AUDIO NOISE REDUCTION IN CABLE SYSTEMS

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ABSTRACT: PART A

A newly developed compandor provides 23 dB of CCIR weighted noise reduction in a transmission medium where starting signal-to-noise ratio is on the order of 45 dB. A two-band approach consisting of the cascade combination of sliding and fixed band compandors provides a cost effective means of maintaining excellent audio quality in stereophonic television transmission. The compandor is discussed in terms of dynamic properties including overshoot performance, sensitivity to time dispersive transmission media, and reduction of correlated noise spectra common in intercarrier television design.

ABSTRACT: PART B

The delivery to the home of program audio in digital form would have many advantages, among them the potential for very high quality and ease of encryption. Conventional high quality digital-audio conversion schemes yield high bit rates at high cost. A conversion scheme will be described which yields high quality sound, modest bit rate, low cost, and inherent resistance to bit errors.

PART A: A HIGH THRESHOLD COMPANDOR FOR MULTICHANNEL TELEVISION SOUND

INTRODUCTION

The advent of multichannel television sound brings with it increased demand for quality in all associated transmission equipment, as well as some specific requirements for audio noise reduction. Addition of a difference channel component and a second language subcarrier to the baseband mono carrier represent a significant compromise in audio quality. This paper is intended to illustrate a cost effective approach providing stereo signal-to-noise performance equivalent to present monophonic transmission.

Existence of three proposals for a transmission system standard made it clear that any compandor developed would have to be designed to handle both de-emphasized white noise, characteristic of an FM-AM system, and de-emphasized triangular noise, characteristic of an FM-FM system. There are also varying degrees of correlated noise spectra such as intercarrier buzz, incidental phase moduation, and impulse noise. Typical modulation levels and bandwidth restrictions for the (L-R) subcarrier and second audio program channel indicate fringe area signal-to-noise ratios as much as 25 dB poorer than mono. Cable systems generally reduce aural carrier power 5 dB with respect to broadcast standards; a reduction in adjacent channel video interference acheived at the expense of audio noise performance. These considerations contributed to the choice of 40 dB as representative of worst case signal-to-noise conditions.

One of the prime requirements during early listening tests was "compatibility". Thinking at that time included processing of both (L+R) and (L-R) signals and required the compressed signal be listenable on monophonic receivers without expanders. B-type noise reduction (1), (2), (3) was suggested as a solution because of the gentle compression slope and modest amount of noise reduction. Unfortunately, those very desirable B-type characteristics limit the effectiveness of the compandor under the present conditions. B-type was designed to reduce noise characteristic of low speed consumer tape processes (predominantly high frequency in nature), and in doing so left a substantial amount of middle frequency noise apparent. The results clearly indicated a need for increased noise reduction effect which extended lower in the middle frequency range.

All noise reduction proponents were in agreement that the issue of compatibility was in conflict with providing the necessary noise reduction. A decision was made to allow compatible processing of the (L+R) signal if a proponent so desired, but to drop the compatibility requirement for (L-R) processing. The proponents were now free to apply more aggressive compandor algorithms and it was at this time that C-type noise reduction (4) was proposed for processing the subcarrier signal.

C-type provides 20 dB of CCIR-weighted noise reduction using two series-connected "sliding band" processors with separated areas of dynamic action. The processors operate at different thresholds to minimize peak compression ratios and undesirable side effects in channels with time variant frequency and phase characteristics.

Listening tests revealed improved middle frequency noise reduction, but not without audible noise modulation artifacts. The channel noise level limited overall noise reduction effect to 15 dB and caused the expander to mistrack. C-type was originally designed for consumer tape systems and was not intended to provide substantial low frequency noise reduction. Therefore, it was not surprising to note the presence of low frequency components of buzz after noise reduction. A new combination approach was needed maintaining the attributes of a high frequency sliding band device, while providing additional low frequency noise reduction to deal with buzz. A vote amongst the compandor proponents resulted in a majority decision to eliminate (L+R) processing due to the fact that (L-R) noise would ultimately limit stereo signal-to-noise performance.

THE COMPANDOR SYSTEM

The new circuit consists of two series-connected compandors operating in separated frequency ranges with differing amounts of noise reduction. The combined output provides increased noise reduction effect in the cross-over region with a modest increase in maximum compression ratio. A block diagram of the compressor is shown in figure 1, with the corresponding expander shown in figure 2. The well documented "dual-path" approach is used in which a dynamically modified signal in the "side-chain" is summed with a dynamically unmodified signal in the "main path". A compressor is formed via in-phase summation, while the complementary expander function is derived via negative feedback.

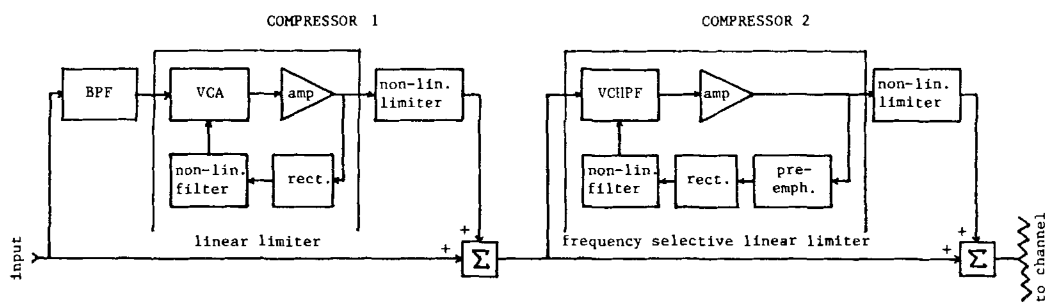

Fig. 1. Compressor block diagram.

