FM BROADCAST AUDIO PROCESSING AND PEAK MODULATION CONTROL

Achieving Maximum Modulation and Effective Loudness in FM Stereo Broadcasting

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INTRODUCTION

Broadcast audio processing is both an engineering and artistic endeavor. The engineering goal is to make most efficient use of the signal-to-noise ratio and audio bandwidth available from the transmission channel while preventing its overmodulation. The artistic goal is set by the audio processing user. It may be to avoid audibly modifying the original program material at all. Or it may be to create a distinct sonic signature for the broadcast by radically changing the sound of the original. Most broadcasters operate somewhere in between these two extremes, with the main goal of audio processing to increase the perceived loudness within the peak modulation constraints of a transmission channel.

Provided that the transmitted signal meets regulatory requirements for modulation control and RF bandwidth, there is no well-defined right or wrong way to process audio. Like most areas requiring subjective, artistic judgment, processing is highly controversial and likely to provoke thoroughly opinionated arguments amongst its practitioners. Ultimately, the success of a broadcast's audio processing must be judged by its results — if the broadcast gets the desired audience, then the processing must be deemed satisfactory regardless of the opinions of audiophiles, purists, or others who consider processing an unnecessary evil.

FUNDAMENTALS OF AUDIO PROCESSING

Loudness is increased by reducing the peak-to-average ratio of the audio. If peaks are reduced, the average level can be increased within the permitted modulation limits. The effectiveness with which this can be accomplished without introducing objectionable side effects (like clipping distortion) is the single best measure of audio processing effectiveness.

COMPRESSION

Compression reduces dynamic range of program material by reducing the gain of material whose average or rms level exceeds the threshold of compression. AGC amplifiers are compressors. Compression reduces the difference in level between the quiet and loud sounds to make more efficient use of permitted peak level limits, resulting in a subjective increase in the loudness of quiet sounds. It cannot make loud sounds seem louder. Compression reduces dynamic range relatively slowly in a manner similar to "riding the gain."

Density is the extent to which the amplitudes of audio signal peaks are made uniform (at the expense of dynamic range). Programs with large amounts of short-term dynamic range have low density; highly compressed programs have high density.

PEAK LIMITING AND CLIPPING

Peak limiting is an extreme form of compression characterized by a very high compression ratio, fast attack time, and fast release time. In modern audio processing, a peak limiter, by itself, usually limits the peaks of the envelope of the waveform, as opposed to individual instantaneous peaks in the waveform. These are usually controlled by clipping. Limiting and clipping, reduce the short-term peak-to-average ratio of the audio.

The main purpose of limiting is to protect a subsequent channel from overload, as opposed to compression, whose main purpose is to reduce dynamic range of the program.

Peak clipping is a process that instantaneously clips off any part of the waveform that exceeds the threshold of clipping. While a peak clipper can be very effective to increase loudness, it causes audible distortion when over-used. It also increases the bandwidth of the signal by introducing both harmonic and intermodulation distortion into its output signal. Therefore various forms of overshoot compensation are used, which is essentially peak clipping that does not introduce significant out-of-band spectral energy into its output.

Limiting increases audio density. Increasing density can make loud sounds seem louder, but can also result in an unattractive, busier, and flatter denser sound. It is important to be aware of the many negative subjective side effects of excessive density when setting controls that affect the density of the processed sound.

Clipping sharp peaks does not produce any audible side effects when done moderately. Excessive clipping will be perceived as audible distortion.

MULTI-BAND COMPRESSION AND FREQUENCY SELECTIVE LIMITING

These techniques divide the audio spectrum into several frequency bands and compress or limit each band separately (although some inter-band coupling may be used to prevent excessive disparity between the gains of adjacent bands). This is the most powerful and popular contemporary audio processing technique, because, when done correctly, it eliminates spectral gain intermodulation. This occurs in a wideband compressor or limiter when a voice or instrument in one frequency range dominates the spectral energy, thus determining the amount of gain reduction. If other, weaker, elements are also present, their loudness may be audibly and disturbingly modulated by the dominant element. Particularly unpleasant effects may occur if the dominant energy is in the bass region, because the ear is relatively insensitive to bass energy, so the loudness of the midrange is pushed down by the dominant bass energy seemingly inexplicably. The best results are obtained with steep crossover slopes allowing more consistency from various program sources. It can also give the "illusion" of an unprocessed "big" sound.

Another type of frequency-selective limiting uses a program-controlled filter. The filter's cutoff frequency, its depth of shelving, or a combination of these parameters, is varied to dynamically change the frequency response of the transmission channel. Such program-controlled filters are most often used as high-frequency limiters to control potential overload due to pre-emphasis in pre-emphasized systems.

EQUALIZATION

Equalization is changing the spectral balance of an audio signal, and is achieved by use of an equalizer. In broadest terms, an equalizer is any frequency-selective network (filter) placed in the signal path. In audio processing, an equalizer is usually a device that can apply a shelving or peaking curve to the audio.

Equalizers are sometimes used on-line in transmission to create a certain sonic signature for a broadcast. Any of the types above may be used. Commercial audio processors may include equalizers for program coloration, or for correcting the frequency response of subsequent transmission links.

SYSTEM TOPOLOGY

A typical audio processing system consists of a slow AGC followed by a multi-band compressor with moderate attack and release time. Correctly-designed multi-band processors have these time constants optimized for each frequency band; the low-frequency bands have slower time constants than the high-frequency bands. This multi-band compressor usually does most of the work in increasing program density.

The amount of gain reduction determines how much the loudness of soft passages will be increased (and, therefore, how consistent overall loudness will be). The broadband AGC is designed to control average levels, and to compensate for a reasonable amount of operator error. It is not designed to substantially increase the short-term program density (the multi-band compressor and peak limiters do that).

Modern audio processing systems usually add other elements to the basic system described above. For example, it is not unusual to incorporate an equalizer to color the audio for artistic effect. The equalizer is usually found between the slow AGC and the multi-band compressor. The multi-band compressor itself can also be used as an equalizer by adjusting the gains of its various bands.

