PEAK MODULATION CONTROL IN DIGITAL TRANSMISSION SYSTEMS
Digital Overshoot: Causes, Prevention, and Cures

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ABSTRACT

What are the causes of overshoot, and how is overshoot different in digital systems? This tutorial answers these questions and will help you avoid the pitfalls in building a digital audio processing and transmission system for FM broadcasting.

INTRODUCTION

Overshoot means at least two different things. In a device designed to control instantaneous levels (like a peak limiter) overshoot occurs when the instantaneous output amplitude exceeds the nominal threshold of peak limiting. In a linear system designed to pass a peak-controlled signal, overshoot occurs when the instantaneous system output amplitude exceeds the instantaneous input amplitude. This assumes that the system has unity gain when measured with a sinewave. Overshoots cause problems because of either (a) overmodulation, or (b) loss of loudness when the modulation level is turned down enough to avoid overmodulation.

Figure 1 shows an overshoot. Most of the signal peaks in this figure are accurately limited to 100%, but one isolated peak extends to about 125% modulation.

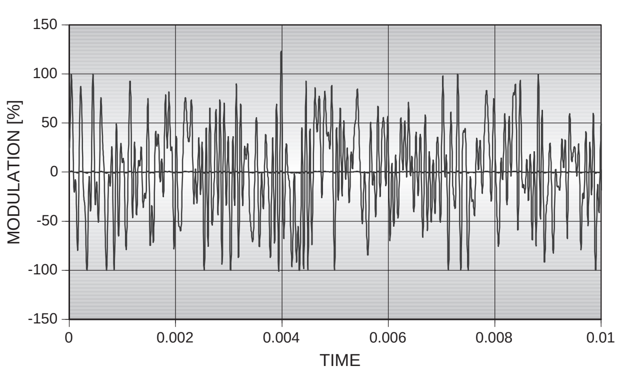

Fig. 1 SINGLE OVERSHOOT WITH COMPLEX INPUT

Overmodulation can occur in both analog and digital systems. Just because a system is digital does not make it immune to overshoot. Like analog systems, a digital audio system also contains subsystems that can cause overshoot, like clippers and filters.

DIAGNOSING OVERSHOOT

How does one find overshoot in a system, and how can we tell the difference between overshoot and sloppy peak control?

Overshoot detection methods are the same for both analog and digital systems, since both technologies ultimately produce the same kind of output: an analog FM signal.

One way to detect the problem is to use a good modulation monitor and a variable persistence oscilloscope. Observe the composite output of the modulation monitor with the oscilloscope. What you should see are frequent peaks reaching 100% modulation and nothing beyond. In a system with overshoot, you will see occasional, low energy peaks extending typically 1-3 dB above the 100% modulation level. The variable persistence feature of the oscilloscope will help you to see these low energy peaks.

If you do not have a variable persistence oscilloscope, you can still detect overshoot by using the peak flasher on the modulation monitor. In a system with no overshoot, the threshold between no flashes and continuous flashes should be abrupt. For example, setting the flasher at 100% should result in almost continuous flashes, and setting it at 103-105% should result in no flashes. However, if the range between continuous flashes and no flashes is 10% or more, then you have an overshoot problem (which could, conceivably, be in the modulation monitor).

OVERSHOOT MECHANISMS

The requirements for peak control and spectrum control tend to conflict in audio processors, which is why sophisticated non-linear filters are required to achieve highest performance. Applying a peak-controlled signal to a linear filter usually causes the filter to overshoot and ring because of two mechanisms: spectrum truncation and time dispersion.

Peak limiters tend to produce waveforms with flat tops, like squarewaves. One can build a square wave by summing its Fourier components together with correct amplitude and phase. Figure 2 shows the first three Fourier components of a squarewave. 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. Simultaneously, the R.M.S. 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.

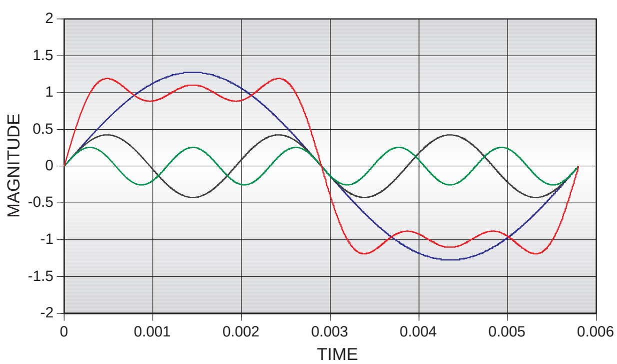

Fig. 2 FOURIER EXPANSION of a SQUAREWAVE

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 ITU-R 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. (A minimum phase filter is one that cannot have any *less* phase shift for a given amplitude response.) Figure 3 shows the amplitude and group delay functions of a minimum phase, elliptic function filter. 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. Figure 4 shows the squarewave response of this filter.

