CHAPTER 17

Comb-Filter Effects

The term *comb filter* has been widely used in the popular audio press as an explanation of delayed reflection effects. Comb filtering is a steady-state phenomenon. It has limited application to music and speech, which are highly transient phenomena. With transient sounds, the audibility of a delayed replica is more the result of successive sound events. A case might be made for proper application of combing to brief snatches of speech and music that approach steady state, but already there is an etymological impasse. The study of the audible effects of delayed reflections is better handled with the generalized threshold approach of Chap. 16.

What Is a Comb Filter?

A filter changes the shape of the frequency response or transfer function of a system. An electronic circuit used to shape the frequency response of a system to achieve a certain desired end could be a filter. A filter could also be a system of pipe and cavities used to change an acoustical system, such as is used in some microphones to adjust the pattern.

In the early days of multitrack recording, experimenters were constantly developing new, different, and distinctive sounds. *Phasing* and *flanging* were popular words among these experimenters.[1] At first multiple-head tape recorders were used to provide delayed replicas of sounds that were then mixed with the original sound to produce some unusual and eerie effects. Currently special electronic circuits are used to generate these delays. Whatever the means, these audible colorations of sound are the result of comb filters.[2]

Superposition of Sound

A sound contractor was concerned about the aiming of his horns in a certain auditorium. The simplest mechanical mounting would cause the beam of one horn to cut across the beam of the other horn. What happens in that bit of space where the two beams intersect? Would the beams tend to spread out? Would sound energy be lost from the beams as one beam interacts with the other? Relax—nothing happens.

Imagine a physics lab with a large, but shallow, ripple tank of water on the lecture table. The instructor positions three students around the tank, directing them to drop stones in the tank simultaneously. Each stone causes circular ripples to flow out from the splash point. Each set of ripples expands as though the other two ripple patterns were not there.

The principle of superposition states that every infinitesimal volume of the medium is capable of transmitting many discrete disturbances in many different directions, all simultaneously and with no detrimental effect of any one on the others. If you were able to observe and analyze the motion of a single air particle at a given instant under the influence of several disturbances, you would find that its motion is the vector sum of the various particle motions required by all the disturbances passing by. At that instant, the air particle moves with amplitude and direction of vibration to satisfy the requirements of each disturbance just as a water particle responds to several disturbances in the ripple tank.

At a given point in space, assume an air particle responds to a passing disturbance with amplitude A and 0° direction. At the same instant another disturbance requires the same amplitude A, but with a 180° direction. This air particle satisfies both disturbances at that instant by not moving at all.

A microphone is a rather passive instrument. Its diaphragm responds to whatever fluctuations in air pressure occur at its surface. If the rate of such fluctuations (frequency) falls within its operating range, it obliges with an output voltage proportional to the magnitude of the pressure involved. In Fig. 17-1, a 100-Hz tone from loudspeaker A actuates the diaphragm of a microphone in free space, and a 100-Hz voltage appears at the microphone terminals. If a second loudspeaker B lays down a second 100-Hz signal at the diaphragm of the microphone identical in pressure but 180 degrees out of phase with the first signal, one acoustically cancels the other, and the microphone voltage falls to zero. If an adjustment is made so that the two 100-Hz acoustical signals of identical amplitude are in phase, the microphone delivers twice the output voltage, an increase of 6 dB. The microphone slavishly responds to the pressures acting on its diaphragm. In short, the microphone responds to the vector sum of air pressure fluctuations impinging upon it. This characteristic of the microphone is intimately involved in acoustical comb-filter effects.

Tonal Signals and Comb Filters

A 500-Hz tone is shown as a line on a frequency scale in Fig. 17-2A. All of the energy concentrated in this pure tone is located at this frequency. Figure 17-2B shows an identical signal except it is delayed by 0.5 ms in respect to the signal of A. The signal has the same frequency and amplitude, but the timing is different. Consider both A and B as acoustical signals combining at the diaphragm of a microphone. Signal A could be a direct signal and B a reflection of A off a nearby sidewall. What is the nature of the combined signal the microphone puts out?

Because signals A and B are pure tones, simple sine waves, both vary from a positive peak to a negative peak 500 times per second. Because of the 0.5 ms delay, these two tonal signals will not reach their positive or negative peaks at the same instant. Often along the time axis both are positive, or both are negative, and at times one is positive while the other is negative. When the sine wave of sound pressure representing signal A and the sine wave of sound pressure representing signal B combine (with due respect to positive and negative

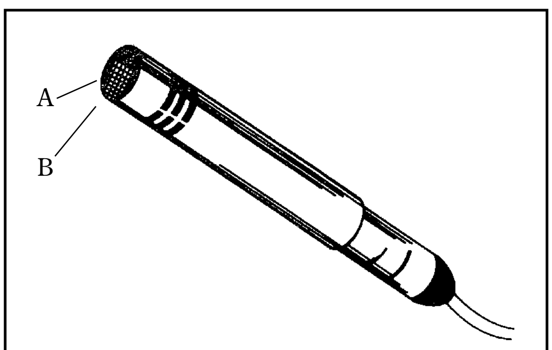


FIGURE 17-1

The microphone diaphragm responds to the vector sum of sound pressures from multiple sources.

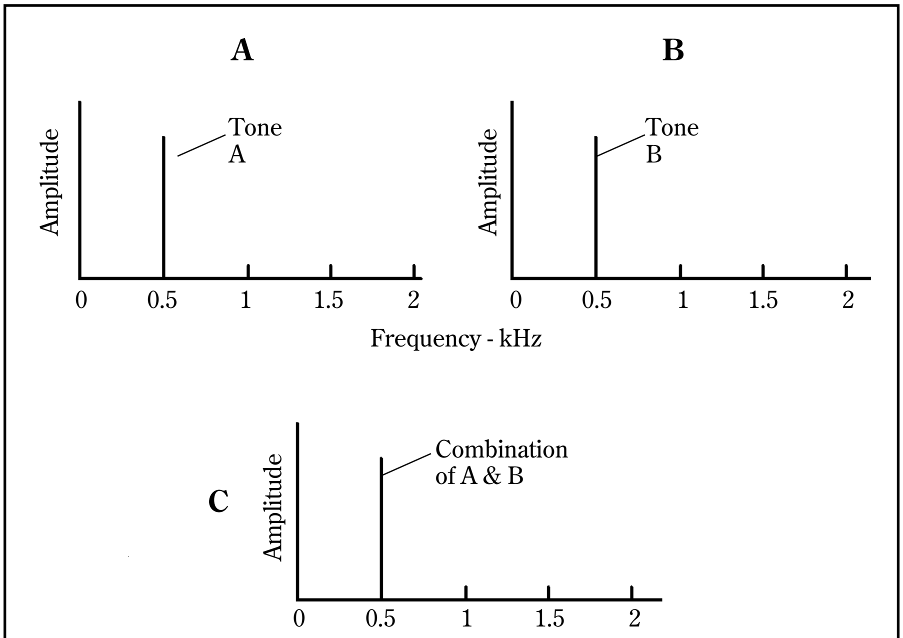


FIGURE 17-2

Tonal signals and comb filters; (A) a sine wave having a frequency of 500 Hz, (B) another sine wave of 500 Hz that is delayed 0.5 ms from A, and C is a summation of A and B. The 500-Hz signal and its delayed counterpart reach their peaks at slightly different times, but adding them together simply yields another sine wave; there is no comb filtering (see Figs. 1-11, 1-13).

