Chapter 17
Distortion and Noise

Audio has made it possible to record sounds and reproduce them with reasonable fidelity, to amplify and modify sounds beyond our unassisted human powers, and to broadcast sounds using electromagnetic radiation. Audio has greatly magnified our abilities to control and use sound. It has also led to the twin evils of noise and distortion.

17.1 Noise

When a sound is recorded and reproduced, or when it is otherwise processed or transmitted electronically, it can be contaminated by noise. Noise is something that is added to the signal. You will recognize noise as the "hiss" that you hear on the telephone or radio. You know that the original speech did not contain that hissing sound. The hiss was added because of imperfection in the recording or transmission path of the speech signal. This kind of noise is broadband; it has components at all frequencies. It is caused by randomness in the electronic components because components do not behave in a completely deterministic way. It is a general rule that when signals are processed in their analog form (not digital) then the more complicated the signal path, the more noise one is likely to add. Sometimes the definition of noise is divided into "broadband noise"—meaning the hiss due to randomness, and "hum"—meaning a low-frequency tonal component.

Hum comes from electrical power. Electricity is distributed around the USA as a 60-Hz (60-cycle per second) sine wave, normally with a voltage between 110 and 120 V. It is said to be alternating current (AC) because it alternates from positive to negative. Audio electronic devices are powered by this AC electricity and some of it may leak into the signal path, either due to inadequate design of the equipment or to deterioration of the electronic components. Furthermore, the power lines radiate

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a 60-Hz electromagnetic field, and this can be picked up by cables or by transducers such as dynamic microphones or tape playback heads. Harmonics of 60 Hz, the second at 120 and the third at 180 Hz, also appear in hum.

17.2 Distortion

A signal processing system generates distortion if the waveform coming out of the system does not have exactly the same shape as the waveform going into the system. Imagine a simple acoustical tone that is transformed into an electrical signal by a microphone. We could display that signal on an oscilloscope to determine its exact shape. Now suppose that the signal is recorded onto tape, transmitted through the telephone wires and finally broadcast across the country. We could examine the final signal with an oscilloscope and determine the final waveform shape. If the electronic processing is accurate, the shape should be the same as the original. If it is not, then the signal has been distorted in some way.

17.2.1 Distortion Not

There are several kinds of signal changes that occur in electronic processing that should not be classified as distortion. A simple change in gain, where the signal is multiplied by a constant, is not distortion. This change does not change the shape of the waveform. The effect of the gain change can be compensated by amplification or attenuation.

Inversion, whereby positive is turned into negative and negative is turned into positive, is equivalent to multiplying the signal by -1 and it is not distortion either. The frequency content of an inverted signal is identical to original. Amplification and inversion appear in parts (b) and (c) of Fig. 17.1.

Other kinds of waveform changes are properly called distortion. Distortion may be linear or nonlinear. We deal with these in turn.

17.2.2 Linear Distortion

Linear distortion can be separated into two categories, amplitude distortion and phase distortion.

Amplitude distortion: Ideally, a signal transmission system should have a flat frequency response. That means that it boosts or attenuates all frequencies equally. Linear distortion occurs when the frequency response of the processing system is not

