for the digital reproduce process is shown in Figure 3. As in the Record block diagram, control and clocking interconnections are omitted for clarity.

The signal from the reproduce head is applied to a low noise, wide bandwidth preamplifier, just as in an analog recorder. The output of the preamplifier is applied to the reproduce equalizer.

Whereas in an analog recorder the purpose of the equalizer is to flatten the amplitude response of the audio signal through-out the pass band of the recorder in accordance with some equalization standard, in a digital recorder the equalizers purpose is to adjust the amplitude and phase response such that zero crossings in the waveform of the reproduce binary signal closely represents the zero crossings in the waveform which was recorded. This is necessary in order to properly detect the value of each binary bit, i.e. a one or a zero. The output of the equalizer is connected to the bit synchronizer and limiter.

The purpose of the bit synchronizer circuitry is to extract the data clocking frequency from the reproduce digital data. This is the same clocking frequency that was used originally to generate the channel code. The limiter limits the positive and negative voltage excursions of the reproduced data.

There are two outputs from the bit synchornizer/limiter. One is the reproduced digital data and the other is the data clock. These are simultaneously applied to the channel decoder which decodes the NRZ information and applies it to the Deformatter/ TBC circuitry.
The function of the deformatter is to separate the basic binary audio data from the overhead information. This is largely a process of serial to parallel conversion.

The time base correction circuitry electronically cancels wow, and flutter components by correcting the timing of each playback sample and also serves to deskew all channels of a recorder to insure phase congruity from channel to channel. One output from the Deformatter is the original binary audio data. The other is the overhead information which will be used in the error detector and corrector.

In the error detector the basic audio data and overhead data are analyzed to determine whether an error has occured. If there has been an error the correction data is used to correct the error. The multitude of mechanisms which can be used to correct or conceal the errors is a subject far more comprehensive than can be covered in this paper.

The output of the error detector and corrector is a series of parallel binary words which should be error-free and a replica of the data that left the original analog to digital converter.

The digital to analog converter takes the binary words and converts them to sequential analog voltage levels. An output sample and hold circuit holds the voltage level constant during each sample period, and serves to eliminate major setting glitches in the D-A converter.

The data formatter in the block diagram shows one output connected to one record head. It is possible to use multiple record heads to record one channel. Therefore less data is put in each track with the result that the bit packing density of each track is decreased.

The output of the digital to analog converter is applied to the output low pass filter. The purpose of this filter is to remove the sampling frequency and its side bands which appear as images of the audio signal. Even though these frequencies are higher than the desired band pass of the recorder they can have an adverse effect upon equipment which may be connected to the output of the digital recorder.

The line output amplifiers can be similar to those found in an analog recorder except for the eceptional dynamic range required.

SIGNIFICANT PERFORMANCE PARAMETERS

There are several performances parameters of a digital audio recorder that deserve special discussion.
Some of these have a direct equivalent in an analog audio recorder and some are new and as yet have no standard way of being defined.

Dynamic Range

In a well designed digital audio recorder it is the A-D converter which theoretically determines the dynamic range and maximum signal to noise ratio of the recorder. The dynamic range is determined by the range of binary numbers (words) which the A-D can generate. Because each bit in the binary number represents a power of 2, a binary number with 16 bits can describe 65,536 discrete numbers. It is the quantizing of the audio signal waveform into these discrete levels that creates the basic system noise. This noise is the error between the actual signal waveform and the accuracy to which the waveform can be described by the allowed range of binary numbers. Talambiras (2) in his paper has described the exact formulas for determining the dynamic range of linear binary encoders. For the purpose of this paper the approximation of 6 dB per bit contained in the binary number will be adequate and will give a figure which is only slightly less than the theoretical maximum. Therefore for a 16 bit linear binary converter the theoretical maximum dynamic range, which is the same as the maximum signal to noise ratio is approximately 96 dB. A 16 bit binary number was specifically chosen for this example because linear converters of 16 bits are the highest resolution converters with the required conversion speed that are technically and economically suitable for digital audio recorders in the near future.

When figures such as 96 dB are described as a recorder's dynamic range it can be misleading to the potential user. A digital system, when it reaches the end of its dynamic range, comes to a hard limit. Figures 4a and 4b show a comparison between a digital and analog recorder on a 2 dB overload. Figure 4c shows the frequency spectra of the waveforms in figures 4a and 4b. The increased content of higher order harmonics in the overloaded digital system, as shown in the lower half of figure 4c, produces a sound which is subjectively worse than that of the overloaded analog system as shown in the upper half of figure 4c. From this it can be seen that it is important to avoid overload in a digital system on peaks of the audio signal.