When a square wave is applied to a linear-phase filter, 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 trailing edge and will be about twice as large. In the first case, the overshoot and ringing are in fact caused by spectrum truncation that 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. Figure 5 compares the squarewave responses of a minimum phase filter and a linear phase filter.

As a worst-case scenario, if we shift the phase of each harmonic in the square wave by 90 degrees (perform a Hilbert Transform), we find that the peak amplitude of the square wave's edges actually becomes, theoretically, *infinite.* In the real world of band-limited audio waveforms this extreme overshoot cannot occur. Nevertheless, it is perfectly possible to have phase shifts introduce up to 6dB of overshoot to peak-limited audio. This is equivalent to 200% modulation!

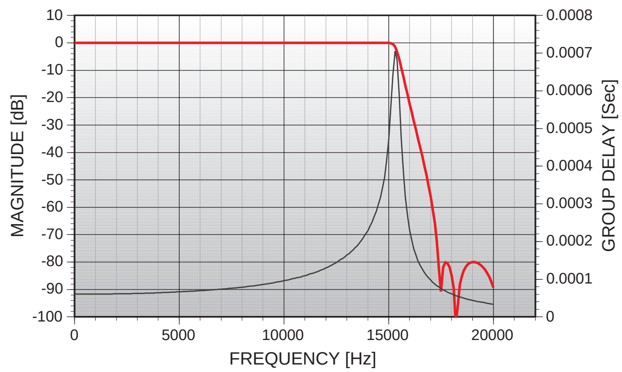

Fig. 3 MINIMUM PHASE LPF FILTER RESPONSE

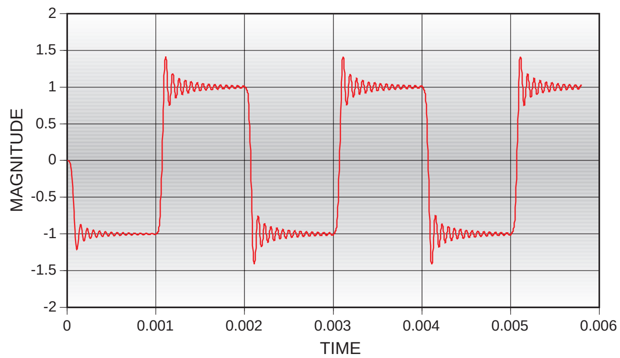

Fig. 4 MINIMUM PHASE LPF SQUAREWAVE RESPONSE

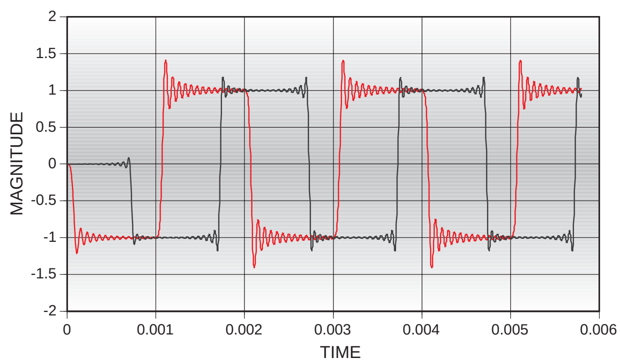

Fig. 5 MINIMUM PHASE & LINEAR PHASE SQUAREWAVE RESPONSES

SAMPLING THEORY

Analog signals consist of a continuous waveform, with a voltage value for every instant in time. Sampled-data or digital signals are only observed at certain instants in time. Not all of the continuous signal values are encoded; only those signal values that occur at the sampling instants are part of the digital signal.

Overshoots and signal peaks may occur at times other than the sampling instants. In other words, the digital samples may straddle the continuous waveform's peaks – the analog peak can "fall between the samples." Consequently, the continuous signal's amplitude may exceed the amplitude of the digital samples. Finding overshoots and determining their amplitudes is therefore a significant problem in digital systems.