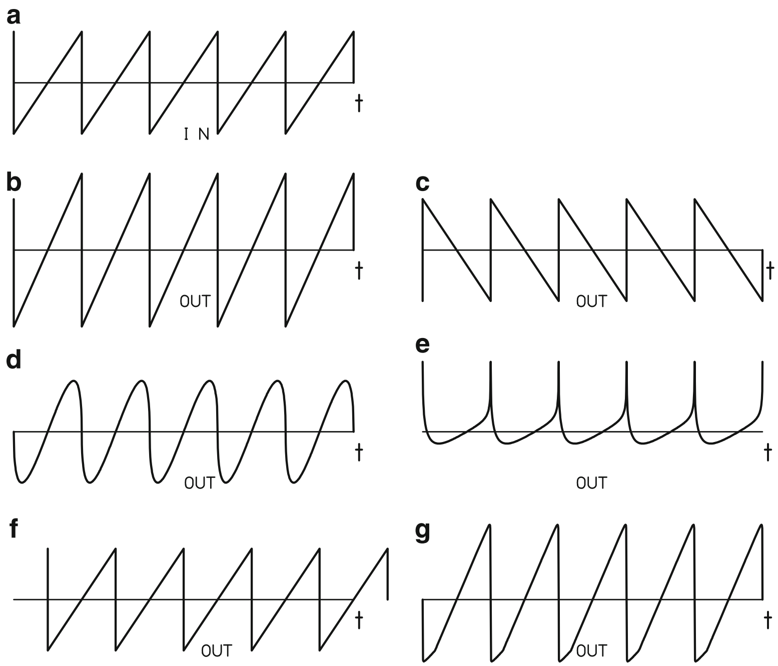

Fig. 17.1 One input and half a dozen outputs from different processing systems: (**a**) The input is a sawtooth wave as a function of time. (**b**) The output of a system with gain—no distortion. (**c**) The output of an inverting system—no distortion. (**d**) The output of a lowpass filter, which reduces the amplitudes of high-frequency components compared to low-frequency components. Consequently the output waveform is smoother than the input—linear distortion. (**e**) The output of a filter that introduces phase distortion. The high-frequency harmonics are just as strong as in the input sawtooth, but their phases are changed, changing the shape of the waveform—linear distortion. (**f**) The output of a time-delay device—no distortion. (**g**) The output of a saturating device that resists extremes, especially negative extreme values tending to cut off the bottom of the waveform—nonlinear distortion

flat—when the system operates like a filter of some sort, giving special emphasis to some frequencies. This effect is known as “amplitude distortion.”

Phase distortion: Ideally, a signal transmission system should have no dispersion. That means that if the system delays a signal at all, then it delays all frequencies equally. Such a delay does not distort the shape of a signal. However, if there is dispersion then some frequencies are delayed more than others, then the shape is changed. The change is known as “phase distortion.”

Linear distortion is not necessarily a serious matter. Every time you adjust the tone controls on a radio you are distorting (or un-distorting) the signal in a linear way. Linear distortion is part of the normal operation of transducers such as microphones. Theoretically, linear distortion can always be made to go away by filtering the signal. For instance, if the nature of the distortion is to reduce the high frequency content then one can overcome that distortion with a high-boost filter. It may not always be easy to compensate for linear distortion if the processing system is very complicated, but in principle one can always do it. All it takes is a sufficiently specialized filter at the receiving end.

17.2.3 Nonlinear Distortion

Although linear distortion may be rather benign, nonlinear distortion is not. It is the most serious kind of distortion because it can be highly unpleasant and there is no simple way to get rid of it. In general, nonlinear distortion cannot be removed by filtering.

Nonlinear distortion by a device or process results in the addition of extra frequencies to the output of the device or process. That is the most significant difference between nonlinear distortion and linear distortion. For instance, if the input to a device has spectral power at frequencies of 100, 223 and 736 Hz then the output of a linear device will have power at only those three frequencies. The amplitudes may be boosted or attenuated differently, the phases may be scrambled, but there won't be any new frequencies. With nonlinear distortion there will be at least a few new frequencies added, and normally a lot of new frequencies will be added.

The Origins of Distortion Imagine a clock pendulum that is slowly oscillating back and forth,smoothly and unobstructed. Clearly the motion is periodic. It is nearly sinusoidal, but not exactly sinusoidal, and a complete description of the motion would include a few harmonics. Now imagine that someone puts up a barrier so that just before the pendulum reaches its maximum displacement to the right it runs into the barrier—bonk! The pendulum motion continues, it even continues to be periodic, but the pattern of motion is obviously changed by the obstruction—the pattern has been distorted. One might say that the pattern has been "clipped" in the sense that the barrier prevents the pendulum from reaching the full extent of vibration.