Therefore it is necessary to set the operating level at least 15 to 20 dB below the level of maximum recorded signal. This then provides a signal to noise ratio, at operating level, of 76 to 83 dB. It is important to note that improvements in the signal to noise ratio of digital audio recorders will only come with an increase in the number of binary bits which represents each digital sample. Therefore if a specific digital recording format contains room for only 16 bits per sample, a modification of that format is required to increase the recorder's signal to noise ratio.

There are methods of digitally encoding the analog samples other than in a purely linear manner. One of these, which can easily be transcoded into a linear form, is that of floating point representation (sometimes known as automatic range coding).

With this method a limited range A-D converter has its input signal prescaled to make best use of the converters limited range. A radix 2 floating point system with one sign bit, 12 mantissa bits and 3 exponent ranging bits can provide a 120 dB dynamic range with a dynamic signal to noise ratio of 78 dB.
All non-linear encoding schemes, of which floating point is an example, trade dynamic signal to noise ratio for either increased dynamic range or for a reduced number of data bits.
A side effect which these systems will have is that the quantizing noise level is a function of the signal level. This is a form of modulation noise.

Time Base Error

All analog recorders contain some time base errors. In audio recorders these errors are typically described as wow and flutter. There is no simple way to remove this error other than through good design of the tape moving and guiding mechanism.

Because in a digital audio recorder each digital sample represents a specific instant in time, during playback circuitry can buffer and electronically correct the rate of the digital samples to the accuracy of a reference crystal oscillator. The result is that there is no wow, flutter or any other longitudinal tape velocity error present in the audio signal.

Distortion

The distortion in a well designed professional analog audio recorder will be the result of the non linearity of the magnetic tape, and will be predominantly 3rd harmonic. In a digital audio recorder the distortion is primarily determined by the linearity of the analog to digital and digital to analog converters. With a dynamic range close to 100 dB most active circuits in the digital recorder are contributing some distortion at high output levels. But certainly the performance of the recorder can be no better than the performance of the converters themselves. The analog to digital converter is the one component within the digital recorder which is being pushed closest to the limit of the present state-of-the-art.

The linearity of converters is usually specified in terms not familiar to the recording industry. Monotonicity, quantizing errors, accuracy and resolution are the parameters with which converters are normally specified. The combination of these specifications will describe a converter's linearity. For example, a 16 bit linear A-D and D-A combination with $\pm$ 1/2 least significant bit linearity would be expected to have no more than 0.003% total harmonic distortion at maximum signal level.

For a converter to be practical for use in a professional recorder product it must not only meet its specifications upon delivery, but it must continue to meet its distortion and accuracy specifications over a long period of time and over an appropriate temperature range. Therefore the long term stability of converter parameters is also significant in determining its suitability for use in a professional product.

Frequency Response

In an analog recorder the high frequency response is determined by the effeciency of the head to reproduce short wavelengths and by the tapes relative short wavelength saturation characteristics. The low frequency response is primarily determined by the shape and size of the reproduce head.

In a properly designed digital recorder the high frequency response is determined by the input and output filters. The low frequency response may be extended to d.c. if desired. The input filter functions to prevent input frequencies at or above the Nyquist frequency from being sampled. For a recorder with a high frequency specification limit of 20 kHz and a sampling frequency of 50 kHz the input filter must attenuate frequencies above 20 kHz such that the response at 25 kHz is at least - 55 to - 60 dB. This can be achieved with careful filter design and still provide less than .5 dB ripple below the cut-off frequency. Some designs will also contain group delay compensation for frequencies close to cut-off.

The purpose of the output filter is to attenuate the sampling frequency and its side bands from the output signal. The frequency and attenuation characteristics of the output filter are normally very similar to those of the input filter except for the frequency compensation which is provided to cancel the frequency loss created by the sampling process. This loss has the form of sin X/X, with unity gain at low frequencies and an attenuation of approximately 4 dB at the Nyquist frequency.

Modulation Noise and Scrape Flutter

One of the more annoying types of noise in an analog recorder is called modulation noise. It is primarily caused by the non-homogeneity of the tapes magnetic coating and by bumps and modules on the tapes surface which create spacing loss. Because the recorded audio signal in a digital audio recorder is encoded in PCM form it is immune to modulation noise. Therefore a digital audio recorder which uses linear PCM encoding will be completely free of modulation noise.

