POLARITY-INDEPENDENT CONTROL OF BROADCAST PROGRAM AMPLITUDES

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This paper describes Inovonics' PIPP™ audio limiter, a patented concept of audio signal processing wherein the positive and negative portions of the audio program waveform are processed independently of one another and subsequently recombined prior to transmission.

The description of PIPP™ limiting is preceded by a brief technical overview of commercial broadcasting, and a discussion of relevant technology with background and methods currently employed.

THE NEED TO CONTROL AUDIO LEVELS IN BROADCASTING

A typical radio transmission consists of a radio frequency (RF) *carrier wave* modulated by an *information signal*. In the commercial radio broadcasting field this information signal is usually speech and music programming, and is generally referred-to as the *program* signal.

Two fundamental methods of radio broadcasting are currently in use worldwide. The first (and oldest) sound-broadcasting technology still in common use is *amplitude modulation,* or *AM,* wherein the amplitude, or strength, of the radio-frequency carrier is varied in accordance with the audio program. The second broadcasting method is *frequency modulation,* or *FM*. In FM broadcasting the amplitude of the carrier wave remains constant, but its frequency is varied in accordance with the audio program.

A third and relatively new radio broadcasting method employs *digital modulation*. While the PIPP™ limiter may well prove to have utility in digital broadcasting, the primary benefit is to conventional analog AM and FM radio services.

The AM and FM transmission methods each have inherent modulation limitations; that is, the extent to which the carrier wave can be modulated by the program signal. In the case of AM, the radio frequency carrier can never assume a magnitude less than zero, or complete carrier cutoff, otherwise referred-to as –100% modulation. Symmetrical positive modulation would normally take the carrier to twice the unmodulated, "resting" value, or to +100%. In practice, an asymmetrical program signal waveform could modulate the carrier to –100%, and something in excess of +100%. In recognition of this possibility and of the slight advantage in coverage area that it affords, the US Federal Communications Commission (FCC) permits commercial AM broadcasts to achieve a positive modulation value of +125%. Some other countries follow this same practice; others impose a strict +100% maximum limit.

FM transmissions differ from AM in that the RF carrier is deviated up and down in frequency by the modulating program signal. Though a carrier frequency of 100MHz could, in theory, be deviated downward to zero frequency and symmetrically upward to 200MHz, not only would the signal occupy an absurdly wide portion of the radio spectrum, but this practice would present insurmountable technical problems in transmission and reception. In practice, a fixed deviation limit is imposed. In the case of commercial FM broadcasting, a standard of ±75kHz is observed the world over. This means that a 100MHz carrier may be deviated upward in frequency to 100.075MHz and downward to 99.925MHz by the audio program signal.

To avoid carrier overmodulation in either AM or FM transmissions, it would be possible simply to adjust the modulating signal to a level that could not possibly exceed the set limits. However, the *dynamic*

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range of a program signal is such that this practice would result in very inefficient utilization of the transmission channel. If levels were preset so that the loudest possible sound modulated the carrier just to the prescribed limit, lower level portions of the program could be lost in transmission system noise, particularly for listeners at greater distances from the station.

"AUDIO PROCESSING"

The dilemma of maintaining low-level sounds at an adequate volume, while at the same time guarding against carrier overmodulation by louder sounds, has given rise to a class of equipment known within the broadcast trade as *audio processors*. These employ the techniques of audio *compression* and audio *peak limiting* to reduce program signal dynamics.

Compression is a function that automatically reduces the dynamic range of the *average value* of the program; that is, it unobtrusively raises the level of softer sounds and decreases louder ones. Not only does this action increase the efficiency of the transmission channel, but it also has the secondary advantage of maintaining a more consistent level of sound in the listening environment. This renders speech more intelligible and music more enjoyable in a noisy workplace or in an automobile.

Though somewhat similar to compression, *limiting* is a separate and distinct level-control function. If compression may be considered a gentle, easy-going level-control operation, then limiting is a veritable 'brick wall.' Limiting prevents *transient peaks* in the audio program signal from overmodulating the carrier. Action of a peak limiter is considerably faster and more precise than that of a compressor.

Audio processors have evolved from the vacuum tube limiters used in the earliest days of radio broadcasting to the sophisticated and complex analog and digital processing systems available to broadcasters today.