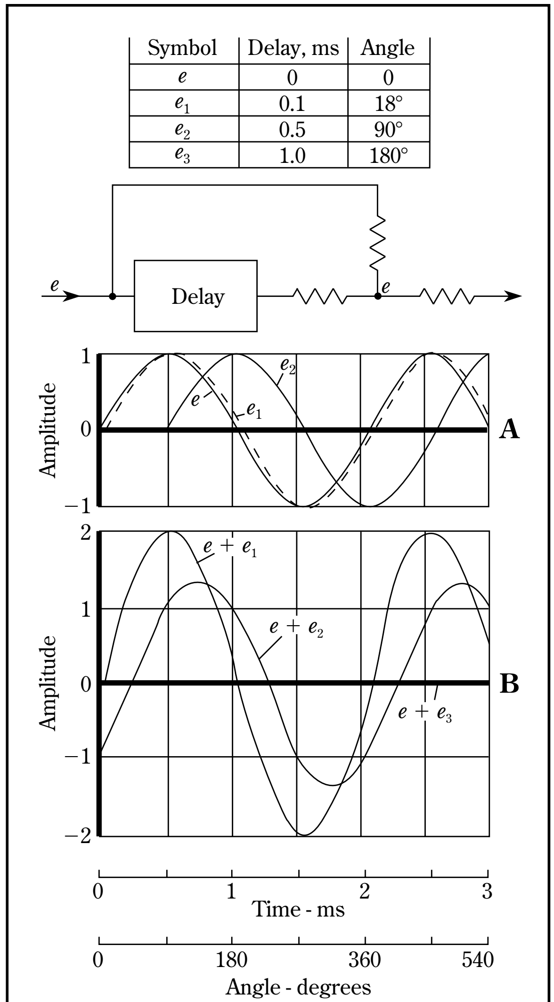
signs) they produce another sine wave of the same frequency, but of different amplitude.

Figure 17-2 treats the two 500-Hz tones as lines on a frequency scale. Figure 17-3 treats the same 500-Hz direct tone and the delayed tone in the more familiar sine-wave form. The delay is accomplished by feeding the 500-Hz tone into a digital delay device and combining the original and the delayed tones in a common three-resistor summing circuit.

In Fig. 17-3A the direct 500-Hz tone is shown originating at zero time. It takes 2 ms for one cycle of a 500-Hz tone ($1/500 = 0.002$ sec). One cycle is also equivalent to 360 degrees. The 500-Hz signal, *e*, is plotted according to the time and degree scales at the bottom of the figure.

A delay of 0.1 ms is equivalent to 18 degrees; a delay of 0.5 ms is equivalent to 90 degrees; a delay of 1 ms is equivalent to 180 degrees. The effect of these three delays on the tonal signals is shown in Fig. 17-3B (later the same delays will be compared with music and speech signals). The combination of e and e_1 reaches a peak of approximately twice that of e (+6 dB). A shift of 18 degrees is a very small shift, and e and e_1 are practically in phase. The curve $e + e_2$, at 90 degree phase difference has a lower amplitude, but still a sine form. Adding e to e_3 (delay 1 ms, shift of 180 degrees) gives zero as adding two waves of identical amplitude and frequency but with a phase shift of 180 degrees results in cancellation of one by the other.

Adding direct and delayed sine waves of the same frequency results in other sine waves of the same frequency. Adding direct and delayed sine waves of different frequencies gives periodic waves of irregular wave shape. *Conclusion:* Adding direct and delayed periodic waves does not create comb filters. Comb filters require signals having distributed energy such as speech, music, and pink noise.



Symbol	Delay, ms	Angle
e	0	0
e_1	0.1	18°
e_2	0.5	90°
e_3	1.0	180°

FIGURE 17-3

An exercise to demonstrate that combining sine waves yields not comb filters, but simply other sine waves. A distributed spectrum is required for the formation of comb filters. 500 Hz sine waves are displayed with delays of 0.1, 0.5, and 1.0 ms to conform to the distributed spectrum cases in Fig. 17-5.

Combing of Music and Speech Signals

The spectrum of Fig. 17-4A can be considered an instantaneous slice of music, speech, or any other signal having a distributed spectrum. Figure 17-4B is essentially the same spectrum but delayed 0.1 ms from Fig. 17-4A. Figure 17-4C is the acoustical combination of the A and B sound pressure spectra at the diaphragm of the microphone. The resulting overall response of Fig. 17-4C appears like a sine wave, but combining spectra is different from combining tonal signals. This sine-wave appearance is natural and is actually a sine-wave shape with the negative loops made positive.

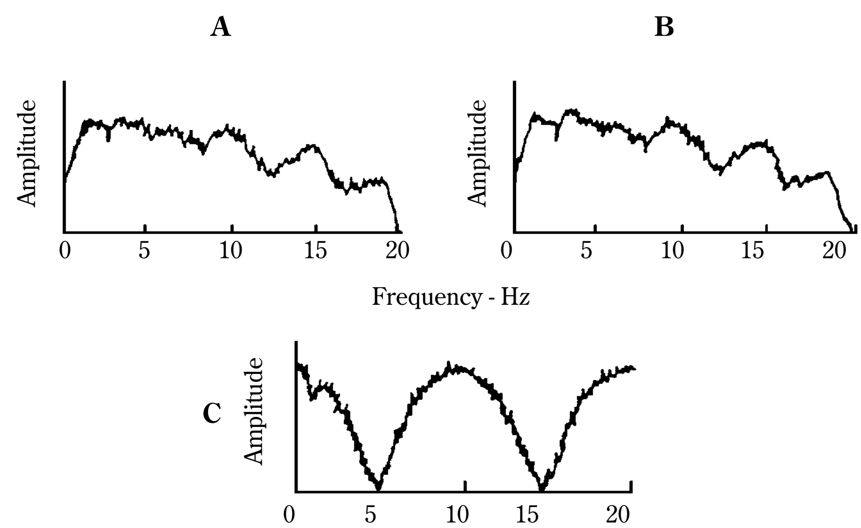


FIGURE 17-4

Combing of signals having distributed spectra; (A) instantaneous spectrum of music signal. (B) A replica of A, which is delayed 0.1 ms from A. (C) A summation of A and C showing typical comb filtering.

