9 Auditory Sensitivity

The ear's extremely wide range of sensitivity is one of the most striking aspects of audition. The preceding chapters emphasized that hearing measurements are affected by psychophysical methods and other nonauditory factors; nevertheless, a reliable picture of auditory sensitivity has been provided by research over the years. Briefly, the ear is sensitive to a range of intensities from about 0 dB SPL (which is an amplitude of vibration of about the size of a hydrogen molecule) to roughly 140 dB (at which pain and damage to the auditory mechanism ensue). This **dynamic range** of the approximately 140 dB corresponds to a pressure ratio of 10 million to 1. In other words, the most intense sound pressure that is bearable is on the order of 10 million times as great as the softest one that is perceivable under optimum listening conditions. In terms of frequency, humans can hear tones as low as 2 Hz (although roughly 20 Hz is required for a perception of "tonality") and as high as about 20,000 Hz. Furthermore, the auditory system is capable of resolving remarkably small temporal differences.

The frequency and intensity sensitivities of the ear interact, affecting each other to a greater or lesser degree. In addition, when the duration of a sound is less than about half of a second, it affects both frequency and intensity sensitivity. Longer durations may be thought of as being infinitely long as far as auditory sensitivity is concerned.

Finally, the ear is able to discriminate small differences in a wide range of stimuli; that is, it has remarkable differential sensitivity—the ability to detect very small differences between similar sounds. This ability applies to all three parameters: intensity, frequency, and time.

So much for sweeping generalizations. Let us now look at some of the details.

ABSOLUTE SENSITIVITY

Minimum Audible Levels

The issue of **absolute sensitivity** is essentially one of describing how much sound intensity is necessary for a typical, normally hearing person to just detect the presence of a stimulus. We must realize at the outset that these values are actually measures of central tendencies (means, medians, and/or modes) that describe a group of ostensibly normal subjects. In addition, it is essential to know how and where the minimum audible sound intensity is measured.

Two fundamental methods have been used to measure the intensity of a minimum audible stimulus (Sivian and White, 1933). The first involves testing a subject's thresholds through earphones, and then actually monitoring the sound pressures in the ear canal (between the earphone and eardrum) that correspond to these thresholds. This procedure yields a measure of **minimum audible pressure (MAP)**. The alternative approach is to seat the subject in a sound field and test his thresholds for sounds presented through a loudspeaker. The subject then leaves the sound field and the threshold intensity is measured with a microphone placed where his head had been. This method measures the **minimum audible field (MAF)**. It is important to dichotomize between the MAP and MAF methods because they result in different threshold values.

Ostensibly, MAP refers to the sound pressure at the eardrum. This quantity is monitored by placing a probe tube in the subject's ear canal. The probe tube passes through the earphone enclosure and leads to a microphone, which measures the sound pressure at the tip of the probe tube. Because it is difficult to place the probe right at the drum (as well as potentially painful and dangerous), the probe is generally located somewhere in the ear canal, as close to the drum as is practicable.

Minimum audible pressures are often stated in terms of the sound pressure generated by an earphone in a standardized 6-cc metal cavity (*6-cc coupler*), which approximates the volume under an earphone on the subject's ear. Such coupler pressures form the reference levels used in audiometric standards (see below). These **coupler-referred MAP** values are more appropriately called **MAPC** to distinguish them from the probe-tube MAP data obtained from actual ear canals (Killion, 1978).

Sivian and White reported the results of their classical MAP and MAF experiments in 1933. Their work was essentially confirmed by Dadson and King (1952) and by Robinson and Dadson (1956), whose data are shown in the lower portion of Fig. 9.1. These curves show monaural MAP and binaural MAF (from a loudspeaker located directly in front of the subject, i.e., at 0° azimuth) as a function of frequency. Monaural MAP values extending to very low frequencies are also shown. The MAP values for frequencies between 10,000 and 18,000 Hz are shown in the figure on an expanded frequency scale. As these MAP and MAF curves clearly show, human hearing is most sensitive between about 2000 and 5000 Hz, and reasonably good sensitivity is maintained in the 100 to 10,000 Hz range. Absolute sensitivity becomes poorer above and below these frequencies.

While the general relationship between auditory sensitivity and frequency is well established, one should be aware that subsequent experiments have provided detailed estimates of absolute sensitivity in the lower and upper frequency ranges. For example, one should refer to Berger (1981) for a detailed analysis of hearing sensitivity in the low-frequency range (50–1000 Hz), and to Schechter et al. (1986) for a detailed analysis of thresholds for the high frequencies (8000–20,000 Hz).

Notice in Fig. 9.1 that the MAF curve falls consistently below the MAP curve. In other words, a lower intensity is needed to reach threshold in a sound field (MAF) than under earphones (MAP). This fact was first demonstrated by Sivian and White (1933), and the discrepancy of 6 to 10 dB is called the "missing

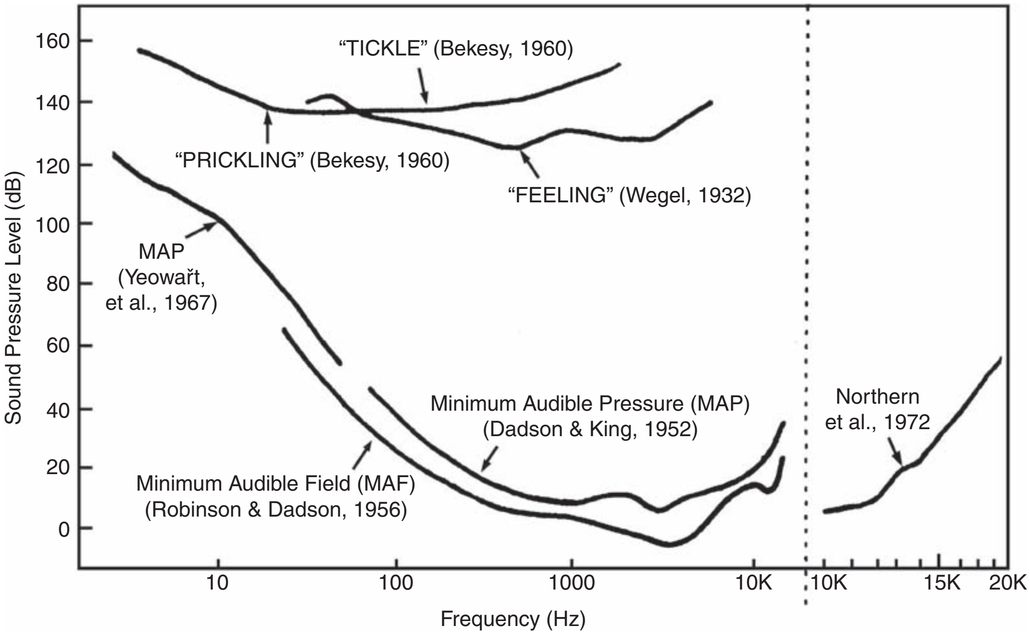

Figure 9.1 Minimal auditory field (MAF) after Robinson and Dadson (1956), and monaural minimal auditory pressure (MAP) after Dadson and King (1952) and Yeowart et al. (1967). The high-frequency MAP curve after Northern et al. (1972) is shown on the expanded scale to the right for clarity. Upper limits of "usable" hearing after Bekesy (1960) and Wegel (1932).

6 dB" (Munson and Wiener, 1952). Sivian and White proposed that the MAP/MAF discrepancy might be due to physiological noises picked up by the ear when it is covered by an earphone. These physiological noises would partially mask (see Chap. 10) the signal presented by the earphone, so that more sound pressure would be needed to achieve MAP than for the unmasked MAF. Although this explanation accounted for part of the missing 6 dB problem, it fell short of accounting for the whole difference.

Subsequent studies have formed the basis for resolving the MAP/MAF difference (Rudmose, 1963, 1982; Villchur, 1969, 1970, 1972; Morgan and Dirks, 1974; Stream and Dirks, 1974; Killion, 1978). This explanation was presented in a cohesive manner by Killion (1978). To begin with, recall from Chapter 3 that diffraction and ear canal resonance enhance the pressure of a free-field signal reaching the eardrum (Shaw, 1974). Thus, a corrected version of the international standard reference MAF curve (ISO-1961) may be converted to eardrum pressure by applying Shaw's (1974) head-related transfer function data. The correction accounts for an apparent error in the low-frequency MAF levels (Killion, 1978). Since binaural thresholds are somewhat better than monaural ones (see Chap. 13), a correction is also made to account for the advantage of binaural MAF over monaural MAP. By accounting for differences between real ear (MAP) and coupler (MAPC) values, the effects of impedance changes and ear canal distortion due to the placement of the earphone, and the effects of physiological noises, the MAP/MAF discrepancy is essentially resolved.

Threshold Microstructure

The MAP and MAF curves in Fig. 9.1 are drawn as smooth curves. It is commonly (and generally implicitly) assumed that an individual's threshold curve is similarly represented by a smooth line. The ubiquitous nature of this assumption is revealed by the fact that both clinicians and researchers make most of their threshold measurements at frequencies that are an octave apart, and very rarely sample at intervals that are less than a half-octave wide. However, this may not be the case (Elliot, 1958; van den Brink, 1970; Cohen, 1982; Long, 1984; Long and Tubis, 1988). Instead, a rippled or jagged configuration is often obtained when thresholds are sampled at numerous frequencies that are measured at very close intervals. Moreover, these patterns are highly repeatable. These irregularities in the threshold microstructure are associated with (particularly spontaneous) otoacoustic emissions, and it is believed that they reflect active processes in the cochlea (Wilson, 1980; Schloth, 1983; Zwicker and Schloth, 1983; Long, 1984; Long and Tubis, 1988; Talmadge et al., 1998, 1999). Discussions of active cochlear processes and otoacoustic emissions may be found in Chapter 4.