Distortion occurs in a loudspeaker driver if you use a finger to prevent the loudspeaker cone from reaching the maximum displacement that the audio signal wants the cone to make. This kind of distortion occurs in a driver, even without interference from the outside, because physical limits in the cone mounting prevent the cone from making very large displacements. If the audio signal sent to a loudspeaker is very large, the signal will try to cause the cone to make extreme displacements and gross nonlinear distortion will occur.

Distortion occurs in audio devices that are purely electronic when the limits of the electronic circuitry are exceeded by a signal that is amplified too much. The usual way to study nonlinear distortion by a process or device is to use one or two sine waves as inputs—thus only one or two frequencies. If there are more, the situation becomes very complicated.

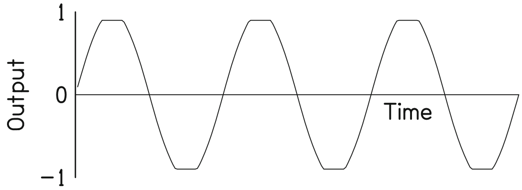

Fig. 17.2 A sine tone suffering symmetrical clipping. Only odd-numbered harmonics (3, 5, 7, ...) are produced by this distortion

Harmonic Distortion If a sine tone is put into a system that clips off the top there is a change in the waveshape. The tone remains periodic because every cycle of the sine tone has its top clipped off in the same way. Therefore the period is not changed by this clipping, but the signal is no longer a sine tone; it is now a complex tone. The complex tone has a fundamental frequency equal to the frequency of the original sine tone, and now it has harmonics. This is called harmonic distortion. This kind of clipping can be expected to generate harmonics of all orders, both even and odd. The second harmonic or the third will normally be the strongest.

If the nonlinear system clips off the top and the bottom of the sine tone symmetrically, as in Fig. 17.2, then the output includes only odd-numbered harmonics as distortion products.

Intermodulation Distortion If you put two sine tones (frequencies f_1 and f_2) into a nonlinear system, you expect there to be harmonic distortion products, with frequencies $2f_1$ and $2f_2$, also $3f_1$ and $3f_2$, and so on. However, there is more. There are also summation tones and difference tones generated by the nonlinearity. Summation tones occur at frequencies generically given by the formula $mf_2 + nf_1$, where m and n are integers. Frequencies $f_2 + f_1$ and $2f_2 + 3f_1$ are examples.

Difference tones occur at frequencies generically given by the formula $mf_2 - nf_1$, where m and n are integers. The most important difference tones are usually the following three: $f_2 - f_1$, $2f_1 - f_2$, and $3f_1 - 2f_2$. In using these formulas, we consider f_2 to be the larger frequency and f_1 to be the smaller. Normally, then, the formulas will lead to positive frequencies for the distortion tones. However, if one of the numbers you calculate turns out to be negative, you can find a legitimate distortion frequency simply by reversing the sign and making it positive.

Of all these distortion products, the difference tones are the worst. They are easy to hear and are the among the most objectionable features of poor audio. As an example, if two sine tones, with frequencies of 2,000 and 2,300 Hz are put into a nonlinear device, the difference tones $f_2 - f_1$, $2f_1 - f_2$, and $3f_1 - 2f_2$ become, 300 Hz, 1,700 Hz, and 1,400 Hz, as shown in Fig. 17.3.

Distortion in the Cochlea The neural transduction that takes place in the cochlea is a highly nonlinear operation. Therefore you expect that the cochlea generates distortion products. For instance, it should generate harmonic distortion. However, the curious nature of wave propagation in the cochlea saves us listeners from that distortion. You will recall that in the cochlea, sounds travel from the high-frequency end (base) toward the low-frequency end (apex). Suppose that there is a 200 Hz tone that is being transduced by the cochlea. Because 200 Hz is a rather

