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REPORT

A NEW FILTERING PROCESS FOR OPTIMAL OVERSHOOT CONTROL

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A NEW FILTERING PROCESS FOR OPTIMAL OVERSHOOT CONTROL

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ABSTRACT

When both peak amplitude (in the time domain) and bandwidth constraints are placed upon a signal as in FM stereo broadcasting, there are conflicts among the requirements for limiting modulation peaks, attenuating components beyond 15 kHz, and maintaining a flat amplitude characteristic to 15 kHz. Previous attempts at overshoot control do not simultaneously satisfy all of the above requirements. Furthermore, some techniques currently in use can cause severe audible distortion under certain programming conditions.

Lowpass filters by design change the frequency distribution of a signal and by consequence change the phase relationships of the same signal. Both changes are causes of overshoot. Elimination of harmonic terms deletes components that serve to reduce the peak amplitude of the signal. Phase distortion rearranges signal components as a function of time to form overmodulating peaks.

Optimal overshoot control must perform all of the following under all programming conditions.

1. Flat frequency response to 15 kHz at all levels up to 100% modulation.
2. High attenuation of frequencies above 15 kHz.
3. Suppress overmodulation due to overshoot to an insignificant level.
4. Insignificant THD and IM distortion at any level up to 100% modulation.
5. No degradation of audio quality.

A new technique that eliminates overmodulation due to overshoot is presented and explained.

I. INTRODUCTION

BACKGROUND: FM stereo radio broadcasting is rapidly becoming a highly competitive medium. This change manifests itself in many ways, including the effort to have a technically superior sound. This objective usually involves a tradeoff between quality and quantity, or fidelity vs. loudness. Most audio processing innovations to date sacrifice some amount of fidelity for some degree of loudness increase. The Harris Dynamic Transient Response (DTR) filter, an integral part of the Harris MS-15 exciter, allows a loudness increase of 2-6 dB (dependent on limiter type) with absolutely no degradation of fidelity.

PRINCIPLES OF STEREO FM: FM stereophonic broadcasting is a frequency domain multiplexed (FDM) system. A left-plus-right (L+R) signal is transmitted in the band 50 Hz-15 kHz. This is the monaural baseband signal. A double sideband suppressed carrier (DSB) signal modulated with left-minus-right information is transmitted at 38 kHz. To properly demodulate the DSB 38 kHz signal, a 19 kHz pilot tone is transmitted with a phase such that when it is frequency-doubled, L-R information can be synchronously detected. The composite stereo signal is shown in Fig. 1.

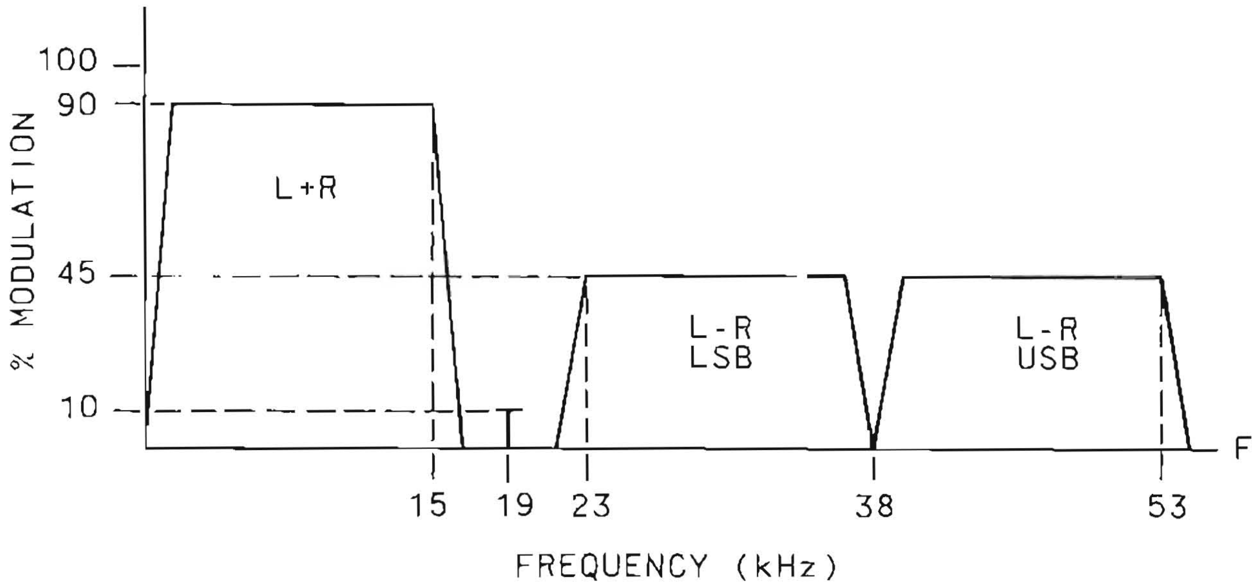


FIGURE 1
FM STEREO MULTIPLEX SIGNAL SPECTRUM

WHY AUDIO LOWPASS FILTERS ARE REQUIRED: There are constraints placed upon the amplitude and bandwidth characteristics of the left- and right-channel audio signals such that the resultant L+R and L-R signals will exceed neither their amplitude bounds nor bandwidth allocations. Otherwise the multiplexed signals would suffer distortion and mutual interference.

To control the amplitude of the L and R channel signals AGC amplifiers, peak limiters, and clipping devices are customarily used. Typically these processors add to the harmonic content of the program, producing a signal which would result in excessive bandwidth. In the better stereo generators, lowpass filters have been included to attenuate harmonics beyond the 15 kHz bandwidth of the system.