Combing of Direct and Reflected Sound

The 0.1 ms delay in Fig. 17-4 could have been from a digital-delay device, or it could have been a reflection from a wall or other object. The spectral shape of a signal will be changed somewhat upon reflection, depending on the angle of incidence, the acoustical characteristics of the surface, etc.

A reflection delayed 0.1 ms will have traveled (1,130 ft/sec) (0.001 sec) = 1.13 ft further than the direct signal. This difference in path length, only about 1 11/32 inch, could result from a grazing angle with both source and listener, or microphone, close to the reflecting surface. Greater delays are expected in more normal situations such as those of Fig. 17-5. The spectrum of Fig. 17-5A is from a noise generator. This is a "shhh" sound similar to the between-station noise of an FM radio receiver. Random noise of this type is used widely in acoustic measurements because it is a continuous signal, its energy is distributed throughout the audible frequency range, and it is closer to speech and music signals than sine or other periodic waves. In Fig. 17-5B, this random noise signal drives a loudspeaker, which faces a reflective surface; a

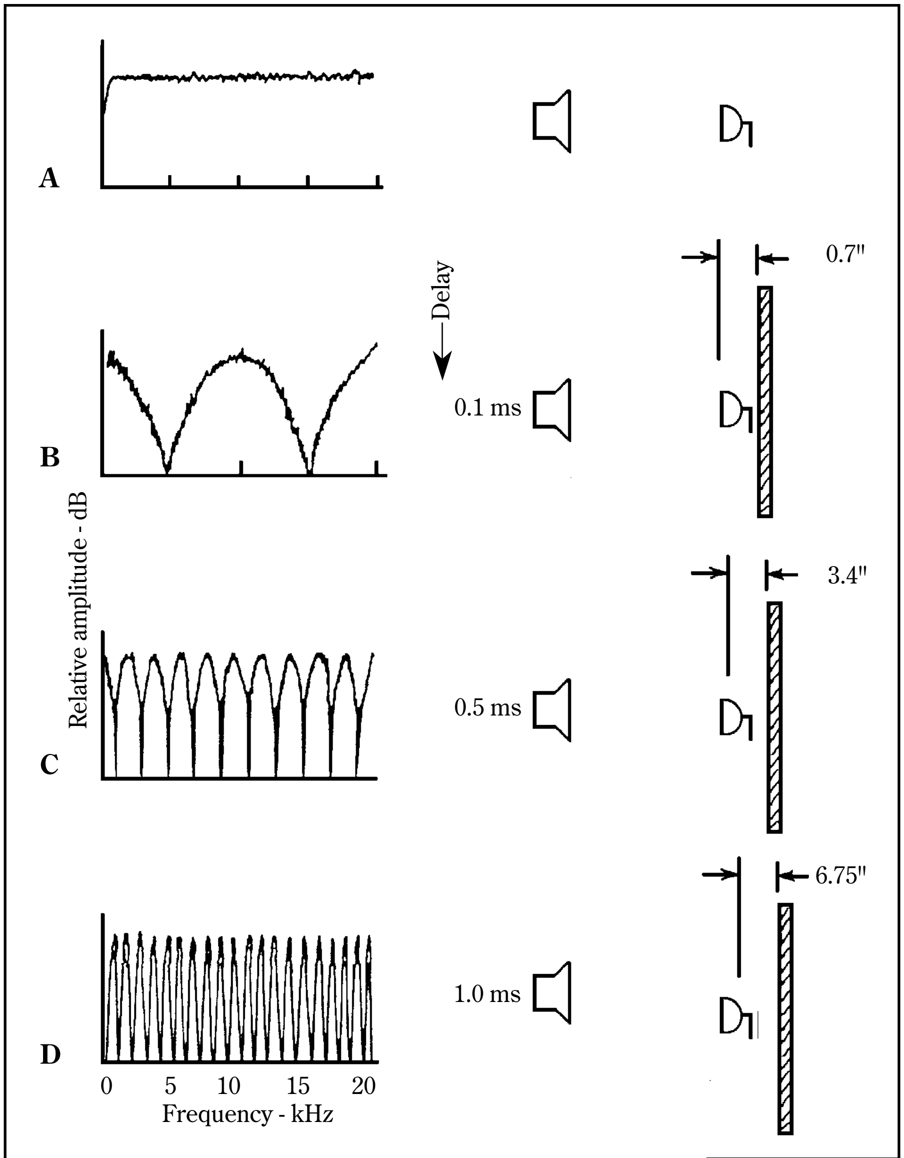


FIGURE 17-5

A demonstration of comb filtering in which direct sound from a loudspeaker is acoustically combined with a reflection from a surface at the diaphragm of a microphone. (A) No surface, no reflection. (B) Placing the microphone 0.7 in from the surface creates a delay of 0.1 ms and the combination of the direct and the reflected rays shows cancellations at 5 and 15 kHz and every 10 kHz. (C) A delay of 0.5 ms creates cancellations much closer together. (D) A delay of 1 ms results in cancellations even more closely together. If t is taken as the delay in seconds, the first null is $1/2t$ and spacing between nulls or between peaks is $1/t$.

nondirectional microphone is then placed at varying distances from the reflective surface.

In Fig. 17-5B, the microphone diaphragm is placed about 0.7 inches from the reflective surface. Interference takes place between the direct sound the microphone picks up from the loudspeaker and the sound reflected from the surface. The output of the microphone shows the comb-filter pattern characteristic of a 0.1 ms delay.

Placing the microphone diaphragm about 3.4 inches from the reflective barrier, as in Fig. 17-5C, yields a 0.5 ms delay, which results in the comb-filter pattern shown. Plotted on a linear frequency scale, the pattern looks like a comb; hence, the name *combfilter.* Increasing the delay from 0.1 to 0.5 ms has increased the number of peaks and the number of nulls five-fold.

In Fig. 17-5D, the microphone is 6.75 inches from the reflective barrier giving a delay of 1.0 ms. Doubling the delay has doubled the number of peaks/nulls once again.

Increasing the delay between the direct and reflected components increases the number of constructive and destructive interference events proportionally. Starting with the flat spectrum of Fig. 17-5A, the far-from-flat spectrum of B is distorted by the presence of a reflection delayed only 0.1 ms. An audible response change would be expected. One might suspect that the distorted spectrum of D might be less noticeable because the multiple, closely spaced peaks and narrow notches tend to average out the overall response aberrations.

Reflections following closely after the arrival of the direct component are expected in small rooms because the dimensions of the room are limited. Conversely, reflections in large spaces would have greater delays, which generate more closely spaced comb-filter peaks and nulls. Thus, comb-filter effects resulting from reflections are more commonly associated with small room acoustics. The size of various music halls and auditoriums renders them relatively immune to audible comb-filter distortions, because the peaks and nulls are so numerous and packed so closely together that they merge into an essentially uniform response.