Peak clippers decrease the peak to average ratio, increasing loudness within the peak modulation constraints of the channel. To decrease clipping-induced distortion, some processors use sophisticated distortion cancelling schemes that remove distortion in frequency bands most likely to be audible to the listener.

Various low-pass filters are often included in the system to limit the bandwidth of the output signal to 15kHz for FM, or to other bandwidths as required by the local regulatory authority. The final low-pass filter in the system is almost always overshoot-compensated to prevent introducing spurious modulation peaks into the output waveform.

PRESERVING THE WAVEFORM FIDELITY OF PROCESSED AUDIO

Highly-processed audio contains many waveform with flat tops that resemble square waves. The waveshape of a square wave is very sensitive to the magnitude and phase response of the transmission channel through which it passes. Deviations from flat magnitude and group delay over the frequency range containing significant program energy will cause the flat tops in the processed program to tilt, increasing peak modulation levels without increasing average levels. This increases the peak-to-average ratio of the wave, reducing the average level (and therefore the loudness) that the channel can accommodate.

Although the audio to the input to an audio processor may be high-pass filtered, the fast peak limiting or clipping processes occurring in the processor are non-linear, producing difference-frequency intermodulation components below the high-pass cutoff frequency of the unprocessed audio. Even if the audio has been high-pass filtered at 30Hz, these intermodulation products may extend down to 5Hz or less. To preserve the shape of the processed wave, these IM products must be passed through the system without being subject to significant magnitude or phase distortion.

Ordinarily, the audio waveform will overshoot less than 1% if the low-frequency cutoff of the transmission system is 0.16Hz or less. This ensures less than 1% tilt of a 50Hz square wave. Although the waveforms of the infrasonic IM products are affected more by this cutoff than power in the audio band, the audio-band power dominates, so the overall waveshape is still adequately preserved when system LF cutoff is 0.16Hz or less. One obvious consequence of this principle is that a system that passes sinewaves flat to 30Hz may severely distort the shape of processed audio unless its LF cutoff is, in fact, far lower.

LOCATION OF SYSTEM COMPONENTS

The best location for the processing system is as close as possible to the transmitter, so that the processing system's output can be connected to the transmitter through a circuit path that introduces the least possible change in the shape of the carefully peak-limited waveform at the processing system's output. Sometimes, it is impractical to locate the processing system at the transmitter, and it must instead be located on the studio side of the link connecting the audio plant to the transmitter. (The studio/transmitter link ["STL"] might be telephone or post lines, analog microwave radio, or various types of digital paths.) This situation is not ideal because artifacts that cannot be controlled by the audio processor can be introduced in the link to the transmitter or by additional peak limiters placed at the transmitter. (Such additional peak limiters are common in countries where the transmitter is operated by a different authority than that providing the broadcast program.).

In this case, the audio output of the processing system should be fed directly to the transmitter through a link which is as flat and phase-linear as possible. Deviation from flatness and phase-linearity will cause spurious modulation peaks because the shape of the peak-limited waveform is changed. Such peaks add nothing to average modulation. Thus the average modulation must be lowered to accommodate those peaks within the carrier deviation limits dictated by government authorities.

This implies that if the transmitter has built-in high-pass or low-pass filters (as some do), these filters must be bypassed to achieve accurate waveform fidelity. A competent modern processing system contains filters that are fully able to protect the transmitter, but which are located in the processing system where they do not degrade control of peak modulation.

The audio received at the transmitter site should be as good quality as possible. Because the audio processor controls peaks, it is not important that the audio link feeding the processing system's input terminals be phase-linear. However, the link should have low noise, flattest possible frequency response from 30-15,000Hz, and low non-linear distortion.

If the audio link between the studio and the transmitter is noisy, the audibility of this noise can be minimized by performing the compression function at the studio site. Compression applied before the audio link improves the signal-to-noise ratio because the average level on the link will be greater. If the STL has limited dynamic range, it may be desirable to compress the signal at the studio end of the STL. To apply such compression, split the processing system, placing the AGC and multi-band compressor sections at the studio, and the peak limiter at the transmitter.

In some countries, the organization originating the program does not have access to the transmitter, which is operated by a separate entity. In this case, all audio processing must be done at the studio, and any damage that occurs later must be tolerated.

If it is possible to obtain a broadband phase-linear link to the transmitter, use the processing system at the studio location to feed the STL. The output of the STL receiver is then fed directly into the transmitter with no intervening processing. A composite STL (ordinarily used for FM stereo baseband) has the requisite characteristics, and can be used to carry the output of the processing system to the transmitter. Because use of a composite STL has so many ramifications, we recommend this only as a means of last resort — installation of the processing system at the transmitter is vastly less complicated.

Where only an audio link is available, feed the audio output of the processing system directly into the link. If possible, request that any transmitter protection limiters be adjusted for minimum possible action — the processing system does most of that work. Transmitter protection limiters should respond only to signals caused by faults or by spurious peaks introduced by imperfections in the link.

REQUIREMENTS FOR STUDIO-TRANSMITTER LINKS ("STL")

If the STL is prior to the audio processor, the STL's signal-to-noise ratio must be sufficient to pass unprocessed audio. If the processor precedes the STL, its frequency response must be flat (±0.1dB) throughout the operating frequency range. The group delay must be essentially constant throughout this range (deviation from linear phase <±10°). Phase correction can be applied to meet the requirement at high frequencies. These requirements are necessary to preserve stereo separation and peak modulation control.

At low frequencies, by far the best way to achieve the specification is to extend the -3dB frequency of the STL to 0.16Hz or lower (as discussed above) and to eliminate any peaking in the infrasonic frequency response prior to the rolloff. Poor AFC-loop design in STL transmitters is all too common, and this is the most likely cause of low-frequency response problems. Such problems can be corrected by applying prior to the STL transmitter equalization that is complementary to existing low-frequency rolloff, such that the overall system frequency response rolls off smoothly at 0.16Hz or below. This solution is much better than clipping the tilt-induced overshoots after the STL receiver because the clipping will introduce non-linear distortion, while the equalizer is distortion-free.