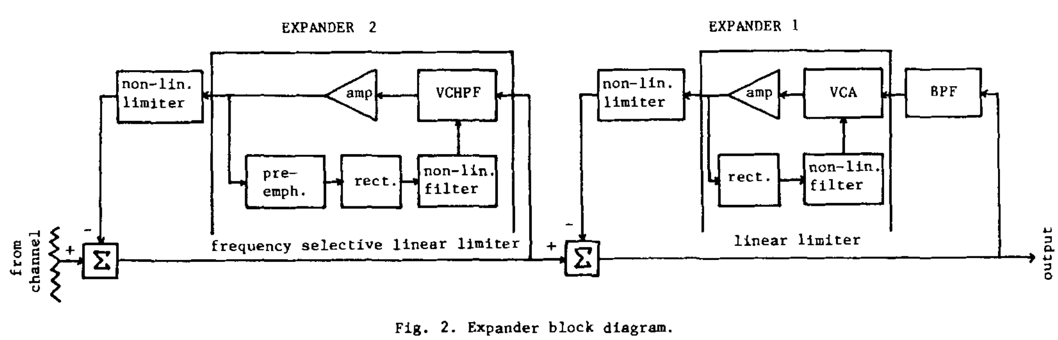

Fig. 2. Expander block diagram.

Compressor 1 is a fixed band device similar to one band of A-type (5) formed by bandpass filtering the input to a voltage controlled attenuator (VCA), in a one-decade range beginning at 80 Hz. An amplifier following the VCA provides the means of adjusting the maximum contribution of the side-chain signal to the compressor output. Attenuation of the VCA is a linear function of control signal which is a full-wave rectified representation of the side-chain output, smoothed by a non-linear filter. Compressor 2 is a sliding band device formed by passing the input signal through a voltage controlled high-pass filter (VCHPF), where the cutoff frequency is threshold-limited to a minimum of 2.2 kHz. An amplifier following the filter again provides a means of adjusting the maximum contribution of the high frequency side-chain to the compressor 2 output. The VCHPF cutoff frequency is a linear function of control signal which is a pre-emphasized, full-wave rectified representation of the high frequency side-chain output, smoothed by a second non-linear filter.

Dynamic Properties

The "bi-linear" nature of the transfer characteristic is illustrated in figure 3. At levels below threshold, side-chain signal dominates the compressor output. As input level increases, the side-chain (a combination of linear and non-linear limiters), contribution becomes less significant and the transfer function approaches unity gain. This configuration provides a effective means of controlling compressor overshoot. Overshoot is the result of time delay in the control path and is equal in amplitude to the change in compressor gain during the transient interval (and thus statistically dependent on source material). Overshoot duration can be minimized by decreasing compressor attack time, but only at the expense of increased susceptibility to expander mistracking (i.e. a compressor with a very fast attack generates additional high frequency components which may extend beyond the channel bandwidth; if these spectral components are attenuated or subjected to phase aberrations, expander mistracking will result).

A non-linear limiter (symmetrical clipper), operates on the side-chain output signal clipping off high level overshoots. The side-chain contribution is reduced at high levels and the resulting distortion (several percent), is short enough in duration (about 1 msec), to be effectively masked by the unaltered signal component of the main path. An identical clipper is used in the expander where complementary subtraction of the distortion products occurs. The relatively small overshoot component at the compressor output (typically 2 dB with average program material), can be handled without significant compromise in the use of channel dynamic range. This also allows the compandor to achieve the full noise reduction effect.

In theory, for transmission paths time dispersive in nature, it is desirable to use rms detection in generation of the compandor control signal. This requires the integration time be long compared to the period of the signal; a direct conflict with good transient performance. In practice, a simple non-linear filter network which combines signal averaging and peak detection provides a sufficiently close approximation without the cost and complexity of rms detection (4).

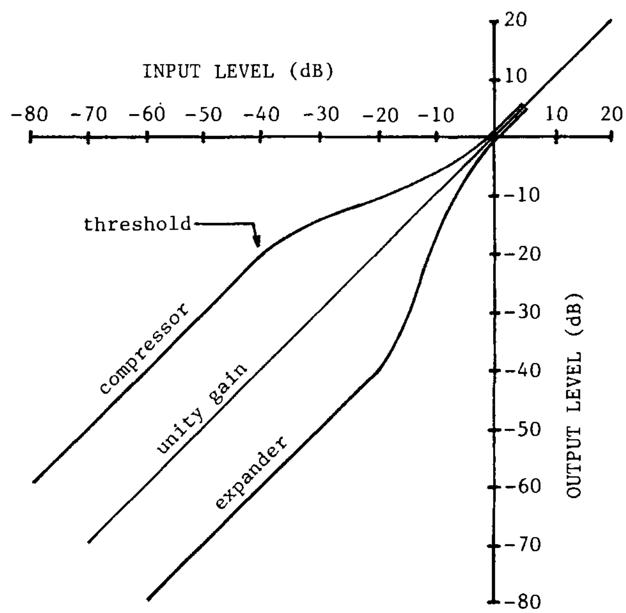

Fig. 3. Input-output characteristics (at 2KHz).