Figure 17-6 illustrates the effect of straining a music signal through a 2 ms comb filter. The relationship between the nulls and peaks of response is related to the piano keyboard as indicated. Middle C, (C4), has a frequency of 261.63 Hz, and is close to the first null of 250 Hz. The next higher C, (C5), has a frequency close to twice that of C4 and is treated favorably with a +6-dB peak. Other Cs up the keyboard will be either discriminated against with a null, or especially favored with a peak in response—or something in between. Whether viewed as fundamental frequencies or a series of harmonics, the timbre of the sound suffers.

The comb filters illustrated in Figs. 17-4, 17-5, and 17-6 are plotted to a linear frequency scale. In this form the *comb* appearance and visualization of the delayed effects are most graphic. A logarithmic-frequency scale, however, is more common in the electronics and audio industry. A comb filter resulting from a delay of 1 ms plotted to a logarithmic frequency scale is shown in Fig. 17-7.

Comb Filters and Critical Bands

Is the human auditory system capable of resolving the perturbations of Fig. 17-5D? The resolution of human hearing is circumscribed by the

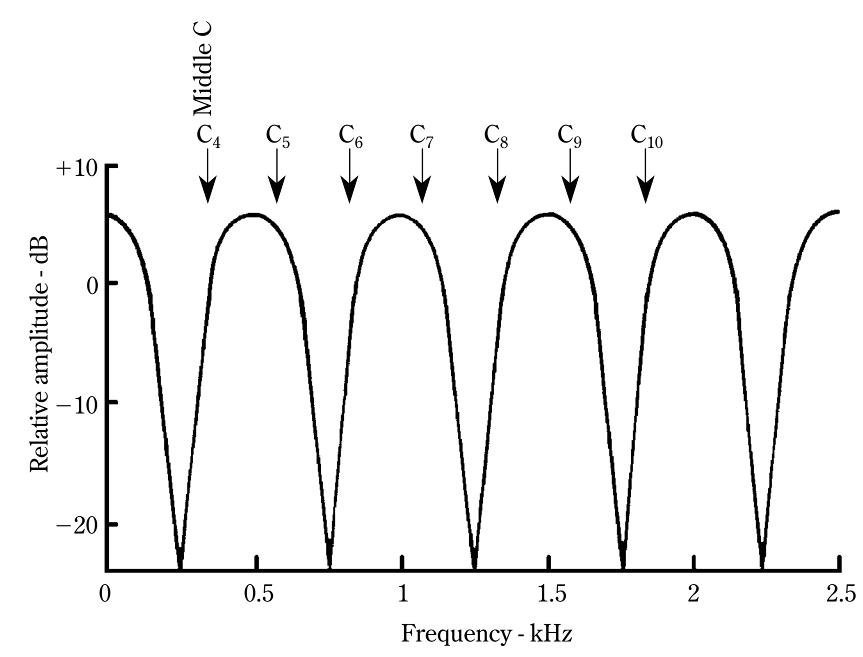


FIGURE 17-6

Passing a music signal through a 2 ms comb filter affects the components of that signal as indicated. Components spaced one octave can be boosted 6 dB at a peak or essentially eliminated at a null, or can be given values between these extremes.

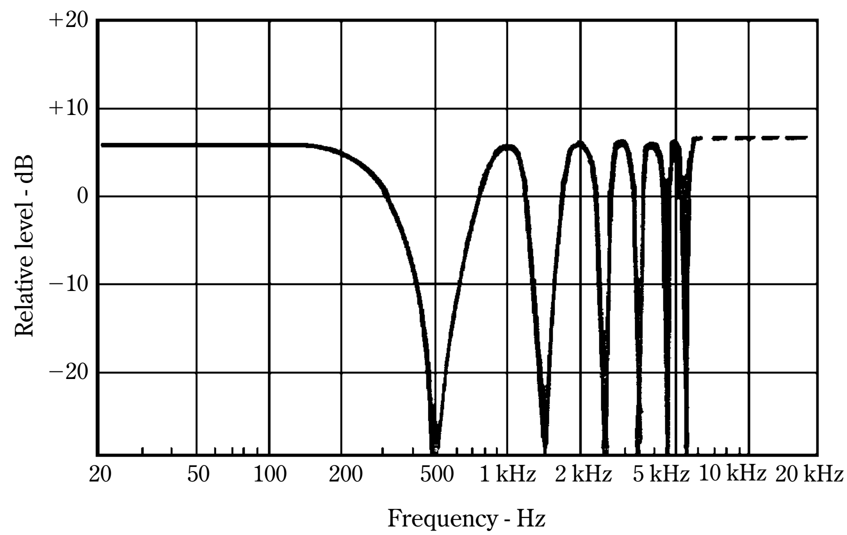


FIGURE 17-7

Up to this point all comb filters have been plotted to a linear scale to demonstrate the origin of the term *comb*. Plotting to the more convenient and familiar logarithmic scale aids in estimating the effects of a given comb on a given signal.

critical band tuning curves of the inner ear. The critical bandwidths at representative frequencies are recorded in Table 17-1. For example, the average critical bandwidth of the human auditory system at 1,000 Hz is about 128 Hz. A peak-to-peak comb-filter frequency of 125 Hz corresponds to a reflection delay of about 8 ms ($1/0.008$ = 125 Hz), which corresponds to a difference in path length between the direct and reflected components of about 9 ft (1,130 ft/sec × 0.008 sec = 9.0 ft). This situation is plotted in Fig. 17-8B. To illustrate what happens for greater delays, Fig. 17-8C is sketched for a delay of 40 ms. Shorter delays are represented by Fig. 17-8A for a delay of 0.5 ms.

Table 17-1 Auditory critical bands.