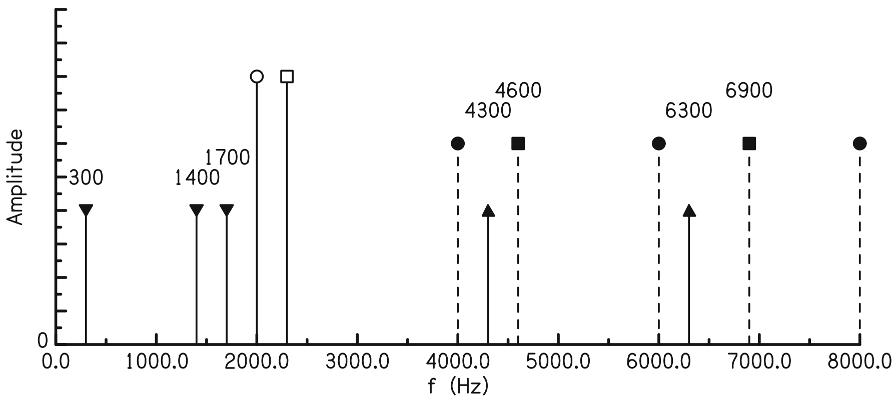

Fig. 17.3 The amplitude spectrum of the output of a nonlinear device when the input to the device is a two-tone signal with components $f_1 = 2,000$ Hz and $f_2 = 2,300$ Hz. The output includes the original frequencies, $f_1 = 2,000$ Hz and $f_2 = 2,300$ Hz, as shown by *open symbols*. The output also includes distortion products: (1) Harmonics of 2,000 Hz, shown by *filled circles*. (2) Harmonics of 2,300 Hz, shown by *filled squares*. (3) Difference tones, shown by *downward pointing triangles*, $f_2 - f_1 = 300$ Hz, $2f_1 - f_2 = 1,700$ Hz, and $3f_1 - 2f_2 = 1,400$ Hz. (4) Summation tones, shown by *upward pointing triangles*, $f_1 + f_2$ at 4,300 Hz, and $2f_1 + f_2$ at 6,300 Hz. There are other distortion components above 8,000 Hz, out of the range of the figure. They include higher harmonics of 2,000 and 2,300 Hz and more summation tones, such as $3f_1+2f_2$

low frequency, this transduction takes place near the apex. That's where harmonic distortion products, 400, 600, 800 Hz, ... are created. But in order to be heard these products need to travel to places tuned to these frequencies, and those places are further toward the base where the 200-Hz component is already present and growing. Those places are busy passing the 200-Hz component and respond mostly to it. Therefore, we are not troubled by harmonic distortion.

On the other hand, difference tones have lower frequencies than the input frequencies and they successfully travel to places further along the basilar membrane where the original tones do not create much excitation. Difference tones can be heard. Clear examples of audible distortion products are described in Exercises 6 and 7.

If you want to investigate difference tones created by the ear's own nonlinear distortion, you should start with two sine tones that have completely independent audio paths, from the generators through the loudspeakers. Otherwise, you can't be sure that the distortion you hear is coming from the ear. It might just be coming from the audio equipment. (You can be pretty sure that the air between two loudspeakers and your ears mixes tones from the loudspeakers in a linear way.) An alternative to two independent loudspeakers is a flute duet, where the conditions of Exercises 6 and 7 can easily occur. Audible distortion is actually one of the very serious problems of flute duets.

Distortion in Rock Guitar Distortion is an essential element of the classic rock and roll sound,especially guitar sound. The earliest form of distortion is known as "overdrive," which just means that somewhere along the electronic processing path the signal has been clipped, as in Fig. 17.2. The clipping does not need to be symmetrical, though it often is because early guitar amplifiers clipped in that way and people liked the sound. The clipping can be gentle in which the peaks and valleys are slightly rounded or it can be harsh where peaks and valleys are seriously cut off. Harsh clipping is characteristic of *fuzz boxes*. Guitar distortion is most effective on a strum of all the strings because playing many notes at once, each note with its harmonics, leads to an uncountably large number of intermodulation distortion products and a thick sound. In modern rock guitar sound, distortion in the form of clipping is combined with filtering (often filtering that changes in time) and other time dependent effects such as *phasing* or *flanging*. Fuzz boxes have evolved into "effects" boxes.