The compandor non-linear filter configurations are identical with time constants optimized for each band of operation. Attack time varies with the dynamic nature of the input signal and is relatively long for transients of moderate level. The resulting overshoots are linearly handled in the transmission path. Large transients cause the filter time constant to adapt to a minimum fixed value shortening the overshoot interval in such a way that modulation products (sidebands), will be masked by the transient itself. The compressor can be modeled as an amplitude modulator which trades dynamic range for signal bandwidth thru multiplication of source and control signal. Attack time must be sufficiently long to maintain psychoacoustic masking of multiplier sidebands and prevent generation of modulation products which extend beyond channel band-limiting filters (attenuation and filter time dispersion cause the expander to generate transient distortion). Compandor release time is critical as well. Designs with single pole control path filters generally require trade-off between response time and distortion caused by ripple. However, the non-linear filter consists of a two pole circuit which reduces ripple distortion and noise modulation products (noise tails due to lengthy release time). A second factor considered in the choice of release time is a psychoacoustic effect whereby sensitivity of the ear is reduced immediately following a transient (5). A release time on the order of 50 msec provides very satisfactory subjective performance.

Band Splitting

Elementary compandors such as those used in telephony are broadband devices consisting of a single signal path. Devices of this nature are best suited for use over narrow portions of the audio spectrum such that the probability of generation of noise modulation artifacts and modulation products is reduced. Wideband compandors exhibit excellent noise reduction properties in the absence of signal, but due to the broadband nature of the detector, the presence of moderate to high level low frequency signals will reduce dynamic action and high frequency noise will become audible (the reverse is also true). Signal pre-emphasis is often used in an attempt to minimize these effects. In contrast, the sliding band (VCHPF) is optimized to take advantage of a second psycho-acoustic phenomena whereby noise in the immediate band surrounding a high level signal is masked in the presence of such a signal, thereby reducing the requirement for noise reduction in that band (5). A signal with dominant mid-frequency energy (f1), will cause the VCHPF cutoff frequency to increase, reducing dynamic action in the region of the signal (where noise is effectively masked), while maintaining noise reduction at higher frequencies (see curve 1 figure 4). A signal in the region of f2 will increase the VCHPF cutoff frequency proportionally more for a given energy level; illustrating the signal adapting nature of the sliding band (see curve 2 figure 4). High frequency noise modulation by low frequency signals is also reduced due to the variable selectivity of the VCHPF. Sliding band technique can be adapted for use at low frequencies but the cost and complexity are not justified in this application. Therefore, a narrow fixed band compandor is cascade connected to provide low frequency noise reduction. The operating bandwidth of the low frequency band is optimized to provide 14 dB of noise reduction with minimal out-of-band modulation effects. Curves illustrating individual and combined compressor contributions (for levels below threshold), are shown in figure 5. First order filter slopes are required in order that summation of the compressor responses result in the smooth overall characteristic shown. The summation process also provides additional middle frequency noise reduction with only a modest increase in peak compression ratio.

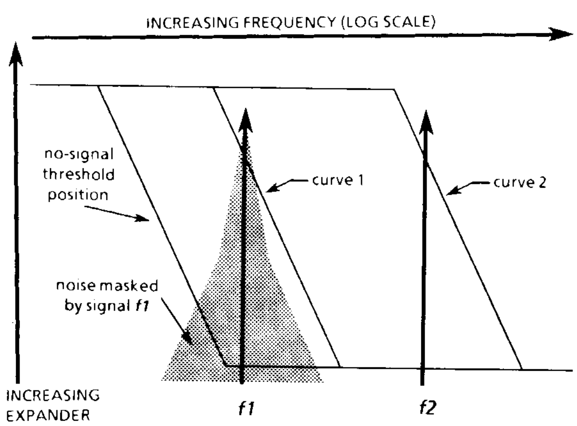

Fig. 4. Signal adapting nature of the sliding band processor.

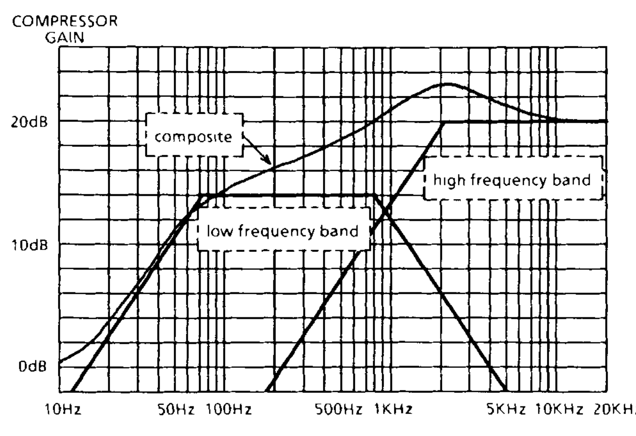

Fig. 5. Compressor gain vs. frequency (below threshold).

