Rappaport

Model AMP-1

Real Time Audio Amplifier

Problem:

Before the 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 time base generated by the steady turning of a crank. Crude as his technique may have been, sound was recorded and quite recognisably reproduced. The limiting factor in Edison's first experiments was not his idea but his hardware. Further refinements in the field of audio could come only from improvement in technique.

Unfortunately, in the more than one hundred years that have elapsed since the birth of recorded music, 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 an audio recording and to a certain extent have stood in the way of development of truly accurate audio components. Although the art and science of sound reproduction has progressed to a point, Edison would never have imagined only after shifting engineering priorities back to a study of the real components of signals will further substantial improvements be realized.

Hence having established that all audio signals can be assessed as a change in amplitude over some period of time or ΔA / Δt, the function of all sound system equipment can be easily defined. For instance, an amplifier performing simple multiplication resulting in an output signal that can be expressed as G (ΔA / Δt) or G ΔA / Δt where G is the gain of the amplifier and ΔA / Δt is the input signal. From this representation it can be seen that all changes in amplitude must be amplified by the same factor and the time base (Δt) 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.

The first type of distortion is amplitude distortion. This is the result of some in the gain factor. G. If the instantaneous value (that is the value at any given time) of the parameter is not absolutely constant, delicate changes in signal amplitude will become distorted.

Such amplitude distortion can assume two forms, harmonic and non-harmonic. Harmonic distortion [the most common 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 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 alter the zero-crossing point.

Non-harmonic amplitude distortions are generally caused by network anomalies. Such phenomena as slew rate limiting, clipping and transient distortion 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 zero.

This leads to the second major family of distortion – time base distortion. Time base distortion occurs when the Δt terms of the signal equation is altered. The zero-crossing 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 the auditory system can more readily detect duration and delay than amplitude.

The AMP-1:

The Rappaport AMP-1 Real Time Audio Power Amplifier has been designed, not for vanishingly small harmonic or intermodulation distortion, but for minimal non-harmonic and time base anomalies. Its function was defined well before its actual circuitry was conceived as constant multiplication of amplitude over a totally non-varying time base. During the AMP-1's development most accepted amplifier design practices had to be ignored on investigation into their implementation, showed circuits with varying multiplication of amplitude, time and signal duration and signal delay. What has resulted is an amplifier circuit which operates optimally and non-varyingly for all signal and load conditions. Where compromises between maximally linear amplitude response and optimal time base performance the design parameters were always adjusted to favor the latter.

The design practice most obviously eschewed in the development of AMP-1 was the use of negative feedback. Negative feedback, quite simply, is the application of a small inverted portion of the amplifier's output to its input terminals. This extra signal is subtracted from the input signal and reduces 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 cancel out some of the errors created by the amplifier circuitry.

This scheme presents two obvious problems. First, all amplifiers introduce some delay to passing a signal from its input to its output and then back to 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 output 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 might not be audible by itself as the prorogation delay 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 capacity before overload. Simply put, an amplifier which used 20db of feedback [a relatively modest amount by current 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 over-driven 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 [SIM].

In addition to the obvious form of feedback induced distortion, there exists another subtle effect of signal regeneration. Because all amplifiers have some forward propagation delay, the feedback portion of the output signal will always lag behind the input. There is a constant introduction of out-of-date information into the amplifier.

Under transient conditions this results in the presentation of an error correction signal intended to reduce the distortion of an input signal that has already passed through the amplifier and is well on its 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 feedback with all the newly created distortions to make yet another trip to the output 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 the AMP-1 operates totally without signal feedback, such distortion regeneration does not take place. The circuit has been designed for maximum linearity without corrective mechanisms and thus responds as easily to transient signals as it does to steady state waveforms. The amplifier makes no attempt to reverse the path of time in order to correct its errors. Those distortions created by the circuit [which are almost entirely harmonic in nature] are allowed to pass only on to the loudspeaker and not back to the input.