17.3 Dynamic Range

The typical electronic signal processor and the typical audio transducer are very linear if the signal is weak. Therefore, it should be easy to avoid the bad effects of nonlinear distortion simply by keeping the signal small. The problem with this approach is that such a small signal may not be large enough compared with the noise added in audio processing. Thus, the signal-to-noise ratio may suffer. The obvious answer to the noise problem is to make the signal larger, but then the system may distort.

A practical example of this dilemma has a singer on stage with a microphone, 50 m away from the microphone amplifier at the back of the hall. If the singer is far from the microphone or sings quietly, the output of the microphone is less than a few millivolts (a few one-thousandths of a volt). The amplifier at the back of the hall needs to multiply that signal by more than 1,000 to get a useful signal of a few volts. In doing so, the amplifier also amplifies the hum and noise picked up by 50 m of microphone cable. That creates a noise problem, characterized by a poor signal to noise ratio. A solution to the noise problem is for the singer to put the microphone close to his mouth and yell. Now the amplifier only needs to amplify by a factor of 100 and the signal to noise ratio improves by 20 dB. Unfortunately, the diaphragm of the microphone cannot readily move to the extreme excursions required by this intense signal source and the microphone causes distortion.

That then, is the nature of the problem. Audio processing must sail the straits between noise on one side and distortion on the other. The width of the straits (to maintain the nautical analogy) is the dynamic range of the system. The dynamic range is the difference, expressed in dB, between the level of an intense signal that causes distortion (say 0.1% distortion) and the level of the noise background. Fortunately, advances in digital electronics have dramatically expanded the dynamic range of audio systems.

Exercises

Exercise 1, Less hum across the pond

European countries use electrical power with a frequency of 50 Hz. Do you expect that "hum" is less audible in European audio gear?

Exercise 2, Too much of a good thing

Why does nonlinear distortion occur if an amplifier is "overdriven."

Exercise 3, Rock 'n' roll

(a) How does nonlinear distortion in the amplifier make chords played on an electric guitar sound "thicker?" (b) If nonlinear distortion makes music sound bad why would the rockers deliberately choose distortion?

Exercise 4, Choose a test signal

You will use your ears to evaluate an audio system for nonlinear distortion. You have a choice of signal sources to put into the system. What do you choose? (a) broadband noise? (b) a complex periodic tone with many harmonics? (c) a sine tone with frequency of 5,000 Hz? (d) a sine tone with frequency of 500 Hz?

Exercise 5, Harmonic distortion

A 1,000-Hz sine tone is put into a nonlinear device causing harmonic distortion. What is the fundamental frequency of those harmonics?

Exercise 6, All the distortion components

You put the sum of two sine tones into a nonlinear device. The sine tone frequencies are 1,000 and 1,200 Hz. What are the frequencies of harmonic distortion products that emerge? What are the frequencies of difference tones that emerge? What are the frequencies of summation tones that emerge? What frequencies are likely to be most audible?

Exercise 7, Worse and worse with up and down

The sine tone frequencies from Exercise 6, 1,000 and 1,200 Hz form the musical interval known as a minor third (See Appendix D if you want to know more about musical intervals.) Suppose the upper frequency is changed from 1,200 to 1,250 Hz. The frequencies of 1,000 and 1,250 Hz make the interval of a Major third. What happens to the difference tone given by $2f_1 - f_2$ when this change occurs?

Exercise 8, Dynamic range

An audio system produces 0.1% harmonic distortion (mostly second and third harmonics) when the level is 90 dB. The noise background is 10 dB. What is the dynamic range of this system?

Exercise 9, Audio amplifier makes toast

This chapter says that electrical power from the outlet in the wall is a 60-Hz sine wave. How is that different from a 60-Hz signal created by a computer sound card and then passed through a power amplifier? If it's not different, why can't you connect a toaster to the power amplifier of your stereo system? (You'd make the connection to the power amplifier just where you would normally connect the loudspeakers.)