Scrape flutter is another source of signal degradation in an analog recorder. It is caused by the tape rubbing upon stationary guides or upon the heads themselves.
Through stick-slip phenomena, longitudinal vibrations are set up in the tape which frequency modulate the reproduced signal. The time base correction circuitry in a digital audio recorder cancels this modulation.
It is probably the elimination of these two forms of signal degradation·which will be the most noticeable improvement in listening quality of digital audio recorders over conventional analog recorders.

OPERATING REQUIREMENTS OF DIGITAL RECORDERS

It is important that the format of the first digital audio recorders allow most of the same operating features as the analog recorders which they will replace. The format must contain the potential for growth to allow for improved recorder performance and for future integration into totally digital recording systems. The basic recording format is the fundamental factor which determines the capability of a digital audio recorder to provide not only the operating features which are initially recognized as necessary but those which will become possible because of advances in digital audio technology and especially in magnetic tape.

Punch-in Recording

Punch-in recording or overdub must be done on the first digital audio recorders in a manner which is similar to that now being done on analog recorders. The subjective effect of a punch-in should be at least as good as that now achieved with analog recorders which are equipped with record and erase delay circuitry to eliminate overlap on punch-in and a gap on punch-out.

In analog recorders a merge is automatically created between an old and a new recording by ramping the erase and bias signals. This is not possible in digital recorders. Therefore a digital recorder must have some form of digital signal processing to create a merge or it must be done externally in peripheral electronics. In either case it is the basic format which will determine the complexity of electronics and procedure required to do subjectively acceptable punch-in recording.

Multiple Digital Generations

One of the prime advantages of digital audio recorders is the ability to duplicate recordings without any deterioration in the quality of the recorded signal. For this to be done, input and output digital ports must be provided to allow interconnection of digital data between a recorders channels or between the recorder and peripheral equipment.

In order to insure that when one channel of a digital recorder is recorded into another the result is in synchronism with the original, an electronic delay must be provided to compensate for the distance between the reproduce and record heads. Simultaneity between channels in a digital recorder must exist just as it does in an analog recorder which has sync recording capability.

Editing

Because of the restrictions which must be placed upon cutting and splicing digital tape, present editing techniques will be replaced by new ones. These new techniques will become acceptable because of the ability to copy a digital recording with no loss in signal quality. To allow for new forms of editing, digital audio recorders must provide not only the digital signals ports, but also the appropriate command and status lines for interconnection to peripheral equipment.

The systems which will be developed for audio editing may be very much like the editing systems now in use for editing video tape recordings.

Recording Time

A digital audio recorder should provide the same recording time as the user is now accustomed with analog recorders. A minimum of 30 mintues should satisfy most applications although more time would be desired. Once again it is the recording format which determines the basic data rate. This in conjunction with the desired bit packing density will determine the tape speed. In turn, the tape speed in conjunction with the maximum reel size will determine the maximum recording time.

SUMMARY

As with any emerging technology, digital audio recording has created some confusion and misunderstanding.
Misinterpretation of performance specifications has led some to believe that digital audio recording may be the answer to all recording problems.
An important advantage which digital audio recorders have is that their performance parameters such as signal to noise, distortion and frequency response are controlled by electronic and not mechanical or physical limitations. But digital recording also provides some limitations which are not present in

analog recorders. The understanding of these limitations by the user is of primary importance so that he may most efficiently and effectively utilize a digital recorder.

A characteristic of digital signal systems is that they fail abruptly, usually without the gradual warning which is typical of deterioration in analog systems. A conservative design approach is required to provide a digital system with a safe operating margin.

It is the recorded bit density which is the single most significant factor in the reliability of a digital recorder. The efforts which are now underway to create a digital audio recorder standard format must consider the reliability, flexibility and potential for performance improvement of that format just as well as that format's basic signal performance capability.
The evaluation of format reliability, though difficult, must be conducted over varied operating conditions.

The limitations and advantages of digital recorders will have a significant effect upon the future configuration and operating procedures in recording studios. For example, mechanical splicing of tape will be replaced by audio editing systems similar to the video editing systems of today.
These systems will provide procedures and techniques which will significantly improve the efficiency and quality of edits.

It is easy to see that the end result will be an all digital studio. This together with the digital audio players that are now being developed for the consumer market, will give the audio enthusiast the potential to experience a fidelity of sound, which was previously available only to the recording engineer in his studio.

