The Negative Effects of Feedback

The Problem

Before his successful invention of recorded music, Thomas Edison had to arrive at the fundamental realization that sound can be entirely characterized in two dimensions. His first cylindrical recording was nothing more than a rough approximation of the changes in pressure amplitude (caused by the modulation of his voice) plotted against a constant timebase (generated by the steady turning of a crank). Crude as his technique may have been, sound was recorded and quite recognizably, reproduced. The limiting factor in Edison's first experience was not his idea, but his hardware; further refinements in the field of audio could come only from improvements in technique.

Unfortunately, in the more than hundred years that have passed since the birth of recorded sound, engineers have clouded the simple definition of sound as changes in amplitude versus time by the use of such irrelevant (to music at least) ideas as frequency, phase, harmonics, intermodulation, etc. Such factors serve only to complicate the basic function of audio recording and reproducing equipment, and, to a certain extent, have stood in the way of the development of truly accurate audio components. Although the art and science of sound reproduction has progressed to a point Edison could never have imagined, only after shifting engineering priorities back to a study of the real components of real signals, will further substantial improvements be realized.

Once, having established that all audio signals can be expressed as a change in amplitude over some period of time, or DeltaA/DeltaT, the function of all sound system equipment can be easily defined. For instance, an amplifier performs simple multiplication resulting in an output signal which can be expressed as G(DeltaA/DeltaT) or GDeltaA/DeltaT, where G is the gain of the amplifier and DeltaA/DeltaT is the input signal. From this representation it can be seen that all changes in amplitude must be magnified by the same factor (i.e. the Gain of the amplifier) and the time base (DeltaT) must remain unchanged, independent of all other considerations. This leads to the discovery of the only two families of real distortions that can and do exist in audio systems.

Such amplitude distortion can assume two forms, harmonic and non-harmonic. Harmonic distortion (the most commonly and easily measured anomaly in audio components) is generally caused by non-linearities in the electrical characteristics of the amplification devices. Such distortion is "harmonic", as the number of zero crossings in the error wave form in an integral multiple of the number of zero crossings in the fundamental. Additionally, the value of the distortion signal will always be zero at the zero crossing point of the fundamental. A small amount of this type of distortion is inaudible as it does not drastically alter the shape of the waveform and does not affect the zero crossing point.

Non-harmonic amplitude distortions are generally caused by network anomalies. Such phenomena as slew rate limiting, clipping, and transient distortions result in non-harmonic distortion components which not only alter the shape of the signal waveform, but can change the zero crossing point, as these elements may have some real value, when the input signal is at zero.

This leads to the second major family of distortion; time base distortion. Time base distortion occurs when the DeltaT term of the signal equation is altered. The zero crossing point displacement described above is a form of time base distortion. Modulation of pulse width, or a change in the delay time between signal "events" also constitute time base distortions. These distortions are the most audible as our auditory system can more easily detect duration and delay than amplitude.

The Audio Note No-Feedback Real Audio Amplifiers.

The Audio Note amplifier range does not make use of any kind of feedback. As a result, they were neither designed for vanishingly small harmonic or low intermodulation distortions, but instead for minimal non-harmonic and time base anomalies.

The Audio Note amplifiers are all using directly heated power triodes in their output stages, and miniature double triodes in their high-gain and driver stages. Their function were defined before their circuitry was conceived, as constant multiplication of amplitude over a totally non-varying time base, with a view to maintaining power output into a varying load.

During the development of these amplifiers using direct heated power triodes, most accepted amplifier design practices had to be ignored, as investigations into their implementation showed circuits with variability of gain with amplitude, time and signal duration, as well as variability of time delay with amplitude, signal duration and signal delay. What has resulted are amplifier circuits which operate optimally and non-varyingly for all signal and load conditions. Where compromises have been necessary between maximally linear amplitude response, and optimum time base performance, the design parameters have always been adjusted to favour the latter. With the superior linearity and load characteristics of the directly heated power triode, whose circuit configurations naturally lend themselves to the defined functions.

The design practices most obviously eschewed in the development of the Audio Note Real Audio amplifiers (using direct heated power triodes) is the use of negative or local feedback. Negative feedback, quite simply, is the application of an inverted portion of an amplifier's output signal to its input terminals. This "extra" signal is subtracted from the input and serves to reduce the effective amplifier gain (as the input signal is then smaller). In addition, steady state distortion is thought to be reduced as the out-of-phase distortion components contained in the feedback signal cancels out some of the errors created by the amplifier circuitry.