SUMMARY

A new companding configuration has been described which provides superior broadband noise reduction characteristics in the presence of high channel noise. This high threshold device functions so as to maintain monophonic audio quality for stereo transmission at the Grade-B service contour limit. In addition, the compandor is ideally suited for use in transmission paths where frequency response characteristics are well defined, and signal-to-noise ratio is as low as 40 dB. Listening tests conducted using very critical program sources including isolated bells, wood blocks and additional highly transient material have shown the compandor to be relatively free of noise modulation artifacts commonly associated with wideband high compression devices.

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PART B: AN ECONOMICAL HIGH PERFORMANCE DIGITAL CODING SCHEME

INTRODUCTION

A good number of analog to digital coding schemes exist, the most common being linear PCM. With adequate sampling rate and number of quantization levels very high quality is obtained. This has become the system of choice for both professional and for highest quality consumer use. However there are many applications where the complexity, performance, and expense of PCM are inappropriate. For several of these applications, the form of encoding known as Adaptive Delta Modulation (ADM) has significant merit. Properly applied, analog noise reduction techniques overcome the inherent performance limitations of this inherently simple and low cost technique.

PCM

The theory of PCM is straightforward and well established. If we sample at more than twice the highest audio frequency, the signal-to-noise ratio of a linear PCM system can be expressed as $\sqrt{1.5} \times 2^n$, where n is the number of bits per sample. Examples of typical systems are the digital compact disc which quantizes to 16 bits at 44.1 kHz and the EIAJ format for recording PCM on video cassettes which employs 14 bit quantization at 44.1 kHz. These systems have bit rates of about 700 kb/s and 600 kb/s, and costs associated with conversion to accuracies better than .006% and .0015%.

In order to reduce both cost and bit-rates, companding techniques have often been applied.

a) Analog companding. Inadequate PCM systems (e.g., 10 bit) can be combined with analog companders. If the noise reduction system is well designed, this approach can work satisfactorily.

b) Instantaneous digital companding. If the analog signal has been quantized to a high precision (14-16 bits), the transmitted bit-rate can be reduced by sending the samples with full resolution only when they are small in magnitude (low level signals). Larger samples are sent with coarser resolution, and quantizing error is therefore increased.

c) Block digital companding. This is similar to b) above, but the reduction in resolution occurs over blocks of data (e.g., 16 or 32 samples) in accordance with the magnitude of the biggest sample within each block. This method gives a greater saving in bit-rate.

Digital companders resemble analog wideband companders in that the level of the quantizing error rises as the signal level rises, and the spectrum of the quantizing error is independent of the nature of the signal; that is, digital companders are fundamentally susceptible to audible noise modulation. Since the spectrum of the quantizing error from a PCM system resembles white noise, which to the human ear is predominantly high frequency, noise modulation is most likely to be audible in the presence of low frequency signals. Our experiments have shown that if the noise above 2 kHz rises to worse than 65 to 70 dB below full-scale in the presence of say a 200 Hz full-scale signal, then noise modulation will be audible on critical musical material. With moderate amounts of pre- and de-emphasis this requirement can be met if the minimum resolution in the presence of full-scale signals corresponds to a 10 bit system.

The British Broadcasting Corporation's NICAM 3, a block range-switching scheme with initial quantization to 14 bits, digitally companded to 10 bits with pre-emphasis, is an example of an optimized system. Digitally companded systems whose resolution drops below 10 bits can be expected to give audible noise modulation under critical conditions.

Digital companding schemes are unattractive for consumer applications because they retain the cost disadvantages of linear PCM, resulting from the need for 14 or 16 bit converters, and the required bit rates remain moderately high (a minimum of about 320 kbits/s per audio channel if noise modulation is to be avoided).

DELTA MODULATION

Delta modulation (DM) can be considered as a special case of differential PCM in which only one bit quantization is performed at a high clock rate. DM is a signal following system. The DM decoder steps up or down with each incoming bit. An encoder is made from a decoder with the addition of a comparator. Figure 1 shows a simple encoder and decoder. The theoretical SNR of a simple DM system is

$$N \geq \frac{\sqrt{3}}{4\pi} \cdot \frac{F^{3/2}}{f_m f_s^{1/2}}$$

where fm is the frequency of a sine-wave at full scale, fs is the audio bandwidth, and F is the sampling frequency. The equality applies when there is no correlation between successive samples; in practice the sampling frequency F is normally much greater than 2fs, so that there is always a high degree of correlation. The resulting value of N is typically 2-3 dB better than that calculated from this formula.

The full scale signal level of a DM system falls at 6 dB/octave with increasing signal frequency; the system is slope limited. A DM system operating at a bit rate on the order of 200-500 kbits/s will have a SNR considerably inferior to a PCM system operating at the same bit rate. Figure 2 shows the quantizing error of a DM system relative to the level of a full-scale signal as a function of the signal frequency with a sampling frequency (and thus bit rate) of 250 kHz. The dotted line shows the equivalent information for a PCM system at a similar bit rate (8 bit, 32 kHz).

