Hi-Fi Stereo Handbook

by

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Preface

Since the first edition of this book was published in 1956 there have been amazing changes and wonderful improvements in sound communication, recording, processing, storage, amplification, control, and reproduction techniques. The constant development of higher-quality units at lower cost per unit has made hi-fi stereo an economic reality for everyone. These remarkable improvements are highlighted by such effects as precision dynamic coupling of phono cartridges to correspondingly improved mediums with up to 50 kHz frequency response, LSI (large-scale integrated-circuit chips containing hundreds of transistors), application of precision frequency counters with digital display, use of digital processing for encoding, recording, storage, detection, and retrieval to produce distortion-free programs, computer and microprocessor circuits for precision tuning and automatic control, advanced high-efficiency speakers, multiple-record changers that can almost think, sophisticated acoustic processing for spatial effects, scientific techniques for acoustic equalizing of any listening chamber, and more—all giving dramatically improved performance with more fidelity and convenience at less cost than was previously available.

It has been more than a century since geniuses of music creation brought forth the need for media, techniques, and equipment to record and store their great works. The world will forever lament that it was not able to preserve the music of the highly disciplined and talented artists of yesterday. At least from now on we can achieve this for ourselves and for future generations.

Modern concepts and standards of high fidelity, or hi-fi, provide for the ultimate in the endeavor toward reproduction of sounds exactly as they were originally created. True fidelity, as you will learn in the first chapter, is only occasionally achieved, but modern recording techniques provide results with maximum control of source, bearing, vocal, instrumental, and other elements arranged with emphasis, presence, balance, and three-dimensional effects, to create a final mixture that can be pleasingly superior to live performances. The art of recording, storing, and reproducing sound electronically has progressed to the point where you, with eyes closed, can almost believe yourself to be sitting fifth row, center, at Lincoln Center.

But once your appetite for high-fidelity surround-sound has been whetted, what then? What equipment should you buy? How much will it cost? And these considerations are only the beginning. To help you this book was prepared as a reference and guide to high-quality sound reproduction.

You will find how to build and install an advanced speaker system, suited to your situation at low cost and with high efficiency and quality. You will find step-by-step instructions on how to equalize sound within your listening area using the latest equipment and techniques. In addition to supplying information which will assist you in planning, selecting, and installing appropriate systems, this book also describes the various system components, plus what they do and how they operate.

Thus, no matter whether the subject is new to you or you are a technician or an experienced hi-fi enthusiast, *Hi-Fi Stereo Handbook* has been written for you. I believe that you will find it informative and highly useful.

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1

Fidelity, Sound, and Distortion

Experts on high fidelity are in universal agreement that there is as yet no exact scientific operational definition for a high-fidelity system. Standards and specified measurements of performance of a system have been impossible to establish because of limitations of the human ear and because of variations in human taste, room acoustics, system distortions, noise, and comparative volume levels.

WHAT *IS* HIGH FIDELITY?

A commonly accepted definition of high-fidelity sound is that it is reproduced sound with a high degree of similarity to the original or "live" sound. High fidelity is felt to be achieved when the sound that is reproduced has negligible distortion from the original, when it has little extraneous noise, and when the volume levels and room acoustical effects are pleasing to hear. This reproduced sound might even be more pleasing to the listener at the output of the system than the original live sound would have been if heard in person at its place of origin.

A reproduction of sound is something like a photograph. The picture cannot carry the original scene to the viewer in every detail. Some features of the picture may be de-emphasized, whereas other features may be emphasized intentionally, or distortion may be introduced for purely aesthetic reasons. Distortions of this sort can greatly improve the illusion that the photographer is trying to create. In the same way, the picture can be spoiled by undesirable distortions and effects, such as poor focus, poor film, or improper lighting of the subject.

Like photography, modern high-fidelity techniques encompass controls for modification of the original (live) sound to compensate for certain defects and make provision to actually improve the effects according to an individual listener's tastes. Undesirable distortions, differences in comparative sound levels, and injection of extraneous noise are also held to a minimum so that the pleasing qualities of the original sound will not be reduced.

In addition, modern concepts of high fidelity take into consideration the listener, his or her ear mechanism, and his or her nervous response, plus his or her listening experience and training.