This scheme presents two very obvious problems. Firstly, all amplifiers introduce some delay to passing a signal from its input, to its output and then back to its input. During this delay period, a feedback amplifier is operating at its natural (referred to as "open-loop") gain. It is not until this initial delay period is over, that the circuit begins to exhibit its intended operating ("closed loop") gain characteristics. There must be, by the very definition of a feedback system, some change in the gain factor G, during the transition from open to closed loop operation. This gain modulation would probably not be audible by itself, as the propagation delays of most good amplifiers are quite small, except that the increased gain of the amplifier during the initialization period results in a decreased maximum input capability before overload. Simply put, an amplifier which utilizes 20 dB of feedback (a relatively modest amount by modern standards) and requires an input of two volts to clip during closed loop operation, would overload with only two tenths of a volt input during the forward delay period. Once the amplifier is overdriven, it may take many times its delay period to become fully restored to normal operation. The distortion created by this condition has been commonly referred to as Transient Intermodulation Distortion (TIM), Dynamic Intermodulation Distortion (DIM), and Slew Induced Distortion (SID).

In addition to this obvious form of feedback induced distortion, there exists another more subtle effect of signal regeneration. Because all amplifiers have some forward propagation delay, the fed back portion of the output signal will always lag behind the input. There is therefore a constant introduction of "out of date" information into the amplifier. Under transient conditions (which is what music is; transients), this results in the presentation of an error correction signal intended to reduce the distortion of an input signal which has already passed through the amplifier and is either already out of the circuit or well on the way out of the circuit. The signal present at the input by the time the feedback has arrived may bear no relation to the previous signal and thus will not be properly acted upon by the regenerated information. The current input signal is then distorted once, through the subtraction of an erroneous feedback waveform, and again by the amplifier. Additionally, the error signal present in feedback is passed through the amplifier and again fed back, with all of the newly created distortions, to make yet another trip through the circuit, until it is allowed to decay through successive attenuation. Thus, a distortion signal which originally may have lasted only a few microseconds, can pass through the amplifier enough times for its effective duration to have exceeded the threshold of human audibility. The mechanism originally designed to reduce audible distortion, actually, under transient conditions, serves to regenerate, emphasize and, in fact, create distortion.

Because our Real Audio triode amplifiers operate totally without signal feedback, such distortion regeneration does not take place. The circuits have been designed for maximum linearity without corrective mechanisms, and thus responds as easily to transient signals as it does to steady state waveforms. The amplifiers make no attempt to reverse the path of time in order to correct their own errors. Those distortions created by these circuits (which are almost entirely harmonic in nature) are allowed to pass only onto the loudspeaker, and not back to the input.

Despite the absence of feedback, the forward propagation delay of all our amplifiers has received much attention. All our output transformers have been designed using this criterion, obviously with a keen eye on cost. It is obvious that if this delay is not absolutely invariant, for all conditions, the DeltaT component of the input signal will not be accurately preserved. Thus, those factors which determine delay have been carefully observed and stabilized. In addition, the operation of all amplification stages at nearly constant power, independent of signal conditions, i.e. Class A operation at every stage, greatly contributes to the symmetry and linearity of our circuits.

It is, however, not enough for an amplifier to operate linearly by itself. In order to minimize audible distortions, the device must be able to operate as well into a real loudspeaker as it does into a laboratory resistive load.

In order to adequately control the cone excursions of the loudspeaker and to optimize power transfer, the effective output impedance of the amplifier should be as far below the impedance of the load as possible. The ratio of these two impedances is referred to a damping factor, usually referenced to an eight ohm speaker. Thus, a damping factor of eighty reflects an amplifier output impedance of one tenth of one ohm. The design of the output transformer is extremely critical, and taps on the output are normally provided to match the load impedance best possible.

A problem in the normal expression of damping factor is that its measurement is performed using steady state signals. This results in a factor relying quite heavily on the action of an amplifier's feedback. The damping ability of an amplifier under transient conditions, before the feedback mechanism has been able to reach, is only accurately expressed as the steady state damping factor divided by the feedback factor. Thus, an amplifier with twenty decibels of feedback and specified damping factor of one hundred, has a damping value of only ten under transient conditions. This not only reduces the amplifier's ability to control the cone movement, but allows voltages created in the speaker voice coil to mix with the output signal and enter the amplifier's feedback system. In this condition, distortions created by the speaker's motion are not only unattenuated, but are emphasized through feedback regeneration.

Audio Note Real Audio, no-feedback amplifier's damping ability remains constant at all signal conditions.

An investigation is underway to fully explain the relationship between damping factor, real world power output and amplifier feedback. This will take some time. In the meantime, the only amplifiers available on the market to fully fulfill the criteria set, amplitude against time, with as little change as possible, are the non-feedback amplifiers from Audio Note.

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