Some inexpensive switching type stereo generators omit the audio lowpass filters in an attempt to eliminate overshoot. With no audio filters, the stereo composite lowpass filter will overshoot instead This is absurd. Not only has the overshoot problem been left unsolved, but the stereo generator is vulnerable to pilot interference and aliasing.

II. CAUSES AND EFFECTS OF OVERSHOOT

MECHANISMS OF OVERSHOOT IN LOWPASS FILTERS: Although the input to a lowpass filter may be accurately amplitude-limited, such is not necessarily the case at the filter's output Ringing and overshoot of the filter can seriously degrade the accuracy of the limiting action. Lowpass filters may overshoot 6 dB (100%) on some signals which are not uncommon at the output of audio processing equipment.

A lowpass filter changes two independent qualities of its input signal In addition to the obvious change of the amplitude vs. frequency characteristic the filter also changes phase relationships among different frequencies in the filter's passband. This is equivalent to stating that different frequencies take different lengths of time to propagate through the filter. Associated with these two changes to the signal are two mechanisms causing overshoot

1. ATTENUATION OF HARMONICS

Consider the ideal case of a lowpass filter with rectangular frequency response and zero time delay. This filter is in fact unrealizable but nevertheless would exhibit overshoot due to elimination of harmonics. The frequency response of this filter is shown in Fig. 2.

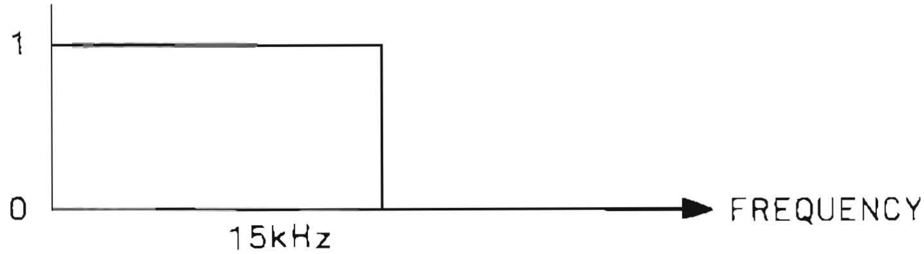


FIGURE 2
IDEAL FILTER FREQUENCY RESPONSE

Assume that the input signal is a 10 kHz squarewave of amplitude A. The Fourier expansion of this signal is:

$$v(t) = A \frac{4}{\pi} \sum_{n=1,3,5,\ldots}^{\infty} \frac{1}{n} \sin(2\pi fnt)$$

where f is frequency.

The squarewave signal has components at the fundamental and odd harmonic frequencies only, i.e., 10, 30, 50, 70, etc. kHz. Since the filter cuts off at 15 kHz only the fundamental (10 kHz) component of the squarewave appears at the filter output. Note that if the squarewave amplitude (A) is one volt, then the peak value of the fundamental component (identically equal to the output signal) is 4/pi or 1.273. This constitutes an overshoot of 27%. The squarewave and its fundamental component are shown superposed in Fig. 3.

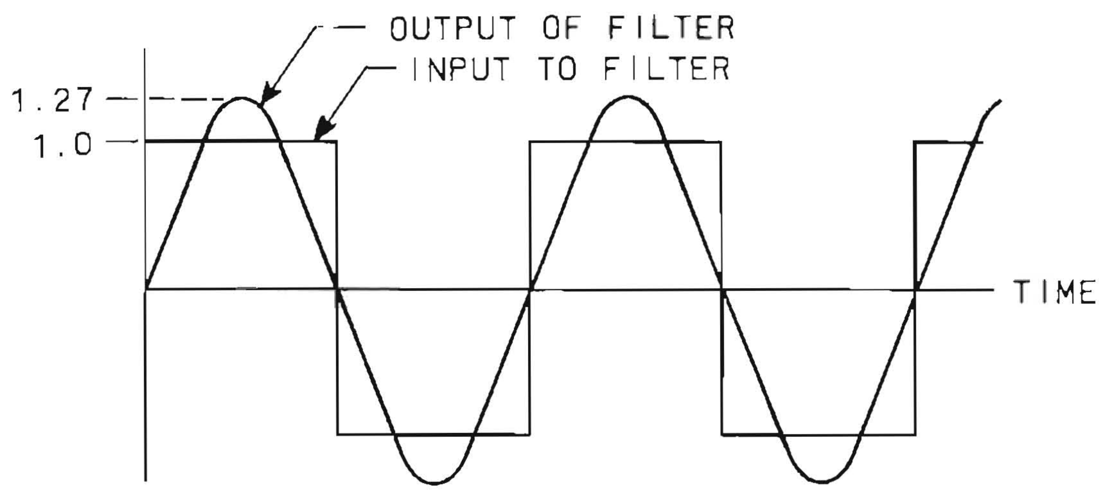


FIGURE 3
SQUAREWAVE AND FUNDAMENTAL COMPONENT

This is only one example of many possible signals that would cause a linear phase lowpass filter to overshoot.

2. NON-UNIFORM TIME DELAY

If different signals propagate through the filter with different time delays, it is possible for input signals separated in time to become coincident at the filter's output. This could result in an overshoot.

Continuing with the example of the squarewave, consider the case where only the fundamental and third harmonic fall within the filter's passband. Squarewaves in the range of 3-5 kHz satisfy this condition. The input and output of the ideal filter discussed in part 1 are shown in Fig. 4.

