CHAPTER 14

MPEG Audio Compression

Have you ever attended a dance and found that for quite some time afterward you couldn't hear much? You were dealing with a type of *temporal masking*!

Have you ever noticed that the person on the sound board at a dance basically cannot hear high frequencies anymore? Since many technicians have such hearing damage, some compensate by increasing the volume levels of the high frequencies, so they can hear them. If your hearing is not damaged, you experience this music mix as too piercing.

Moreover, if a very loud tone is produced, you also notice it is impossible to hear any sound nearby in the frequency spectrum — the band's singing may be drowned out by the lead guitar. If you've noticed this, you have experienced *frequency masking*!

MPEG audio uses this kind of perception phenomenon by simply giving up on the tones that can't be heard anyway. Using a curve of human hearing perceptual sensitivity, an MPEG audio codec makes decisions on when and to what degree frequency masking and temporal masking make some components of the music inaudible. It then controls the quantization process so that these components do not influence the output.

So far, in the previous chapter, we have concentrated on telephony applications — usually, LPC and CELP are tuned to speech parameters. In contrast, in this chapter, we consider compression methods applicable to general audio, such as music or perhaps broadcast digital TV. Instead of modeling speech, the method used is a *waveform* coding approach — one that attempts to make the decompressed amplitude-versus-time waveform as much as possible like the input signal.

A main technique used in evaluating audio content for possible compression makes use of a *psychoacoustic model* of hearing. The kind of coding carried out, then, is generally referred to as *perceptual coding*.

In this chapter, we look at how such considerations impact MPEG audio compression standards and examine in some detail at the following topics:

- Psychoacoustics
- MPEG-1 Audio Compression
- Later MPEG audio developments: MPEG-2, 4, 7, and 21

14.1 PSYCHOACOUSTICS

Recall that the range of human hearing is about 20 Hz to about 20 kHz (for people who have not gone to many dances). Sounds at higher frequencies are *ultrasonic*. However, the

frequency range of the voice is typically only from about 500 Hz to 4 kHz. The dynamic range, the ratio of the maximum sound amplitude to the quietest sound humans can hear, is on the order of about 120 dB.

Recall that the decibel unit represents ratios of intensity on a logarithmic scale. The reference point for 0 dB is the threshold of human hearing — the quietest sound we can hear, measured at 1 kHz. Technically, this is a sound that creates a barely audible sound intensity of 10^{-12} Watt per square meter. Our range of magnitude perception is thus incredibly wide: the level at which the sensation of sound begins to give way to the sensation of pain is about 1 Watt/m^2, so we can perceive a ratio of 10^{12}!

The range of hearing actually depends on frequency. At a frequency of 2 kHz, the ear can readily respond to sound that is about 96 dB more powerful than the smallest perceivable sound at that frequency, or in other words a power ratio of 2^{32}. Table 6.1 lists some of the common sound levels in decibels.

14.1.1 Equal-Loudness Relations

Suppose we play two pure tones, sinusoidal sound waves, with the same amplitude but different frequencies. Typically, one may sound louder than the other. The reason is that the ear does not hear low or high frequencies as well as frequencies in the middle range. In particular, at normal sound volume levels, the ear is most sensitive to frequencies between 1 kHz and 5 kHz.

Fletcher-Munson Curves. The Fletcher-Munson equal-loudness curves display the relationship between perceived loudness (*in phons*) for a given stimulus sound volume (*Sound Pressure Level*, in dB), as a function of frequency. Figure 14.1 shows the ear's perception of equal loudness. The abscissa (shown in a semi-log plot) is frequency, in kHz. The ordinate axis is sound pressure level — the actual loudness of the tone generated in an experiment. The curves show the loudness with which such tones are perceived by humans. The bottom curve shows what level of pure tone stimulus is required to produce the perception of a 10 dB sound.

All the curves are arranged so that the perceived loudness level gives the same loudness as for that loudness level of a pure tone at 1 kHz. Thus, the loudness level at the 1 kHz point is always equal to the dB level on the ordinate axis. The bottom curve, for example, is for 10 phons. All the tones on this curve will be perceived as loud as a 10 dB, 1,000 Hz tone. The figure shows more accurate curves, developed by Robinson and Dadson [1], than the Fletcher and Munson originals [2].