Upper Limits of Hearing

Just as we may conceive of the minimum audibility (MAP and MAF) curves as the lower limit of hearing sensitivity, the upper limits of hearing may be thought of as the sound pressure levels (SPLs) that are too loud or cause some kind of unpleasant sensation. These criteria are actually quite different. *Uncomfortable loudness* is usually what we mean by the upper limit of usable

Table 9.1 Reference Equivalent Threshold Sound Pressure Levels (RETSPLs) for Various Earphones in Decibels of Sound Pressure Level (dB SPL re: 20 μPa) in Appropriate Measurement Couplers

	Supra-aural receivers in 6-cc [NBS-9A] coupler		Circumaural receivers in flat-plate coupler		Insert receivers (etymotic ER-3A & EARtone 3A) in:		
Frequency (Hz)	**Telephonics TDH-49 & 50**	**Telephonics TDH-39**	**Sennheiser HDA200**	**Koss HV/1A**	**HA-2 coupler**	**HA-1 coupler**	**Occluded ear simulator**
125	47.5	45.0	30.5	–	26.0	26.5	28.0
250	26.5	25.5	18.0	–	14.0	14.5	17.5
500	13.5	11.5	11.0	–	5.5	6.0	9.5
750	8.5	8.0	6.0	–	2.0	2.0	6.0
1000	7.5	7.0	5.5	16.0	0.0	0.0	5.5
1500	7.5	6.5	5.5	–	2.0	0.0	9.5
2000	11.0	9.0	4.5	–	3.0	2.5	11.5
3000	9.5	10.0	2.5	–	3.5	2.5	13.0
4000	10.5	9.5	9.5	8.0	5.5	0.0	15.0
6000	13.5	15.5	17.0	–	2.0	−2.5	16.0
8000	13.0	13.0	17.5	16.5	0.0	−3.5	15.5
9000	–	–	18.5	21.0	–	–	–
10,000	–	–	22.0	25.5	–	–	–
11,200	–	–	23.0	24.5	–	–	–
12,500	–	–	28.0	26.0	–	–	–
14,000	–	–	36.0	33.0	–	–	–
16,000	–	–	56.0	51.0	–	–	–

Source: Based on ANSI S3.6-2004.

hearing. It refers to the level at which a sound is too loud to listen to for any appreciable period of time and is often referred to as the **uncomfortable loudness level (UCL)** or **loudness discomfort level (LDL)**. The LDL is associated with sound pressure levels approximating 100 dB (Hood and Poole, 1966, 1970; Hood, 1968; Morgan, Wilson, and Dirks, 1974), with higher mean LDLs of about 111 to 115 dB SPL reported by Sherlock and Formby (2005). On the other hand, the upper curves in Fig. 9.1 show that SPLs of about 120 dB or more produce sensations variously described as feeling, tickle, touch, tolerance, discomfort, or pain. Notice that these *unpleasant sensations* are actually *tactile* rather than auditory. High levels can produce temporary changes in hearing sensitivity and even permanent hearing loss and are discussed later in this chapter.

REFERENCE LEVELS

One might now ask what constitutes a reasonable conception of normal hearing sensitivity for the population as a whole. That is, how much SPL does the average person who is apparently free of pathology need to detect a particular sound? The answer permits standardization of audiometric instruments so that we may quantify hearing loss relative to "what normal people can hear."

Prior to 1964, several countries had their own standards for normal hearing and audiometer calibration based upon locally obtained data. For example, the 1954 British Standard (1954) was based upon one group of studies (Wheeler and Dickson, 1952; Dadson and King, 1952, whereas the 1951 American Standard (ASA-1951) reflected other findings (USPHS, 1935–1936; Steinberg et al., 1940). Unfortunately, these standards differed by about 10 dB, and the American Standard was too lax at 250 and 500 Hz, and too strict at 4000 Hz (Davis and Kranz, 1964). This situation was rectified in 1964 with the issuance of Recommendation R389 by the International Organization for Standardization (ISO-1964). This standard is generally referred to as ISO-1964. It was based upon a round-robin of loudness-balance and threshold experiments involving American, British, French, Russian, and German laboratories, and as a result equivalent reference SPLs were obtained for the earphones used by each country. These reference levels were subsequently incorporated into the S3.6 standard disseminated by the American National Standards Institute (ANSI S3.6-1969; now ANSI S3.6-2004).

The reference values for pure tone signals presented from various kinds of earphones are shown in Table 9.1.[1] These values are called **reference-equivalent threshold sound pressure levels (RETSPLs)** and are based on the current versions of the *American National Standard Specification for Audiometers* (ANSI S3.6-2004), essentially corresponding to the ISO 389 standards

[1] The testing room must be very quiet in order to obtain auditory thresholds as low as those in Tables 9-1 to 9-3 (corresponding to 0 dB HL in Fig. 9.2). The maximum permissible noise levels for this purpose (ANSI S3.1—1999 [R2003]) are summarized in octave bands in Appendix 9-1 and in third-octave bands in Appendix 9-2.

Table 9.2 Sound Field Reference Equivalent Threshold Sound Pressure Levels (RETSPLs) in Decibels of Sound Pressure Level (dB SPL re: 20 μPa) at a Point Corresponding to the Listener's Head When Narrow Bands of Noise or Frequency Modulated Tones are Presented from Loudspeakers Located at Various Azimuths

	Loudspeaker azimuth			
	0° (front)		45° (side)	90° (side)
Center frequency[a] (Hz)	**Monaural[b]**	**Binaural[c]**	**Monaural[b]**	**Monaural[b]**
125	24.0	22.0	23.5	23.0
250	13.0	11.0	12.0	11.0
500	6.0	4.0	3.0	1.5
750	4.0	2.0	0.5	−1.0
1000	4.0	2.0	0.0	−1.5
1500	2.5	0.5	−1.0	−2.5
2000	0.5	−1.5	−2.5	−1.5
3000	−4.0	−6.0	−9.0	−6.5
4000	−4.5	−6.5	−8.5	−4.0
6000	4.5	2.5	−3.0	−5.0
8000	13.5	11.5	8.0	5.5

[a]Center frequencies of the narrow bands of noise or frequency-modulated tones used as test signals.
[b]Listening with one ear.
[c]Listening with both ears.
Source: Based on ANSI S3.6-2004.

(ISO-389-1-5,7, 1994a, 1994b, 1994c, 1998a, 1988b, 2005). Representative reference values for signals presented from loudspeakers are shown in Table 9.2 (ANSI S3.6-2004; ISO-389-7, 2005). Notice that these signals are narrow bands of noise or frequency modulated tones, which are employed because pure tones are subject to problems due to standing waves when used for sound field testing. In addition, separate values are provided for listening with one ear (monaurally) to sounds presented from different loudspeaker directions, and for listening with two ears (binaurally) when the speaker is located in front of the listener. Table 9.3 shows the reference values used when hearing is measured by bone conduction (ANSI S3.6-2004; ISO-389-3, 1994b). These values are called **reference-equivalent threshold force levels (RETFLs)** because they express the equivalent force (in dB re: 1 μN) on a measuring device known as a *mechanical coupler* or *artificial mastoid*, which corresponds to 0 dB HL when the bone-conduction vibrator is placed on a person's mastoid or forehead.

Hearing Level

Because each of the SPLs in Table 9.1 corresponds to minimum audibility, we may think of them as all representing the same **hearing level**. Thus, each RETSPL may also be referred to as 0 dB hearing level (0 dB HL) for its respective frequency. For example, the reference level for a 1000-Hz tone (for TDH-49 earphones) is 7.5 dB SPL so that 0 dB HL corresponds to 7.5 dB SPL at 1000 Hz. At 250 Hz, more sound pressure is required to reach the normal threshold so that 0 dB HL equals 26.5 dB SPL at this frequency. The relationship between SPL and HL is illustrated in Fig. 9.2. Figure 9.2a shows the minimally audible (threshold) values in dB SPL as a function of frequency. As in Fig. 9.1, intensity increases upward on the y-axis. Figure 9.2b shows the same information in dB HL. Notice that the minimum audible values (0 dB HL) all lie along a straight line in terms of hearing level. In other words, the HL scale calls each zero reference SPL value "0 dB HL," so that thresholds can be measured in comparison to a straight line rather than a curved one.

The graph in Fig. 9.2b is the conventional **audiogram** used in audiology. Actually, the term "audiogram" may legitimately be used to describe any graph of auditory sensitivity as a function of frequency. By convention, increasing intensity (which indicates a hearing loss) is read downward on the y-axis when thresholds are plotted in dB HL.

Table 9.3 Reference Equivalent Threshold Force Levels (RETFLs) for Bone-Conduction Vibrators, Expressed in Decibels (dB) re: 1 μN Measured on a Mechanical Coupler (Artificial Mastoid)

Frequency (Hz)	Vibrator at mastoid	Vibrator at forehead
250	67.0	79.0
500	58.0	72.0
750	48.5	61.5
1000	42.5	51.0
1500	36.5	47.5
2000	31.0	42.5
3000	30.0	42.0
4000	35.5	43.5

Source: Based on ANSI S3.6-2004.

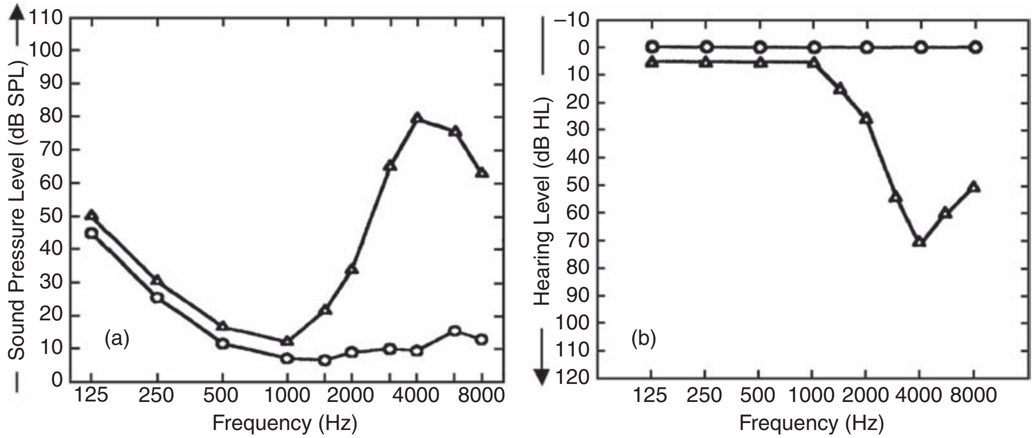

Figure 9.2 Audiograms showing normal hearing (*circles*) and a hearing loss in the high frequencies (*triangles*), expressed in (a) dB SPL and (b) dB HL. Note that intensity is shown downward on the clinical audiogram in dB HL.