Digital STL systems using lossy bit-rate-reduction schemes will not successfully pass peak-limited audio. The lossy compression adds large amounts of quantization noise to certain frequency bands, as determined by a psychoacoustic analysis of the program material. This added noise will cause the peak level of the peak-limited audio to increase substantially. For example, measurements have shown that APT-X™ at 256kps introduces as much as 3db of overshoot with processed audio and ISO/MPEG Layer II at 384kbps introduces as much as 1dB. Overshoots increase markedly as bit rate is reduced. While these overshoots can be clipped or limited, such processing will cause audible side-effects. On the other hand, if the audio processor is located at the transmitter, its input can be fed without difficulty from an STL using a lossy bit-rate-reduction scheme because it is unnecessary to preserve the waveshape of unprocessed input audio. Fig. 0 shows the peak output level of a digital STL system being fed from a peak controlled signal. Note the lack of peak control. This results in a loudness loss of more than 3dB!

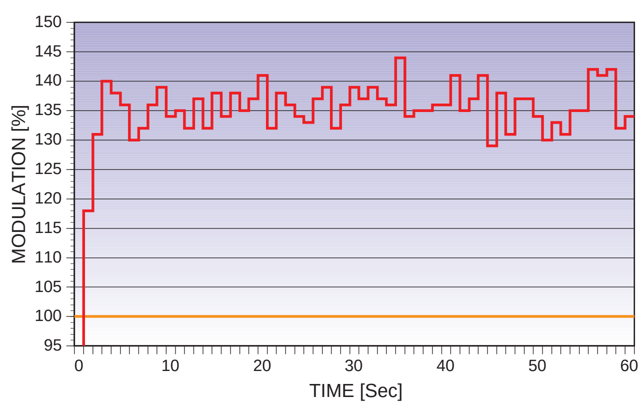

Fig. 0 DIGITAL STL SYSTEM PEAK MODULATION
MOSELEY ASSOCIATES, INC. DSP-6000 APT-X™ 256kbps Stereo

PEAK MODULATION CONTROL

The audio processor must control the peak modulation of the RF carrier to the standards required by the governing authority, such as the FCC in the United States. In FM, the peak deviation of the carrier must be controlled so that the modulation monitor specified by the governing authority does not indicate overmodulation. Because the rules often permit the modulation monitor to ignore very brief overshoots, the instantaneous peak deviation might exceed the peak modulation as indicated on the modulation monitor.

The requirements for peak control and spectrum control tend to conflict, which is why sophisticated non-linear filters are required to achieve highest performance. Applying a peak-controlled signal to a linear filter almost always causes the filter to overshoot and ring because of two mechanisms: spectrum truncation and time dispersion. One can build a square wave by summing its Fourier components together with correct amplitude and phase. Analysis shows that the fundamental of the square wave is approximately 2.1dB higher than the amplitude of the square wave itself. As each harmonic is added in turn to the fundamental, a given harmonic's phase is such that the peak amplitude of the resulting waveform decreases by the largest possible amount. Simultaneously, the rms value increases because of the addition of the power in each harmonic. This is the fundamental theoretical reason why simple clipping is such a powerful tool for improving the peak-to-average ratio of broadcast audio: clipping adds to the audio waveform spectral components whose phase and amplitude are precisely correct to minimize the waveform's peak level while simultaneously increasing the power in the waveform.

If a square wave (or clipped waveform) is applied to a low-pass filter with constant time delay at all frequencies, the higher harmonics that reduce the peak level will be removed, increasing the peak level and with it the peak-to-average ratio. Thus even a perfectly phase-linear low-pass filter will cause overshoot. There is no sharp-cutoff linear low-pass filter that is overshoot-free: overshoot-free spectral control to FCC or CCIR standards must be achieved with filters that are embedded within the processing, such that the non-linear peak-controlling elements in the processor can also control the overshoot.

If the sharp-cutoff filter is now allowed to be minimum-phase, it will exhibit a sharp peak in group delay around its cutoff frequency. Because the filter is no longer phase-linear, it will not only remove the higher harmonics required to minimize peak levels, but will also change the time relationship between the lower harmonics and the fundamental. They become delayed by different amounts of time, causing the shape of the waveform to change. This time dispersion will therefore further increase the peak level.

When a square wave is applied to a linear-phase filter, overshoot and ringing will appear symmetrically on the leading and trailing edge of the waveform. If the filter is minimum-phase, the overshoot will appear on the leading edge and will be about twice as large. In the first case, the "overshoot and ringing" are in fact caused by spectrum truncation which eliminates harmonics necessary to minimize the peak level of the wave at all times; in the second case, the overshoot and ringing are caused by spectrum truncation and by distortion of the time relationship between the remaining Fourier components in the wave.

OVERSHOOTS IN COMPOSITE STL SYSTEMS AND FM EXCITERS

It is well known that processed, peak-controlled program material causes most composite STL systems and FM exciters to produce overshoot in FM composite baseband signals. The heavier the processing the more they overshoot. This overshoot occurs even in properly band-limited systems that utilize STL paths without multipath. Accordingly, loudness is compromised because average modulation must be reduced to prevent illegal peak overmodulation caused by this overshoot. Previous attempts to eliminate this overshoot have degraded system performance.

It is essential that the measuring equipment have transient accuracy at least as good as the equipment being measured. Many popular modulation monitors introduce false tilt and overshoot into their readings. To avoid such inaccuracy, we used a Belar Electronics Wizard™ FM Modulation Analyzer -- a known-accurate instrument -- to plot peak modulation versus time on the output of an aggressive audio processing system and several contemporary STL systems and FM exciters. The results reveal the extent of the problem. The same program material and time segment was used for all plots. Fig. 1 shows the audio processor output peak modulation directly. Figs. 2 and 3 show the peak modulation of STL System 2 and FM Exciter 2 respectively. Note that both the STL System and FM Exciter suffer from overshoot causing over-modulation. To prevent the resulting over-modulation, the modulation level must be reduced over 13%, losing almost 1.5dB of loudness!