The idea is that a tone is produced at a certain frequency and *measured* loudness level, then a human rates the loudness as it is perceived. On the lowest curve shown, each pure tone between 20 Hz and 15 kHz would have to be produced at the volume level given by the ordinate for it to be perceived at a 10 dB loudness level [1]. The next curve shows what the magnitude would have to be for pure tones to each be perceived as being at 20 dB, and so on. The top curve is for perception at 90 dB.

For example, at 5,000 Hz, we perceive a tone to have a loudness level of 10 phons when the source is actually only 5 dB. Notice that at the dip at 4 kHz, we perceive the sound as being about 10 dB, when in fact the stimulation is only about 2 dB. To perceive the same

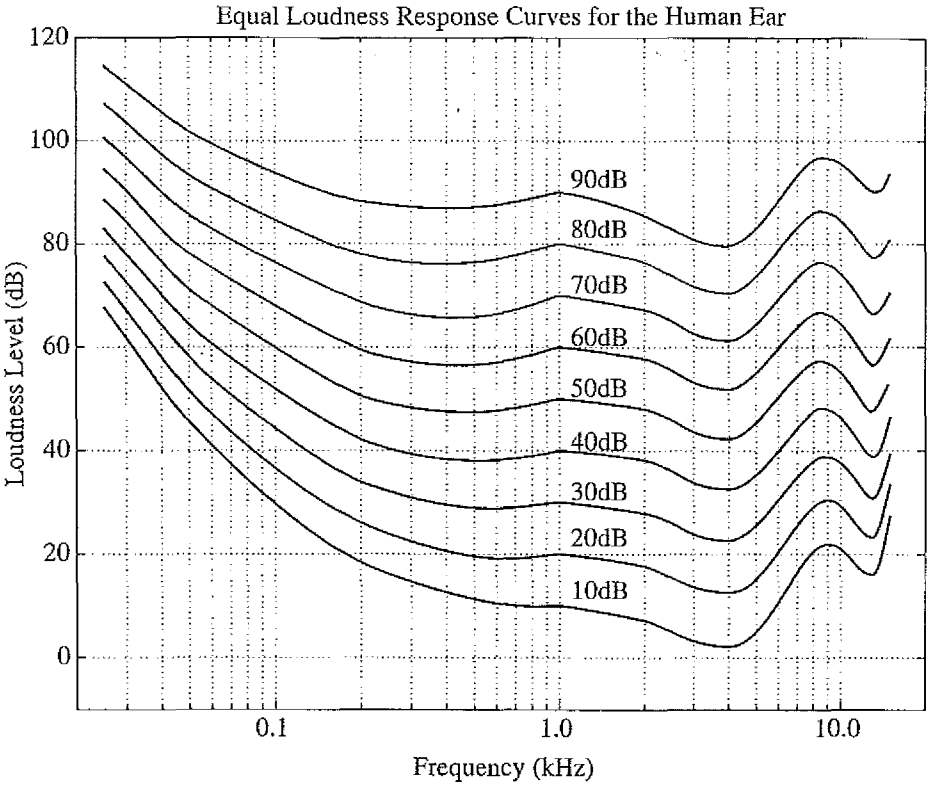

FIGURE 14.1: Fletcher-Munson equal loudness response curves for the human ear (remeasured by Robinson and Dadson).

effective 10 dB at 10 kHz, we would have to produce an absolute magnitude of 20 dB. The ear is clearly more sensitive in the range 2 kHz to 5 kHz and not nearly as sensitive in the range 6 kHz and above.

At the lower frequencies, if the source is at level 10 dB, a 1 kHz tone would also sound at 10 dB; however, a lower, 100 Hz tone must be at a level 30 dB — 20 dB higher than the 1 kHz tone! So we are not very sensitive to the lower frequencies. The explanation of this phenomenon is that the ear canal amplifies frequencies from 2.5 to 4 kHz.