Now, suppose that we measure the thresholds of a person whose cochlea has been partially damaged by excessive noise exposure. This kind of trauma often appears as a hearing loss in the higher frequencies. The triangles in Fig. 9.2 show the impaired thresholds in terms of both SPL and HL. The difference in dB between the impaired thresholds and the reference values (circles) is the amount of hearing loss at that frequency. For example, our hypothetical patient has a threshold of 5 dB HL at 1000 Hz. This means that he requires 5 dB HL to just detect the tone, as opposed to only 0 dB HL for a normal person. In SPL, this corresponds to a threshold of 12 dB (i.e., the 7.5 dB RETSPL) for 0 dB HL plus the 5 dB hearing loss. Had the threshold been 40 dB HL, the corresponding value would have been 47.5 dB SPL. Similarly, the 70 dB HL threshold at 4000 Hz is equivalent to 80.5 dB SPL (70 dB over the 10.5 dB RETSPL). As one might expect, audiometers are calibrated to dB HL values by measuring the output of the earphone in SPL at each frequency and then converting to HL by subtracting the appropriate RETSPL shown in Table 9.1.

Effects of Duration

Thus far we have been considering tones lasting for about 1 s or more. From the standpoint of audition, such durations may be viewed as infinite. Auditory sensitivity is altered, however, for durations much shorter than 1 s. Extremely short durations, on the order of 10 ms or less, result in transients that spread energy across the frequency range. These transients will confound the result of an experiment if they are audible (Wright, 1960, 1967), so that special care is needed in the design and interpretation of studies using short durations.

The perception of **tonality** appears to change in a somewhat orderly manner as the duration of a very short tone burst is increased (Burck et al., 1935; Doughty and Garner, 1947). A click is heard at the shortest durations, then a click with tonal qualities (*click pitch*) at slightly longer durations. For frequencies below about 1000 Hz, a *tonal pitch* is perceived when the duration of the tone burst is long enough for the subject to hear several cycles (periods) of the tone. Thus, the duration threshold for tonality decreases from about 60 ms at 50 Hz to approximately 15 ms at 500 Hz. Above 1000 Hz, the threshold for tonality is essentially constant and is on the order of about 10 ms.

Absolute sensitivity decreases when the duration of a stimulus becomes much shorter than 1 s, and the nature of this phenomenon reveals an interesting property of the auditory system. Two observations are routinely encountered (Hughes, 1946; Zwislocki, 1960; Small et al., 1962; Campbell and Counter, 1969; Watson and Gengel, 1969). First, for durations up to roughly 200-300 ms, a 10-fold (decade) change in duration can offset an intensity change on the order of about 10 dB. In other words, reducing the duration of a tone burst at threshold from 200 to 20 ms (a decade reduction) reduces sensitivity to the degree that the intensity must be increased by 10 dB to re-attain threshold. Alternatively, the threshold intensity decreases by about 10 dB when the duration of a tone burst is increased from 20 to 200 ms. Second, durations longer than about 1/3 s are treated by the ear as though they are infinitely long. That is, increasing or decreasing durations that are longer than approximately 300 ms does not change the threshold level. These principles are shown in idealized form in Fig. 9.3.

The phenomenon under discussion is called **temporal integration** or **temporal summation**. It demonstrates that the ear operates as an energy detector that samples the amount of energy present within a certain time frame (or window). A certain amount of energy is needed within this time window for the threshold to be attached. This energy may be obtained by using a higher intensity for less time or a lower intensity for more time. The ear integrates energy over time *within* an integration time frame of roughly 200 ms. This interval might also be viewed as a period during which energy may be stored and can

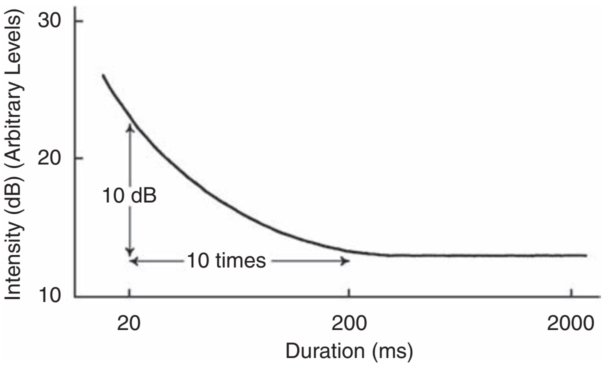

Figure 9.3 Idealized temporal integration function showing that a 10-times (decade) change in duration is offset by an intensity change of about 10 dB for stimulus durations up to about 200–300 ms.

be measured as a time constant τ (Plomp and Bouman, 1959). Energy available for longer periods of time is not integrated with the energy inside the time window. This additional energy thus does not contribute to the detection of the sound, so that the threshold does not change durations longer than 200 ms. Photographers might think of this situation as analogous to the interaction of a camera's f-stop (intensity) and shutter speed (duration) in summating the light energy for a certain film speed (integration time): The lens opening and shutter speed may be traded against one another as long as the same amount of light is concentrated upon the film. The trade-off between intensity and duration is illustrated conceptually in Fig. 9.4.

Figure 9.5 shows the effect of frequency upon temporal integration at threshold. Thresholds for shorter durations are shown relative to the threshold levels obtained for 512 ms, which are represented by the horizontal line. Notice that although temporal integration occurs for all frequencies shown, the functions become flatter (i.e., the time constant τ for integration becomes shorter) as frequency increases from 250 to 4000 Hz.

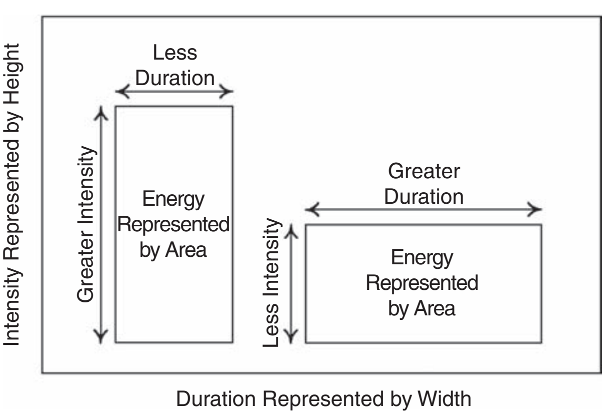

Figure 9.4 Artist's conceptualization of temporal integration depicting the trade-off between stimulus intensity and duration.

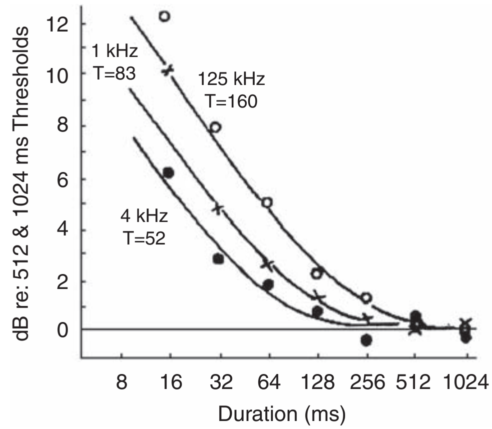

Figure 9.5 Effect of frequency on temporal integration. *Source*: Adapted from Watson and Gengel (1969), with Permission of *J. Acoust. Soc. Am.*

Temporal integration is observed at higher levels as well as at absolute threshold. Temporal summation of loudness is discussed in Chapter 11, and Chapter 3 covers this topic with respect to the acoustic reflex.

DIFFERENTIAL SENSITIVITY

Having examined the lower and upper bounds of hearing, we may now ask what is the smallest perceivable difference between two sounds. This quantity is called either the **difference limen (DL)** or the **just noticeable difference (jnd)**. These terms will be used interchangeably in this text. The DL is the smallest perceivable difference in dB between two intensities (**ΔI**) or the smallest perceivable change in hertz between two frequencies (**Δf**). We may think of the jnd in two ways. One is as the absolute difference between two sounds, and the other is as the relative difference between them. The latter is obtained by dividing the absolute DL by the value of the starting level. Thus, if the starting level I is 1000 units and the DL or ΔI is 50 units, then the relative DL ΔI/I is 50/1000 = 0.05. This ratio, **ΔI/I**, is called the **Weber fraction**.

A point about absolute versus relative DLs should be clarified before proceeding. The frequency DL or Δf is an absolute difference in hertz, as opposed to the relative frequency DL obtained by dividing Δf by the starting frequency f. Suppose it is necessary to change a 1000-Hz tone (f) by a 3.6 Hz Δf in order for a particular subject to just detect the frequency difference. His absolute frequency DL is thus 3.6 Hz, whereas his relative DL is 0.0036. However, the situation is different when the intensity DL is given in decibels, which is usually expressed as **ΔI in dB** or $10\log\frac{I+\Delta I}{I}$. Since decibels are actually ratios, ΔI in dB is really a relative value. (This is why ΔI and I were expressed as "units" in the above example.) Both ΔI/I and ΔI in dB are commonly encountered in the literature. Let's use two examples to illustrate the relationship between ΔI/I and ΔI in dB or $10\log\frac{I+\Delta I}{I}$.

If $\Delta I/I = 1.0$, then ΔI in dB would be $10\log\frac{1+1}{1} = 10\log 2$, or 3 dB. When $\Delta I/I$ is 0.5, then ΔI in dB $= 10\log\frac{1+0.5}{1} = 1.76$dB.