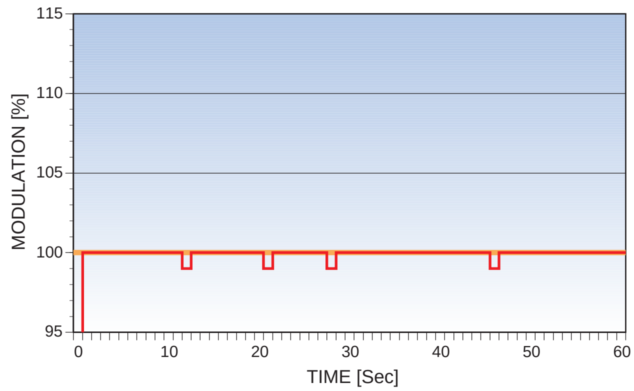

Fig. 1 AUDIO PROCESSOR OUTPUT PEAK MODULATION

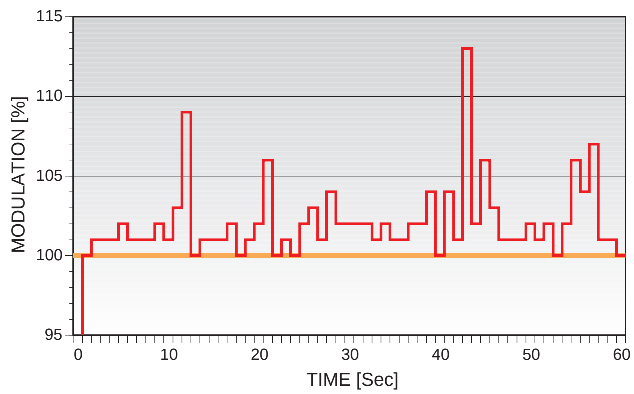

Fig. 2 STL SYSTEM 2 PEAK MODULATION-Before Modification
MOSELEY ASSOCIATES, INC. PCL-6010/C

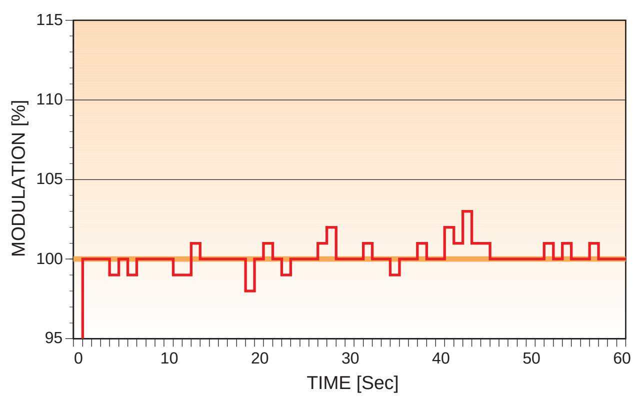

Fig. 3 FM EXCITER 2 PEAK MODULATION-Before Modification
CONTINENTAL ELECTRONICS MFG. CO., INC. 802A

THE FM STEREO SYSTEM

The world-standard FM stereo "pilot-tone" system encodes the sum of the channels (L+R) in the frequency range of 30-15,000Hz in the stereo baseband — the "stereo main channel." It encodes the difference between the channels (L-R) on a double-sideband suppressed-carrier sub-channel centered at 38kHz, and occupying 23kHz to 53kHz in the stereo baseband — the "stereo sub-channel." A pilot tone at 19kHz tells the receiver that a stereophonic transmission is being received, and provides a phase and frequency reference to permit the receiver to regenerate the 38kHz sub-carrier to use in its stereo demodulator. Any energy that appears in the frequency range from 30 to 19,000Hz caused by a signal in the stereo sub-channel is termed "sub-channel-to-main channel crosstalk." Any energy that appears in the frequency range from 19 to 57kHz caused by a signal in the stereo main channel is termed "main-channel-to-sub-channel crosstalk."

When the stereo encoder is driven by a pure right-only or left-only signal, "stereophonic separation" can be measured at the stereo decoder as the ratio between the desired and undesired signal levels, where the "desired" signal is the signal appearing in the decoder output channel corresponding to the channel driven at the encoder, and the "undesired" signal is the signal caused by the desired signal that appears at the remaining output.

Ideally, crosstalk is non-existent and stereophonic separation is infinite. In practice, both linear and non-linear errors cause these characteristics to deteriorate.

In the linear domain, separation and crosstalk are mathematically orthogonal. Phase and frequency errors that cause one to deteriorate will not affect the other. For example, phase or frequency errors in the composite signal channel will cause separation to deteriorate, but cannot affect crosstalk, since the stereo main and sub-channels are already separated in frequency and changes in phase or amplitude response in the composite channel cannot affect this frequency separation. Conversely, mismatches between the linear response of the left and right signal paths prior to the stereo encoder will cause crosstalk, but cannot affect separation.

Non-linearities in the composite channel can cause both separation and crosstalk to deteriorate because such errors cause harmonic and intermodulation distortion that introduce new frequencies into the baseband. These new frequencies are likely to inject power into a part of the baseband spectrum that will be decoded by the stereo decoder in spatial locations different than the locations of the original sound sources. Further, these new frequencies are perceived by the ear not as changes in spatial localization, but as highly offensive distortion.

This is somewhat analogous to "aliasing distortion" in a sample-data audio system. In such a system, any input frequencies greater than one-half of the sampling frequency (the "Nyquist frequency") are encoded with the wrong frequency: they "fold around" the Nyquist frequency and appear at the decoder as frequencies unrelated to the program material that produced them. The ear perceives this "aliasing" as offensive distortion.

PREVIOUS NON-LINEAR SOLUTION

The most common technique for reducing FM composite baseband signal overshoot has been composite baseband clipping. Composite clipping has been disparaged because it causes signal degradation and because early implementations that clipped the pilot could violate the FCC rules. No implementation prevents dynamic signal degradation.