AUTOMATIC AUDIO LEVEL CONTROL TECHNIQUES

Compressors and limiters operating in the analog domain have common operating principles and utilize similar electronic circuitry. The following discussion of audio level control techniques will refer to analog implementations that can apply equally to the functions of compression or limiting in general terms. Examples using analog electronic circuitry are perhaps more easily and universally understood by broadcast engineers than the comparable numerical calculation (software) routines used in digital processing systems, which can take many forms to achieve identical results.

From this point forward it is assumed that the reader has a basic understanding of electronics as it applies to professional sound and broadcasting equipment.

Most audio level controllers operate in the *feedback* mode. A program signal is presented to the input of a variable-gain amplifier, and an associated control circuit monitors the amplifier output and automatically controls the amplifier gain to keep the output level constant. In a broadcast *limiter,* the output signal is restricted to the predetermined or mandated *maximum instantaneous (peak) value.* In a *compressor,* circuitry generally maintains a predetermined *ratio* between the *average values* of the input and output signals. A simplified diagram of an elementary analog peak limiter is shown in Figure 1. Circuit action is detailed in the text that follows.

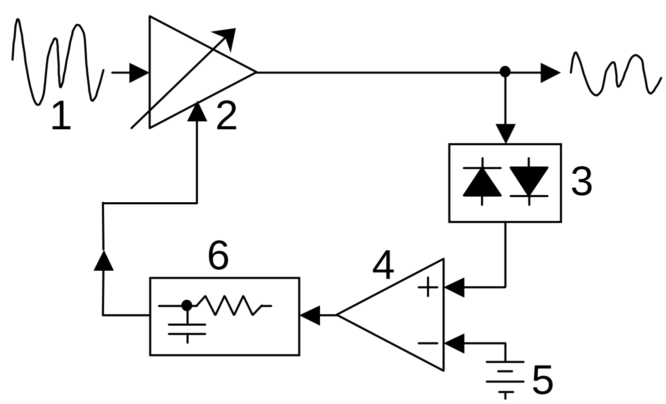

Figure 1 – Simple Audio Limiter

An input program signal **1** is presented to a variable-gain amplifier stage **2**. The output of the variable-gain stage is rectified by a full-wave, peak-rectification circuit **3** and fed to a comparator amplifier **4**. A fixed reference voltage **5** representing the desired maximum peak value of the output signal is presented to the other input of the comparator. When the input signal peak level reaches the fixed threshold value, the comparator generates a DC error voltage. This error voltage is filtered by a network **6** to establish certain circuit time constants, and then is applied to the variable-gain amplifier stage **2** to reduce the signal level. The time-constant network **6** sets the *attack* and *release* characteristics of the circuit; that is, how quickly the circuit responds to an input overload and the time required for the circuit to recover from the overload and restore gain to the initial figure.

The variable-gain amplifier **2** of the example is a device for the linear control of an AC signal by a DC voltage or current. This function may be implemented in a number of ways. "Variable-mu" vacuum tubes were among the first devices to be used, as well as "variolossers" fabricated from semiconducting metallic oxides. In more recent history, electro-optic attenuators, field-effect transistors, "Gilbert cell" and other 'multiplier' circuits, and a variety of monolithic voltage-controlled amplifiers, or VCAs, have been put to use as gain-control elements. In every case, a critical and desired quality of the gain-control device is to effect *linear* reduction of the program audio signal, and to introduce as little distortion as possible into the program signal waveform.

WAVEFORM ASYMMETRY

The human voice, solo musical instruments and most other sounds that occur in nature exhibit wave shapes that are *asymmetrical*. Unlike pure tones, or *sinewaves*, it is the harmonic content of these waveforms, often typified by various degrees of asymmetry, which gives a particular tone its character. For example, a trumpet note compared with the same note from a piano.

Figure 2 illustrates a typical asymmetrical voice signal waveform. The particular wave shape shown was the actual oscilloscope display presented while a drawn-out "ooooo" sound (as in the word 'smooth') was spoken into a microphone.