An important concept in psychophysics is known as **Weber's law**, which states that the value of $\Delta I/I$ (the Weber fraction) is a constant (k) regardless of stimulus level, or

$$\frac{\Delta I}{I} = k$$

Similarly, ΔI in dB or $10\log\frac{I+\Delta I}{I}$ would also be the same across stimulus levels. A classic conceptual illustration of Weber's law is the number of candles one must add to a number of candles that are already lit in order to perceive a difference in the amount of light (Hirsh, 1952). If 10 candles are lit, then only one more will produce a jnd of light (DL = 1). However, if there are originally 100 candles then 10 must be added to result in a perceptible difference, and to notice an increase in the light provided by 1000 candles, 100 must be added. Thus, the absolute value of the DL increases from 1 to 100, whereas the Weber fraction has remained constant, at k = 0.1 (since 1/10 = 10/100 = l00/l000 = 0.1), illustrating Weber's law.

Intensity Discrimination

Early differential sensitivity experiments (Knudsen, 1923) were plagued by audible transient noises due to the abrupt switching on and off of the stimuli, making it unclear whether subjects were responding to the stimulus or to the audible transient. In his classical study, Riesz (1928) overcame the switching noise problem by using **amplitude modulation (AM)** to produce intensity differences, as illustrated in Fig. 9.6a. Amplitude modulation was produced by simultaneously presenting two tones of slightly different frequencies, resulting in a tone that **beats** (fluctuates in intensity) at a rate equal to the difference in frequency between the two original tones. For example, combining a 1000-Hz tone with a 1003-Hz tone results in three beats per second, which Riesz found to be the optimal rate for

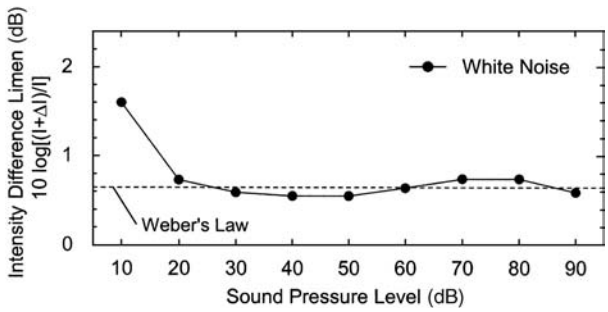

Figure 9.7 The intensity difference limen in decibels shown as a function of stimulus level for white noise based on the data of Houtsma, Durlach, and Braida (1980). The horizontal dashed line has been drawn through the data to represent Weber's law.

measuring intensity DLs in his study. To find the DL, Riesz's subjects adjusted the amplitude of one of the two beating tones until the beats became minimally audible. The intensity difference between the two tones was then taken as the measure of the DL. Technological advances made it possible for later studies to measure intensity DL by using pulsed pure tones, as illustrated in Fig. 9.6b.

The size of ΔI in dB is shown as a function of stimulus level for white noise in Fig. 9.7 and for pure tones in Fig. 9.8. Weber's law predicts that ΔI in dB should be the same at all stimulus levels, represented by the dashed horizontal lines in these figures. Weber's law appears to hold for broadband stimuli like white noise (e.g., Houtsma et al., 1980; Wojtczak and Viemeister, 2008). For example, Fig. 9.7 shows that the Weber fraction for white noise (expressed as $10\log\frac{I+\Delta I}{I}$) is constant at about 0.6 to 0.8 dB except for the faintest stimulus level.

The situation is different for narrow band signals like pure tones, for which the Weber fraction decreases somewhat as the stimulus level increases (Riesz, 1928; McGill and Goldberg, 1968a, 1968b; Viemeister, 1972; Moore and Raab, 1974;

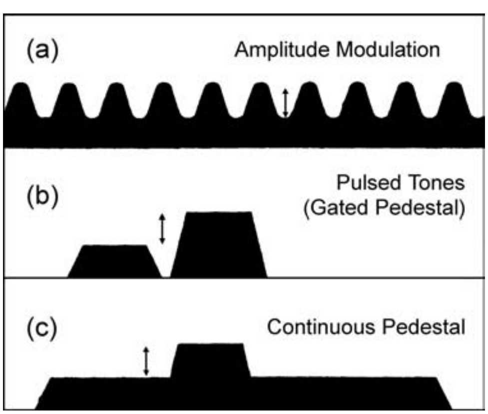

Figure 9.6 Artist's conceptualizations of various methods used to obtain the DL for intensity (see text): (a) amplitude modulation, (b) pulsed tones or gated pedestal, and (c) continuous pedestal.

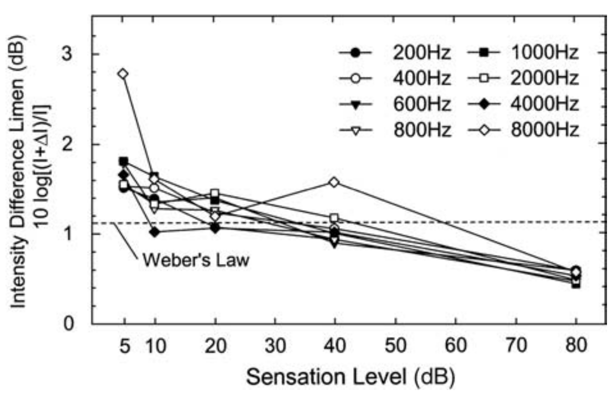

Figure 9.8 The intensity difference limen in decibels shown as a function of stimulus level for pure tones based on the data of Jesteadt, Wier, and Green (1977). The horizontal dashed line has been drawn through the data to represent Weber's law.

Jesteadt et al., 1977; Houtsma et al., 1980; Florentine et al., 1987; Viemeister and Bacon, 1988; Turner et al., 1989; Stellmack, Viemeister, and Byrne, 2004; Wojtczak and Viemeister, 2008). For example, Figure 9.8 shows that the Weber fraction decreases with increasing intensity from about 1.7 dB at a sensation level (SL) of 5 dB to about 0.5 dB at 80 dB SL. This slight deviation from Weber's law has come to be known as the **near miss to Weber's law** (McGill and Goldberg, 1968a, 1968b).[2]

Rabinowitz et al. (1976) combined and summarized the results for differential sensitivity at 1000 Hz. Their analysis suggested that Weber's law holds between 10 and 40 dB SL, although differential sensitivity changes with sensation level above and below this range. Viemeister and Bacon (1988) similarly suggested that the function relating the relative DL for intensity to sensation level can be approximated by a horizontal line segment between about 20 and 50 dB SL and a sloping one for higher levels. Informative discussions if Weber's law and the near miss are provided by, for example, Florentine et al. (1987), Viemeister and Bacon (1988), Green (1988), and Wojtczak and Viemeister (2008).

Riesz (1928), using the amplitude modulation method, reported that the Weber fraction is frequency dependent, becoming smaller as frequency increased from 35 Hz up to about 1000 Hz. The Weber fraction remained more or less constant for the frequencies above this, at least for SLs above 20 dB. However, this result has not been confirmed by subsequent studies (e.g., Harris, 1963; Schacknow and Raab, 1973; Penner et al., 1974; Jesteadt et al., 1977). For example, Jesteadt et al. (1977) found that ΔI/I does not change with frequency so that a single straight line could be used to show ΔI/I as a function of sensation level. The similarity of the functions relating ΔI in dB to sensation level at various frequencies can be seen Fig. 9.8. Florentine et al. (1987) investigated intensity DLs over a wide frequency range from 250 to 12,000 Hz. They did find a frequency dependency for the high frequencies, but not for the frequencies up to about 4000 Hz (similar to the findings by Jesteadt et al.).

Some of the variability existing in the intensity DL data may be due to the use of alternative methods of presenting the stimuli. Recall here that the DL experiment basically asks whether the subject can hear a difference (which is equal to ΔI) between a baseline signal presented at intensity of I and a more intense signal presented at an intensity of (I + ΔI). We might call the baseline signal the pedestal. There are two general ways to present the increment.

One approach involves leaving the pedestal (I) on all the time and to add an intensity increment (ΔI) on top of it at various times. This is the **continuous pedestal** method and is shown schematically in lower frame (c) of Fig. 9.6. Alternatively, a pedestal alone (I), which may be presented for some period of time, and then be turned off, followed after a brief interval by the presentation of the pedestal together with the increment on top of it (I + ΔI). This strategy may be called the **gated pedestal** method and is shown in the middle frame (b) of Fig. 9.6. Turner, Zwislocki, and Filion (1989) pointed out that the continuous pedestal method is analogous to the types of listening conditions used in early DL studies (e.g., Riesz, 1928; Lüscher and Zwislocki, 1949; Jerger, 1952), while the pulsed-tone methods used by Jesteadt et al. (1977) and other more recent studies (e.g., Florentine and Buus, 1981; Florentine et al., 1987; Viemeister and Bacon, 1988; Turner et al., 1989) involve the gated pedestal technique.

Quite a few studies have directly or indirectly compared intensity DLs using the continuous and gated pedestal methods (e.g., Campbell and Lasky, 1967; Green, 1969; Green et al., 1979; Carlyon and Moore, 1986a; Viemeister and Bacon, 1988; Turner et al., 1989; Bacon and Viemeister, 1994). The general finding has been that smaller intensity DLs are produced by the continuous pedestal method than by the gated pedestal approach. Representative mean results from the study by Turner et al. are shown in Fig. 9.9 for stimuli presented at three frequencies. The reason(s) for the gated-continuous difference is not

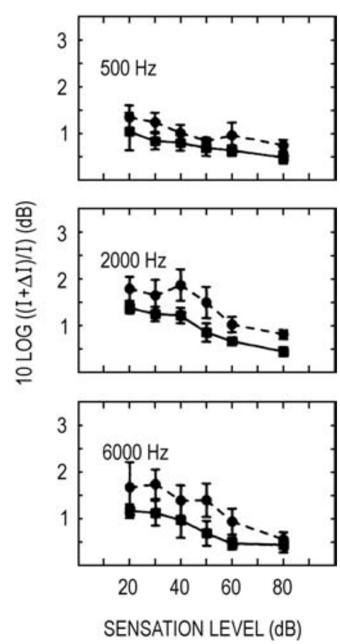

Figure 9.9 Intensity DLs in decibels (as $10\log\frac{1+\Delta I}{1}$) as a function of sensation level at three frequencies for the continuous (*squares/solid lines*) versus gated (*circles/dashed lines*) pedestal methods. *Source*: From Turner et al. (1989), with permission of *J. Acoust. Soc. Am.*

[2] A "severe departure" from Weber's law occurs under certain conditions (see Carlyon and Moore, 1984, 1986a, 1986b; Moore, 1984, 1986a,b). Here, a large increase in the Weber fraction is found at about 55-65 dB SPL for very brief high-frequency signals presented under the gated (versus continuous) pedestal condition, and for the detection of a signal in a band-reject masker, expressed as the signal-to-noise ratio.

clearly established. Turner et al. (1989) have suggested that it might involve a short-term memory effect, and that (citing data by Gescheider et al.) it is probably not a specifically auditory phenomenon because similar findings are also found for touch.