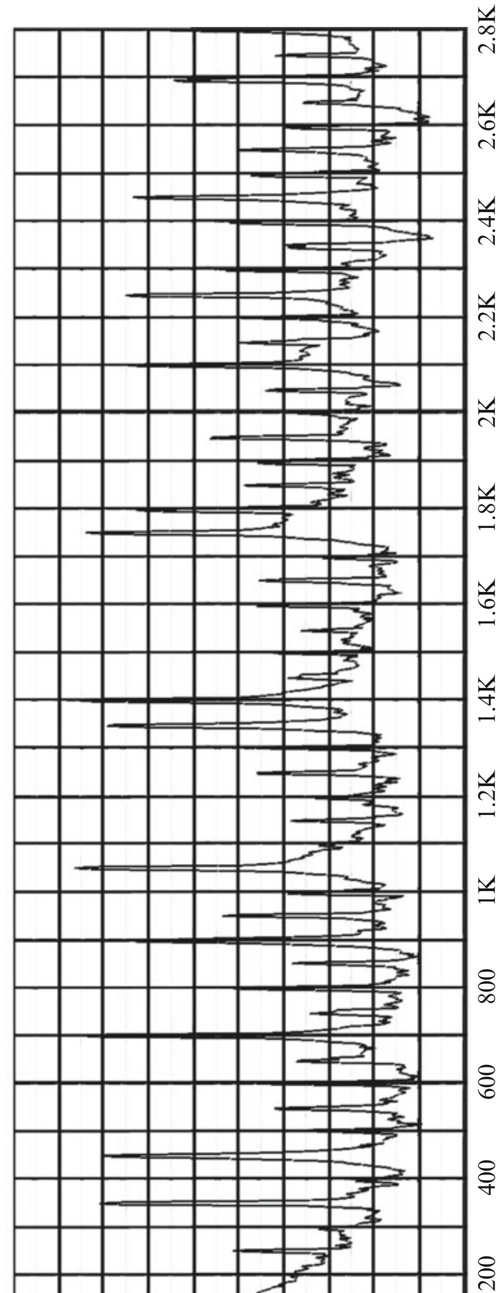

Fig. 17.4 The spectrum of a recorded car horn. (It was a Buick.) The measured sound level was 98 dB, as measured with an A-weighted sound level meter

Exercise 10, Thinking like an acoustician—The car horn

Acousticians try to find physical explanations for human perceptions of sound qualities. The sound of a car horn is an example. The acoustician thinks that the perception can characterized by a loudness and a tone color.

1. Assuming that the spectrum is broadband, having both low and high frequencies, an intuitive approximate sense of the loudness will be gained from a decibel reading on a sound level meter. That is a single, easy measurement to make: Put a sound level meter 3 m in front of a car, blow the horn, and read the meter.
2. The tone color will be quantified by recording the sound electronically and studying its spectrum, as shown in Fig. 17.4. The horizontal axis shows frequency in Hertz. The vertical axis shows the levels of the spectral components in decibels, but the scale is arbitrary. No problem. You don't really need the absolute values of dB on the spectral plot because you already have a sense of scale from the reading you made with the sound level meter. Instead, you are concerned with the relative levels of the different components. It is typical to take the tallest peak as a reference level. Then all the other peaks have levels that are negative. In the case of the car horn, however, it makes sense to take the component at 350 Hz as a standard for the level because it is both a tall peak and a low-frequency peak.

(a) First, consider the frequencies of tall peaks in the spectrum from 0 to 2,800 Hz. Do you see some mathematical relationships in the list of frequencies? Do you see harmonics? What does this tell you about the car horn sound?
(b) If you find more than one fundamental frequency, what does this suggest about how car horns are made or installed?
(c) If you find that there are two fundamental frequencies, determine the ratios of those frequencies. Compare with the ratios in Appendix D to determine the musical interval in the car horn sound.
(d) Can you account for all the peaks in the spectrum by assuming that they are harmonics?
(e) Can you see some distortion components? What are their frequencies and how can you account for them?

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