Center frequency (Hz)	Width of critical band* (Hz)
100	38
200	47
500	77
1,000	128
2,000	240
5,000	650

*Calculated equivalent rectangular band as proposed by Moore and Glasberg.[3]

The relative coarseness of the critical band cannot analyze and delineate the

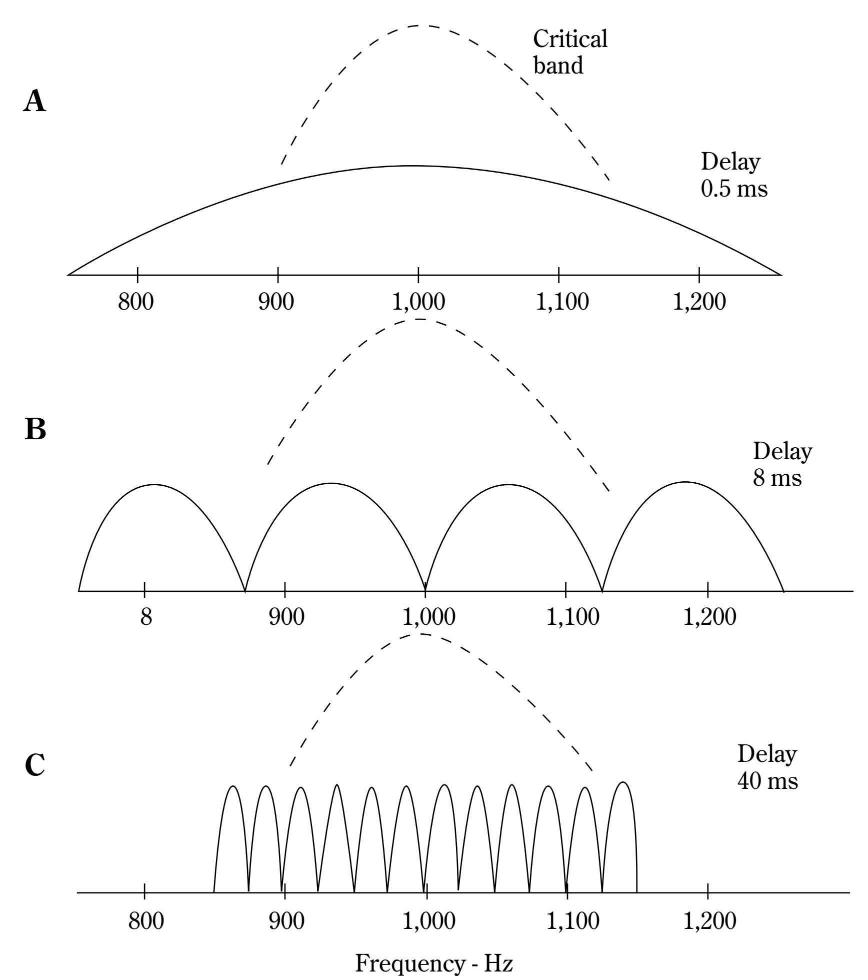


FIGURE 17-8

The audibility of combing is an important but not a well-understood factor. To assist in estimating the perceptual importance of comb filters, they are compared to the auditory critical band effective at a frequency of 1,000 Hz. (C) At a delay of 40 ms the width of the critical band is so coarse, relatively, that no analyzing of the comb filter is possible. (A) On the other hand the width of the auditory critical band is comparable to the comb peak at 0.5 ms delay. (B) is an example in between A and C. This would seem to confirm the observation that in large spaces (long delays) comb filters are inaudible, while they often are very troublesome in small spaces (short delays).

numerous peaks and nulls resulting from a 40-ms delay (Fig. 17-8C). Therefore, the human ear would not be expected to interpret response aberrations resulting from 40-ms combing as a coloration of the signal. On the other hand, the combing resulting from the 0.5 ms delay (Fig. 17-8A) could be delineated by the ear's critical band at 1,000 Hz resulting in a perceived coloration of the signal. Figure 17-8B illustrates an intermediate example in which the ear is marginally able to analyze the combed signal. The width of the critical bands of the auditory system increases rapidly with frequency. It is difficult to imagine the complexity of the interaction between a set of critical bands and a constantly changing music signal, with diverse combing patterns from a host of reflections. Only carefully controlled psychoacoustical experiments can determine whether the resulting colorations are audible (Chap. 16).

Comb Filters in Stereo Listening

In the standard stereo listening arrangement, the input signals to each ear come from two loudspeakers. These signals are displaced in time with respect to each other because of the loudspeaker spacing; the result is the generation of comb-filters. Blauert indicated that comb-filter distortion is not generally audible.[4] The auditory system has the ability of disregarding these distortions as the perception of timbre is formed. This is called *binaural suppression of colorations of timbre*; however, no generally accepted theory exists to explain how the auditory system accomplishes this.[4] Distortion can be heard by plugging one ear; however, this destroys the stereo effect. Comparing the timbre of signals from two loudspeakers producing comb-filter distortion and one loudspeaker (that does not) will demonstrate that stereo comb-filter distortion is barely audible. The timbre of the two is essentially the same. Furthermore, the timbre of the stereo signal changes little as the head is turned.

Coloration and Spaciousness

A reflected wave reaching the ear of a listener is always somewhat different from its direct wave. The characteristics of the reflecting wall vary with frequency. By traveling through the air, both the direct and reflected components of a sound wave are altered slightly, due to the air's absorption of sound, which varies with frequency. The amplitudes and timing of the direct and reflected components differ. The human auditory system responds to the frontal, direct component somewhat differently than to the reflection from the side. The perception of the reflected component is always different than the direct component. The amplitudes and timing will be related, but with an interaural correlation less than maximum.

Weakly correlated input signals to the ears contribute to the impression of spaciousness. If no reflections occur, such as when listening outdoors, there is no feeling of spaciousness (see Fig. 16-4). If the input signals to the ears are correct, the perception of the listener is that of being completely enveloped and immersed in the sound. The lack of strong correlation is a prerequisite for the impression of spaciousness.

Combing in Stereo Microphone Pickups

Because two microphones separated in space pick up a sound at slightly different times, their combined output will be similar to the single microphone with delayed reflections. Therefore, spaced microphone stereo-pickup arrangements are susceptible to comb-filter problems. Under certain conditions the combing is audible, imparting a *phasiness* to the overall sound reproduction, interpreted by some as room ambience. It is not ambience, however, but distortion of the time and intensity cues presented to the microphones. It is evident that some people find this distortion pleasing, so spaced microphone pickups are favored by many producers and listeners.

Audibility of Comb-Filter Effects

Chapter 16 showed that spaciousness is the result of reflections combining with the direct signal. This chapter demonstrated that combining a signal with a close replica of itself delayed a small amount creates comb filters. The audibility of comb filters is thus clearly stated in the Olive-Toole thresholds of Fig. 16-4. Only through psychoacoustical measurements of this type can the audibility of comb-filters be determined.