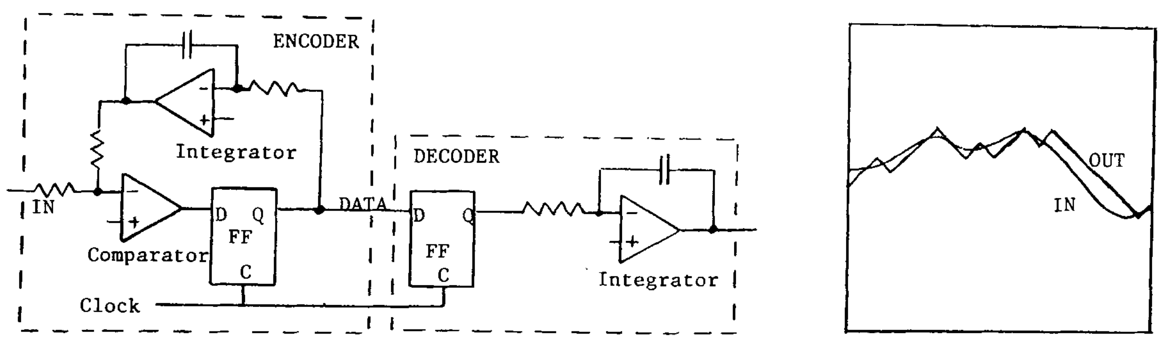

Figure 1: Basic Delta Modulation System

ERROR FEEDBACK

A well known flaw of DM is that the quantizing error resulting from sine-wave inputs does not resemble random white noise, but contains discrete frequencies, or birdies. Error feedback is a technique where the output of an imperfect transmission path is compared with its input and the discrepancy due to the imperfection is added to the input signal in such a sense as to reduce the discrepancy. Since DM is a grossly oversampled system much of the quantizing error lies above audio frequencies. The application of error feedback in the audio band reduces the quantizing error (including the birdies) at audio frequencies at the expense of increased error at frequencies above audio where it is of no significance. In practice, birdies are reduced to inaudibility and audio band noise is reduced by about 10 dB as shown in Fig. 2.

COMPANDED DM

The dynamic range of DM can be improved by the application of companding techniques. An ideal compander would adjust system gain so that the signal is always converted at full scale, whatever its amplitude or frequency content. This objective cannot be achieved by an analog compression-expansion system controlled by the analog signal itself, since it implies an infinite compression ratio, and therefore an infinite expansion ratio (requiring infinite precision for proper tracking). A digital (or quantized) compander in which the step-size adopts one of a discrete number of values, can provide an approximation to the ideal characteristic. Unfortunately this requires very high precision, comparable with PCM system, because errors in step-size lead to discontinuities in the slope of the reproduced audio, and hence additional error.

However, by using a hybrid analog-digital approach, an almost ideal characteristic is achieved. The step-size is made continuously variable at a syllabic rate under the control of the bit-stream. When the data contains long strings of 1's or 0's, indicating that the coder is lagging behind the input signal, the circuit tends to increase the step-size until the strings are broken up. The step-size varies in accordance with the short term peak value of the slope, adapting so that the modulator is always close to full-scale. High precision is not required as discrepancies between the encoder and decoder lead to a small low frequency amplitude modulation of the signal. An unusual property of Adapting Delta Modulation (ADM) is that there is no inherent maximum or minimum of the control signal level, so in theory the dynamic range is infinite. In practical circuits limitations are introduced by amplifier noise sources and power supply rails.

PROGRAM-ADAPTING PRE-EMPHASIS

The subjective performance of ADM with error feedback can be enhanced by a program-adapting response shaping system. Consider the following signal conditions:

a) When signal amplitudes are small, high frequency emphasis in the coder and de-emphasis in the decoder can improve the subjective noise level by reducing the noise power in the part of the spectrum where the human ear is most sensitive.

b) When the input signal contains large amplitudes at low frequencies alone, a degree of low frequency attenuation in the coder, and corresponding boost in the decoder reduces the extent of step-size adaptation and therefore reduces the change in high frequency noise due to the increases in step-size.

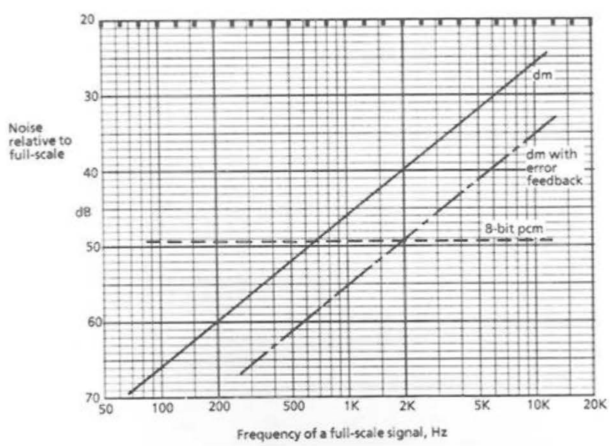

Figure 2: Ratio of maximum signal to noise as function of frequency of a full-scale (sine-wave) signal.