In summary, differential sensitivity for intensity follows Weber's law for wide-band noise and becomes slightly more acute with increasing sensation level in a manner that is a "near miss" to Weber's law for narrow-band stimuli like pure tones.

Frequency Discrimination

The early work (Knudsen, 1923) on differential sensitivity for frequency, like that on intensity discrimination was plagued by transient noise problems associated with the presentation of the stimuli. Shower and Biddulph (1931) circumvented this problem by using **frequency-modulated (FM)** tones as the stimuli. In other words, the test tone was varied continuously in frequency at a rate of twice per second. The subject's task was to detect the presence of a modulated tone as opposed to a steady tone. The DL was taken as the smallest difference in frequency that produced a perceptible modulation of the original tone. Since Shower and Biddulph's classic study included a very wide range of frequencies (62–11, 700 Hz) and sensation levels (5–80 dB), it has remained the most widely cited study of differential frequency sensitivity for many years. However, subsequent studies using pulsed tones have generally resulted in better (smaller) DLs at low frequencies and poorer (larger) DLs at higher frequencies than were found with the FM tones (Harris, 1952; Rosenblith and Stevens, 1953; Henning, 1967; Nordmark, 1968; Moore, 1973; Jesteadt and Wier, 1977; Wier et al., 1977; Nelson et al., 1983). We shall return to this point below. The most likely reason for the discrepancy is that frequency modulation results in a stimulus with a complex spectrum, so that we really cannot be sure what serves as the basis for the subject's responses.

Wier et al. (1977) reported the results of an extensive frequency-discrimination study using pulsed pure tones from 200 to 8000 Hz at sensation levels between 5 and 80 dB. They took the DL to be the smallest frequency difference **Δf** that the subject could detect 71% of the time. Fig. 9.10 shows some of their results at four sensation levels. The important observations are that Δf becomes larger as frequency increases, and that Δf becomes smaller as sensation level increases. Sensation level is relatively more important at low frequencies than at high ones, where the curves tend to converge. The best (smallest) values of Δf—on the order of 1 Hz—occur for low frequencies presented at about 40 dB SL or more. The DL increases substantially above about 1000 Hz so that Δf at 40 dB SL is roughly 16 Hz at 4000 Hz and 68 Hz by 8000 Hz. Figure 9.10 also shows that Δf is not simply a monotonic function of frequency; it does not always get larger as frequency increases. We see a departure from a monotonically rising function between 200 and 400 Hz. (There are also rather dramatic peaks in the vicinity of 800 Hz, although their origin is unclear.)

Other studies using pulsed tones at various frequencies and sensation levels are in essential agreement with the findings of

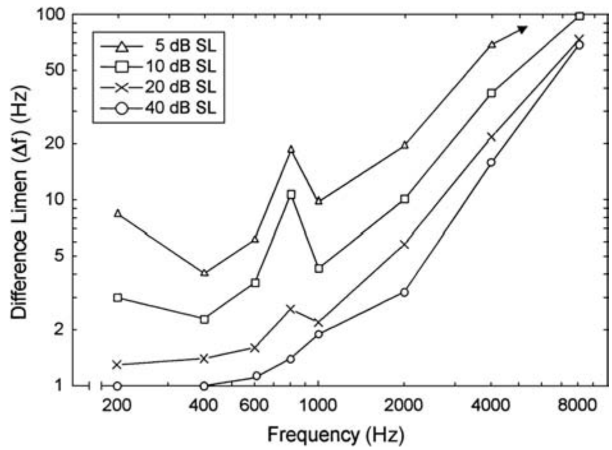

Figure 9.10 The frequency difference limen Δf is shown as a function of frequency at sensation levels of 5, 10, 20, and 40 dB, based the on data of Wier, Jesteadt, and Green (1977).

Wier et al. (Harris, 1952; Moore, 1973; Jesteadt and Wier, 1977; Nelson et al., 1983). Nordmark's data (1968) are in agreement with those of Wier et al. when the latter are corrected for differences in experimental methodology (Wier et al., 1976). Nelson et al. (1983) replicated the Wier et al. study using a somewhat different methodology. On the basis of their data, they developed a general equation to predict frequency discrimination given the frequency and level of the stimulus. This approach also predicted the data of Weir et al. extremely well and was also able to successfully estimate earlier frequency DL data of Harris (1952).

Relative differential sensitivity for frequency is shown as the Weber fraction **Δf/f** for the data of Wier et al. (1977) in Fig. 9.11. Notice that Δf/f improves (becomes smaller) as SL increases and

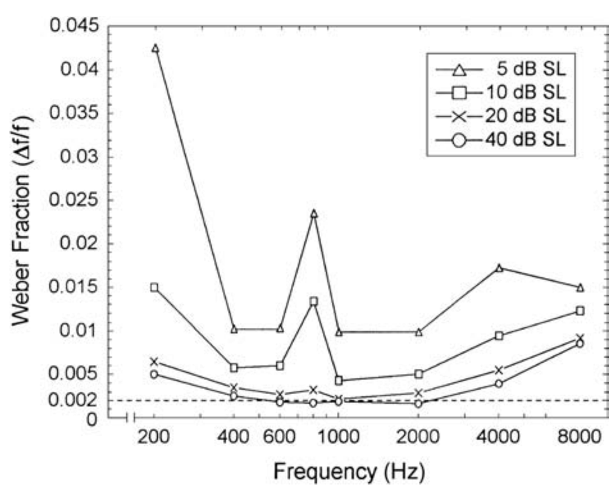

Figure 9.11 Differential sensitivity Δf/f is shown as a function of frequency at sensation levels of 5, 10, 20, and 40 dB, based on the data of Wier, Jesteadt, and Green (1977).

is about 0.006 or less for frequencies as high as 4000 Hz when the tone level reaches 20 dB SL. The Weber fraction becomes as small as approximately 0.002 for frequencies between 400 and 2000 Hz at 40 dB SL, which corresponds to a frequency difference of just 0.2%. The value of $\Delta f/f$ is relatively constant for moderate sensation levels between about 400 and 2000 Hz, but becomes larger at higher and lower frequencies. In summary, then, $\Delta f/f$ is a somewhat complex function of both frequency and level, unlike ΔI in dB, which appears to depend principally upon stimulus level alone for a reasonably wide range of frequencies.

Profile Analysis

The discrimination of changes in spectral configuration is called **profile analysis** (Green, 1983, 1988). These differences in spectral shape are important factors contributing to the distinction between sound qualities or timbers (Chap. 12) and among speech sounds (Chap. 14).

Profile analysis experiments involve asking subjects to listen for a difference in the level of one component of a complex sound compared to the other components of the sound. This is a lot like an intensity DL, but instead of listening for a level difference between two sounds, the comparison is being made across frequencies within the same sound. The typical approach uses a two-alternative forced choice method, as illustrated in Fig. 9.12. Both intervals contain complex sounds made up of the same component frequencies. All of the components are equal in level in the comparison sound, illustrated by the left (interval 1) spectrum for trial a in the figure. In the target sound, all but one of the components are equal in level, but the remaining component is higher in level, as in the right (interval 2) spectrum for trial a. The larger component (highlighted with a thicker line) is the *signal* and the equal-level components are called the *background*. The subject's task is to choose the interval containing the target signal (interval 2 in this case). In this context, the threshold is the smallest level increment necessary to detect the signal above the background.

A special technique is needed to induce the subject to listen for a level difference across frequencies between the signal and the background instead of a difference between the overall levels of the two sounds. The approach is called **roving levels**, and it involves randomly varying the overall levels of both the comparison and target sounds over a large range (usually about 40 dB) from trial to trial. This concept is easily appreciated by seeing how the amplitudes of the spectra change across the six trials illustrated in Fig. 9.12. The ensuing jumble of levels prevents the subject from choosing the target sound by comparing its level to that of the comparison sound. In addition, the listener is given feedback (indicating which interval contained the target sound) after each response. As a result, the subject learns to pick the target sound based on the across-frequencies comparison between the signal and background (i.e., the spectral profile).

Figure 9.12 illustrates the typical profile analysis experiment, in which the components are equally spaced logarithmically in frequency, the signal (usually about 1000 Hz) is the center frequency, and the background frequencies all have the same level. Although our discussion focuses upon this basic arrangement, the student should be aware that others have also been used, such as multiple-component signals (e.g., Green and Kidd, 1983; Bernstein and Green, 1987; Green et al., 1987), jagged spectra (e.g., Kidd et al., 1991; Lenze and Richards, 1998), and signals involving decrements in level (e.g., Heinz and Formby, 1999).

Let us summarize several of the principal features of profile analysis. The interested student should refer to the papers cited and Green's (1988) classical book, *Profile Analysis*. Discriminations in profile analysis appear to be most sensitive (thresholds are lowest) when the signal frequency is in the midrange of the sound's spectrum, usually in the vicinity of about 500 to 2000 Hz (Green and Mason, 1985; Green et al., 1987). Thresholds become lower as the range of frequencies in the sound (bandwidth) widens and as the number of components within that range (spectral density) gets larger (Green et al., 1983, 1984; Bernstein and Green, 1987). However, the threshold becomes poorer due to masking (Chap. 10) when adding components that are close in frequency to the signal (Green et al., 1983; Bernstein and Green, 1987). Thresholds do not appear to be affected by the phase relationships among the components (Green et al., 1984; Green and Mason, 1985). In addition, Green and Mason (1985) found that the size of the increment (in dB) needed for a 1000-Hz signal to be discriminated from the background stayed the same for a range of background components from 30 to 70 dB SPL (although the increment became smaller at 80 dB). Thus, the intensity discrimination involved in profile analysis appears to follow Weber's law over a fairly wide range of levels. This is similar to what we saw for ΔI in dB for broadband noise earlier in the chapter.