The composite baseband clipper is a non-linear device. Thus, it generates distortion and aliasing products that contaminate the composite baseband signal, degrading dynamic stereo separation and causing audible dynamic distortion. It also produces distortion products in the sub-carrier region, reducing or destroying the sub-carrier's market value and reducing revenue potential.

While it is possible to lowpass-filter the clipped baseband (thus protecting the sub-carriers), such filtering does nothing to eliminate intermodulation distortion in the stereo baseband region below 57kHz, and will also tend to increase peak modulation, partially negating the peak control provided by the composite clipper. Such filters do not protect the 19kHz pilot tone from interference caused by the clipper-induced distortion, which can cause problems in receivers' stereo decoders.

If the FM exciter is the source of overshoot (as opposed to the STL), or if the clipper precedes the STL, then the composite baseband clipper cannot control overshoot. Instead, it can actually increase overshoot because the clipping process produces increased amounts of infrasonic intermodulation distortion products.

Some have argued that composite baseband clipping increases loudness more than audio clipping in the left and right channels. But this loudness increase is accompanied by degraded dynamic stereo separation and crosstalk. To preserve dynamic stereo performance, the spectra of the stereo main channel and sub-channel must be completely isolated: the main channel must not have any energy above 19kHz, and the sub-channel must not have any energy below 19kHz.

One consequence of such frequency separation is this: in a system that achieves high dynamic separation and low crosstalk, it must be impossible for the system's final filter/limiter to reproduce any approximation to a square wave if the square wave's fundamental frequency is higher than one-third the cut-off frequency of the low-pass filter prior to stereo encoding (typically 15kHz). This is because the third harmonic of the square wave is three times the frequency of the fundamental, so the low-pass filter removes it (and all higher harmonics too); any square wave above 5kHz will emerge from the receiver as a sine wave. Because they generate spurious harmonic and intermodulation products, composite baseband clippers do not meet this criterion and thus compromise dynamic stereo performance.

Composite clipping has one potential advantage. Conventional wisdom holds that the peak modulation of the composite baseband is the greater of the left or right channel levels, plus the pilot. However, this is only an approximation because the pilot is correlated in phase with the 38kHz suppressed sub-carrier. This causes the total composite modulation to decrease slightly when the left and right channels are unequal in level. Assuming 10% pilot injection and holding the left channel at 100% modulation, decreasing the right channel from 100% to 0% modulation will cause the composite modulation to decrease by 2.8%. Perfectly accurate peak limiting in the audio domain, prior to stereo encoding, can only control the composite modulation to an accuracy of -2.8%/+0%. Only a process that is aware of the total peak composite modulation (including the pilot and any subcarriers) can control composite modulation accurately. Since composite baseband clipping controls the peak deviation of the composite signal precisely (assuming the pilot is also clipped), it can theoretically be louder than peak limiting in the audio domain. But the "advantage" is an imperceptible 0.24dB at best! Composite baseband clippers that do not clip the pilot (which are the only clippers legal for use in the U.S.) do not eliminate the interleave error, and therefore produce no loudness advantage at all!

Left channel modulation 100%.
Right channel modulation from 0% to 100%.
10% pilot injection.
Peak deviation of composite shown with 100% normalized to L=R with pilot present.

Right % Modulation	Composite % Modulation
100.0%	100.00%
99.5%	99.79%
99.0%	99.59%
98.0%	99.26%
97.0%	98.99%
96.0%	98.77%
95.0%	98.59%
94.0%	98.45%
93.0%	98.33%
92.0%	98.14%
90.0%	98.06%
80.0%	97.65%
70.0%	97.48%
60.0%	97.39%
50.0%	97.33%
40.0%	97.29%
30.0%	97.26%
20.0%	97.24%
10.0%	97.22%
0.0%	97.20%

Fig. 4 INTERLEAVING ERROR IN THE FM STEREO SYSTEM

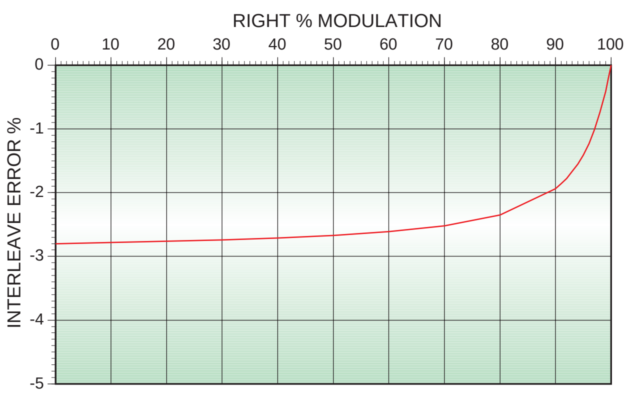

Fig. 5 INTERLEAVING ERROR IN THE FM STEREO SYSTEM

Fig. 4 shows the exact interleaving failure due to pilot summation in the FM stereo system. Fig. 5 is a plot of the interleaving error in % of modulation versus right channel modulation, with the left channel modulation held constant at 100%.

In these digital times, bad audio quality has become unacceptable to formerly unsophisticated consumers. It is absurd that composite baseband clippers are being used to degrade system performance below that of some of the least expensive receivers! Composite clipping is a very easy, unsophisticated method of increasing apparent loudness of the broadcast signal, but it compromises quality in a way that is unacceptable to any broadcaster trying to compete with CD or the newer recordable digital media.

OVERSHOOT SOURCE

Extensive computer modeling and analysis of several current-generation composite STL systems and FM exciters has revealed that the overshoot problem is not in the high frequency domain (as previously assumed), but instead at infrasonic frequencies. All of the systems modeled have infrasonic peaks in their frequency response, and/or have insufficient low-frequency response to accurately reproduce a processed composite baseband signal. Some of the systems even suffer from marked non-linearity, having different frequency response at different modulation levels at very low frequencies, aggravating the problem. This poor low-frequency transient response can be caused by incorrectly designed AFC loops and/or deficient low-frequency response of the composite baseband amplifiers. Figs. 6, 7, and 8 show the system response of three popular composite STL systems. Figs. 9, 10, and 11 show the system response of three popular FM exciters. Note the radical differences in low-frequency response, and how the response of a given system depends on its RF operating frequency. It's no surprise or myth that each of these modulators has its own sonic signature as well. (This might explain why legend has it that certain FM channels sound better than others!) Note also that of the systems modeled here, the newer systems do not have better performance. Contrary to popular belief, science does not always march forward!