Psychophysical reactions and imagination contribute to the realism of high-fidelity reproduction. The word "presence" is used to describe the degree of realism of the reproduced sound. This term suggests that the reproduction is so real that the listener can feel the presence of the source that is causing the live sound, even though that source is many miles away or even extinct. Furthermore, psychologists have shown that the trained human mind will fill in missing sounds that should appear in a musical rendition, even though these sounds are not present in the reproduction.

The application of the term "high fidelity," then, is largely a personal matter. Everyone can be a hi-fi expert—at least as far as his or her own tastes in equipment and quality of sound reproduction are concerned.

SOUND

The word "sound" is used in different ways. In the psychoacoustical sense, "sound" means to the listener the sensation of hearing audible vibrations conveyed from any medium (such as air, usually) through the ear to the brain. As used in physics, however, "sound" means the external cause of the sensation. In hi-fi, we are concerned with both meanings.

Sources of sound are bodies in vibration. Vibrations of a low-note, bass-viol string can actually be seen. The sounds caused by such a source have only a few vibrations per second. They are therefore called *low-frequency sounds*, *low notes*, or *lows*. On the other hand, the tinkles of a glass or of a musical instrument such as a triangle have comparatively many vibrations per second, and these objects are said to vibrate at *high frequencies*. Such sounds are known as *high notes* or *highs* in the range of human hearing sensations.

Frequencies of sounds audible to the average person range between 20 and 20,000 hertz (cycles per second). The average human hearing system has certain characteristics of receiving, converting, and interpreting sound that are important considerations in the techniques of producing high-fidelity impressions.

Loud sounds are heard with good fidelity over a comparatively wide frequency spectrum. In other words, highs, lows, and in-between-frequency sounds are heard in proper relation to each other when all these sounds are loud; however, when the volume is reduced, the ear tends to attenuate (or cut down) the highs and lows but leaves the in-betweens proportionately louder. This is demonstrated by the curves shown in Fig. 1-1.

The normal human hearing system has directional characteristics, receiving all sounds best from the forward position (source in front of observer) and with the ability to distinguish the direction of the source or of a reflection. The ear can distinguish several sounds of different frequencies at the same time to a high degree. It is possible for the ear to distinguish

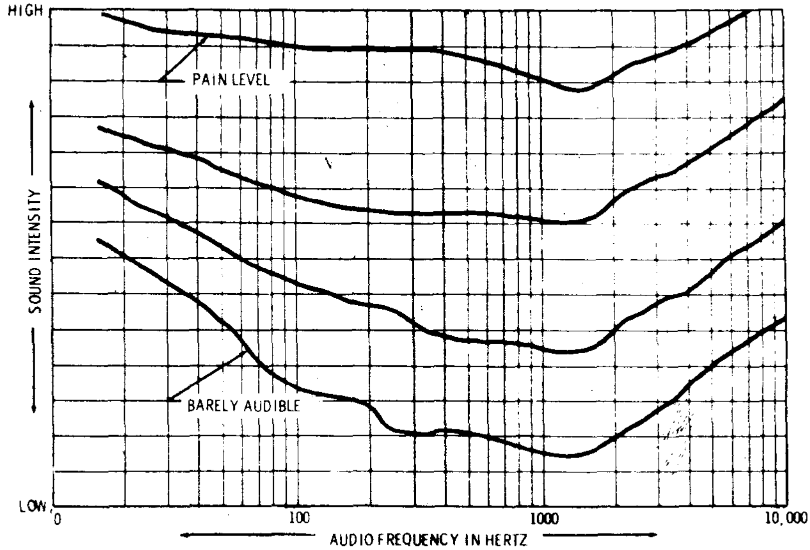


Fig. 1-1. Sound intensity required at different frequencies to produce uniform response in the sensory system of the ear.

between sounds with frequency differences as small as 3 hertz.

Human hearing does not respond linearly to changes in sound intensity (sound pressure) but more nearly logarithmically. That is, a tenfold increase in sound intensity appears to the human to be only about a onefold increase. Thus a natural unit for measuring relative sound intensity would be one that represents about the smallest change in sound level that most people can detect.This unit is called the decibel (dB) and 0 dB corresponds to a sound level just audible. Fig. 1-2 shows the intensities in decibels of familiar sounds.