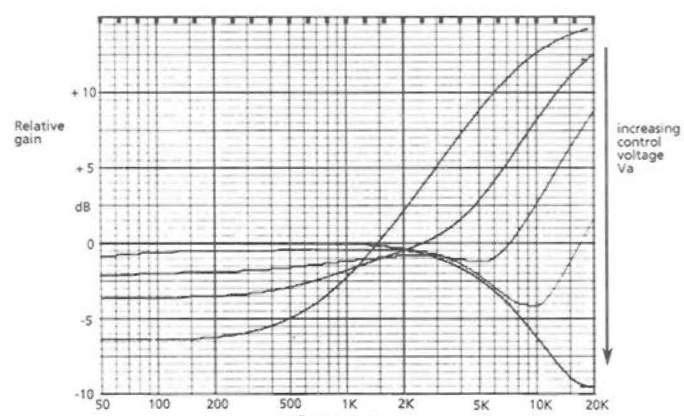

Figure 3: Typical set of responses of adaptive pre-emphasis
(measured results)

c) When the signal contains high amplitudes at high frequencies, the effect of both the high frequency emphasis and low frequency attenuation in the coder is to boost the low frequency noise in the reproduced signal; it is therefore desirable to reduce these response changes, or even to invert their directions.

Hence a family of emphasis characteristics of the form shown in figure 3 give good results. Since the significant factor in determining the required characteristics is the slope of the input signal, which in turn is directly proportional to the step-size, it is possible to control the emphasis with the same information (derived from the bit-stream) that controls the step-size.

As with any signal-controlled syllabic compander, the adapting pre-emphasis of ADM might be expected to display a lag between the onset of a large amplitude signal and the adaptation of the circuit, with a resultant need to compromise between speed of response and modulation distortion. In the adaptive pre-emphasis this compromise is greatly reduced by using a two-path configuration, in which the pre-emphasized components of the signal present in both encoder and decoder are subjected to non-linear complementary overshoot suppression, similar to that used in the Dolby analog noise reduction systems.

PRACTICAL IMPLEMENTATION

Figure 4 shows the block diagram of a practical encoder. The decoder would use the same functional blocks except that the comparator (10) and the error feedback (8 and 9) blocks are not required. This system has been built using standard off-the-shelf op-amps and CMOS logic. For mass production a large scale I.C. is envisioned. A feasibility study conducted for us by Silicon Systems Inc., the California IC company known for its expertise in switched capacitor techniques, has concluded that the whole converter can be condensed to one chip with an area of 20mm^2, requiring only a few non-critical external capacitors and a source of clock. The I.C. would be capable of switching between encode and decode modes.

PERFORMANCE

The frequency response of the system is determined almost entirely by the input and output filters, which can be simple two- or three-pole low pass filters. Assuming a sampling frequency of 250 kHz, a response up to 16 kHz (or even 20 kHz) can be achieved.

The overload point of the system is determined by the supply rails. The noise level is proportional to the level of the control signal, which in principle can decay to zero yielding an infinite dynamic range. Practical

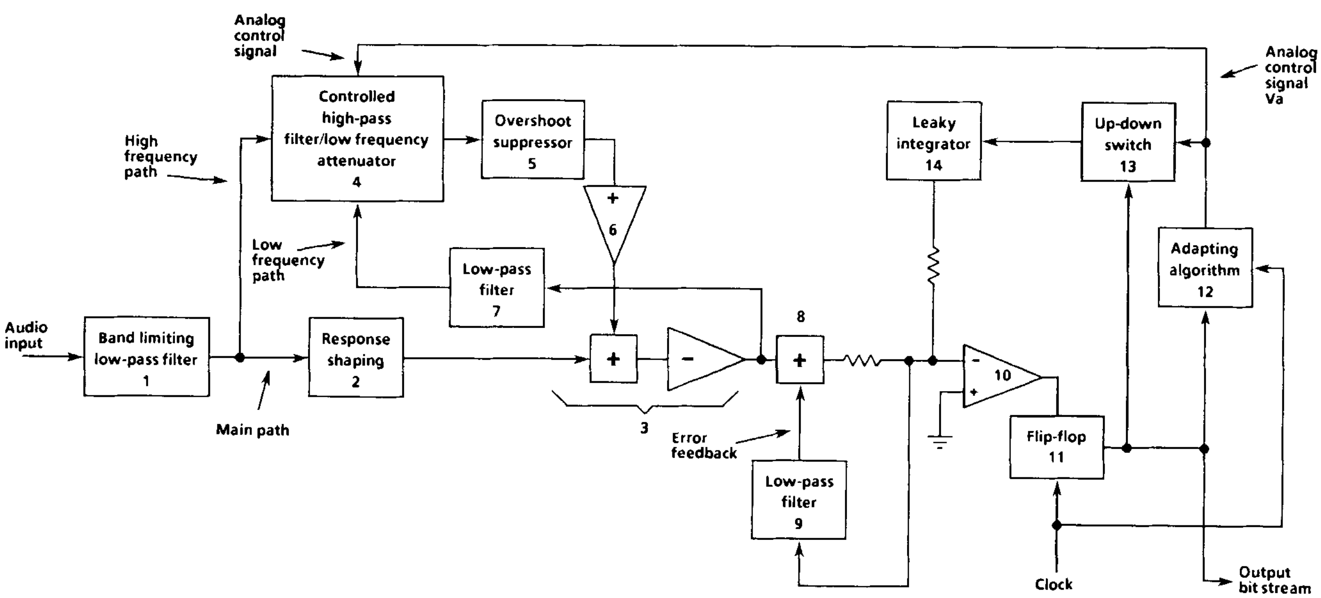

Figure 4: Practical Encoder

circuits of course contain noise sources so the control signal will remain large enough to encode the encoder's own noise sources into the bit stream. The combination of the encoder and decoder amplifier noise sources limit the dynamic range. Using typical I.C. op-amps the dynamic range is on the order of 100 dB.