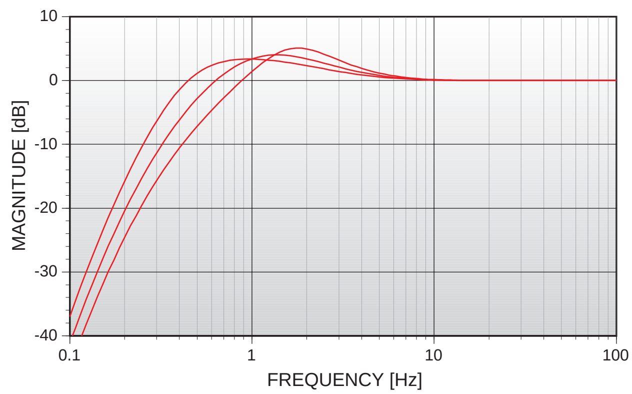

Fig. 6 STL SYSTEM 1 INFRASONIC FREQUENCY RESPONSE
MOSELEY ASSOCIATES, INC. PCL-606/C

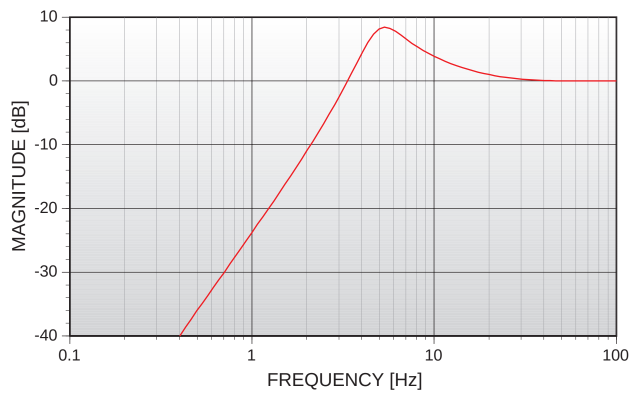

Fig. 7 STL SYSTEM 2 INFRASONIC FREQUENCY RESPONSE
MOSELEY ASSOCIATES, INC. PCL-6010/C

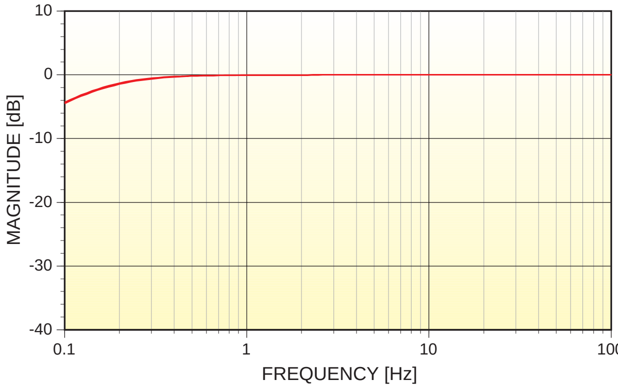

Fig. 8a STL SYSTEM 3 INFRASONIC FREQUENCY RESPONSE
TFT, INC. 8300 - Original Design

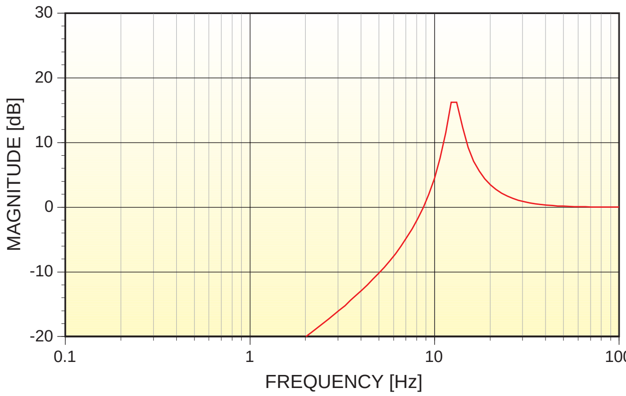

Fig. 8b STL SYSTEM 3 INFRASONIC FREQUENCY RESPONSE
TFT, INC. 8300 - Updated Design