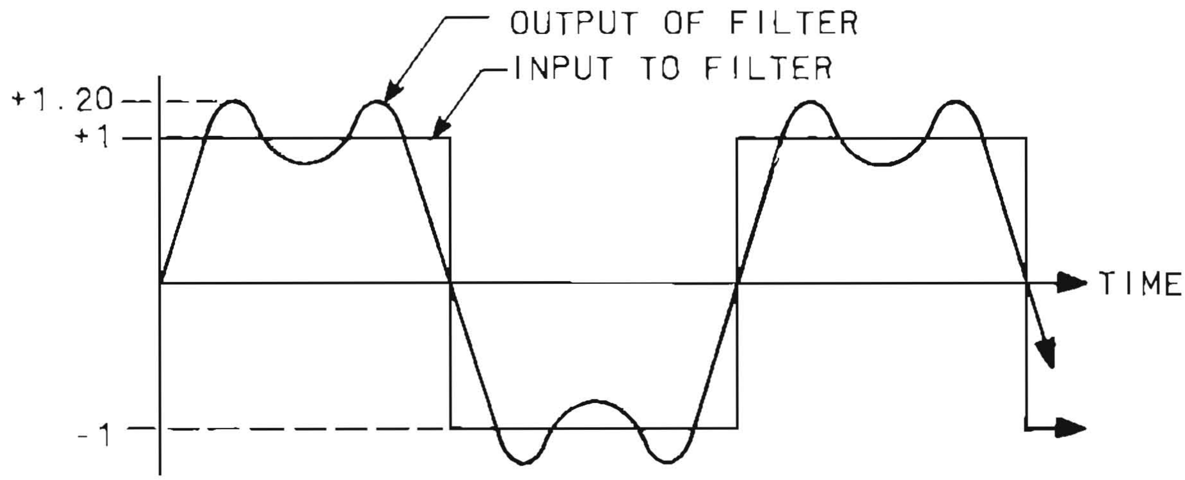


FIGURE 4

3-5kHz SQUAREWAVE RESPONSE: FUNDAMENTAL PLUS 3rd HARMONIC

Overshoot is 20.0%. Since such a filter is impossible to build, the response of Fig. 4 in general cannot be produced. Rather, time delay will vary as a function of frequency, thereby upsetting the phase relationship between the fundamental and third harmonic. If the phase of the third harmonic is shifted 180 degrees relative to the fundamental, the waveform of Fig. 5 results. Overshoot is 70% or 4.6 dB.

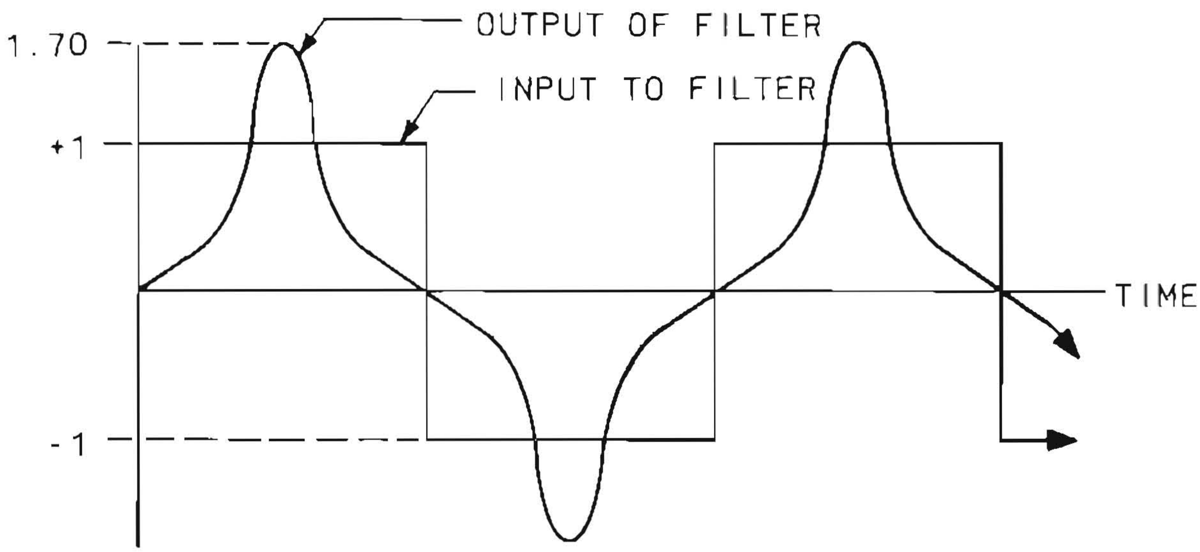


FIGURE 5

EFFECT OF PHASE DISTORTION

EXAMPLES: A typical filter specification may require frequency response to be flat within ±0.1 dB from 0-15 kHz and -50 dB at 19 kHz and above. By far the most practical filter meeting these specifications will be an elliptic function type filter. This filter exhibits a very sharp rate of cutoff and a highly nonuniform time delay characteristic.

A seventh order filter meeting these specifications has group delay (time delay) of approximately 43 microseconds which is uniform from DC to 3.5 kHz, 45 microseconds at 5 kHz, 53 microseconds at 7.5 kHz, 62 usec at 10 kHz, increasing to 238 microseconds at 15 kHz which corresponds to 1,-285 degrees of phase distortion (3½ rotations). Therefore the squarewave response of Fig. 5 is certainly possible with this filter.

Time delay generally increases with frequency within the passband of an elliptic filter. Minimum time delay is at DC while maximum time delay within the passband occurs at the cutoff frequency A test signal has been devised which causes filters to overshoot primarily as a function of their time delay distortion. The test signal consists of a sinewave burst immediately followed by a DC step signal. The sinewave will accumulate maximum time delay (238 usec.) while the DC step signal will accumulate a minimal time delay (43 usec.). At the filter's output the sinewave will coincide with the beginning of the DC step signal. This phenomenon is shown in Fig. 6 Note that the overshoot is 100% (6 dB)!