Temporal Resolution

The importance of being able to make fine temporal discriminations should become obvious when one realizes that speech is made up of signals that change rapidly over time. We will briefly look at several general aspects of the temporal realm. The first deals with temporal resolution, the second with the nature of successiveness and temporal order, and the last is the difference limen for duration.

Temporal resolution refers to the shortest period of time over which the ear can discriminate two signals. One way to measure this period is by asking a subject to discriminate between signals that are exactly the same except for a phase difference. Green (1971, 1973a, 1973b) has referred to this quantity as **temporal auditory acuity** or **minimum integration time**. The latter phrase suggests that the ear's most sensitive temporal discriminations also provide an estimate of the shortest time period within which the ear can integrate energy. We could think of this time period as the "low end" of the scale for temporal integration, as discussed earlier in the chapter.

Temporal resolution has been studied using a variety of approaches, such as the temporal modulation transfer function and gap detection (e.g., Patterson and Green, 1970; Ronken,

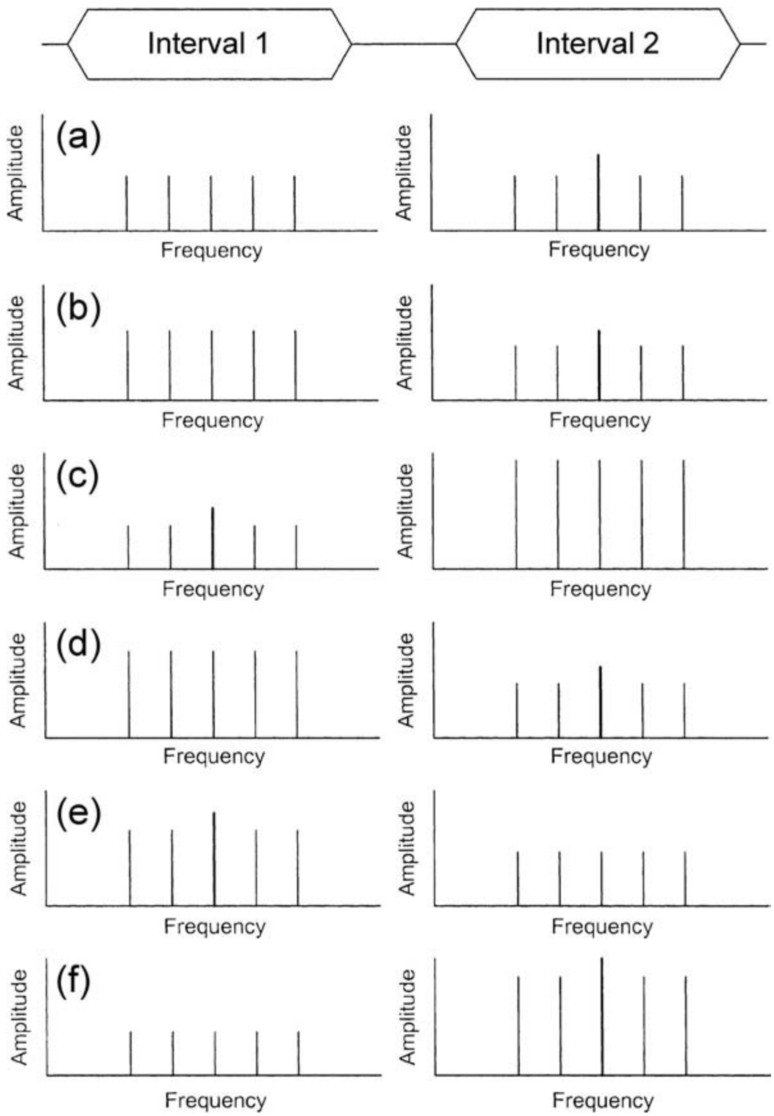

Figure 9.12 The typical profile-analysis experiment employs a two-interval forced choice procedure with roving levels. The time course of the two intervals is shown at the top of the figure, and the spectra of the sounds in intervals 1 and 2 are shown for six trials, labeled (a)–(f). The subject's task is to indicate which interval contains a level difference between the signal frequency (indicated by the thicker line) and the background frequencies. Notice that the overall levels of the comparison and target sounds are varied from trial to trial.

1970; Green, 1971, 1973a, 1973b, 1985; Viemeister, 1979; Forest and Green, 1987; Formby and Muir, 1988). The **temporal modulation transfer function** (**TMTF**) addresses the ability to detect the presence of amplitude modulation in a sound (e.g., Viemeister, 1979; Bacon and Viemeister, 1985; Formby and Muir, 1988). The basic method is illustrated in Fig. 13a, which shows the envelopes of two noises presented one after the other in a two-interval forced-choice paradigm. One of these signals is a noise that has been subjected to **sinusoidal amplitude modulation** (**SAM**), and the other signal is the same noise without modulation. The subject's task is to select the amplitude modulated noise. This is a measure of temporal acuity because the amplitude fluctuations are occurring over time: the faster the **modulation rate** or **frequency** (the number of modulations per second), the closer together in time are the fluctuations (Fig. 9.13b). The listener's sensitivity for hearing the presence of amplitude modulation is measured in terms of **modulation depth**, or how deep the amplitude fluctuations must be in order for them to be detected (Fig. 9.13c).

Typical TMTFs are illustrated in Fig. 9.14 using data from two representative studies and show how much modulation depth is needed for modulation to be detected by the listener at different

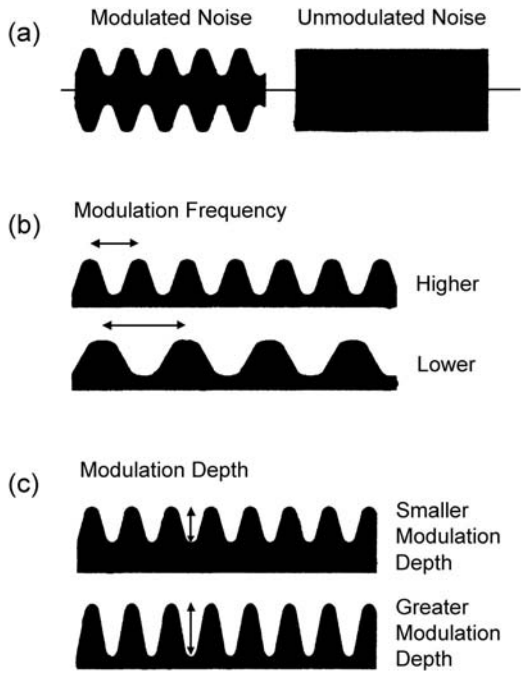

Figure 9.13 Artist's conceptualizations of (a) amplitude-modulated versus unmodulated noises, (b) different modulation rates, and (c) different modulation depths.

modulation frequencies. Modulation depth is usually expressed in percent or decibels. When expressed in decibels, 0 dB corresponds to 100% and 0 dB, with smaller modulation depths given in decibels below 0 dB. For example, −6 dB would be 50%, −12 dB would be 25%, and −20 dB would be 10% modulation. Moreover, TMTFs are plotted with maximum (100% or 0 dB) modulation depth and the bottom of the y-axis. The figure shows that the TMTF has a low-pass configuration. The ability to detect modulation is most sensitive for modulation frequencies up to about 50 Hz, and then decreases considerably above about 100 Hz.

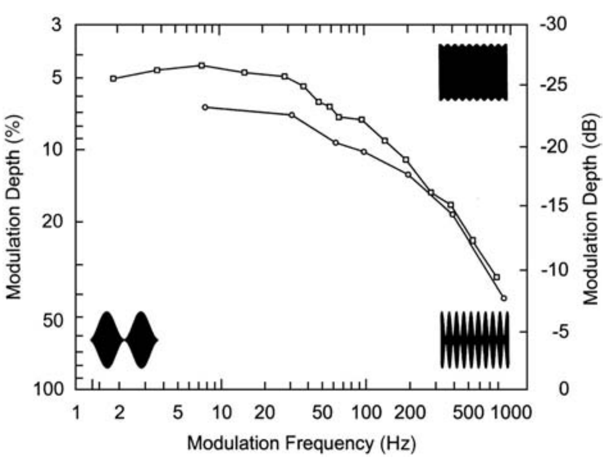

Figure 9.14 Examples of temporal modulation transfer functions based on the data of Bacon and Viemeister (1985; *squares*) and Formby and Muir (1988; *circles*). The inserts are artist's conceptualizations of stimulus waveforms to show that modulation frequency increases from left to right along the x-axis, and modulation depth decreases from bottom to top on the y-axis.

A lucid appreciation of temporal resolution is provided by the **gap detection** technique, which has been employed by numerous investigators since it was introduced by Plomp in 1964 (e.g., Penner, 1977; Fitzgibbons, 1983; Shailer and Moore, 1983; Fitzgibbons and Gordon-Salant, 1987; Forest and Green, 1987; Formby and Muir, 1988; Moore et al., 1992; Schneider et al., 1994; Trehub et al., 1995; Phillips, Taylor, Hall, et al., 1997; Lister, Besing, and Koehnke, 2002; Phillips and Smith, 2004; Elangovan and Stuart, 2008). The basic strategy of the gap detection experiment is actually quite straightforward. Suppose we have a continuous burst of noise lasting 500 ms. We could "chop out" a short segment in the center of the noise lasting, say, 10 ms. We now have a (leading) noise burst lasting 245 ms, followed by a 30 ms silent period, followed by a (trailing) 245 ms noise burst. Hence, we have a *gap* lasting 10 ms surrounded in time by leading and trailing noise bursts. Three different gap durations are illustrated schematically in Fig. 9.15. The subject is asked whether he hears the gap, hence, the paradigm is called gap detection. The duration of the gap is varied according to some psychophysical method (see Chaps. 7 and 8) in order to find the shortest detectable gap between the two noise bursts, which is called the **gap detection threshold (GDT)**. Thus, the GDT reflects the shortest time interval we can resolve, and it is taken as a measure of temporal resolution.