Since the noise is a function of the signal (as with any companded system, analog or digital), the signal-to-noise ratio is not the same as the dynamic range. Realistic assessment of noise in the presence of signals requires consideration of masking. We have made measurements of high frequency noise in the presence of low frequency signals, and low frequency noise in the presence of high frequency signals. Figures 5 and 6 give examples of these measurements, together with the limits of audiblity. It is apparent that the noise modulation is unlikely to be audible on critical material. For perspective, calculated points for the NICAM 3 system are indicated.

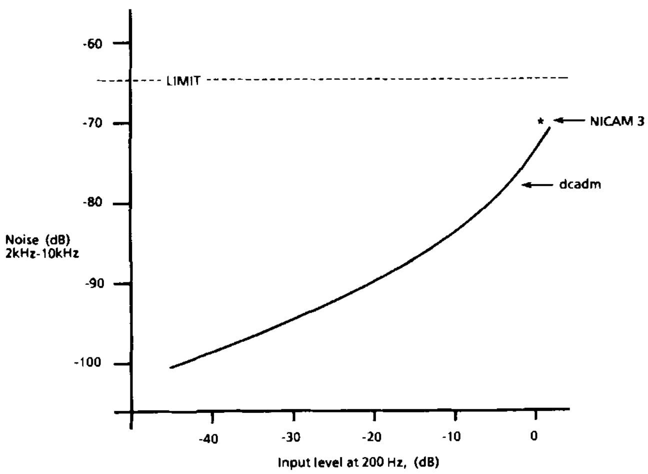

Figure 5: High frequency noise v level of low frequency signal
(measured results, 0dB = full-scale)

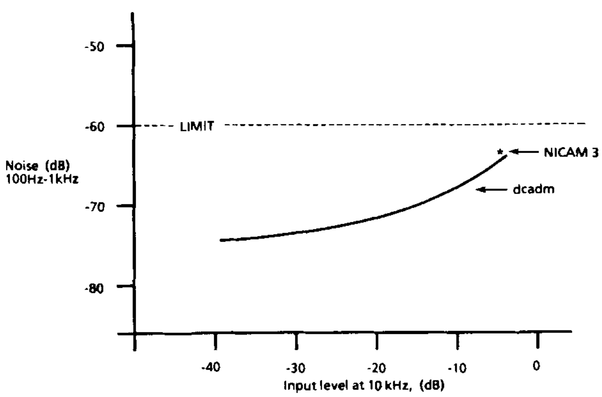

Figure 6: Low frequency noise v level of high frequency signal
(measured results, 0dB = full-scale)

Figure 7 shows the dynamic performance of the system. The test signal is a high frequency transient. The overshoot suppression eliminates the normal transient distortion resulting from the finite response time of a syllabically adapting system; the rounding at the beginning of the tone burst is largely the result of the low pass filters.

DIGITAL DATA ERRORS

The reason for converting audio signals from an analog into a digital representation is to assure more reliable transmission of the signals, since digits may be transmitted with essentially no error. Error free transmission has a price however, in that excessive amounts of bandwidth or power may be required. Since a practical system should be designed for efficient use of the channel, there will typically be some finite error rate.

Digital errors may be dealt with in several ways. If the error rate is low enough, the errors may simply be ignored, and the resulting sound quality degradation tolerated. One method to deal with the errors is to detect which bits are in error by means of a parity or algebraic check code. The unreliable bits can then be detected and discarded (turning the bad data into erasures instead of errors) and some type of error concealment can be employed in the DAC. The ideal method of handling errors is to transmit extra redundant bits along with the audio bits so that errors may be detected and actually corrected in the digital domain. However, an Error Detection And Correction (EDAC) system entails considerable complexity, and will itself fail when the raw error rate is sufficiently high, thus requiring error concealment in the DAC. In the design of a digital transmission system the effect of digital errors on audio quality should be considered. The nature of the degradation of various types of A-D-A conversion systems when subjected to digital errors can have a significant impact on overall system design.

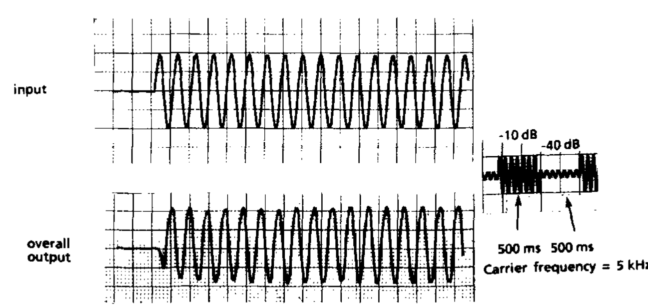

Figure 7: Typical response to a high level transient

The effect of digital data errors or erasures on PCM systems is well known. An isolated one bit error can result in an impulse of amplitude ranging from negligible (for an LSB error) to half of full scale (for an MSB error) as shown in figure 8. Even a low error rate can obviously have a severe impact on a PCM system. Isolated erasures, however, can be handled quite comfortably by a PCM system since straight forward linear interpolation can replace an erased value. Provided these erasures do not occur too frequently, the effect of this error concealment is quite innocuous. Linear interpolation can only conceal isolated erasures and not burst erasures; if burst erasures are likely to occur, interleaving is usually employed so that a burst error is transformed into a repetitive error.