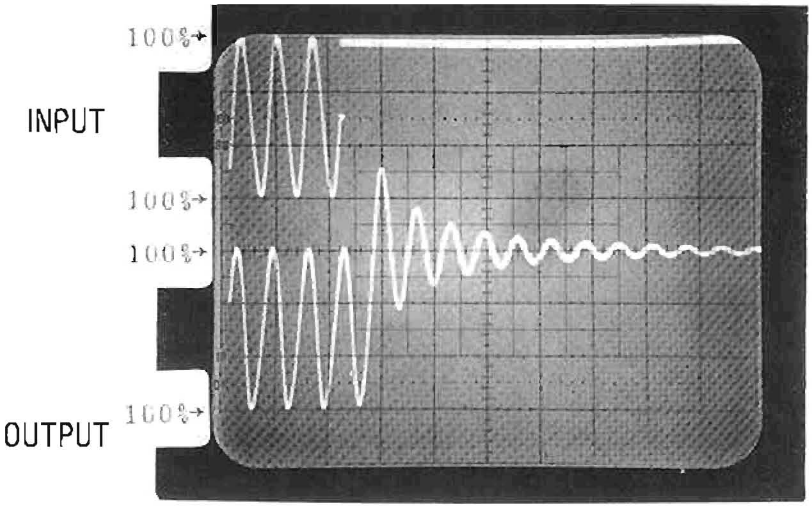


FIGURE 6

SINE/STEP FILTER RESPONSE

Through a combination of effects (both attenuation of harmonics and nonuniform time delay) a myriad of signal types can cause a typical elliptic lowpass filter to overshoot. A low frequency squarewave response is shown in Fig. 7.

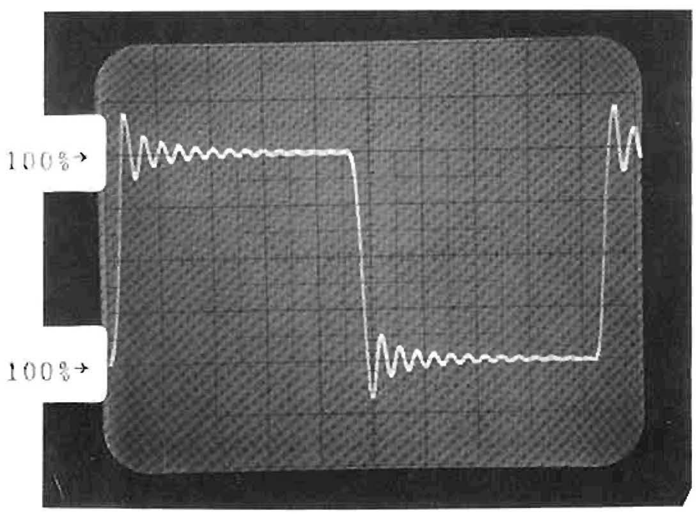


FIGURE 7

TYPICAL FILTER

SQUAREWAVE RESPONSE

The waveforms of Fig. 6 and Fig. 7 are common with certain types of music and/or certain types of FM limiters. To offset overshoot audio levels are simply turned down to a point where overshoots "of frequent recurrence" do not exceed 100% modulation. This can mean a sizeable reduction in modulation effectiveness, usually on the order of 2.5 - 6 dB!

III. DEVELOPMENT OF A SOLUTION

NEED FOR A NEW APPROACH: There have been several previous approaches to the problem. Although existing systems do control overshoot, they also contribute unwanted side effects to the signal.

One method for overshoot control uses a delay line and an AGC stage. This system can cause gain "pumping" Another popular system uses alternate clipping and filtering combined with a complementary high frequency boost and cut. This system suffers from excessive intermodulation distortion and a high frequency rolloff which is dependent upon signal level.

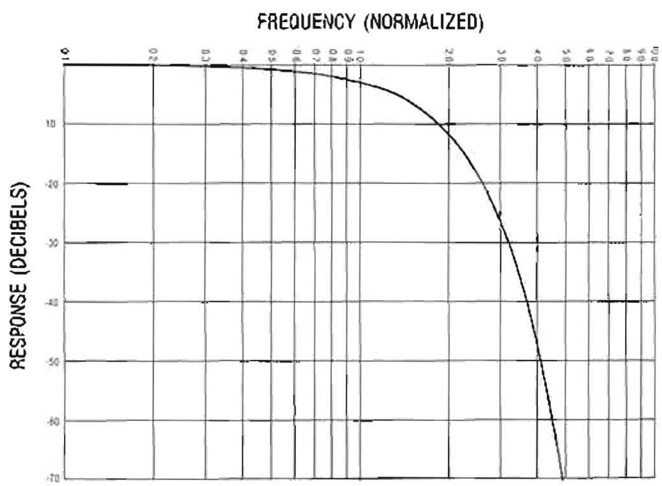
CONSTRAINTS: It is clearly desirable to have a filter which will eliminate harmonics above 15 kHz yet preserve the peak amplitude-limited nature of its input signal. Note that it is not necessary to have a filter that does not overshoot. Ringing and overshoot are completely unobjectionable provided that the overshoots do not exceed the 100% modulation level. From this point on, the term "overshoot" will denote only overshoots above 100% modulation. The filter requirements are

1. Frequency response flat ±0.5 dB 20 Hz-15 kHz at all levels up to 100% modulation.
2. Attenuation above 19 kHz inclusive: 50 dB minimum.
3. Overshoots not exceeding 102% modulation
4. Filter shall be transparent to steady state sinewave signals: THD and IM distortion 0.1% or less.
5. Any effect of eliminating overmodulating overshoots shall be inaudible.