The essential finding of GDT experiments is that auditory temporal resolution is on the order of 2 to 3 ms. Such GDTs are obtained when the noise signal contains the higher frequencies and when these are presented at levels that are adequately audible. That the ear can make temporal discriminations as small as about 2 ms is a consistent finding for the various approaches that have been used to study temporal auditory acuity, and is analogous to what Hirsh (1959) Hirsh and Sherrick (1961) described as auditory **successiveness** (versus simultaneity). Interested students will find detailed discussions of gap detection parameters in many sources (e.g., Fitzgibbons, 1983; Shailer and Moore, 1983; Buus and Florentine, 1985; Green, 1985; Forest and Green, 1987; Phillips, 1999; Elangovan and Stuart, 2008).

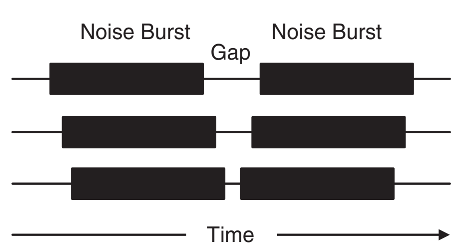

Figure 9.15 Artist's conceptualization of the gap detection paradigm with three different gap durations.

It is noteworthy, however, that 2 to 3 ms GDTs are found when the two sounds separated by the gap are the same, as in Fig. 9.16a. In contrast, GDTs are much longer (typically exceeding 20 ms) when the leading and following sounds differ in various ways, such as in terms of their spectra and/or durations (Phillips et al., 1997; Lister et al., 2002; Phillips and Smith, 2004; Elangovan and Stuart, 2008). Figure 9.16b shows an example, in which the two sounds differ in frequency. These examples reveal a distinction between with-channel and across-channel processing (e.g., Phillips, 1999; Elangovan and Stuart, 2008): Hearing the gap simply involves detecting a discontinuity between the two sounds when the two sounds are the same (i.e., within the same auditory filter channel), as in frame **a**, which can be accomplished by the peripheral auditory system. In contrast, the comparison is more complex when the two sounds are different (i.e., in different auditory channels), which requires central processing across auditory channels. Similarly longer temporal intervals are needed for **perceived temporal order**, which involves determining which of two different sounds came first (e.g., "high" and "low") as in Fig. 9.16b, and for detecting differences in the onset times of two otherwise simultaneous tones, as in Fig. 9.16c (e.g., Hirsh, 1959; Hirsh and Sherrick, 1961; Pisoni, 1977).

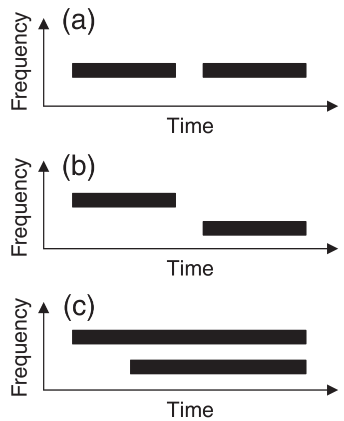

Figure 9.16 Examples of stimulus arrangements for tasks involving (a) gap detection when both signals are the same (within channel), (b) gap detection when the two sounds differ in frequency (between channels), and (c) detection of temporal onset time between two otherwise simultaneous signals differing in frequency. Arrangement (b) also shows the arrangement of a temporal order task, where the listener indicates whether the higher or lower signal came first.

Temporal Discrimination

Earlier in this chapter, we saw that ΔI depends mainly upon intensity and that Δf is appreciably affected by both intensity and frequency. Differential sensitivity for the duration of a signal has also been investigated, although not as extensively as the other two parameters. The general finding is that the **difference limen for duration (ΔT)** becomes smaller as the overall duration decreases (Small and Campbell, 1962; Abel, 1972; Sinnott et al., 1987; Dooley and Moore, 1988). Abel (1972) studied ΔT for stimulus durations between 0.16 and 960 ms, using various bandwidths of noise from 200 to 300 Hz wide as well as 1000 Hz tone bursts. She presented subjects with two intervals, one containing a standard stimulus duration (T) and the other containing a slightly longer duration (T + ΔT). The subject listened to the two intervals (which were presented randomly) and indicated the one with the longer-duration signal. The smallest time difference correctly detected that 75% of the time was taken as the DL for duration ΔT. As Figure 9.17 shows, ΔT decreases from about 50 ms at durations of 960 ms to on the

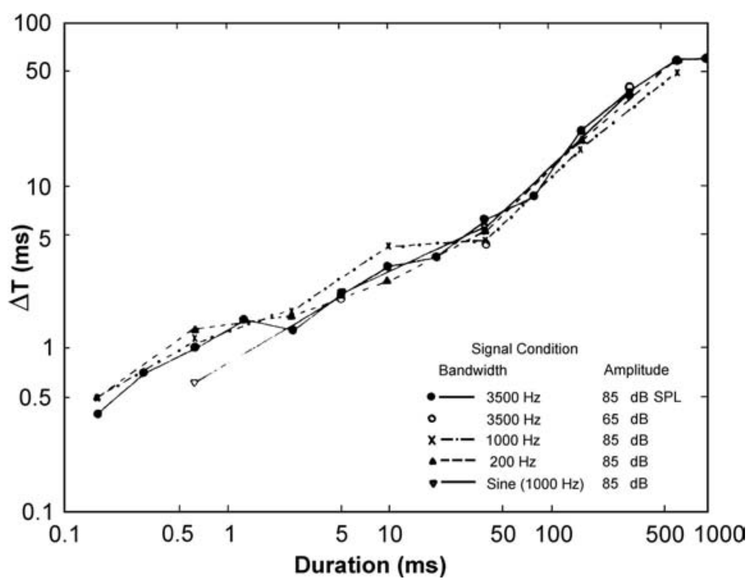

Figure 9.17 Values of ΔT as a function of duration from 0.16 to 960 ms. *Source*: From Abel (1972), with permission of *J. Acoust. Soc. Am.*

order of 0.5 for durations of less than 0.5 ms. Differential sensitivity in terms of the Weber fraction ΔT/T is not a constant, but changes with duration so that ΔT/T is about 1.0 at 0.5 to 1 ms, roughly 0.3 at 10 ms, and approximately 0.1 from 50 to 500 ms. The results were essentially independent of bandwidth and intensity. Observations by Sinnott et al. (1987) and Dooley and Moore (1988) were in essential agreement with these findings.

Stimulus Uncertainty

Figure 9.18 illustrates a discrimination experiment in which the subjects are presented with pairs of tonal sequences (Watson et al., 1975, 1976; Watson and Kelly, 1981). Each sequence involves 10 brief tones arranged in a certain pattern of frequencies, one after the other. The sequences are the same in both intervals of each trial, except that the frequency of one of the tones may be changed in the second interval. The listener's task is indicate whether the two sequences are the same or different. The smallest frequency difference that allows the subject to tell the two sequences apart constitutes a frequency DL, but the outcome is clearly being affected by the complexity of the task as well as by the frequency difference itself. (It is also possible to change more than one of the frequencies, or to vary the amplitude or duration.)

The nature of the task can be altered in various ways. The position of the tone being changed (shown by arrows in the figure) may be the same in every trial, such as always second (trial *a*) or always ninth (trial *b*), or it might occur in different positions from trial to trial. It is also possible for the overall pattern of the tones to be the same in every trial (e.g., *a* and *b*), or for the patterns to change from trial to trial (e.g., b and c). Stimulus arrangements that keep changing from trial to trial present the listener with considerably more uncertainty than trials that are always the same. In fact, the general observation has been that discrimination performance deteriorates when the listener must deal with greater amounts of stimulus uncertainty compared to lesser degrees of uncertainty about the stimuli (e.g., Watson et al., 1975, 1976; Watson and Kelly, 1981; Howard et al., 1984; Leek et al., 1991; Neff and Jesteadt, 1996; Kidd et al., 2002).

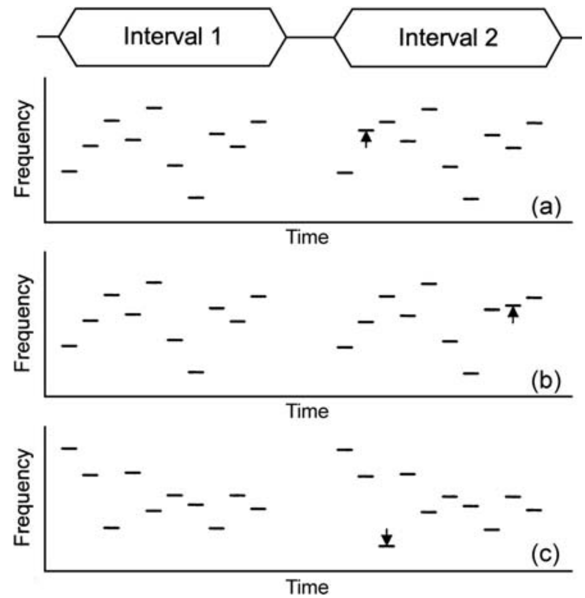

Figure 9.18 Schematic illustration of discrimination experiments in which subjects must determine whether pairs of 10-tone sequences are the same or different based on a frequency difference affecting one tone in the sequence. The frequency changes are highlighted by arrows and occur for the 2nd tone in trial a, the 9th tone in trial b, and the 3rd tone in trial c. Notice that the overall tonal patterns are similar in trials a and b, and is different in trial c.

Figure 9.19 illustrates the dramatic influence of stimulus uncertainty on the sizes of the DLs for frequency (Δf), intensity (ΔI), and time (ΔT), based on data reported by Watson and Kelly (1981). The term "informational masking" is used to describe the deterioration in perceptual performance due to uncertainty about the stimuli and will be revisited in Chapter 10.