Figures

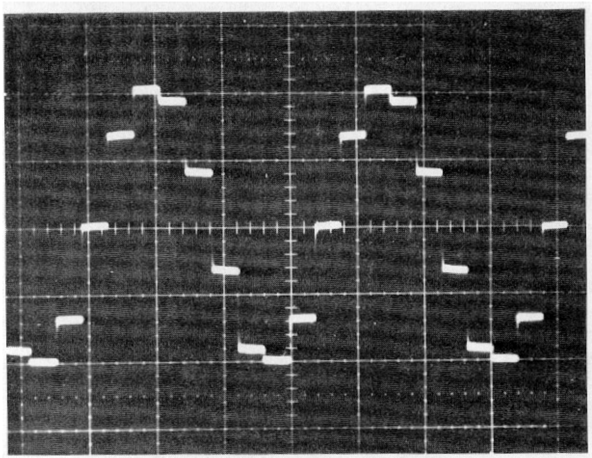
References

1. Alastair Heaslett, "Some Criteria for the Selection for Sampling Rates in Digital Audio Systems", AES Digital Audio Standards Report dated Jan. 12, 1978

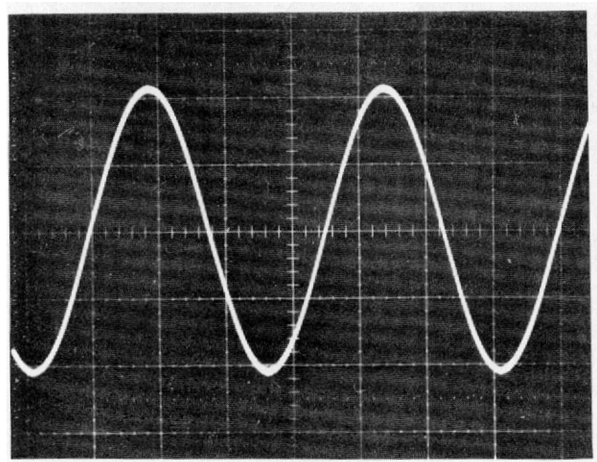
2. Robert Talambiras, "Some Considerations in the Design of Wide-Dynamic-Range Audio Digitizing Systems", AES Preprint 1226 (A-1).

FIGURE 1

SAMPLED AND FILTERED WAVEFORMS OF A 6 kHz SINEWAVE AT OPERATING LEVEL.