BESSEL FILTER UNSATISFACTORY: One filter that does not overshoot is the Bessel type. The Bessel filter has maximally flat time delay for a minimum phase filter However, its frequency response is inadequate. It has a very gradual rate of cutoff shown in Fig. 8.

FIGURE 8

IDEAL BESSEL FILTER RESPONSE (GAUSSIAN FILTER)



NON MINIMUM PHASE FILTER UNSATISFACTORY: If one were to start with a filter with sufficiently sharp cutoff and attempt to find a phase function resulting in minimum overshoot, the result would be a lowpass filter that approximates linear phase, yet still overshoots. Even in the case of the ideal filter discussed under "Causes and Effects of Overshoot", the filter still overshoots.

FILTER MUST BE NONLINEAR: It would appear that there is no filter that satisfies all the above conditions. There is no linear time-invariant filter that satisfies all the above conditions. The statement that we can tolerate overshoots below a certain level implies that a nonlinear filter may work. That is, the filter may have one set of characteristics up to a certain level and other characteristics above that level. The requirements that the action be inaudible and that the filter be transparent to sinewaves (no harmonic or intermodulation distortion) imply that the filter must be perfectly linear up to 100% modulation. It is feasible to have a filter which is linear up to 100% modulation and nonlinear only when an overshoot above 100% is imminent.

IV. SOLUTION: DYNAMIC TRANSIENT RESPONSE FILTER

THEORY, IDEAL CASE: Assume that we have two identical lowpass filters The filters have a cutoff frequency of 15 kHz with infinite attenuation above and zero attenuation below, and a uniform time delay of 100 microseconds for all frequencies. (Such a filter is of course non-causal and impossible to build.) Consider the situation of Fig 9 where the filters are cascaded, that is, the output of one filter drives the second.

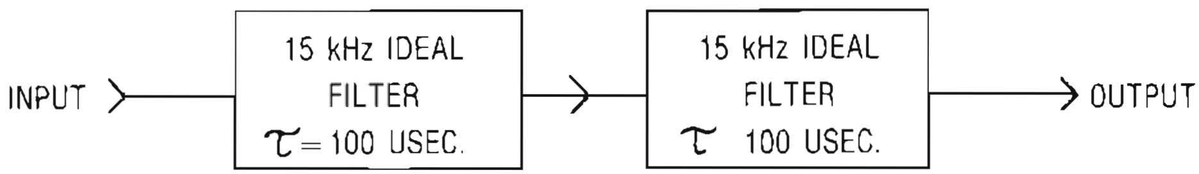


FIGURE 9

CASCADED IDEAL FILTERS

After application of a signal to the first filter, the output appears 100 usec. later with all components above 15 kHz removed. Phase relationships and amplitudes of components below 15 kHz are preserved, the only change to the signal will be the elimination of components above 15 kHz and 100 usec. of time delay When the first filters's output is applied to the second, the second filter will function only as a 100 usec. delay line. Since there are no components above 15 kHz at the second filter's input, the second filter does not change the signal except for the addition of time delay. Therefore, **the first filter predicts the output of the second filter.** This prediction technique is employed in the Harris DTR filter.

IMPLEMENTATION: The DTR filter is a system which comprises two lowpass filters, an allpass filter (phase equalizer), and nonlinear compensation circuitry. A block diagram is given in Fig. 10.

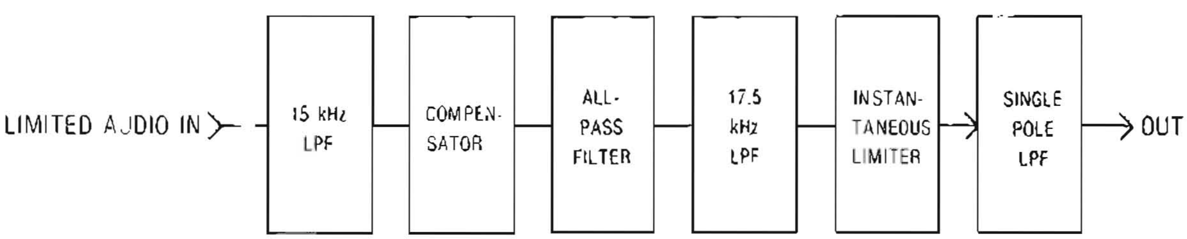


FIGURE 10

DYNAMIC TRANSIENT RESPONSE LOWPASS FILTER

The first filter (at extreme left) in Fig. 10 is a seventh order elliptic type with a cutoff frequency of 15 kHz. The second filter is also seventh order elliptic but the cutoff frequency is 17.5 kHz. The second lowpass filter is preceded by an allpass filter which linearizes the lowpass filter's phase from DC to the cutoff frequency of the first filter (15 kHz). The combination of the allpass filter and the second lowpass filter presents an approximately uniform time delay of 100 usec. between DC and 15 kHz.

Since the passband of the first filter is contained within the linear phase passband of the second filter, the second filter changes neither the phase nor amplitude relationships of the first filter's output but only adds time delay. Therefore, the first filter predicts the output of the second filter. If the first filter overshoots, the second filter will overshoot 100 microseconds later. The inverse is also true if the first filter does not overshoot, the second filter will not overshoot

The compensator functions only when the first filter overshoots. The compensator is designed to take the appropriate evasive action to stop the second filter's predicted overshoot. This nonlinear action is discussed below.

EVASIVE ACTION: SUBTRACTING OVERSHOOTS: A block diagram of the compensator is shown in Fig. 11.