TEMPORARY THRESHOLD SHIFT

It is not uncommon to experience a period of decreased hearing sensitivity, which lasts for some time, after being exposed to

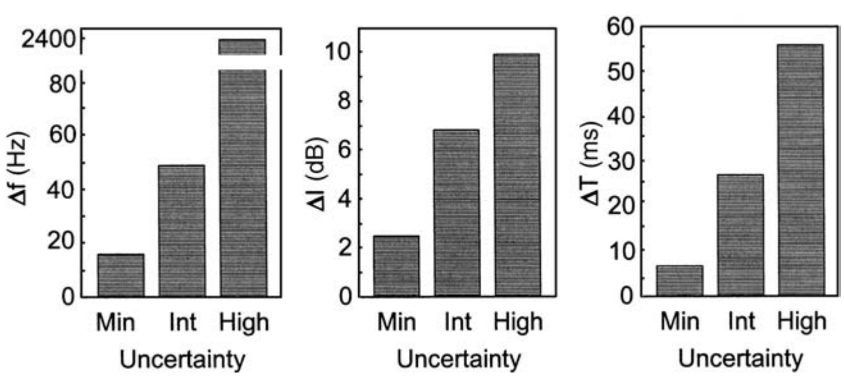

Figure 9.19 Increasing amounts of stimulus uncertainty [minimal (min), intermediate (int) and high] cause Δf, ΔI, and ΔT to increase considerably (informational masking). *Source*: Based on data by Watson and Kelly (1981).

high sound intensities, for example, after leaving a rock concert. This temporary shift in auditory threshold may last as long as roughly 16 h or more, improving gradually. The phenomenon is quite descriptively called **temporary threshold shift (TTS)** or **poststimulatory fatigue**.

Temporary threshold shift appears to be a manifestation of temporary changes in the hair cells as a result of exposure to the fatiguing stimulus. As one might expect, excessive and/or long-standing exposures may result in permanent threshold shifts, reflecting pathological changes or destruction of the hair cells and their associated structures. From the practical standpoint, the amount of TTS produced by exposure to a given fatiguing stimulus has been used as a predictor of individual susceptibility for noise-induced hearing loss. However, this approach is not unchallenged. Because space permits only brief coverage of TTS, the reader is referred to other sources for reviews of this topic and related areas (e.g., Elliott and Fraser, 1970; Kryter, 1985; Ward, 1973, 1991; Miller, 1974; Henderson et al., 1976; Melnick, 1991; Schmiedt, 1984; Saunders et al., 1985; Clark, 1991; Hamernik et al., 1991; Gelfand, 2001).

It has long been known that TTS is related to the stimulus intensity (Hirsh and Bilger, 1955; Ward et al., 1958, 1959a, 1959b; Mills et al., 1970). Exposure levels below approximately 80 dB SPL are often described as **effective quiet** because they do not appear to produce any TTS. Above effective quiet, the amount of threshold shift increases as stimulus intensity is raised, as illustrated in the upper frame of Fig. 9.20 (Mills, JH, Gengle, RW, Watson, CS, Miller, JD, 1970). For a given intensity, the amount of TTS increases with the duration of the fatiguing stimulus in a manner that is proportional to the logarithm of exposure time. In addition, higher exposure levels produce greater amounts of threshold shift. However, the amount of

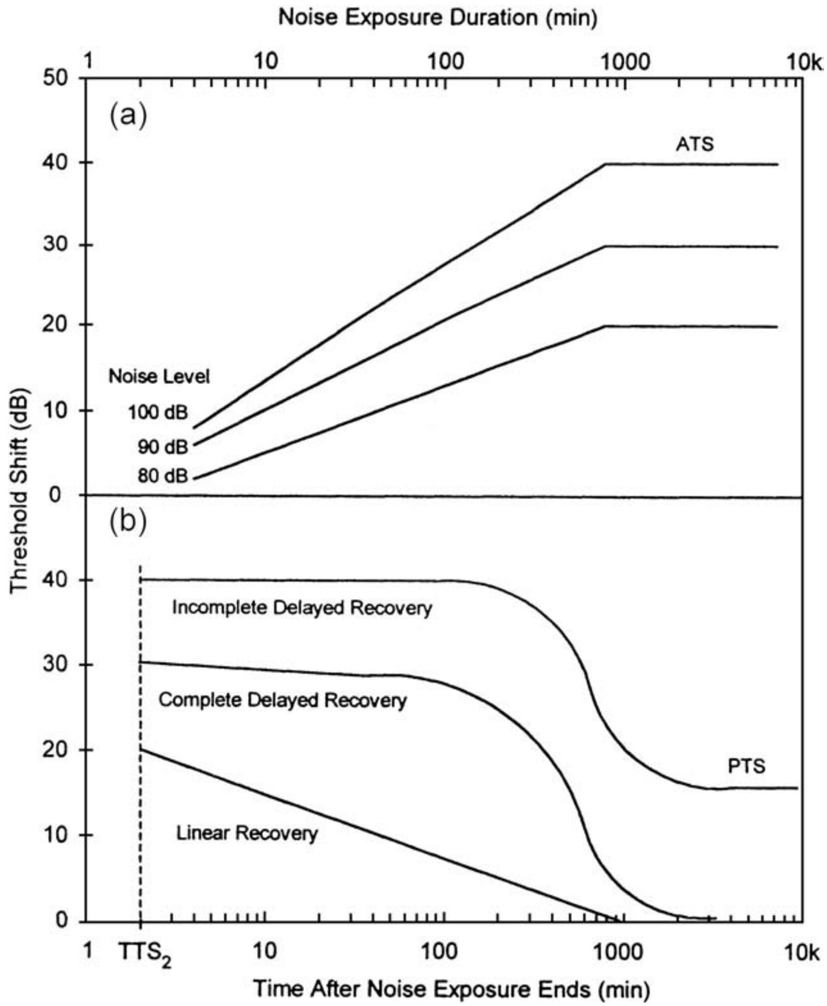

Figure 9.20 *Upper frame*: development of temporary threshold shift as a function of exposure duration (parameter is exposure level). *Lower frame*: patterns of recovery from temporary threshold shift with time after exposure ends. See text.

threshold shift eventually stops increasing after approximately 8 to 16 hours of exposure, when a maximum amount of TTS is achieved, called **asymptotic threshold shift (ATS)**.

It appears that higher frequency stimuli result in more TTS than do lower frequency fatiguers (Ward, 1963). The amount of TTS is smaller for intermittent sounds than for continuous stimulation (Ward et al., 1958, 1959a), and it appears to be related to the total or average time that the fatiguer is "on" during the course of stimulation (Ward, 1973). The frequency range over which TTS occurs becomes wider as the stimulus level is raised, and this affect is asymmetrical in that the higher frequencies (up to roughly 4000–6000 Hz) are the most severely affected. Temporary threshold shift reaches a maximum at a higher frequency than the fatiguer frequency, generally about one-half to one-octave above (Ward, 1962; Elliott and Fraser, 1970; Ward, 1973; Miller, 1974; Schmiedt, 1984; Saunders et al., 1985). Physiological work has also revealed that the greatest amount of TTS measured for single auditory neurons occurs when the animal is exposed to high-level sounds a half-octave below the neuron's characteristic frequency (Cody and Johnstone, 1981). The basis of this phenomenon appears to be that the location of the maximum basilar membrane vibration shifts in a basal (higher frequency) direction by an amount equivalent to about a half-octave at high levels of stimulation (Johnstone et al., 1986).

The course of recovery from TTS may be measured at various times after the fatiguing stimulus has been turned off. The recovery course is rather complicated within about 2 minutes following the offset of the fatiguer, during which time it is nonmonotonic with a "bounce" occurring in the vicinity of the two-minute point (Hirsh and Ward, 1952; Hirsh and Bilger, 1955). This bounce is a reversal of the recovery after which the TTS begins to decrease again with time. For this reason, the amount of TTS produced by a given exposure is typically measured at the two-minute point and is therefore called **TTS_2**. The course of recovery of temporary threshold shift is then measured beginning from TTS_2. Three types of recovery are illustrated in the lower panel of Fig. 9.20. The normal course of recovery is seen as a straight line and progresses at a rate that is proportional to the logarithm of the time since exposure (Ward et al., 1959a, 1959b). Greater amounts of TTS begin their recovery after some delay. Both of these recovery patterns are complete, meaning that hearing sensitivity eventually returns to its pre-exposure level. In contrast, it is possible that large amounts of TTS may not resolve completely, leaving the person with a **permanent threshold shift (PTS)** or noise-induced hearing loss.

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APPENDIX 9.1

Maximum Permissible Room Noise Levels[a] (As Octave Band Levels in dB) Required for the Measurement of Thresholds As Low As the Reference Levels in Tables 9.1 to 9.3 and 0 dB HL in Fig. 9.2

Octave band center frequency (Hz)	Ears covered with		Ears not covered
	Supra-aural receivers	Insert receivers	
125	39[b] (35[c])	67[b] (59[c])	35[b] (29[c])
250	25	53	21
500	21	50	16
1000	26	47	13
2000	34	49	14
4000	37	50	11
8000	37	56	14

[a]Different room noise levels apply when the ears are covered versus uncovered because the amount of noise entering the ear is reduced by earphone muffs and insert receivers.
[b]Applies when 250 Hz is the lowest frequency tested.
[c]Applies when 125 Hz is the lowest frequency tested.
Source: Based on ANSI S3.1-1999 [R2003].

APPENDIX 9.2

Maximum Permissible Room Noise Levels[a] (As Third-Octave Band Levels in dB) Required for the Measurement of Thresholds As Low As the Reference Levels in Tables 9.1 to 9.3 and 0 dB HL in Fig. 9.2

Third-octave band center frequency (Hz)	Ears covered with		Ears not covered
	Supra-aural receivers	Insert receivers	
125	34[b] (30[c])	62[b] (54[c])	30[b] (24[c])
250	20	48	16
500	16	45	11
800	19	44	10
1000	21	42	8
1600	25	43	9
2000	29	44	9
3150	33	46	8
4000	32	45	6
6300	32	48	8
8000	32	51	9

[a]Different room noise levels apply when the ears are covered versus uncovered because the amount of noise entering the ear is reduced by earphone muffs and insert receivers.
[b]Applies when 250 Hz is the lowest frequency tested.
[c]Applies when 125 Hz is the lowest frequency tested.
Source: Based on ANSI S3.1-1999 [R2003].