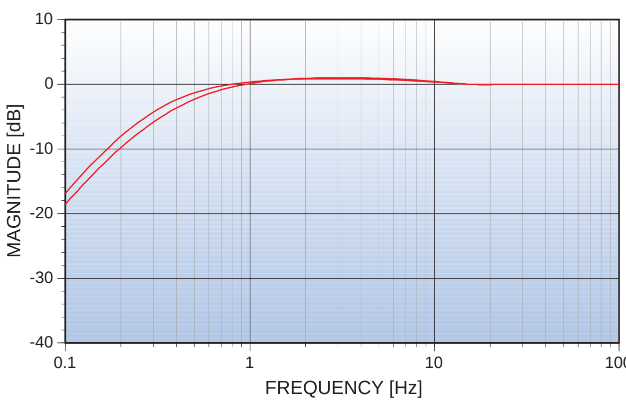

Fig. 9 FM EXCITER 1 INFRASONIC FREQUENCY RESPONSE
BROADCAST ELECTRONICS, INC. FX-50

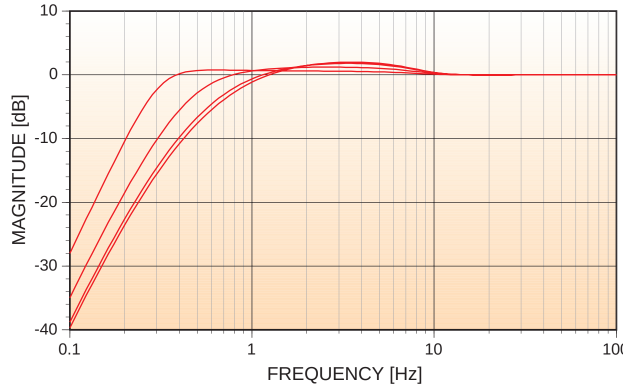

Fig. 10 FM EXCITER 2 INFRASONIC FREQUENCY RESPONSE
CONTINENTAL ELECTRONICS MFG., CO., INC. 802A

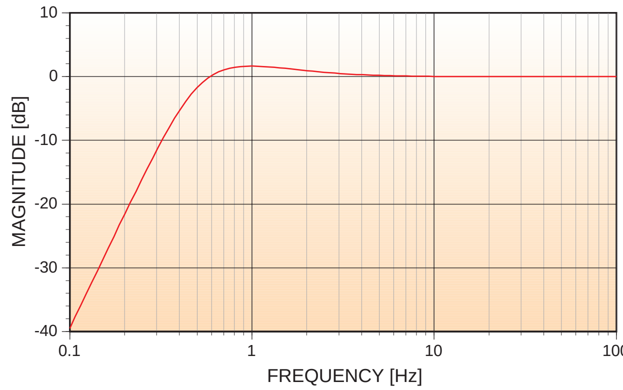

Fig. 11 FM EXCITER 3 INFRASONIC FREQUENCY RESPONSE
CONTINENTAL ELECTRONICS MFG., CO., INC. 802B

Another important consideration is the overall system performance of the various components in the composite signal path. When these components are cascaded, the system response deteriorates. The amount of degradation depends on input and output impedance interactions in the composite baseband amplifiers and how many systems are cascaded, so the audible performance of a system is not always simply the sum of the performance of its parts. Fig. 12 shows the total system response of STL 2 and Exciter 3. Note how the response has further degraded. (Once again, this might explain why legend contends that certain combinations of equipment sound better than others.)

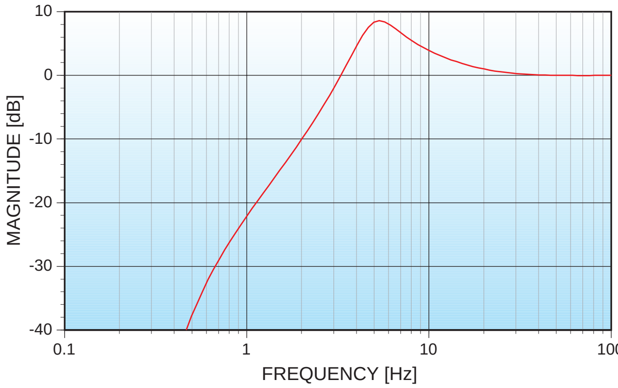

Fig. 12 STL SYSTEM 2 AND EXCITER 3 TOTAL SYSTEM RESPONSE

PEAK-WEIGHTED MONITORING

Recently, on the basis of a controversial interpretation of the FCC Rules & Regulations, devices have been introduced that change the way that broadcasters measure modulation. By delaying the response of the peak indicator, such a modulation monitor ignores peaks of less than 1 millisecond duration (assumed to be overshoots), and does not indicate over-modulation under these conditions. Although this technique of modulation measurement is under close investigation by the industry (for reasons not relevant here), it does not ignore the type of composite baseband overshoot described above. Instead, provided the peak indicator circuitry has been designed correctly, the modulation monitor accurately measures this overshoot because its duration is far longer than the delay of the peak indicator. So the composite path must still be free from infrasonic overshoot to preserve peak control providing maximum loudness with minimum distortion. Fig. 13 shows STL system 2 peak modulation, peak-weighted with 9 cycles at 10kHz. Note there is little difference between Fig. 13 and Fig. 2, which once again, is the exact same program material and segment.

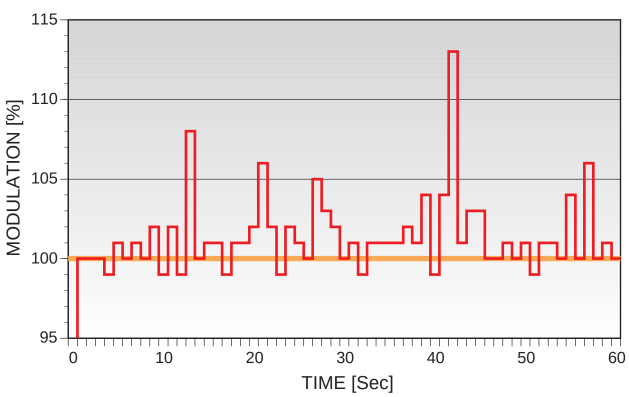

Fig. 13 STL SYSTEM 2 PEAK MODULATION
Peak Weighted, 9 cycles/10kHz

RESPONSE REQUIREMENTS

If we model the system (to a first-order approximation), as a high-pass filter with a single dominant pole, we can use the equation in Fig. 14 to compute the percentage of overshoot when a square-wave of frequency F (Hz) is applied to the system input. This overshoot will occur regardless of whether the low-frequency rolloff is in the composite channel, or in the left and right audio channels after peak limiting. In the former case, the rolloff can also compromise low-frequency separation. To achieve less than 1% overshoot with a 50Hz square-wave (a reasonable criterion for good peak control), the dominant pole must be located at 0.16Hz or lower with no peaking! Good 10Hz square-wave response does not predict low overshoot because a peak in the region below 10Hz can phase equalize the 10Hz fundamental while simultaneously distorting the phase and amplitude of the components below 10Hz. If more than one low-frequency roll-off element is cascaded in the composite path, each element's cut-off frequency must be substantially below 0.16Hz. Fig. 15 shows the minimum required low-frequency response.