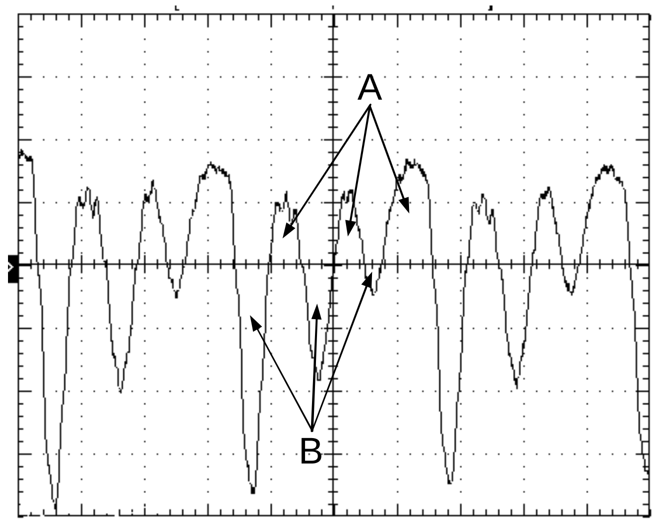

Figure 2 –Asymmetrical Speech Waveform

The central horizontal *baseline* represents the zero-voltage point or signal ground-potential reference. The AC waveform seeks a relationship with this baseline that contains equal included *areas* above and below it. The total of the integrated *areas* labeled **A** and **B** in one complete cycle of Figure 2 are equal, which explains the greater *amplitude* of the narrower negative peaks relative to the wider positive-going excursions.

When an asymmetrical audio signal is presented to a typical audio limiter (e.g. Figure 1), the limiter will react to the highest peak value, be it positive or negative, owing to the limiter's full-wave rectification of the output signal sample.

Figures 3, 4 and 5 illustrate the effect of conventional limiting on three waveform examples. In these examples 3 divisions above the baseline correspond to +100% modulation and 3 divisions below the baseline correspond to –100% modulation. The example speech waveforms shown are those actually displayed on the screen of an oscilloscope, thus accounting for slight variations between duplicated examples.

In Figure 3 a symmetrical signal (a pure sine wave) having equal amplitude and included areas in both the negative and positive directions is applied to the limiter. Either waveform excursion establishes limiting action, and modulation in this instance achieves –100% and +100%.

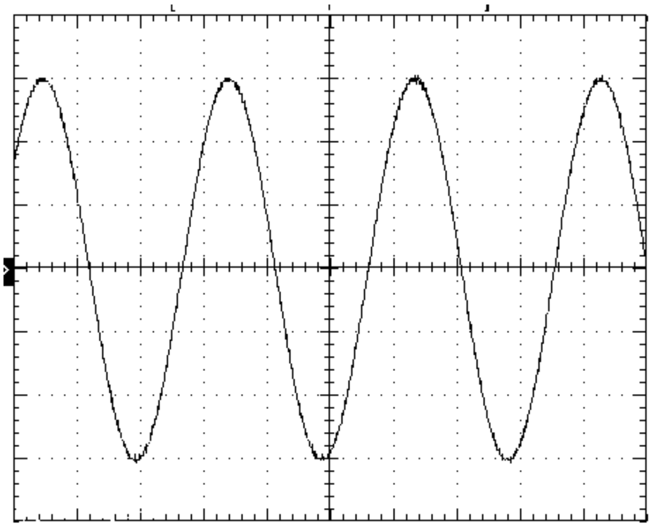

Figure 3 – Symmetrical Waveform

Figure 4 is an asymmetrical speech signal similar to the example shown in Figure 2. The higher negative amplitude reaches the limiting threshold and establishes limiter gain and the modulation limit. Negative modulation reaches –100%, but positive modulation peaks reach only about +50%.

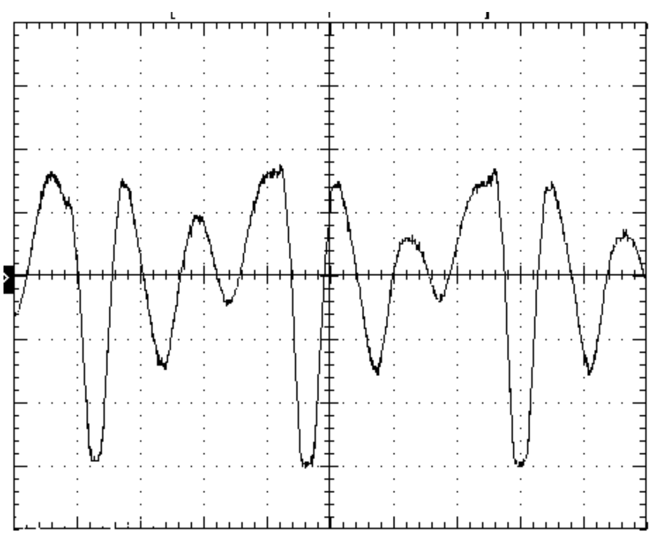

Figure 4 – "Negative" Asymmetrical Waveform

In practice, phase of a speech or music program signal is arbitrary. An asymmetrical signal has a 50/50 chance of reaching the limiter with predominating waveform peaks extending in either the positive or the negative direction.