Figure 6 shows a random, band-limited signal and the samples that represent it. Notice that while some of the samples fall on the signal peaks, most do not. Figure 7 shows an expanded view of part of the same signal in Figure 6.

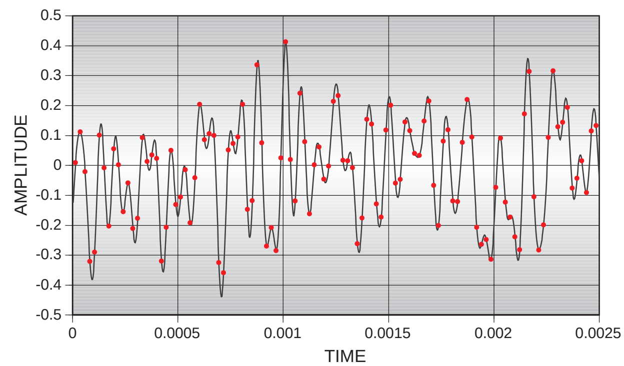

Fig. 6 SAMPLING MISSES THE PEAKS

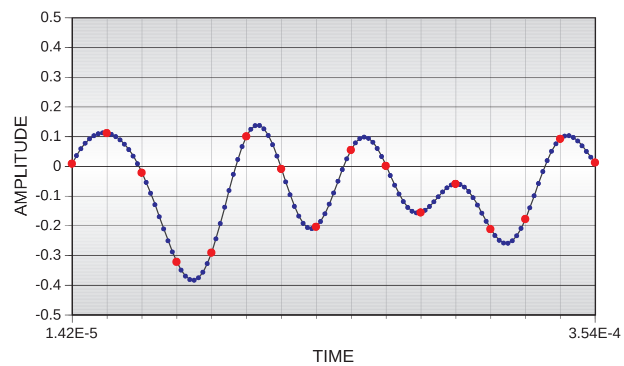

Fig. 7 SAMPLING MISSES THE PEAKS – Expanded View

When a digital signal is converted back to analog, the continuous signal must be "filled in" between the samples. This is called "reconstruction." A lowpass filter called a "reconstruction filter" accomplishes this. A theoretically ideal reconstruction filter has a gain of unity to the Nyquist frequency, a gain of zero above it, and constant group delay at all frequencies. The reconstruction filter is an *essential and crucial* part of a sample-data system. Reconstruction is not just a matter of "connecting the dots" between the samples. Instead, the reconstruction filter does a sophisticated interpolation between the samples. Ideally, it uses a knowledge of all samples, past and future, to do this interpolation.

A perfect analog reconstruction (using a "brick wall" filter) will produce one and only one waveform, which passes through all of the sampled points. In other words, the ideal reconstructed waveform is unique and predictable. It will be *exactly the same* as the original analog waveform at the input to the system, provided that this waveform was filtered with an "anti-aliasing" filter so that it has no energy above the Nyquist frequency.

Such a "brick wall" filter is physically unrealizable. It would be infinitely large and have an infinite time delay. So all physical reconstruction filters are approximations. It is easy to make them have constant time delay (be linear-phase) by using "oversampling" techniques where most of the filtering is done in the digital domain before the D/A converter. The compromise occurs in their magnitude response. Such filters have small frequency response ripples in their passband, which does not extend fully to the Nyquist frequency. Their gain in the stopband is not zero. To reduce cost many commercial realizations do not start the stopband at the Nyquist frequency, but slightly above. This can cause further errors.

Practical reconstruction filters are only flat to 80-95% of the Nyquist frequency, and they vary from system to system. While they do not exactly reconstruct the original waveform, typical ones used in commercial practice come very close.

Figure 8 shows an ideal "brick wall" type reconstruction filter cutting off at 20 kHz, along with a practical reconstruction filter cutting off at a somewhat lower frequency.

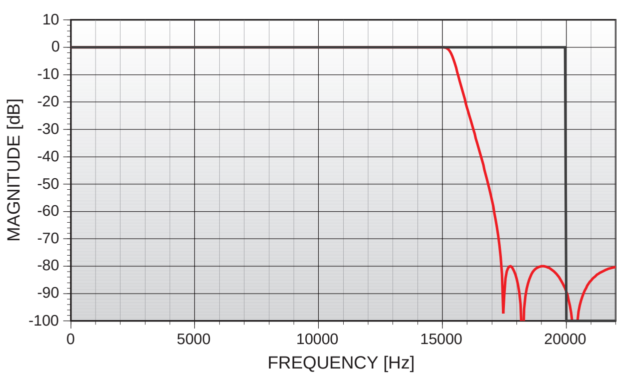

Fig. 8 IDEAL AND PRACTICAL RECONSTRUCTION FILTERS

THE REAL WORLD

As an example, let us consider a digital audio system with a sampling rate of 44.1 kHz. Depending on the architecture of this 44.1 kHz sampled system, it might have a frequency response of 15 to 20 kHz, or some other value less than 22.05 kHz. For the sake of argument, let us assume that our system is linear phase and has a bandwidth of 20 kHz.