$$\text{Overshoot \%} = 100\,[1 - exp(P/2F)\,]$$

where P = - (1/RC)
where F = Frequency in Hz

Fig. 14 OVERSHOOT EXPRESSION

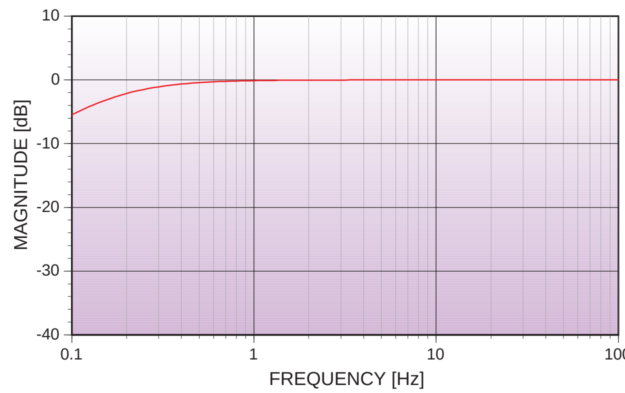

Fig. 15 MINIMUM LOW-FREQUENCY RESPONSE REQUIREMENT

LINEAR SOLUTION

Since poor infrasonic frequency response causes composite baseband overshoot, it can be eliminated by the proper design of the AFC circuitry and composite baseband amplifiers. It can also be corrected (although not as accurately) by an infrasonic equalizer that flattens the very low-frequency response of the composite signal path. Both methods are linear solutions to linear problems. The first method is preferred. Unlike composite baseband clippers, linear correction produces no distortion or aliasing products, ensuring maximum loudness without side-effects.

Highly optimized modifications to most current generation FM exciters and STL systems permit these units to pass the most highly-processed composite baseband signal while adding less than 1% overshoot — often an improvement of 10:1 or more, resulting in a 1dB loudness advantage with almost any audio processing. This can be a very cost-effective solution, offering better sonic performance than digital alternatives because there is no data compression to potentially cause distortion. These modifications offer better performance than all of the latest unmodified analog equipment known to us. Fig. 16 shows the optimized response of the STL System 2, shown in Fig. 7. Fig. 17 shows the optimized response of the FM Exciter 2, shown in Fig. 10. Figs. 18 and 19 show the results of the optimization of STL System 2 and FM Exciter 2 respectively by indicating tight peak modulation control. Note the before optimization peak control in Figs. 2 and 3.

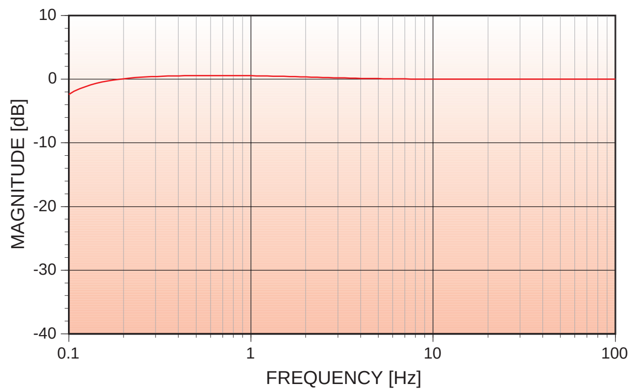

Fig. 16 STL SYSTEM 2 OPTIMIZED RESPONSE
MOSELEY ASSOCIATES, INC. PCL-6010/C

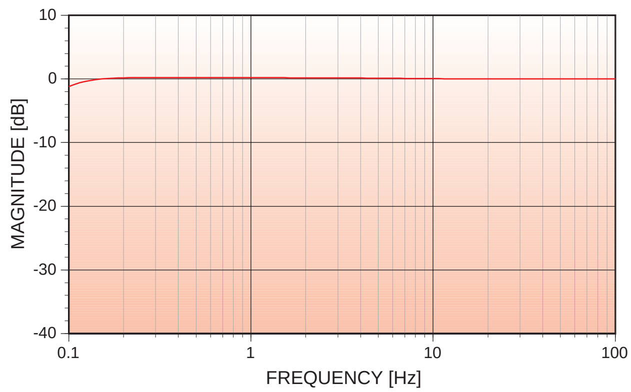

Fig. 17 FM EXCITER 2 OPTIMIZED RESPONSE
CONTINENTAL ELECTRONICS MFG., CO., INC. 802B

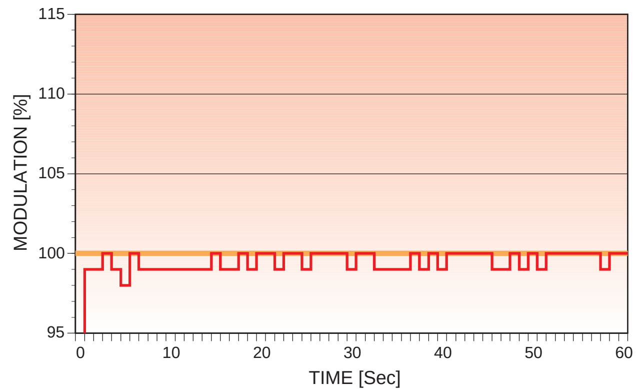

Fig. 18 STL SYSTEM 2 PEAK MODULATION - After Modification
MOSELEY ASSOCIATES, INC. PCL-6010/C

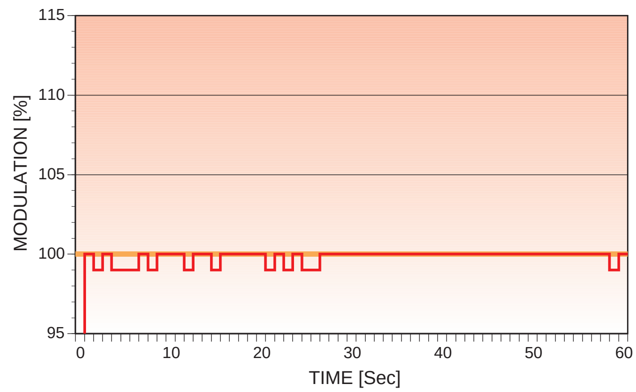

Fig. 19 FM EXCITER 2 PEAK MODULATION - After Modification
CONTINENTAL ELECTRONICS MFG., CO., INC. 802B

Getting a signal as loud as possible under existing government regulations requires attention to every link in the audio chain. One or more weak links can noticeably decrease the loudness and competitiveness of the signal. In broadcast audio, as in most other endeavors, attention to detail separates the winners from the losers.