In Figure 5 the asymmetrical waveform example has been subjected to a 180-degree phase reversal. This could be the normal and expected result of an amplifier phase inversion or simply due to the reversal of microphone or program line connections. The higher peak that initiates the limiting action is now positive-going. Positive modulation now reaches +100%, but negative modulation is on the order of 50%.

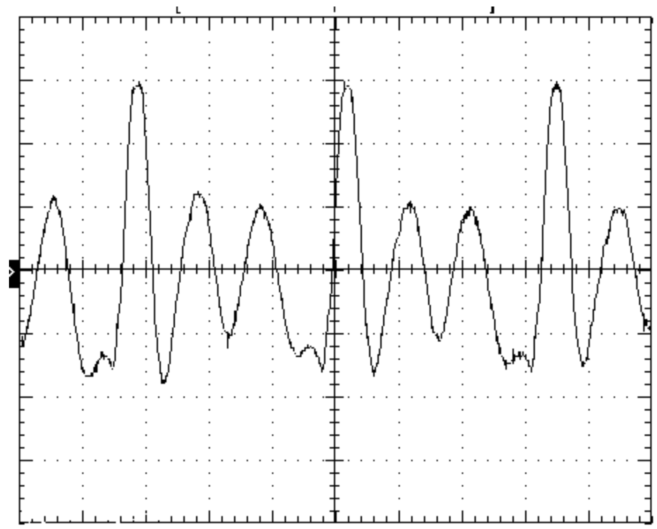

Figure 5 – "Positive" Asymmetrical Waveform

POLARITY-INDEPENDENT PEAK PROCESSING – THE PIPP™ CONCEPT

Figures 6, 7 and 8 show three modulating waveforms that are similar to the previous examples. The difference in these next cases is that the limiter has acted *independently* above and below the signal baseline, assigning *separate gain factors* to positive and negative waveform excursions.

The limited symmetrical sinewave in Figure 6 is essentially identical to Figure 3. Having no asymmetry, the sine wave is not modified by the polarity-independent limiting. It retains the same equal amplitude above and below the baseline. In other words, a pure tone is not affected by PIPP™ limiting, PIPP™ limiting does not generate *measurable* distortion.

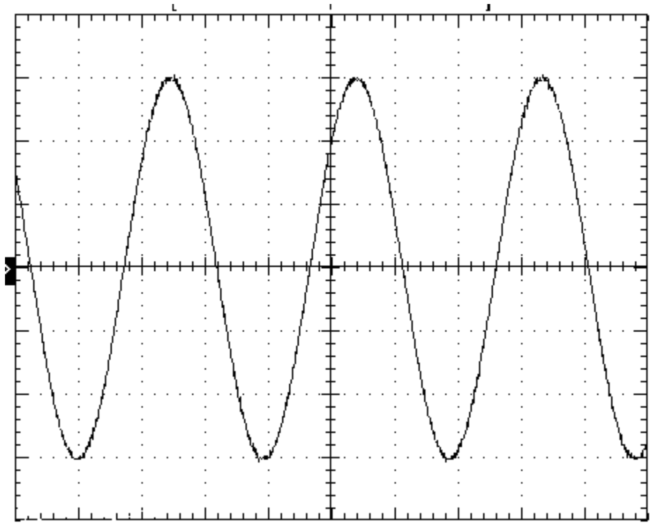

Figure 6 – Symmetrical Waveform With Polarity-Independent Limiting

Figure 7 shows the same 100% negative modulation also afforded by conventional limiting seen in Figure 4. However, the polarity-independent limiter has increased circuit gain above the baseline to give 100% positive modulation as well.