If we apply an input signal with no energy above the 20 kHz bandwidth of our 44.1 kHz example system, then the system will transmit all frequencies with no amplitude change and no relative phase change. In other words, there will be no alteration of the input signal. It follows that there will be no overshoot.

The upper trace in Figure 9 shows the bandwidth of our hypothetical audio system, and the lower trace shows the applied input signal's bandwidth. Since the input signal bandwidth is less than the system bandwidth, there is no overshoot.

However, let us consider what happens if we apply a 44.1 kHz sampled signal with a 21 kHz bandwidth to our 20 kHz bandwidth system. Our system will attenuate the energy that appears between 20 and 21 kHz. This will change the peak amplitude of our signal, and may cause overshoot.

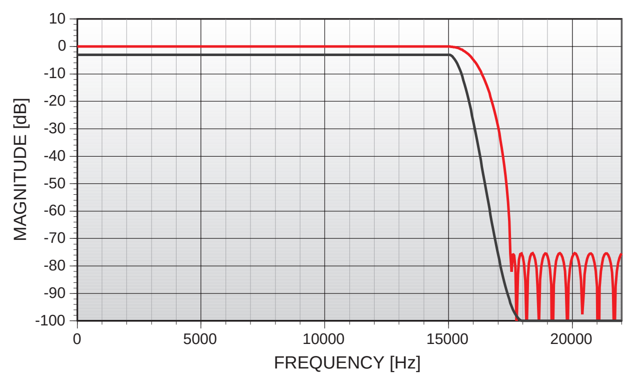

Fig. 9 INPUT BANDWIDTH LESS THAN SYSTEM BANDWIDTH: NO DIGITAL OVERSHOOT

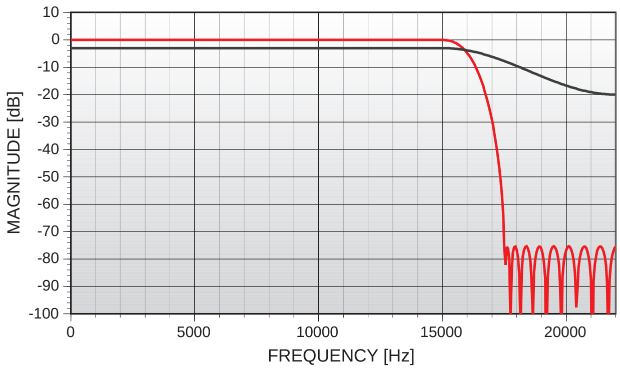

Fig. 10 INPUT BANDWIDTH GREATER THAN SYSTEM BANDWIDTH: CAUSES DIGITAL OVERSHOOT

Figure 10 depicts this condition. Notice that there is significant energy that is below Nyquist, but beyond the system bandwidth. Truncation of this energy will create overshoot.

The lower trace in Figure 11 shows the peak amplitude of an accurately limited signal at 100% modulation. The upper trace shows the peak modulation of a 15 kHz band limited system when driven with a signal of wider bandwidth. Spectrum truncation causes 111% overmodulation in this example.