Comb Filters in Practice

Example 1: Figure 17-9 illustrates three microphone placements that produce comb filters of varying degree. A close source-to-microphone distance is shown in Fig. 17-9A. The direct component travels 1 ft and the floor-reflected component travels 10.1 ft (see Table 17-2). The difference between these (9.1 ft) means that the floor reflection is delayed 8.05 ms (9.1/1130 = 0.00805 sec.). The first null is therefore at 62 Hz with subsequent null and peak spacing of 124 Hz. The level of the

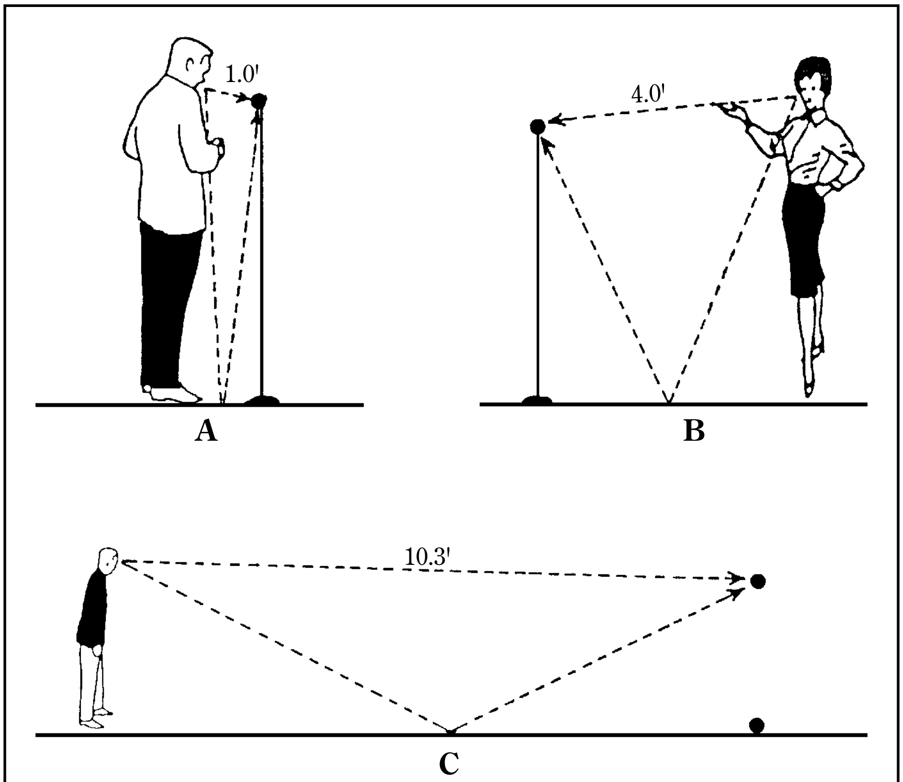


FIGURE 17-9

Common microphone placements compared with respect to production of comb filters (see Table 17-1). (A) Reflection 20 dB down, minimum comb-filter problems. (B) Reflection only 8 dB down, comb-filter problems expected. (C) Reflection almost same level as direct, comb-filter problems certain. A microphone on the floor of (C) would reduce the difference in path length between direct and reflected components (and the combing) to almost zero.

Table 17-2 Comb-filter situations (Refer to Fig. 17-9).

Fig 8-13	Path length, ft.		Difference		First null 1/2t	Pk/null spacing 1/t	Refl. level
	Direct	Refl.	Ft.	(t) ms.	Hz	Hz	dB
A	1.0	10.1	9.1	8.05	62	124	−20
B	4.0	10.0	6.0	5.31	94	189	−8
C	10.3	11.5	1.2	1.06	471	942	−1

reflection is -20 dB referred to the direct component (20 log 1.0/10.1 = 20 dB).

Similar calculations for Fig. 17-9B and C are included in Table 17-2. In A the direct component is 10 times stronger than the floor reflection. The effect of the comb filter would be negligible. Figure 17-9C has a reflection almost as strong as the direct, and the comb-filter effect would be maximum. Figure 17-9B is intermediate between A and C. A microphone is shown on the floor in Fig. 17-9C. A floor bounce would occur, but the difference between the direct and reflected path length would be very small, essentially eliminating the comb filter.

Example 2: Two microphones on a podium, Fig. 17-10, are very common. Are they used as stereo microphones? Stereo reproduction systems are quite rare in auditoriums. The chances are very good that the two microphones are fed into the same mono system and thus become an excellent producer of comb-filter effects. The common excuse for two microphones is "to give the speaker greater freedom of movement" or "to provide a spare microphone in case of failure of one." Assuming the microphones are properly polarized and the talker is dead center, there would

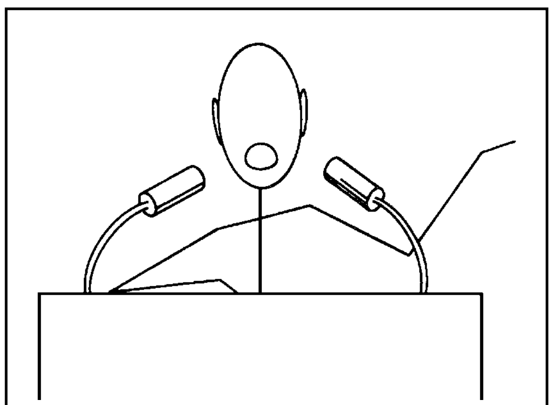


FIGURE 17-10

An infamous example of comb-filter production, two microphones feeding into the same mono amplifier with a sound source that moves about.

be a helpful 6-dB boost in level. Assume also that the microphones are 24 in apart and the talker's lips are 18 in from a line drawn through the two microphones and on a level with the microphones. If the talker moves laterally 3 in, a 0.2 ms delay is introduced, reducing important speech frequencies. If the talker does not move, the speech quality would probably not be good, but it would be stable. Normal talker movements shift nulls and peaks up and down the frequency scale with quite noticeable shifts in quality.

Example 3: A common situation with comb-filter possibilities is the singing group with each singer holding a microphone (Fig. 17-11), and each microphone fed to a separate channel but ultimately mixed together. The voice of A, picked up by both microphones, is mixed, producing comb filters resulting from the path difference. Each singer's voice is picked up by all microphones but only adjacent singers create noticeable comb filters. Experiments reported by Burroughs[5] indicate that if singer A's mouth is at least three times farther from singer B's microphone than from A's own microphone, the

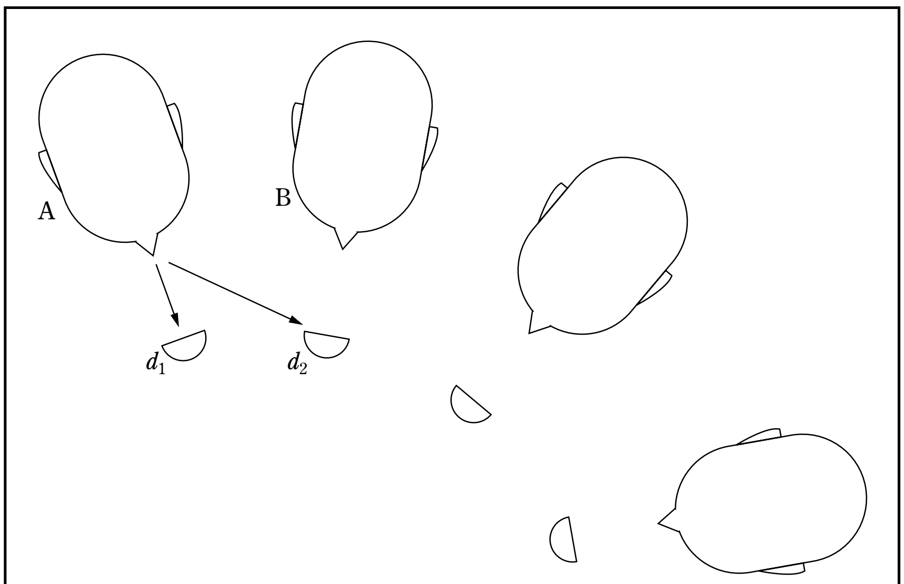


FIGURE 17-11

For group singing, if d_2 is at least three times as great as d_1, the comb-filter effect is minimized.

comb-filter effects are overshadowed by other problems. This 3:1 rule works because maintaining this distance means that delayed replicas are at least 9 dB below the main signal. This assures that comb-filter peaks and nulls are +/− 1 dB or less in amplitude and thus essentially imperceptible.