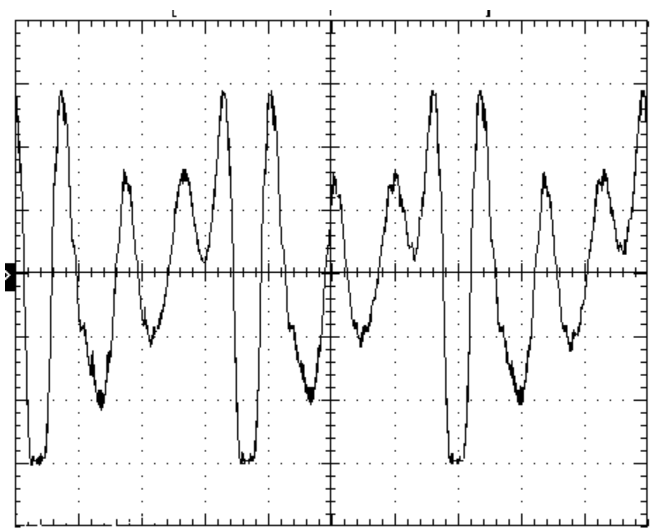

Figure 7 – "Negative" Asymmetrical Waveform With Polarity-Independent Limiting

Similarly, Figure 8 shows the same 100% positive modulation above the baseline that appears in Figure 5. Polarity-independent limiting has now imparted greater gain to the negative waveform excursions, resulting in 100% negative modulation.

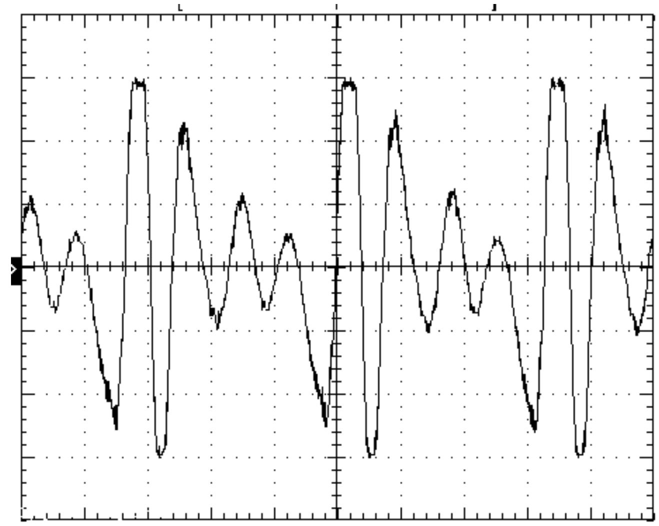

Figure 8 – "Positive" Asymmetrical Waveform With Polarity-Independent Limiting

The next two oscilloscope screen images illustrate the utility of PIPP™ limiting in a typical example of speech transmission. Once again, full modulation is indicated by 6 divisions peak-to-peak, three divisions below the baseline representing –100% modulation and three divisions above the baseline representing +100% modulation. In both instances the word "hello" was spoken into a microphone, which in the first example (Figure 9) feeds a conventional limiter, and in the second example (Figure 10) feeds a Polarity-Independent Limiter that uses an actual analog implementation of the PIPP™ concept.

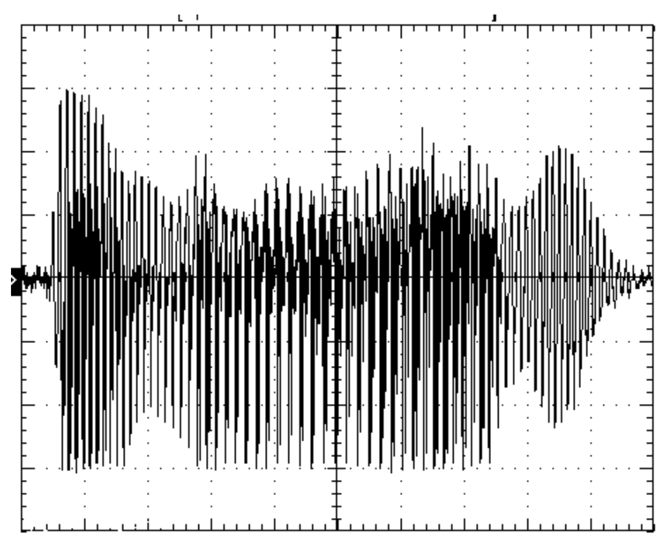

Figure 9 – The Spoken Word "HELLO" With Conventional Processing

In Figure 9 the carrier would not be fully modulated in the positive direction. In fact, the long-term average value of positive modulation is on the order of only 50%. But because negative speech waveform excursions are maintained at a –100% value by the conventional limiting circuitry, overall transmission efficiency is actually about 75%.