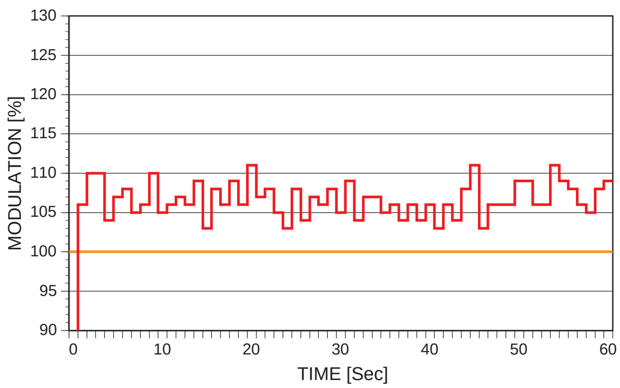

Fig. 11 PEAK MODULATION 15 kHz SYSTEM w/15.3 kHz INPUT

Spectral truncation can easily occur in a digital sample rate converter (SRC) when converting from a higher to a lower sample rate – for example, 44.1kHz to 32kHz. To avoid aliasing, the sample rate converter must apply a lowpass filter to the signal before conversion to the lower rate. If this occurs, the filter will introduce overshoots if it truncates signal energy. However, if the signal at the higher rate has previously been bandlimited so that its energy is contained entirely within the passband of the SRC's lowpass filter, then the SRC will theoretically introduce no overshoot. (In practice, it will introduce a very negligible amount because of passband ripple in the frequency response of the SRC's lowpass filter.) For example, converting from 32kHz to 48kHz sample rate and then back to 32kHz will introduce almost no overshoot. This is because the original 32kHz signal was already appropriately bandlimited, so the 48kHz to 32kHz conversion did not have to remove any signal energy.

Some digital FM exciters use 38 kHz as an internal sampling rate for obvious reasons – the sampling rate is equal to the stereo subcarrier frequency. An FM stereo generator therefore has a Nyquist frequency of 19 kHz – the absolute upper limit of frequency response. However, the practical limitations of pilot protection and real-world SRC lowpass filtering dictate that the frequency response will be limited to something between 15 and 17.5 kHz. So what happens if we apply our hypothetical 44.1 kHz sampled, 20 kHz bandwidth input signal to an FM exciter? We get overshoot because of spectrum truncation. But if the bandwidth of our input signal were already limited to 15 kHz, then there would be no overshoot.

If a filter in the system does not have linear phase, then this can cause time dispersion. The phases of the various Fourier components of the waveform become shifted with respect to each other so they no longer line up in time. This will cause changes in peak level. If the waveform in question is highly processed, the changes will almost certainly be in the direction of increasing the peak level. This effect aggravates the effect of the spectrum truncation discussed above. Non-linear phase filters therefore cause even more overshoot than their linear phase brethren.

Another way to create problems is to perform simple digital clipping, where every sample exceeding a fixed positive or negative threshold is replaced by a sample at that threshold. Just as is the case in the analog world, clipping produces significant amounts of harmonic energy. If we start with a 20 kHz bandwidth signal in our 44.1 kHz sampled system and clip, we can easily produce harmonics to 60 kHz or more. Clipper-induced harmonics above 22.05 kHz will alias to lower frequencies. But most importantly, digital clipping can produce energy that goes all the way to the 22.05 kHz Nyquist frequency (with some of that energy introduced by aliasing).

Many downstream DSP operations (such as sample rate conversion) involve some kind of filtering or bandwidth reduction. As soon as a downstream lowpass filter receives our digitally clipped signal, overshoot may result.

The last of these lowpass filters is the reconstruction filter. This means that even if every sample is constrained to a threshold value by a digital clipper, the output of the reconstruction filter in the analog domain can overshoot beyond this threshold. Another way of looking at this is to realize that the digital clipper only operates on samples of the waveform. It is not smart enough to anticipate what the analog output of the reconstruction filter will be between the samples. In most cases, there will be peaks in the analog output of the reconstruction filter that are higher in amplitude than any of the clipped samples emerging from the digital clipper. So clipping in the digital domain does not perfectly control peak levels in the analog domain.

The problem is not caused by the filter -- it is caused by clipping in a way that “breaks the rules” by producing out of band energy, meaning energy beyond the 20 kHz bandwidth of the system. In other words, clipping makes high frequency energy that the filter must remove. When this happens, overshoot appears at the output of the filter.

The challenge to a DSP programmer is therefore to create signals that are limited in both bandwidth and amplitude. It is easy to do one or the other, but doing both simultaneously requires some special techniques. An ordinary lowpass filter will control the bandwidth, but not the amplitude. A clipper will control the amplitude of the individual samples, but not the bandwidth.

EMPHASIS FILTERS

In the analog world, there are many standards for pre-emphasis, de-emphasis, equalization, etc. These standards are based on realizable transfer functions of RLC filters. With common analog implementations the amplitude response entirely defines the phase response because the filters are “minimum phase.” If a pre-emphasis network is cascaded with its complementary de-emphasis network, then not only do the amplitude responses of the two networks provide unity gain, but the phase responses also subtract, producing linear phase. Thus, ideal cascaded analog pre-emphasis and de-emphasis networks will not cause overshoot.