Note that as the overall loudness increases, the curves flatten somewhat. We are approximately equally sensitive to low frequencies of a few hundred Hz if the sound level is loud enough. And we perceive most low frequencies better than high ones at high volume levels. Hence, at the dance, loud music sounds better than quiet music, because then we can actually hear low frequencies and not just high ones. (A "loudness" switch on some sound systems simply boosts the low frequencies as well as some high ones.) However, above 90 dB, people begin to become uncomfortable. A typical city subway operates at about 100 dB.

14.1.2 Frequency Masking

How does one tone interfere with another? At what level does one frequency drown out another? This question is answered by masking curves. Also, masking answers the question of how much noise we can tolerate before we cannot hear the actual music. Lossy audio data compression methods, such as MPEG Audio or Dolby Digital (AC-3) encoding, which is popular in movies, remove some sounds that are masked anyway, thus reducing the total amount of information.

The general situation in regard to masking is as follows:

- A lower tone can effectively mask (make us unable to hear) a higher tone.
- The reverse is not true. A higher tone does not mask a lower tone well. Tones can in fact mask lower-frequency sounds, but not as effectively as they mask higher-frequency ones.
- The greater the power in the masking tone, the wider its influence — the broader the range of frequencies it can mask.
- As a consequence, if two tones are widely separated in frequency, little masking occurs.

Threshold of Hearing. Figure 14.2 shows a plot of the threshold of human hearing, for pure tones. To determine such a plot, a particular frequency tone is generated, say 1 kHz. Its volume is reduced to zero in a quiet room or using headphones, then turned up until the sound is just barely audible. Data points are generated for all audible frequencies in the same way.

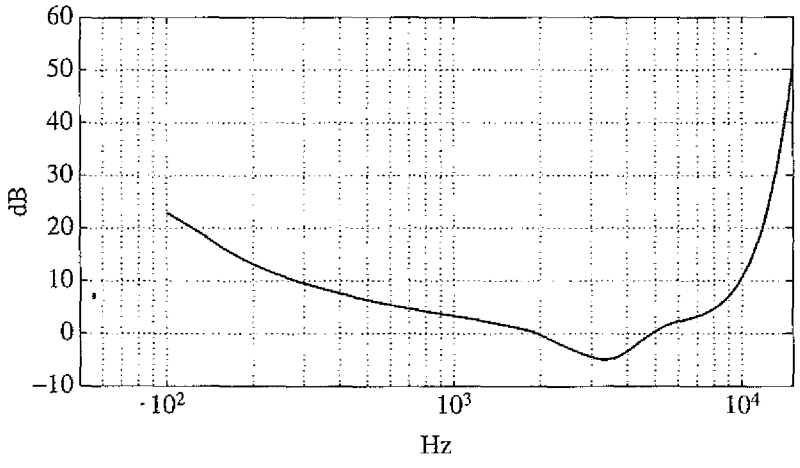

FIGURE 14.2: Threshold of human hearing, for pure tones.

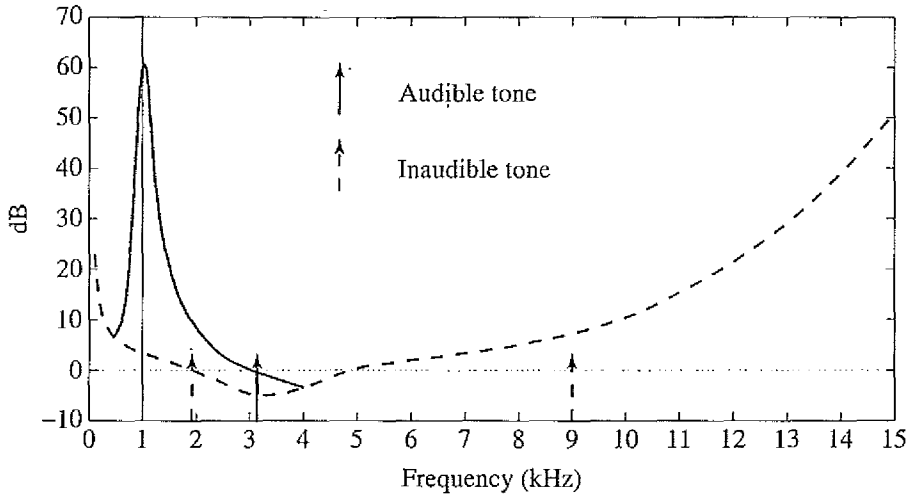

FIGURE 14.3: Effect on threshold of human hearing for a 1 kHz masking tone.