Despite the absence of feedback, the forward propagation delay of the AMP-1 has been given much attention. It is obvious that if this delay is not absolutely invariant for all conditions that the Δt component of the input signal will not be accurately preserved. Thus, those factors which determine delay have been carefully observed and stabilized. In addition to the operation of all amplification stages of nearly constant power independent of signal conditions, the thermal operating characteristics of the devices employed are carefully controlled. All transistors are maintained at an intentionally high temperature. Unlike conventional amplifiers that maintain semiconductor junctions at a low heat level until stressed, the AMP-1 holds its devices at temperatures rarely exceeded in normal operation. Instead of constantly forcing transistors to heat up and slowly cool until exercised again the AMP-1 keeps them hot even when the signal conditions turn them off by thermo-coupling them to a heatsink which remains very warm at all times. The resulting constant junction temperature not only maintains invariant delay properties but reduces transistor degradation due to rapid thermal cycling. Additionally, because the elevated temperature results in a freer transfer of energy in the transistor chip [largely due to an increased Brownian Motion] the devices operate far more linearly.

All operating parameters of the AMP-1 are further stabilized through the use of bias regulation systems. One such servo circuit maintains the low-level voltage gain portion of the amplifier at its optimum operating point by deriving its bias reference from an inverted replica of the stage's output voltage. Any change in DC level at the input of the regulation circuit results in a compensatory DC shift of the circuit bias point.

A second servo mechanism eliminates DC offset at the amplifiers output terminal. Any shift in output voltage is sensed by the circuit and a corrective potential is applied to the input of the high current stages. This system not only eliminates the effects of thermal drift but also protects against damage to loudspeakers in the event that a DC voltage is applied to the amplifier's input. Both servo circuits guard against the regeneration of AC components through the use of a unique filtration mechanism [patents applied for].

In order to ensure that the intrinsic linearity of the AMP-1 is not affected by the speed of the input signal, all circuitry has been designed to be quite fast. The inherent slew rate capability [greater than five hundred volts per microsecond] and bandwidth [greater than five hundred kilohertz] of the circuit prevent the occurrence of any signal slope limiting other than that prescribed by the one hundred kilohertz filter built into the amplifier's input buffer.

It is 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 resistive load. Despite the nominal impedance ratings of audio speaker systems, the demands on an amplifier vary greatly with signal content. A speaker rated at eight ohms may often appear as a much lower impedance.

In order to drive such low impedance loads an amplifier must be able to deliver adequate current as well as voltage. Developing twenty volts rms across an eight-ohm load requires two and one half amps rms or three and one half amps peak but the same twenty volt rms would require a peak of twenty eight amps instantaneously to drive a one ohm load. An amplifier's power supply must be able to supply such energy levels on demand and its output stages must be capable of delivering this power to the loudspeakers. The short-term peak current capability of each channel of the AMP-1 is forty amps, thus enabling it to deliver its full output voltage into any loudspeaker load. In addition, the high current power supplies for each channel are completely isolated from one another to ensure that current demands placed on one side of the amplifier do not affect the performance of the other.

In order to adequately control the cone excursions of the loudspeaker and to optimize power transfer the effective output impedance of the amplifier must be far less than the impedance of the load. The ratio of these two impedances is referred to as damping factor usually reference to an eight-ohm speaker. Thus, a damping factor of eighty, reflects an amplifier output impedance of one tenth of one ohm.

A problem in the expression of this characteristic, however, is that the 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 react is only accurately expressed as the steady state damping factor divided by the feedback factor. Thus, an amplifier with twenty decibels of feedback and a 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 cone motion, but also allows voltages created in the speaker voice coil to mix with the output signal and enter the feedback system. In this condition, distortions created by speaker motion are not only not attenuated but are emphasized through feedback regeneration.

Because the AMP-1 has no feedback its damping factor had to be kept high through the design of an output stage of inherently low impedance. Additionally, because of the low inductance of the output circuitry the amplifier's damping ability remains constant for all signal conditions.

As with all Rappaport products the AMP-1 is warranted against defects in material and workmanship for a period of three years from date of purchase.

Specifications:
27V RMS before clipping
40A peak short term peak current capability
Voltage Gain 23.5db
Power consumption Idle 500W; Peak 3500W
Overall dimensions 19" W, 15.5" D, 8.75" H
Net weight 75 lbs