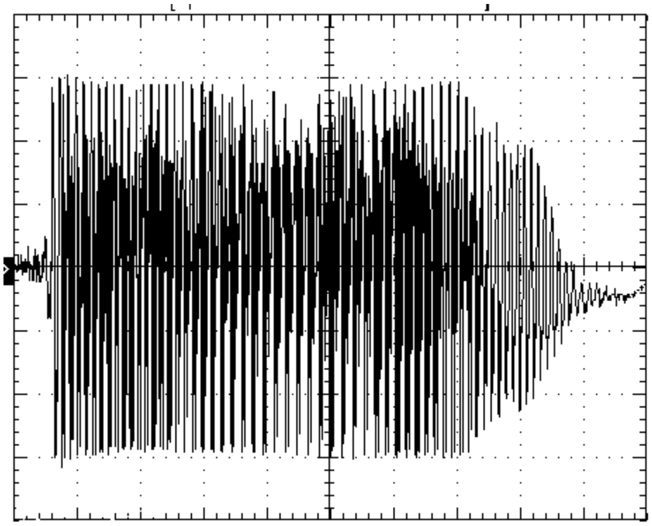

Figure 10 – The Spoken Word “HELLO” With Polarity-Independent Processing

In Figure 10 polarity-independent processing has again maintained negative modulation at 100%, but has increased the positive modulation to 100% as well. Overall transmission efficiency would be improved in this case from the 75% of Figure 9 to very nearly the theoretical maximum of 100%.

ANALOG IMPLEMENTATION OF PIPP™ LIMITING

A simplified schematic diagram depicting a simple implementation of PIPP™ limiting by analog circuitry is shown in Figure 11.

The program signal **1** is presented to two “absolute value” circuits, **2** and **3**. These split the incoming signal into positive and negative components; that is, the portion of the signal above, and the portion of the signal below, the input signal baseline. The positive and negative components of the signal are independently acted upon by two separate variable-gain stages, **4** and **5**.

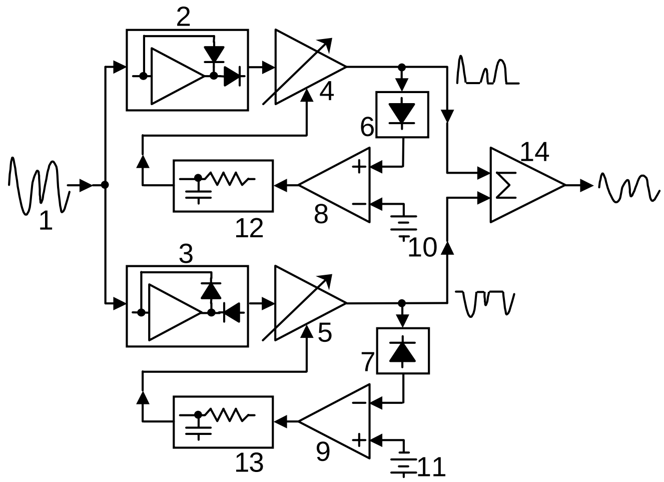

Figure 11 – Polarity-Independent Limiter

A positive peak rectification circuit **6** monitors the positive component of the program signal and a negative peak rectifier **7** monitors the negative component. Each rectifier has an associated comparator amplifier, **8** and **9**, that gives an error voltage representing the difference between the program signal peak level and the desired maximum permitted level established by fixed references **10** and **11**. The positive and negative error voltages are separately filtered by networks **12** and **13** and are applied to the respective variable-gain amplifiers **2** and **3**, which independently reduce the positive and the negative values of the input program signal, respectively. Combining amplifier **14** sums the independently limited positive and negative components, restoring the program signal.

The simplified circuit of Figure 11 illustrates only one manner in which the PIPP™ concept may be implemented. As explained in the description of the simple limiter circuit (Figure 1), a variety of analog circuits may be utilized for general audio program limiting, and these apply equally to the PIPP™ concept. Feed*forward*, as well as feed*back*, limiting may be used, and any of a number of variable-gain devices may be called into play.