Unfortunately, in real world analog FM you *cannot* make a complementary pair of analog 50 or 75μs pre-emphasis and de-emphasis filters. This is because the de-emphasis filter is usually a simple RC rolloff that has zero gain at infinite frequency. To be truly complementary, the pre-emphasis filter would have to have infinite gain at infinite frequency. Such a filter is unrealizable in the analog domain (it would have to have more zeros than poles).

Digital creates further complications. DSP programmers generally use a kind of filter called ”infinite impulse response” or IIR to perform emphasis functions. IIR filters have the advantage that their phase functions are similar, but not identical to the analog RLC networks they emulate. Therefore, if a designer produces an IIR filter that performs, for example, a 75μs pre-emphasis curve, the digital network will usually be very accurate in matching the amplitude response. Nevertheless, there may be significant phase errors. Depending on the sampling rate and other factors, the phase errors may be 10-20 degrees with respect to an analog network at higher audio frequencies. A 10-degree error creates a maximum overshoot in the order of 1.6% (0.138dB); a 20-degree error creates a maximum overshoot in the order of 6% (0.506dB)[1]. So depending on the spectral content of the program material, this error may create overshoots requiring the reduction of average modulation by an audible amount. Clearly, DSP designers need to pay attention to the phase response of the emphasis networks they design, as well as the amplitude response.

We would like to make a simple proposal in this regard: that DSP designers of emphasis networks match not only the complement of the amplitude response, but also the complement of the phase response of the equivalent analog de-emphasis network. This proposal provides an unambiguous definition of the transfer function, and it allows systems integrators to mix analog and digital components in a way that will minimize overshoot. For example, if an analog audio processor is used

[1] For a test case consisting of a sinewave and its third harmonic in-phase, where the amplitude of the third harmonic has been adjusted to give the minimum peak level of the overall waveform, the correct amplitude of the third harmonic is 0.11111, assuming that the amplitude of the fundamental is 1. Phase-shifting the third harmonic by 10 degrees increases the peak level by 1.524%; phase shifting it by 20 degrees increases the peak level by 3.559%.

with flat audio outputs[2], a digital FM exciter conforming to this proposal will reapply pre-emphasis, but it will match both the amplitude and the phase response of the analog de-emphasis network present at the output of the analog audio processor. Furthermore, if the analog audio processor produces audio that is limited in both amplitude and bandwidth (to 15 kHz), there will be no inherent overshoot in the combined system. The digital FM exciter will exactly complement the audio processor's emphasis network, and will otherwise have linear phase and flat amplitude response over the 15 kHz bandwidth of the audio processor's output[3].

LOW FREQUENCY EFFECTS

Although there are no major differences between analog and digital systems when it comes to highpass filtering, it can still affect overshoot and should be addressed. Highpass filtering is used primarily to remove "numerical DC" which can be introduced by A/D converter offset, integer truncation, and other causes. If a system is entirely DC coupled, DC offsets will cause a digital FM exciter to go off frequency, possibly in violation of government regulations rules. Moderate frequency errors caused by DC offsets can significantly increase distortion in automobile radios, which often use relatively narrow IF filters.

As is the case with high frequency overshoot, AC coupling should affect neither the phase nor the amplitude of the lowest frequencies to be transmitted. (Arguably, if we change neither the phase nor the amplitude of the lowest frequency transmitted, we have done nothing to the sound quality by removing DC.) Although one might think that a 5-20 Hz cutoff on a highpass filter would be adequate for audio purposes, it is not. Phase shifts extend several octaves above the cutoff frequency of a minimum phase highpass filter. To limit tilt on a 20 Hz squarewave to less than 1%, a 0.05 Hz cutoff frequency is recommended.

Many converters have built-in highpass filters with cutoff frequencies higher than 0.16Hz and will therefore introduce overshoot. Converters must therefore be carefully qualified for use in broadcast transmission work.

Figure 12 shows what happens when an accurately limited audio signal is highpass filtered at 5 Hz. The 5 Hz highpass filter still has significant phase shift at frequencies several octaves higher. For example, at 50 Hz, our 5 Hz highpass filter still has 6 degrees of phase shift – enough to cause significant overshoot. There are two traces on this diagram – a lower trace at 100% which indicates the accurately limited signal peaks, and another higher trace which indicates where the highpass filtered signal peaks are actually falling: up to 115%.