The point of the threshold of hearing curve is that if a sound is above the dB level shown — say it is above 2 dB for a 6 kHz tone — then the sound is audible. Otherwise, we cannot hear it. Turning up the 6 kHz tone so that it equals or surpasses the curve means we can then distinguish the sound.

An approximate formula exists for this curve, as follows [3]:

$$Threshold(f) = 3.64(f/1000)^{-0.8} - 6.5e^{-0.6(f/1000-3.3)^2} + 10^{-3}(f/1000)^4 \quad (14.1)$$

The threshold units are dB. Since the dB unit is a ratio, we do have to choose which frequency will be pinned to the origin, (0, 0). In Equation (14.1), this frequency is 2,000 Hz: $Threshold(f) = 0$ at $f = 2$ kHz.

Frequency Masking Curves. Frequency masking is studied by playing a particular pure tone, say 1 kHz again, at a loud volume and determining how this tone affects our ability to hear tones at nearby frequencies. To do so, we would generate a 1 kHz *masking tone* at a fixed sound level of 60 dB, then raise the level of a nearby tone, say 1.1 kHz, until it is just audible. The threshold in Figure 14.3 plots this audible level.

It is important to realize that this masking diagram holds only for a single masking tone: the plot changes if other masking tones are used. Figure 14.4 shows how this looks: the higher the frequency of the masking tone, the broader a range of influence it has.

If, for example, we play a 6 kHz tone in the presence of a 4 kHz masking tone, the masking tone has raised the threshold curve much higher. Therefore, at its neighbor frequency of 6 kHz, we must now surpass 30 dB to distinguish the 6 kHz tone.

The practical point is that if a signal can be decomposed into frequencies, then for frequencies that will be partially masked, only the audible part will be used to set quantization noise thresholds.

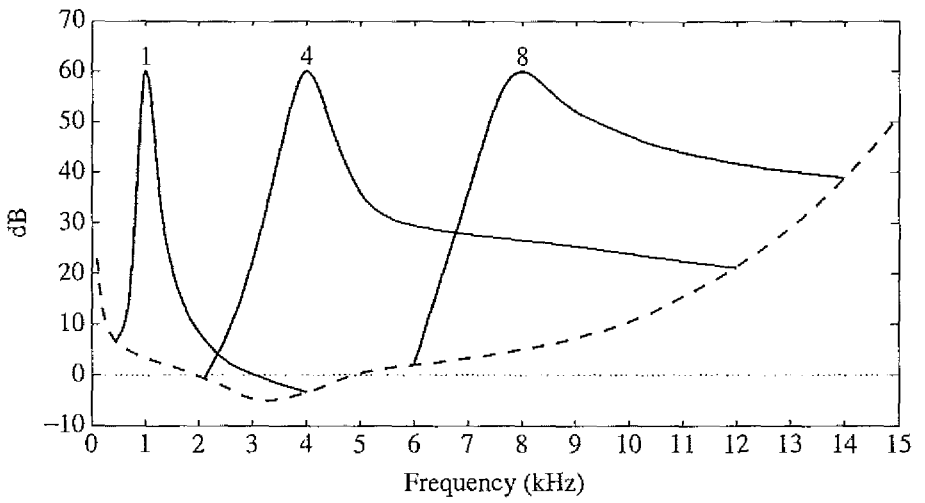

FIGURE 14.4: Effect of masking tones at three different frequencies.