Example 4: Dual mono loudspeakers, one on stage left and the other on stage right, or variations of this theme, are quite common (Fig. 17-12). Two sources radiating identical signals create comb filters over the audience area. On the line of symmetry (often down the center aisle) both signals arrive at the same time and no comb filters are produced. Equi-delay contours range out from stage center over the audience area, the 1-ms delay contour nearest the center line of symmetry, and greater delays as the sides of the auditorium are approached.

Example 5: Multi-element loudspeakers can have their own comb-filter sources. In Fig. 17-13 it is apparent that frequency f_1 is radiated by both bass and mid-range units, that both are essentially equal in magnitude, and that the two radiators are physically displaced. These are the ingredients for comb-filter production in the audience area.

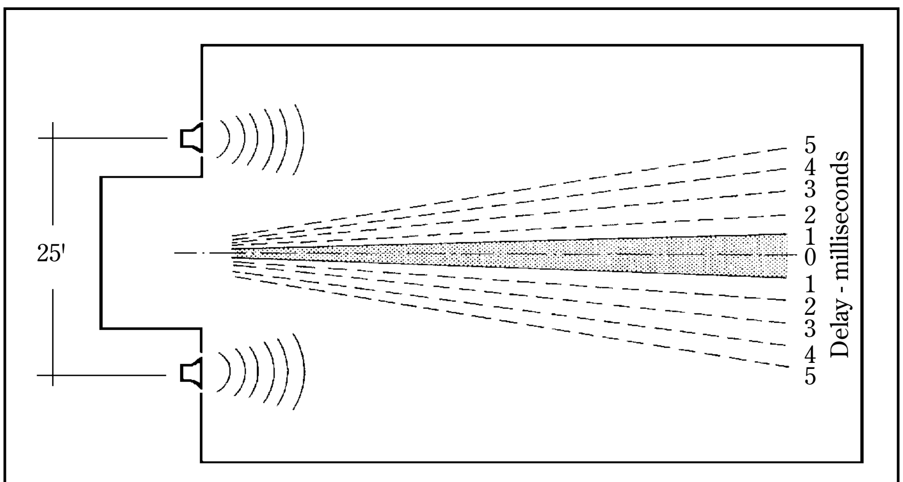


FIGURE 17-12

In the common split system in which two loudspeakers radiate identical signals, zones of constructive and destructive interference result which degrade sound quality in the audience area.

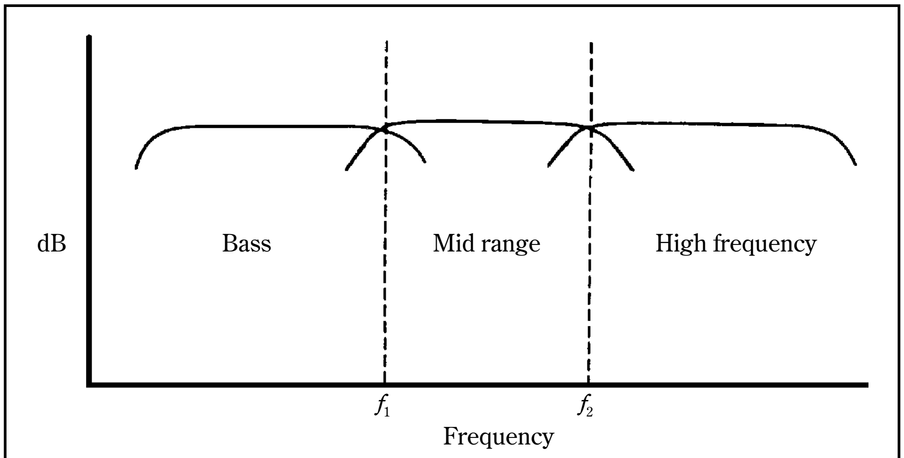


FIGURE 17-13

Comb-filter distortion can occur in the crossover region of a multielement loudspeaker because the same signal is radiated from two physically separated units.

The same process is at work between the mid-range and the tweeter units. Only a narrow band of frequencies is affected, the width of which is determined by the relative amplitudes of the two radiations. The steeper the crossover curves, the narrower the frequency range affected.

Example 6: Permanently mounted microphones may be flush-mounted with the advantage of an approximate +6-dB gain in sensitivity due to the pressure rise at the table surface. Another advantage is minimizing comb-filter distortions. In Fig. 17-14, a direct ray from the source activates the microphone diaphragm, which is shielded from reflections.

Estimating Comb-Filter Response

Remembering a few simple relationships enables you to estimate the effect of comb filters on the response of a system. If the delay is t seconds, the spacing between peaks and the spacing between nulls is $1/t$ Hz. For example, a delay of 0.001 second (1 ms) spaces the peaks $1/0.001 = 1{,}000$ Hz, and the nulls will also be spaced the same amount (Table 17-3).

The frequency at which the first null (i.e., the null of lowest frequency) will occur is $1/(2t)$ Hz. For the same delay of 0.001 *s,* or 1 ms,

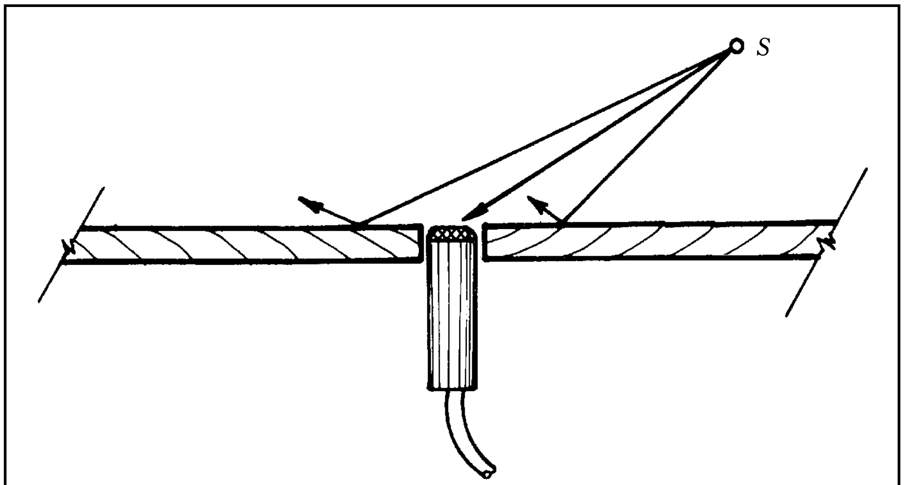


FIGURE 17-14

The flush-mounted microphone. Sounds from the source *S* that strike the surface do not reach the microphone, thus avoiding comb-filter effects. Another advantage of this mounting is an increase of level due to the pressure buildup near the reflecting surface.