DIGITAL IMPLEMENTATION OF THE PIPP™ CONCEPT

Sound waveforms may be translated into a continuous stream of numerical values, which subsequently may be mathematically manipulated to perform various audio-processing functions. This manipulation is referred-to as Digital Signal Processing, or DSP.

Any of a virtually unlimited variety of 'software routines' can be applied in implementing PIPP™ limiting using DSP techniques.

In its conversion to a digital data stream, a sound waveform is first *sampled* at a fixed rate, this rate being at least twice the highest audio frequency to be converted. Popular sampling rates used in broadcast audio are 32kHz, 44.1kHz and 48kHz. The result of sampling is a continuum of discrete, instantaneous values that describe the audio waveform.

Each waveform sample is *quantized*, or assigned a specific numerical value corresponding to its amplitude at the instant of sampling. Accuracy in recovering the original audio waveform depends on system *resolution,* or system *word length*; that is, the number of discrete quantization levels used to express the audio signal. The number of digital 'bits' used to express each quantization level determines system resolution and, hence, analog-to-digital conversion accuracy. Broadcast-quality audio systems are commonly 16-bit systems.

A quantization level is an *exact* 'digital address,' thus the sampled value is assigned the closest fixed level. *Binary coding* used in digital audio systems assigns 2^{16}, or 65,535 quantization levels in a 16-bit system. Assuming a linear coding scheme, this means 32,767 discrete values above the signal's resting baseline address of '32,768,' and 32,767 discrete values below it.

Referring back to Figure 11, the block diagram of analog PIPP™ implementation, the polarity-independent variable-gain amplifiers **2** and **3** become arithmetic *multiplication* functions for numerical values above and below the signal baseline, respectively. The stream of numerical values is monitored to make independent mathematical determinations of the positive and negative peak values of the incoming audio waveform. This information is then used to assign an independent multiplication factor for all numbers above 32,768, and another multiplication factor for numbers below 32,768. These multiplication factors are time varying, being constantly updated according to the controlling software routine. To impart gain, a sample would be multiplied by a number greater than 1. To affect attenuation the sample would be multiplied by a decimal fraction of one. These ongoing, continuous computations are so programmed as to maintain audio program level peaks at the ±32,767 system limits, regardless of the initial relationship between the positive and negative program peak excursions.

CLAIMS, ADVANTAGES AND CRITICISMS OF PIPP™ LIMITING

The net product of PIPP™ implementation as described is an amplitude-limited audio program signal that will modulate an RF carrier to its greatest capacity, regardless of program waveform geometry. The primary advantage of this action is to ensure the most effective and efficient use of the transmission channel, be it AM, FM, or a digital transmission, whether it is a commercial radio broadcast or perhaps a point-to-point, 2-way communication. This can translate to a more intelligible signal in the presence of interference, an increase in transmission coverage area, or a reduction in the power requirement for a radio transmitter that is used more efficiently. Depending on the actual program signal source and various other circumstances, the advantage provided by PIPP™ limiting in any of these instances can vary between negligible and appreciable.

An obvious criticism of the PIPP™ concept would be the assertion that its action alters the 'natural' relationship between the negative and positive components of the information waveform, or the relationship between a fundamental sound frequency and its overtones, primarily even-order harmonics. This would necessarily imply the intro-

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duction of unwanted distortion into the audio signal, a valid inference that can't be argued. Certainly, any change to waveform symmetry does indeed introduce even-order harmonic distortion. Still, only a sinewave (pure tone) is completely without distortion components in the first place, and because a sine wave is perfectly symmetrical, PIPP™ limiter action would leave a sinewave in its natural, undistorted state, indeed as shown in Figure 6.

Nonetheless, speech, music and other sounds in nature can be said to contain "native distortions" which legitimately appear in the guise of overtones and harmonics. These serve to characterize a particular sound and help a listener distinguish sounds of identical pitch. For instance, a note struck on a piano will not have the same character as that same note played on a saxophone. This is due entirely to the harmonic structure of the musical instrument waveform. PIPP™ limiting simply tends to amplify the *character* of the sound, perhaps making a cello more "cello-like," and adding certain richness or fullness to the human voice.

The PIPP audio processing concept was first implemented in Inovonics' Model 718 *(DAVID-III)* FM-Airchain Processor, and subsequently used in the Model 719 *(DAVID-IV)* all-digital processing system as well.