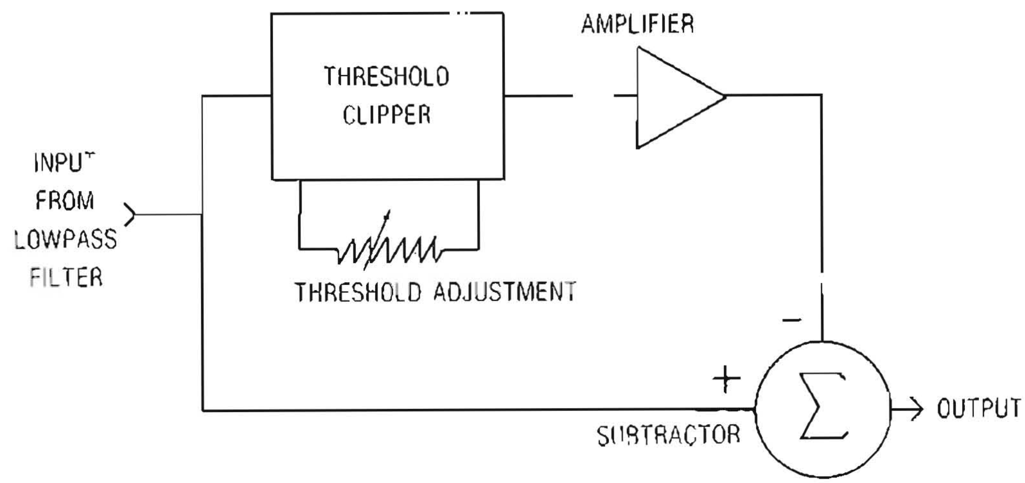
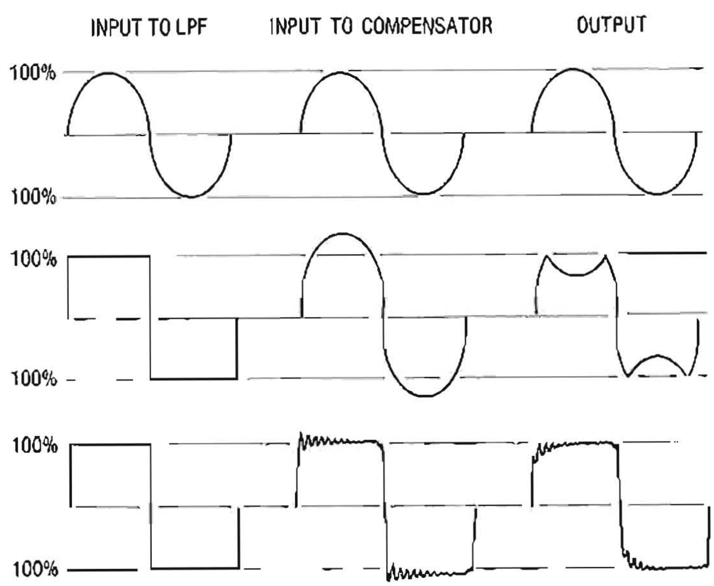


FIGURE 11
COMPENSATOR

Up to the 100% modulation level, the compensator causes no change to the signal. If the first filter overshoots above the 100% modulation level, that part of the signal which exceeds the threshold will be separated from the input and applied to the amplifier. The overshoots, having been separated and amplified, are subtracted from the original signal. Typical inputs and outputs are shown in Fig 12.



The effect of this action is to subtract a component at the frequency of the offending overshoot

The output of the compensator contains frequency components above and below 15 kHz The compensator subtracts signal components below 15 kHz to eliminate overshoot Incidental extraneous components above 15 kHz are generated in the process. These high frequency products must be removed in such a manner that phase relationships are preserved. For this reason the processed signal is filtered by an allpass/lowpass combination which approximates linear phase to 15 kHz.

INSTANTANEOUS LIMITER/FILTER: Due to several approximations in the process, the second filter may occasionally overshoot several percent. An instantaneous limiter follows the second filter for this reason. Harmonics generated by this device are very low in amplitude since overshoot has been nearly eliminated. The limiter drives a single pole filter which does not overshoot. Even severely processed audio results in harmonic components that are at least 60 dB down.

V. RESULTS

OPERATION: The DTR filter is highly effective in its operation. Overmodulation due to overshoot has been reduced to 2%, approximately the accuracy of a modulation monitor. Operation is possible with **any** limiter and any program material including Dolby® processed audio. Setup is easy All one does is to apply a signal that is known not to overshoot at 100% peak modulation. This can be a 400 Hz sinewave. Using this signal as a reference the compensation thresholds are set to a level corresponding to 100% modulation. LED indicators are provided to indicate when an overshoot is compensated, thereby facilitating setup.

Extensive listening and A-B tests have shown that on all type of programming the DTR filter produces no audible effect. Unlike other techniques using AGC, delay lines, or conventional clippers, the DTR filter takes action if and only if an overshoot is imminent. Because the energy contained in the overshoots is inappreciable, deletion of that energy is imperceptible to the ear. The modulation monitor, however, does not respond to average energy but rather peak voltage. The perception of the modulation monitor (and the FCC) is unlike that of the ear and **does** respond to low energy, high instantaneous amplitude transients.

The overshoot compensator removes the transients above 100% modulation. It does no more and no less.

WAVEFORMS: When a 600 Hz squarewave is applied to a conventional 15 kHz elliptic function lowpass filter the output rings and overshoots as shown by the top trace of Fig. 13.