Critical Bands. The human hearing range naturally divides into *critical bands*, with the property that the human auditory system cannot resolve sounds better than within about one critical band when other sounds are present. Hearing has a limited, frequency-dependent resolution. According to [4], "In a complex tone, the critical bandwidth corresponds to the smallest frequency difference between two partials such that each can still be heard separately.... the critical bandwidth represents the ear's resolving power for simultaneous tones or partials."

At the low-frequency end, a critical band is less than 100 Hz wide, while for high frequencies, the width can be greater than 4 kHz. This indeed is yet another kind of *perceptual nonuniformity.*

Experiments indicate that the critical bandwidth remains approximately constant in width for masking frequencies below about 500 Hz — this width is about 100 Hz. However, for frequencies above 500 Hz, the critical bandwidth increases approximately linearly with frequency.

Generally, the audio frequency range for hearing can be partitioned into about 24 critical bands (25 are typically used for coding applications), as Table 14.1 shows.

Notwithstanding the *general* definition of a critical band, it turns out that our hearing apparatus actually is somewhat tuned to *certain* critical bands. Since hearing depends on physical structures in the inner ear, the frequencies at which these structures best resonate is important. Frequency masking is a result of the ear structures becoming "saturated" at the masking frequency and nearby frequencies.

Hence, the ear operates something like a set of band-pass filters, which each allows a limited range of frequencies through and blocks all others. Experiments that show this are based on the observation that a constant-volume sound will seem louder if it spans the boundary between two critical bands than it would were it contained entirely within one critical band [5]. In effect, the ear is not very discriminating *within* a critical band, because of masking.

TABLE 14.1: Critical bands and their bandwidths.

Band #	Lower bound (Hz)	Center (Hz)	Upper bound (Hz)	Bandwidth (Hz)
1	-	50	100	-
2	100	150	200	100
3	200	250	300	100
4	300	350	400	100
5	400	450	510	110
6	510	570	630	120
7	630	700	770	140
8	770	840	920	150
9	920	1000	1080	160
10	1080	1170	1270	190
11	1270	1370	1480	210
12	1480	1600	1720	240
13	1720	1850	2000	280
14	2000	2150	2320	320
15	2320	2500	2700	380
16	2700	2900	3150	450
17	3150	3400	3700	550
18	3700	4000	4400	700
19	4400	4800	5300	900
20	5300	5800	6400	1100
21	6400	7000	7700	1300
22	7700	8500	9500	1800
23	9500	10500	12000	2500
24	12000	13500	15500	3500
25	15500	18775	22050	6550

Bark Unit. Since the range of frequencies affected by masking is broader for higher frequencies, it is useful to define a new frequency unit such that, in terms of this new unit, each of the masking curves (the parts of Figure 14.4 above the threshold in quiet) have about the same width.

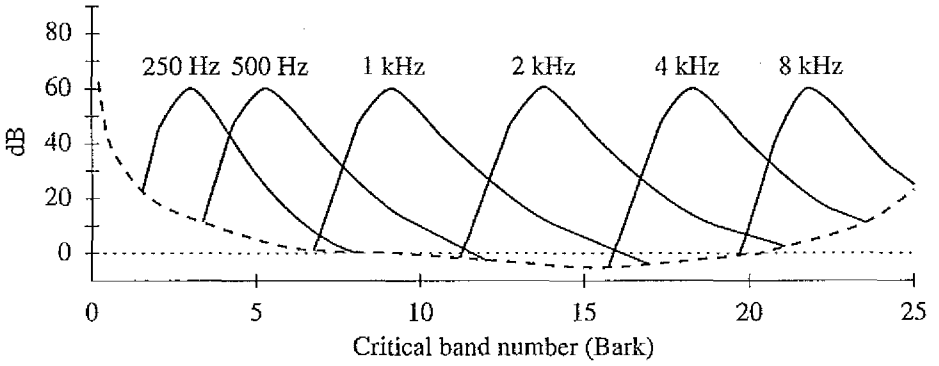

FIGURE 14.5: Effect of masking tones, expressed in Bark units.

The new unit defined is called the *Bark*, named after Heinrich Barkhausen (1881–1956), an early sound scientist. One Bark unit corresponds to the width of one critical band, for any masking frequency [6, 7]. Figure 14.5 displays critical bands, with the frequency (the abscissa) given in Bark units.