Table 17-3 Comb-filter peaks and nulls.

Frequency of Delay (ms)	Spacing between nulls lowest null (Hz)	Spacing between peaks (Hz)
0.1	5,000	10,000
0.5	1,000	2,000
1.	500	1,000
5.	100	200
10.	50	100
50.	10	20

the first null will occur at $1/[2 \times 0.001] = 500$ Hz. For this 1-ms delay, you can almost figure out the system response in your head; the first null is at 500 Hz, nulls are spaced 1,000 Hz, and peaks are spaced 1,000 Hz apart. Of course, there is a peak between each adjacent pair of nulls at which the two signals are in phase. Adding two sine waves with the same frequency, the same amplitude, in phase, doubles the amplitude, yielding a peak 6 dB higher than either component by itself

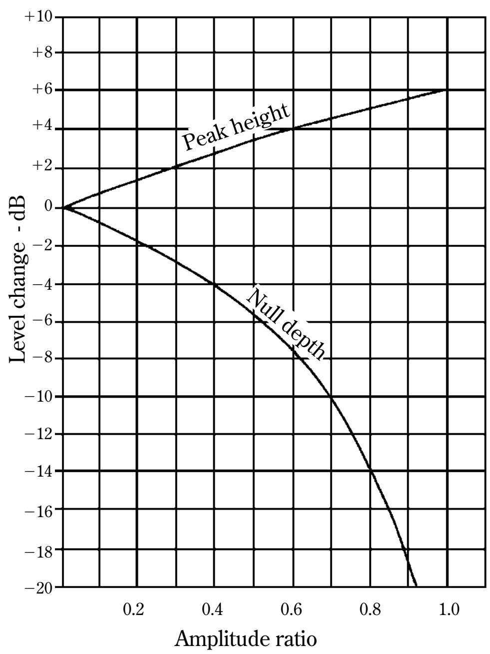


FIGURE 17-15

The effect of amplitude ratios on comb-filter peak height and null depth.

(20 log 2 = 6.02 dB). The nulls, of course, will be at a theoretical minimum of minus infinity as they cancel at phase opposition. In this way, the entire response curve can be sketched as the phase of the two waves alternates between the in-phase and the phase-opposition condition down through the spectrum.

An important point to observe is that the $1/(2t)$ expression above gives a null at 500 Hz, which robs energy from any distributed signal subject to that delay. A music or speech signal passing through a system having a 1-ms delay will have important components removed or reduced. This is nothing short of signal distortion, hence the common phrase *comb-filter distortion.*

If the mathematics of the $1/t$ and the $1/(2t)$ functions seem too laborious, Figs. 17-15 and 17-16 are included as graphical solutions.

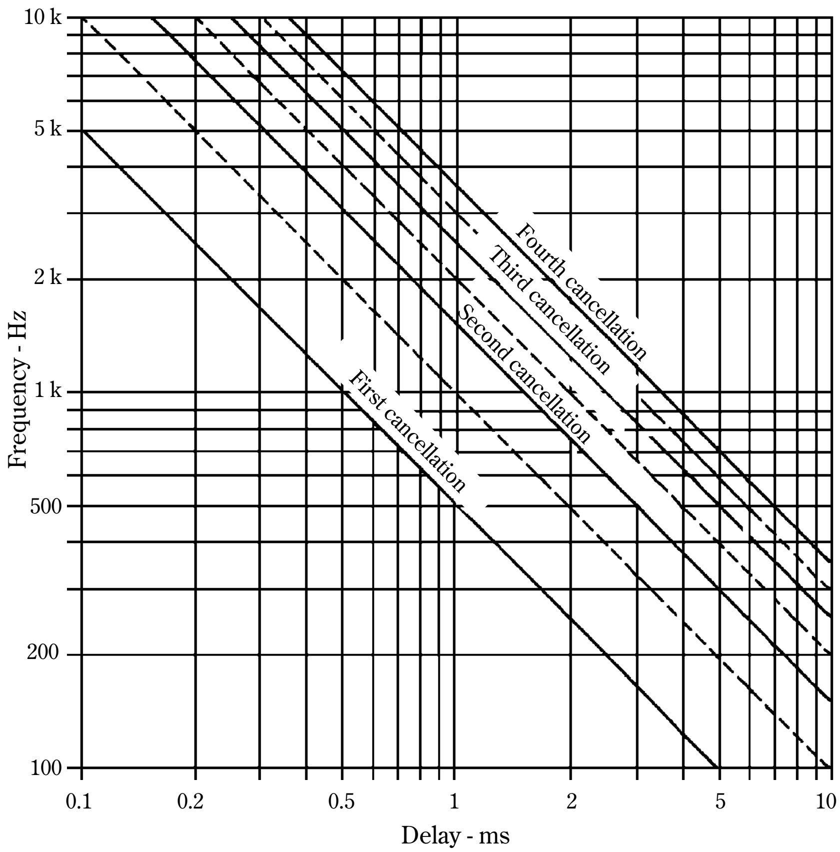


FIGURE 17-16

The magnitude of the delay determines the frequencies at which destructive interference (cancellations) and constructive interference (peaks) occur. The broken lines indicate the peaks between adjacent cancellations.

Endnotes

[1]Bartlett, Bruce, *A Scientific Explanation of Phasing (Flanging),* J. Audio Eng. Soc., 18, 6 (1970) 674–675.

[2]*The New Stereo Soundbook* by Ron Streicher and F. Alton Everest, Audio Engineering Associates, 1029 N. Allen Ave., Pasadena, CA 91104.

[3]Moore, Brian C.J. and Brian Glasberg, *Suggested Formulae for Calculating Auditory-Filter Bandwidths,* J. Acous. Soc. Am., 74, 3 (Sept 1983) 750–753.

[4]Blauert, Jens, *Spatial Hearing,* (1983), Cambridge, MA, MIT Press, 325–326.

[5]Burroughs, Lou, *Microphones: Design and Application,* Plainview, NY, Sagamore Publishing Co., (1974), Chapters 10 and 11.