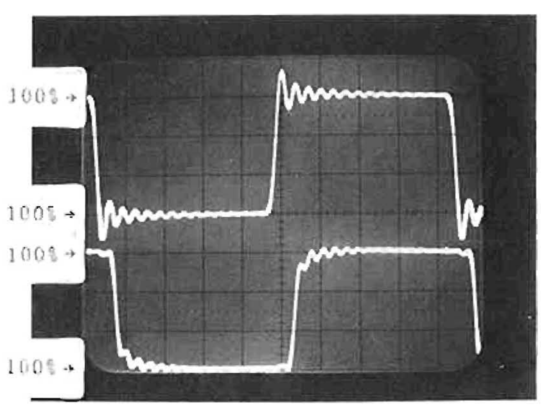


FIGURE 13

SQUAREWAVE RESPONSES

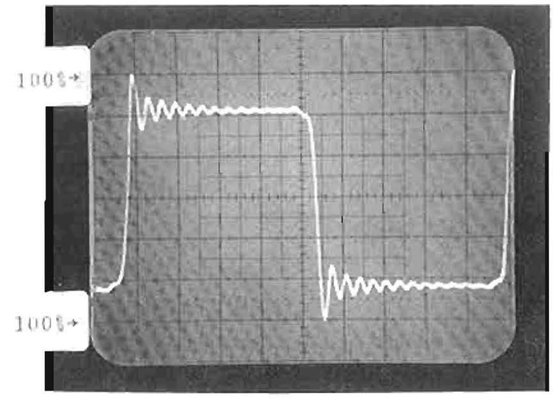
The DTR filter is shown in the bottom trace. There is a small amount of ringing, however, none of this causes overmodulation.

Fig. 14 shows the same squarewave but applied at 65% modulation. The system is completely linear at this level and no compensation is taking place

FIGURE 14

DTR FILTER RESPONSE:

SQUAREWAVE AT 65% MODULATION

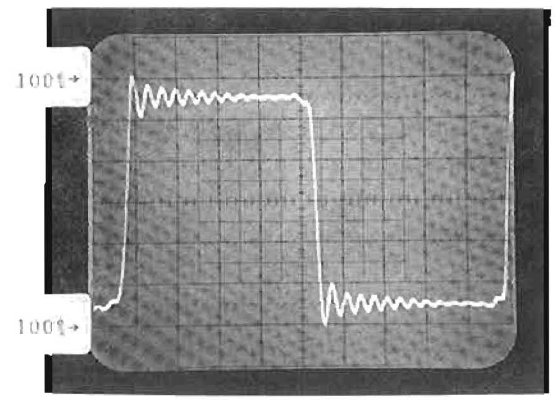


In Fig. 15 the level has been increased to 90% modulation. Here there is some action taking place to limit the first cycle of ringing to 100%.

FIGURE 15

DTR FILTER RESPONSE:

SQUAREWAVE AT 90% MODULATION

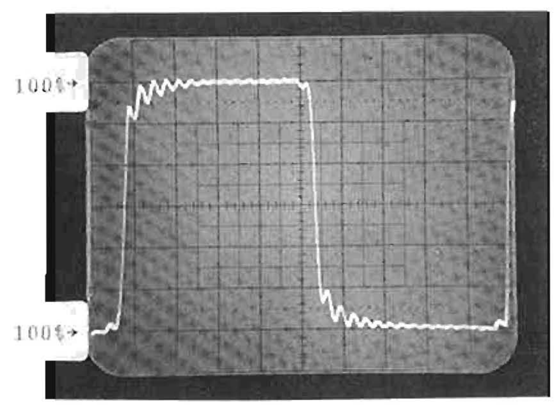


Finally in Fig. 16 the overshoot compensator is completely enabled as the squarewave is applied at 100% modulation.

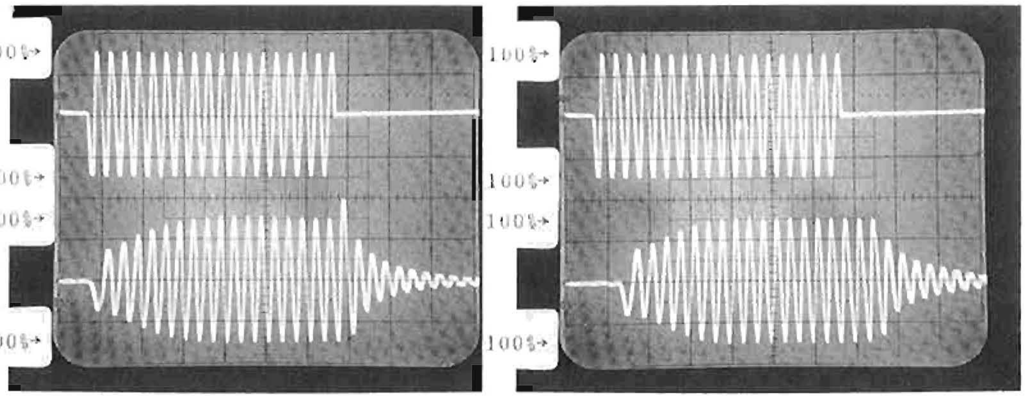
FIGURE 16

DTR FILTER RESPONSE:

SQUAREWAVE AT 100% MODULATION



Tone bursts also demonstrate the capabilities of the DTR filter. Fig. 17 shows a 15.0 kHz tone burst input signal, the output of a conventional filter, and the output of the DTR filter.