Since sound may be produced by electrical means, as in hi-fi systems, decibels are used to define and compare levels of power, voltage, and current. For power and sound levels,

$$\text{dB} = 10 \log_{10}\left(\frac{P_1}{P_0}\right) \quad \text{or} \quad 10 \log_{10}\left(\frac{S_1}{S_0}\right)$$

where power level P_1 is compared to power level P_0, or sound intensity S_1 is compared to sound intensity S_0. By convention P_0 is usually taken to be one watt, and S_0 is taken to be the threshold of hearing. When P_0 is one watt, the expression is "decibels referred to one watt" and is written dBW.

Table 1-1 shows power levels and their corresponding dBW values. For current and voltage levels,

$$\text{dB} = 20 \log_{10}\left(\frac{I_1}{I_0}\right) \quad \text{or} \quad 20 \log_{10}\left(\frac{V_1}{V_0}\right)$$

where current I_1 is compared to current I_0 or voltage V_1 is compared to voltage V_0.

The ear can to some degree detect relative phase changes of sound. This means that a small increase or decrease in the frequency, or pitch, of one note in relation to the frequencies of other notes simultaneously played can be detected.

Requirements for the components of a high-fidelity system that can please many individual ears of various listening tastes are quite exacting because of the fineness of the mechanism of the human hearing system.

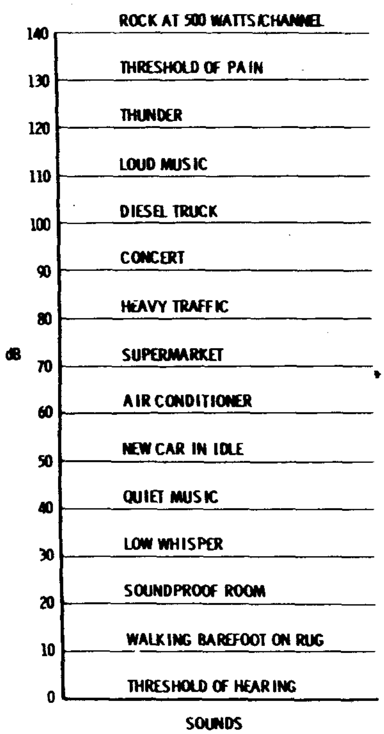


Fig. 1-2. Typical sound intensities in decibels.

DISTORTION OF SOUNDS IN HIGH-FIDELITY SYSTEMS

The best high-fidelity systems are substantially less than perfect. The ways in which the audio-frequency output sound differs from an input or a desired ideal output sound are classified as *distortion.* A complete high-fidelity system may be divided into functional sections, as illustrated in Fig. 1-3. Distortion may be created in one or more of these sections. If more than one section is causing distortion, the final output sound may reflect the sum of the distortions from all distorting sections; however, in other cases, a section may be purposely designed to introduce distortion of a type which compensates for inherent distortion in another section. For example, bass and treble boost circuits can be used to offset (at least partially) the falling-off of response of a speaker at the highest and lowest frequency portions of the response range. Compression, equalization, dynamic range expansion, time delay, reverberation, pulse modulation, digital processing, and control techniques are purposely introduced distortions to the original naked audio signal in modern audio processing; they provide control and improvement of the final effective output sound from the total audio system. These techniques may even take into account the human hearing system.

Table 1-1. Power Levels in dBW

Watts	dBW	Watts	dBW	Watts	dBW
1.00	0	10.0	10	100	20
1.25	1	12.5	11	125	21
1.6	2	16	12	160	22
2.0	3	20	13	200	23
2.5	4	25	14	250	24
3.2	5	32	15	320	25
4.0	6	40	16	400	26
5.0	7	50	17	500	27
6.3	8	63	18	630	28
8.0	9	80	19	800	29

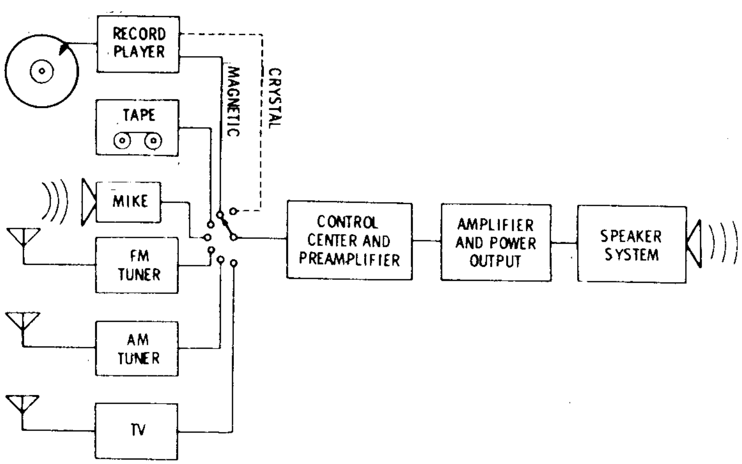


Fig. 1-3. Block diagram of a high-fidelity system.