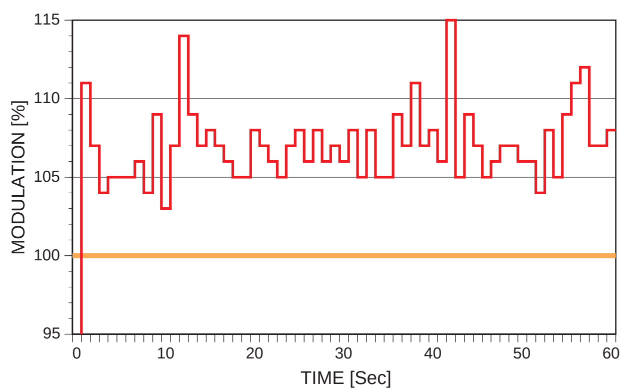

Fig. 12 PEAK MODULATION – System LF Cutoff 5Hz

When the highpass filter's cutoff is lowered to 0.16 Hz, the overshoot disappears as shown in Figure 13. If there is more than one highpass filter in the signal path, the cutoff frequency of the individual filters must be chosen such that the total system cutoff frequency is no higher than 0.16Hz.

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[2] By "flat" output, we mean an output to which de-emphasis has been applied in the digital domain. In FM audio processors, peak limiting is applied to the pre-emphasized signal. Therefore, the output of the audio processor is pre-emphasized unless explicitly passed through a de-emphasis filter.

[3] Note to DSP designers: An efficient way to design IIR emphasis networks is to first optimize an IIR transfer function to match the desired amplitude function, with no consideration for the phase. Then, determine the difference between the actual phase function and the desired phase function, and design an IIR allpass network that corrects the phase function. The cascaded emphasis network plus its associated phase equalizer will emulate both the amplitude and phase function of an analog RLC emphasis network, plus a constant delay term. For a 75µs pre-emphasis or de-emphasis filter, a second-order allpass will typically equalize the phase error to less than 2 degrees. This error does not cause audible loss of loudness.

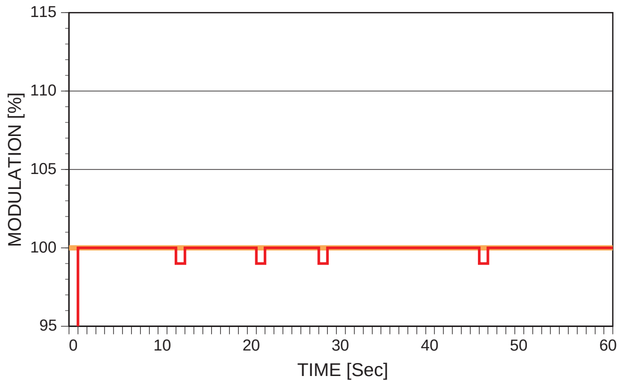

Fig. 13 PEAK MODULATION – System LF Cutoff 0.16Hz

LOSSY COMPRESSION SYSTEMS

Lossy digital audio compression systems generally cause significant overshoot. The design requirements for a digital audio compression system conflict with what we need to prevent overshoot. A lossy compression system is intended to preserve the perceived sound quality as much as possible. This means throwing away part of the signal. Lossy compression systems minimize the audibility of the changes to the audio signal. Unfortunately, preserving perceived sound quality is not the same thing as preserving the audio waveform's exact shape. The result is usually overshoot.

Figure 14 shows the overshoot present in an audio compression system. When presented with processed audio accurately limited to 100% modulation, the lossy compression system overshoots to as high as 144% in this example.

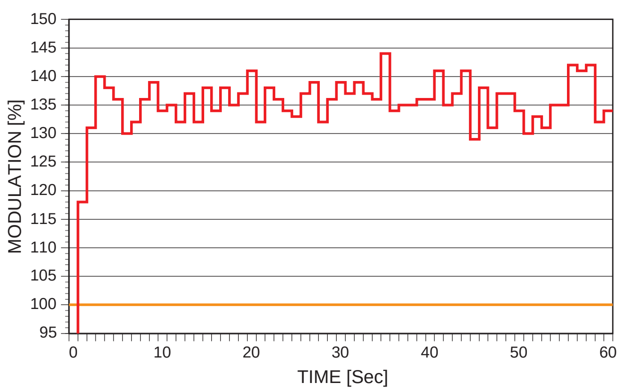

Fig. 14 PEAK MODULATION – APT-X™ 256kbps Stereo

Lossy compression systems are not intended to preserve waveform shapes, so if overshoot is an issue then processing should occur *after* the compression system. However, this introduces another conflict: a lossy compression system may not effectively mask its own artifacts if its output is processed. Processing, after all, changes the peak to average ratio, energy distribution as a function of frequency, etc. In such a situation, the best tradeoff would be to use a compression system that applies only minimal compression. No compression at all would be preferable.