The effects of digital errors or erasures on an ADM system are quite different from a PCM system. An isolated one bit error will cause the decoder to integrate in the wrong direction for one clock cycle, as shown in figure 9. This has the effect of adding a small step to the audio waveform. The amplitude of this step is never large because every bit has equal weight--one quantum of resolution. Additionally, the spectrum of this disturbance is subjectively less annoying than that with PCM since it is shifted towards lower audio frequencies.

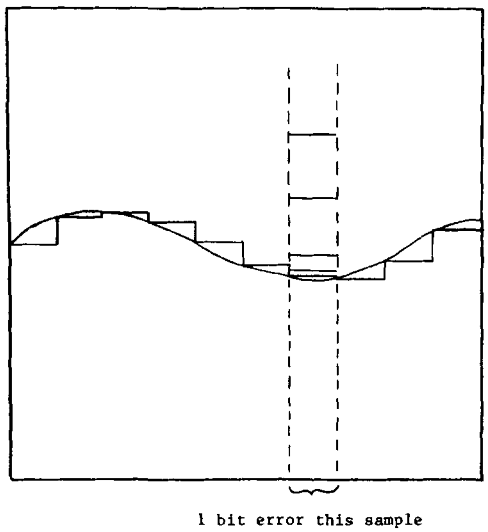

Figure 8a: Effect of a one bit error in a PCM system

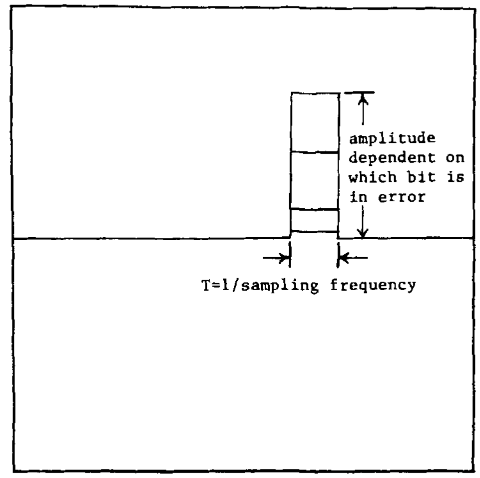

Figure 8b: Error generated by a one bit PCM error

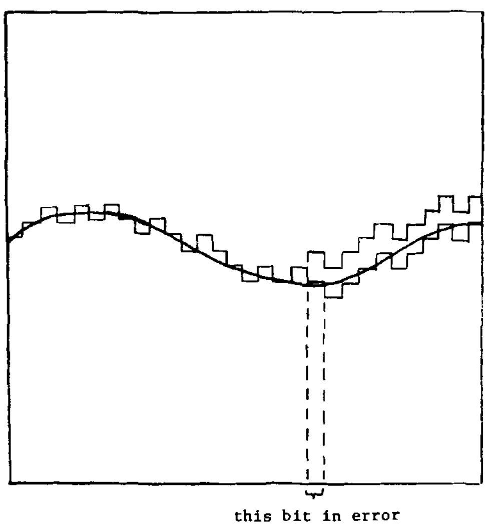

Figure 9a: Effect of a one bit error in a DM system

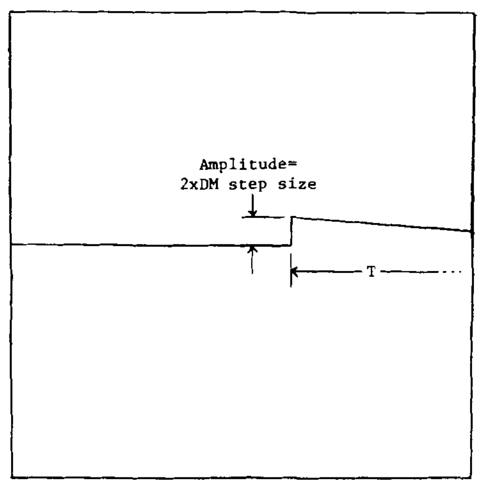

Figure 9b: Error generated by a one bit DM error
(The error decays at a rate set by the integrator leak)

The innocuous effect on reproduced audio of an ADM bit error can yield some very great system advantages. Consider a digital transmission system with a bit error rate of 1E-5. This would lead to very audible impulses occurring a couple of times per second with PCM coding, but would be barely audible with ADM. The PCM would require some form of error control in the transmission system; the ADM would not.

APPLICATIONS

The performance of a digitally controlled ADM system with program-adapting pre-emphasis as described are similar to those of an optimally designed digitally companded PCM system such as NICAM 3. However the ADM operates at a somewhat lower bit-rate (250 kbits/s vs. 322) and eliminates the need for the expensive 14-bit converters and high-order anti-aliasing filters. The ADM tolerance for errors means a perfect transmission channel is not required. ADM is therefore very attractive for consumer applications, including direct distribution of program to the home via cable.

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