The high-fidelity system is somewhat like a chain, which is likely to be limited in overall performance by its weakest section; but the chain analogy breaks down in the preceding case, in which the distortion introduced by one section may be used to compensate for distortion in another.

The speaker system is the weakest link in the high-fidelity equipment chain. Its inherent limitations arise primarily from the fact that two conversions of energy must take place: (1) the conversion of electrical energy from the amplifier output stage to mechanical energy in the motion of the diaphragm or cone, and (2) the conversion of mechanical motion to acoustic energy (sound) suitable to the listener's ear. Such energy conversion is known as *transduction*, and the devices which effect it are known as *transducers*. Input devices such as phono pickups and microphones are also transducers and have many of the same weaknesses as speakers, though to a lesser degree because of the relatively low power levels at which they operate. Input devices provide transduction between sound input (or physical motion of a phono-pickup needle) and electrical output, just the reverse of the action in speakers.

The amplifier portion of the system can also contribute distortion if not properly designed or properly used, or not in proper working order. The voltage-amplifier stages are basically the least troublesome. Being of the resistance-coupled type, they usually have good response over the required frequency range with very little distortion. The power-amplifier stage and the output transformer which couples it to the speaker system are ordinarily important contributors to the overall distortion in the system.

Let us consider what we expect from an ideal system and how such a system would perform. The specifications of an ideal system are not easy to state, because, even in the actual attendance of a listener at a concert, the location of his or her seat, the arrangement of the orchestra, and the acoustics of the hall can greatly influence just exactly how the music sounds to the listener. Most of the tastes and reactions of the listener are conditioned by experience and are too complex to be classified in any complete manner.

To keep our discussion concrete and practical, therefore, we must concentrate on those electrical and physical features which distinguish a given system from other systems and which, in the most direct way, provide the information needed by the prospective purchaser of such a system.

The most generally accepted concept of perfection in a high-fidelity system is that which envisions reproduction sounding to the listener exactly as though he or she were present at the location of the original source of the music at the time this music was being recorded or transmitted. Seldom, if ever, will a system approach such a condition, but this is the earnest objective of the high-fidelity enthusiast.

Imperfections are generally classified according to their effects on performance. These effects are as follows:

1. Frequency distortion
2. Amplitude distortion

3. Spatial distortion
4. Phase distortion
5. Transient distortion

Frequency Distortion

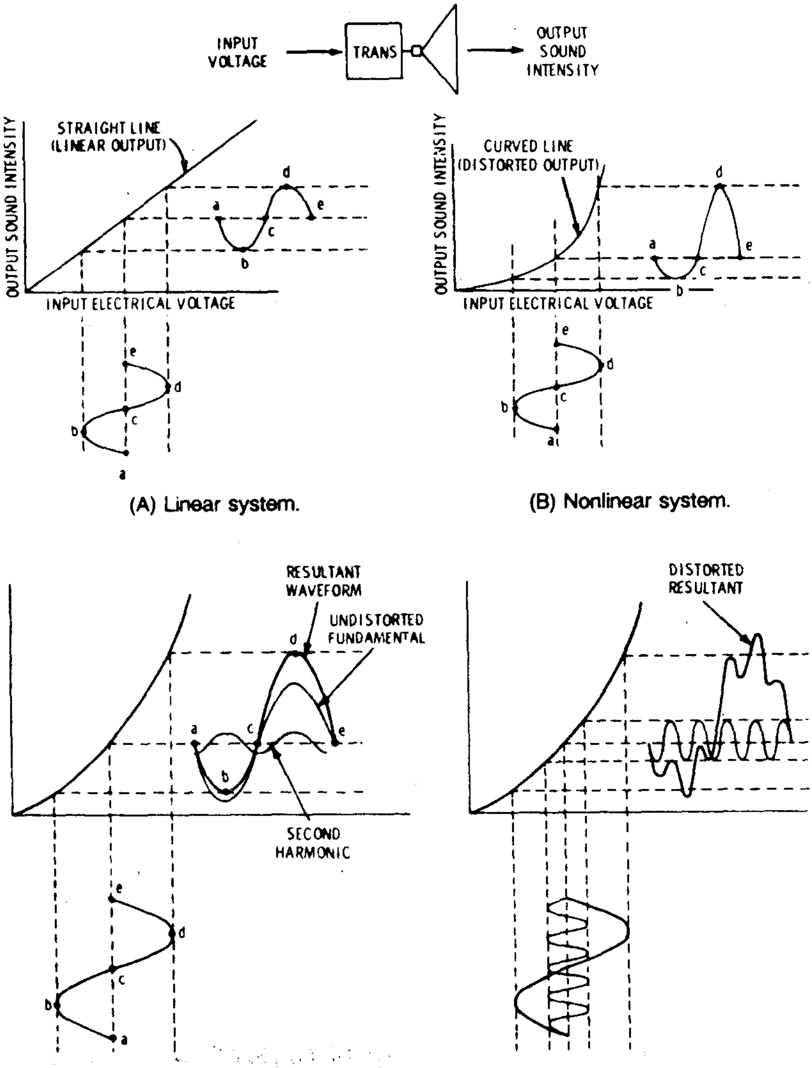
Frequency distortion is the variation of sound output intensity with frequency, for constant input intensity. This ordinarily has the effect of limiting the range of sound frequencies which can be usefully reproduced, or at least of reducing the relative amplitude of certain frequency components so much that the sound loses its naturalness. Frequency distortion may arise electrically in the amplifier, in the transformer which couples it to the speaker system, and in the speaker voice coil. Frequency distortion may arise mechanically in the diaphragm or cone, in its mounting and orientation, and acoustically in the transfer from the diaphragm to the space into which the sound is radiated. Input and storage devices such as tapes, records, phono pickups, tuners, and microphones can also introduce frequency distortion.

Amplitude Distortion

Amplitude distortion is the failure of the instantaneous amplitude (intensity) of the sound output of any or all frequencies to be directly proportional to the instantaneous amplitude of the electrical signal input. An ideal system would have an output-versus-input amplitude relation which is plotted as a straight line and is thus *linear.* For example, in a linear system, doubling the voltage (or current) of the electrical input would double the intensity of the sound output. Tripling the electrical input would triple the output. Any variation (an increase or decrease) of the input will cause a corresponding proportional change in the output. Since in this ideal system the output is always directly proportional to the input, no amplitude distortion is introduced, and the waveform of the sound-pressure (intensity) output is an exact replica of the electrical voltage or current input waveform.

Practically, however, some amplitude distortion is introduced at some point or points in every system, so that the input-output relation cannot be plotted as a straight line; the relation is thus nonlinear to some degree. For this reason, this type of distortion is often referred to as *nonlinear distortion.*

The effects of linearity on a sine-wave input are shown in Fig. 1–4. The characteristic of the linear system is illustrated in Fig. 1–4A. With the straight-line characteristic, the ratio between input voltages before and after any change is the same as the ratio between resulting sound intensities. For example, the variation *a-b-c* on one side of zero is the same as variation *c-d-e* on the other side, for both input and output. On the other hand, this is not true for the nonlinear system illustrated in Fig. 1–4B. There, the curvature of the characteristic is such that portion *a-b-c* of the input signal produces a much smaller variation of output sound intensity than does portion *c-d-e.* The output waveform is therefore distorted. Its nonsinusoidal characteristic indicates that harmonic distortion (a form of amplitude distortion) has been introduced.



(A) Linear system.

(B) Nonlinear system.

(C) Harmonic distortion.

(D) Intermodulation distortion.

Fig. 1-4. Effects of linear and nonlinear systems on sine-wave inputs.

Nonlinearity in the output section of audio systems can result from poor or limited design in the output transformer, from changes in the radiation efficiency of the diaphragm or cone due to flexing with amplitude, from a change of effective magnetic-flux density with a change in the motion of the voice coil or diaphragm, and from nonlinearity of the air. (The ratio between compressed volume and compressing force is not constant with variation of level.) Poorly designed input devices, amplifiers, and control sections can introduce nonlinearity as the sound passes through the system.

There are two main types of amplitude distortion: harmonic distortion and intermodulation distortion.