PREVENTION: THE RULES

In summary, these are "the rules" for avoiding digital overshoot:

1. Control bandwidth to something less than Nyquist. Producers of digital audio processing systems can avoid overshoot problems by not producing energy that goes all the way to Nyquist. In an FM system, the bandwidth should be at least 15 kHz but the necessity of pilot protection requires limiting the bandwidth to something less than 19 kHz – typically 17.5 kHz maximum. The exact bandwidth will be determined by the particular digital exciter being used and by the sample rate used in the digital link to the exciter. Many links (including some of the new "uncompressed" links) operate at 32kHz. 32kHz systems must be *strictly* limited to 15kHz audio bandwidth to prevent overshoot. In any event, limiting audio bandwidth to 15 kHz will avoid problems caused by spectrum truncation.

2. Simultaneously control the amplitude of the reconstructed waveform. Digital audio processors must control not just the amplitude of the digital audio samples, but also the amplitude of the continuous waveform they represent.

3. Cascaded emphasis networks should match each other in both phase and amplitude. In particular, the phase response of a digital emphasis network should match that of the equivalent analog network.

4. Any minimum phase AC coupling should have a corner frequency of 0.05 Hz or lower.

When the stereo pilot tone is absent, the peak level of the FM stereo composite baseband signal is equal to the higher of the left or right audio channels. Therefore, controlling the peak level of the individual channels will correctly control the peak level of the composite if the system has ideal performance.

Adding the pilot tone changes the performance very slightly because of “interleaving” between the pilot tone and the 38kHz subcarrier. If you start with a single channel at 100% peak modulation and then add the second channel, also at 100% modulation, the peak level of the baseband (including pilot tone) will increase by roughly 2.7%. This is only 0.23dB, and is small enough to be negligible.

So we conclude that if "the rules" are obeyed in processing the left and right audio signals, then the digital composite waveform produced in a DSP stereo generator will be predictable, and its peak modulation will be controlled within a worst-case 0.23dB window. Obeying "the rules" means that there is really no need for a digital composite interface. If a STL only needs to transmit digital audio rather than digital composite, there is a huge saving in both bit rate and dollars. 16-bit stereo audio at 32 kHz, for example, only requires a payload bit rate of 1.024 megabits/second. But non-subsampled digital composite, for example at a sampling rate of 152 kHz and 16 bits would require a payload bit rate of 2.432 megabits - over double what is needed for transmission of stereo digital audio. So, obeying "the rules" not only avoids overshoot, but can also reduce operating and equipment costs.

If these simple rules are followed, then systems integrators will be able to connect a digital audio processor to an uncompressed STL followed by a digital FM exciter, and *no overshoot* will occur downstream from the audio processor. Industry-standard AES/EBU connections can be used. There will be no need for the use of brutish blunt instruments (such as composite clippers). Conversely, if the rules are broken, then overshoots will appear downstream, and it may not be easy to identify the true source of the problem.

INTERIM CURES

Until all digital and analog audio processors obey the rules, there will be circumstances where overshoots appear downstream because of improper audio processing. If this is the case, then overshoot compensation can be applied in the FM exciter itself. There are several digital FM exciters with this capability. Overshoot compensation in FM exciters (or any system) must also obey the rules, otherwise the problem will reappear downstream.

CONCLUSION

Digital systems differ from analog in three ways that affect overshoot. (1) Digital systems are *mostly* linear phase, which eliminates the time dispersion source of overshoot. (2) Emphasis networks are usually the only part of digital systems that are *not* linear phase. Nevertheless, use of complementary phase functions will eliminate overshoot. (3) Digital sampling may miss the peaks, so special DSP algorithms are required to control peaks and eliminate downstream overshoots.

Proper digital processing can prevent digital overshoot The need for composite clippers, overshoot clippers, etc. can be eliminated simply by controlling bandwidth, controlling amplitude of the reconstructed waveform, and maintaining phase linearity. Since composite clippers, overshoot limiters, and other non-linear processing can only degrade audio quality, the system designer has an excellent incentive to follow the rules, because doing so will pay benefits in the form of a cleaner, louder on-air sound.