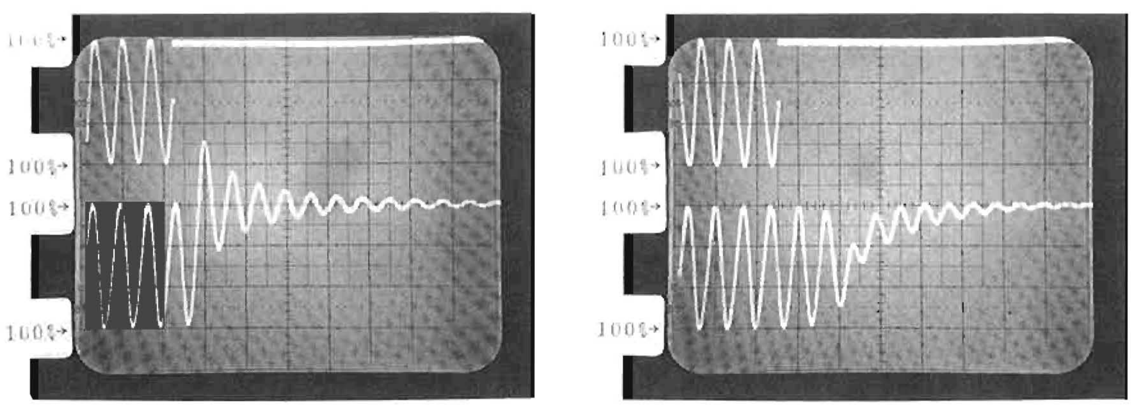


A. CONVENTIONAL FILTER B. DTR FILTER

TOP TRACE: INPUT
BOTTOM TRACE: OUTPUT

TONE BURST TESTS
FIGURE 17

One of the most difficult tests to be contrived for a filter is the signal described under "Examples", Causes and Effects of Overshoot. The signal consists of a sinewave at 100% modulation at a frequency very close to cutoff (15.0 kHz) applied for a time sufficiently long to ensure steady-state filter response, followed immediately by a transition to a step signal. The sinewave signal, being close to cutoff, will accumulate a long time delay with respect to the low-frequency step. At the filter's output, the signals will coincide at the transition. Fig. 18a shows the input and output of a conventional filter. Overshoot is 100%. If the modulation level of the transmitter were to be turned down to accommodate such overshoots without causing overmodulation, a full 6 dB of signal would be lost! The compensated filter response to the same test signal is shown in Fig. 18b. Overshoot is less than 2%. Notice that the DTR filter has no effect on the sinewave.



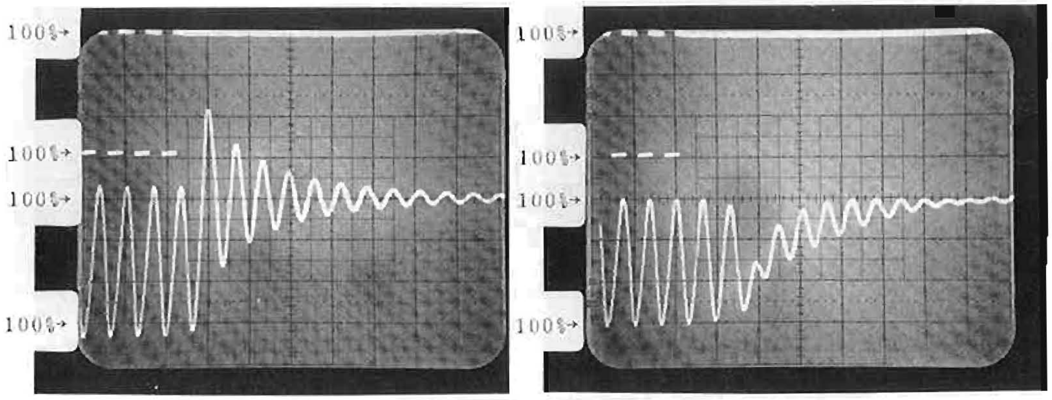
A. CONVENTIONAL FILTER B. DTR FILTER

TOP TRACE: INPUT
BOTTOM TRACE: OUTPUT

SINE/STEP TESTS
FIGURE 18

It is possible to cause even more overshoot by substituting a squarewave for the sinewave part of the test signal of Fig. 18. Fig. 19a shows the resultant squarewave/ step test signal applied to a conventional filter which overshoots 150%, or 8 dB!

The response of the DTR filter to the same test signal is shown in Fig. 19b.



A. CONVENTIONAL FILTER B. DTR FILTER

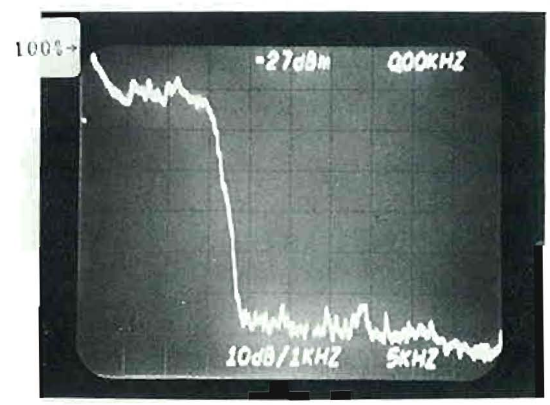
TOP TRACE: INPUT
BOTTOM TRACE: OUTPUT

SQUAREWAVE/STEP TESTS
FIGURE 19

This kind of signal is not uncommon with many types of limiters. Bass note attacks simultaneous with sibilance have been known to cause 100% overshoots when processed by a clipping-type limiter.

DYNAMIC SPECTRUM: Fig. 20 shows the output spectrum of the DTR filter under near "worst case" programming conditions. The input signal had extreme high frequency content, and was processed by several limiters, the last being a hard clipper. The spectrum analyzer, a Tektronix 7L5, is equipped with a microprocessor and memory which was used to produce a miximum-hold display

FIGURE 20
DYNAMIC SPECTRUM



Over a 3 minute period the output spectrum of the DTR filter was continuously monitored; each time that a frequency contained more energy than previously, the display would be updated. Therefore, the display of Fig. 20 does not represent a typical instantaneous spectrum but rather the maximum spectral amplitude reached over a 3 minute period.

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