Harmonic Distortion—Harmonic distortion results from the fact that passage of a signal through a nonlinear system generates frequency components not present in the original signal and having frequencies which are integral multiples (1, 2, 3, 4, etc., times) of the frequency of the signal from which they are generated. For example, a nonlinear system to which a pure sine-wave electrical signal of 400 Hz is applied would generate and radiate sound energy at such frequencies as 800, 1200, and 1600 Hz in addition to that at 400 Hz. Fig. 1-4C illustrates harmonic distortion due to a nonlinear system. Notice that the output waveform, which is "flattened" somewhat on the negative alternation and "peaked" somewhat on the positive alternation, is a *combination of the undistorted fundamental and its second harmonic. If both* alternations had been "flattened," the output would have contained the fundamental and the third harmonic. More complex waveforms contain higher-order harmonics.

Intermodulation Distortion—When two pure sine-wave signals of different frequencies are applied to a good speaker system, they should have no effect on one another and should appear separate and distinct as sound output components. In a nonlinear system, however, the two signals heterodyne in the same way as the oscillator and incoming signals in the mixer of a superheterodyne receiver; they produce new undesired frequency components with frequencies equal respectively *to the sum and difference* of the frequencies of the original sine-wave signals. The harmonics arising from harmonic distortion are also obtained, along with frequency components of the sums and differences of these harmonics. This mixing process is similar to that used in modulation; hence its designation is *intermodulation distortion.*

Introduction of intermodulation distortion by nonlinearity is illustrated in Fig. 1-4D. In this figure two input signals are applied to a nonlinear system. At this point, the two signals intermodulate and produce a waveform with both harmonic and intermodulation distortion. Although the original frequency components are still present, the output signal also contains new distortion components *with frequencies which* are *respectively* equal to the sums and differences of the frequencies of the applied input signals. Harmonic-distortion components are also present. The same factors in audio-system design and construction which cause nonlinearity and produce harmonic distortion also cause intermodulation distortion.

Spatial Distortion

Spatial distortion is the difference between the audio sound output from the reproducing system and the audio sound input to the ear. It may be caused by room-resonance standing waves, absorption or reflection of frequencies from walls, floor, ceiling, and objects or holes in a room, or by changing relationships between system components, the listener, and the enclosure. It may manifest itself to the listener as a real or apparent wrong location of the source of sound. This arises either from a narrow directivity characteristic of the speaker or from the failure of the system to simulate the true spatial distribution of the sources of sounds being reproduced. The directivity characteristic is a feature of the speaker system alone; on the other hand, spatial location of sound sources to simulate the original can be obtained in the speaker system only when the original sound is transmitted or recorded with this in mind. Binaural and stereophonic techniques are examples of the latter.

Phase Distortion

Phase distortion is distortion resulting when the different frequency components are reproduced in improper time relation to each other. The causes of phase distortion are generally the same as the causes of frequency distortion, and substantial frequency distortion is practically always accompanied by phase distortion. One type of phase distortion is the Doppler effect, which is exemplified by the apparent change in pitch of a train whistle as the train passes by.

Transient Distortion

Transient distortion is failure of a system to exactly follow sudden large changes in sound level. If the speaker system is not properly designed, pulses of sound energy tend to shock the system into oscillation at its natural frequency. The flywheel effect of these oscillating circuits causes the oscillation to continue after the true pulse which excited it has ceased. This effect is often referred to as *hangover.*

Two common forms of transient distortion are transient intermodulation distortion and slew induced distortion. These two distortions are related to the response time of individual audio circuits. Each circuit in a hi-fi system has a certain response mode, which, among other things, includes its rate of output voltage rise versus its rate of excitation voltage rise; that is, the lag of response to the input signal. This lag is measured in microseconds. It is caused partly by the inherent characteristics of the circuit and its components but mostly by the negative feedback provided for purposes of reducing other types of distortion.

The time lag of the feedback signal is caused by the impedance of the feedback loop from the original input through the amplifying stages and back to the return point. If the feedback could be returned to the input instantaneously, the overall lag would be negligible, but zero time delay is not possible. The greater the negative feedback, the greater the effect of such delay on transient intermodulation distortion and its slewing rate.

The *slew rate* of a circuit is a measure of voltage rise with respect to the elapsed time in microseconds of the rise. The voltage is that of the leading edge of an output pulse responding to an instantaneous standard input pulse. The slew rate is expressed in volts per microsecond (V/μs) for 80 percent of