A) WAVEFORM AT OUTPUT
OF D-A CONVERTER



B) WAVEFORM AT OUTPUT
OF OUTPUT FILTER

FIGURE 2

SIMPLIFIED BLOCK DIAGRAM, RECORD SIGNAL FLOW

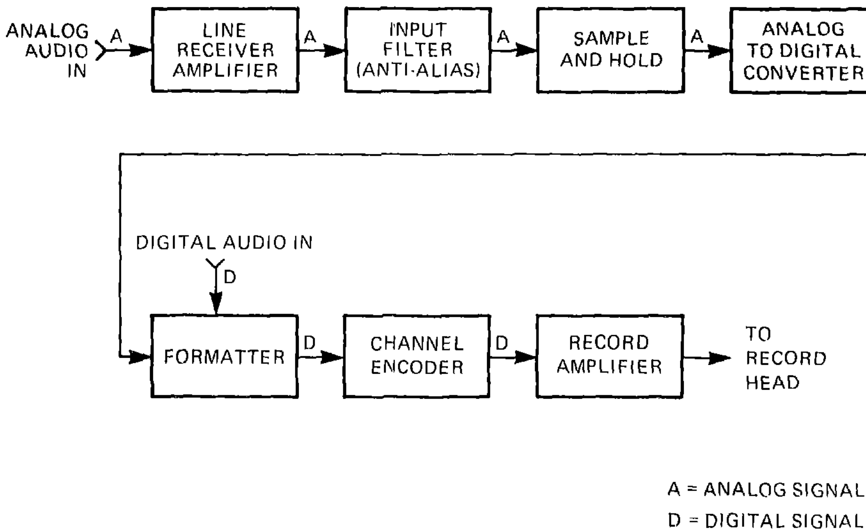


FIGURE 3

SIMPLIFIED BLOCK DIAGRAM, REPRODUCE SIGNAL FLOW

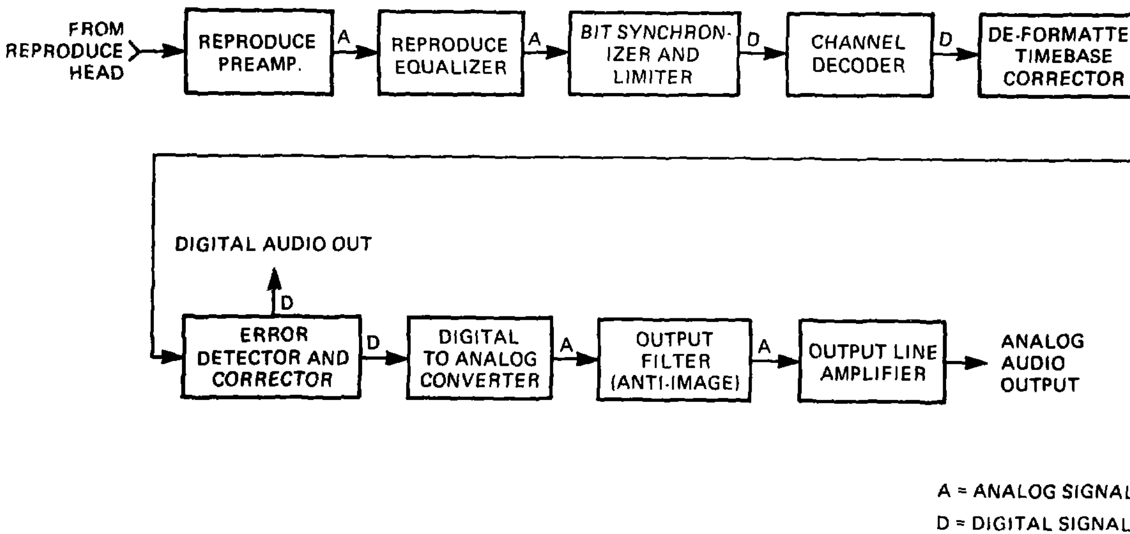
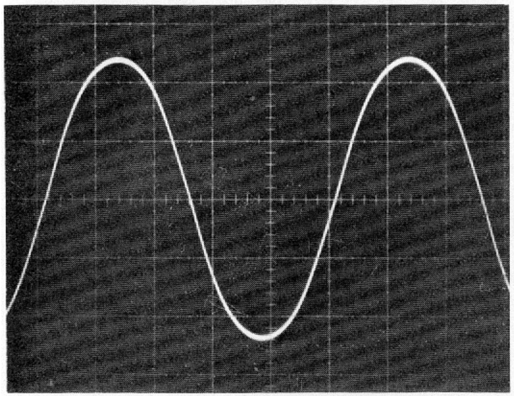
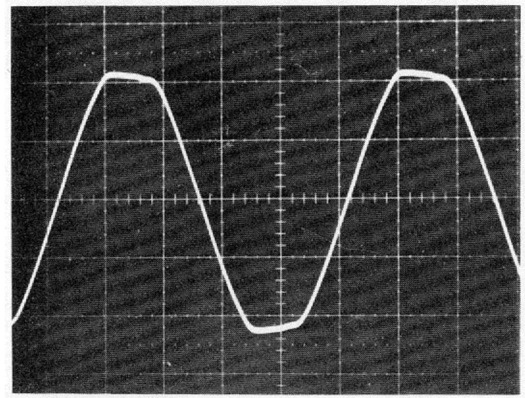


FIGURE 4

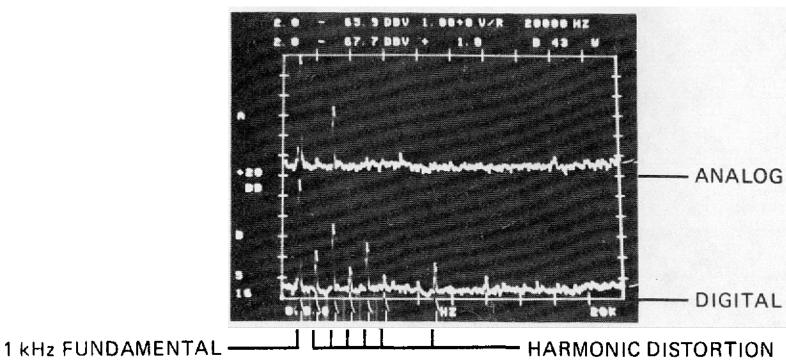
OVERLOAD CHARACTERISTICS WITH INPUT 2dB ABOVE MAXIMUM INPUT SIGNAL LEVEL



A) 1 kHz SINEWAVE FROM ANALOG SIGNAL SYSTEM



B) 1 kHz SINEWAVE FROM DIGITAL SIGNAL SYSTEM



C) FREQUENCY SPECTRA FROM ANALOG AND DIGITAL SIGNAL SYSTEMS