The conversion between a frequency f and its corresponding critical-band number b, expressed in Bark units, is as follows:

$$\text{Critical band number (Bark)} = \begin{cases} f/100, & \text{for} f < 500 \\ 9 + 4\log_2(f/1000), & \text{for} f \geq 500 \end{cases} \tag{14.2}$$

In terms of this new frequency measure, the critical-band number b equals 5 when $f =$ 500 Hz. At double that frequency, for a masking frequency of 1 kHz, the Bark value goes up to 9. Another formula used for the Bark scale is as follows:

$$b = 13.0 \arctan(0.76f) + 3.5 \arctan(f^2/56.25) \tag{14.3}$$

where f is in kHz and b is in Barks. The inverse equation gives the frequency (in kHz) corresponding to a particular Bark value b:

$$f = [(\exp(0.219 \times b)/352) + 0.1] \times b - 0.032 \times \exp[-0.15 \times (b-5)^2] \tag{14.4}$$

Frequencies forming the boundaries between two critical bands are given by integer Bark values. The critical bandwidth (df) for a given center frequency f can also be approximated by [8]

$$df = 25 + 75 \times [1 + 1.4(f^2)]^{0.69} \tag{14.5}$$

where f is in kHz and df is in Hz.

The idea of the Bark unit is to define a more perceptually uniform unit of frequency, in that every critical band's width is roughly equal in terms of Barks.

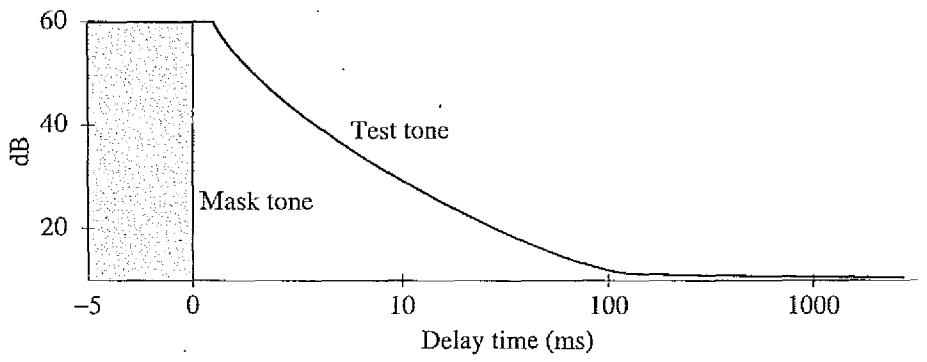

FIGURE 14.6: The louder the test tone, the shorter the amount of time required before the test tone is audible once the masking tone is removed.

14.1.3 Temporal Masking

Recall that after the dance it takes quite a while for our hearing to return to normal. Generally, any loud tone causes the hearing receptors in the inner ear (little hairlike structures called *cilia*) to become *saturated*, and they require time to recover. (Many other perceptual systems behave in this temporally slow fashion — for example, the receptors in the eye have this same kind of "capacitance" effect.)

To quantify this type of behavior, we can measure the time sensitivity of hearing by another masking experiment. Suppose we again play a masking tone at 1 kHz with a volume level of 60 dB, and a nearby tone at, say, 1.1 kHz with a volume level of 40 dB. Since the nearby test tone is masked, it cannot be heard. However, once the masking tone is turned off, we can again hear the 1.1 kHz tone, but only after a small amount of time. The experiment proceeds by stopping the test tone slightly after the masking tone is turned off, say 10 msec later.

The delay time is adjusted to the minimum amount of time such that the test tone can just be distinguished. In general, the louder the test tone, the less time it takes for our hearing to get over hearing the masking tone. Figure 14.6 shows this effect: it may take up to as much as 500 msec for us to discern a quiet test tone after a 60 dB masking tone has been played. Of course, this plot would change for different masking tone frequencies.

Test tones with frequencies near the masking tone are, of course, the most masked. Therefore, for a given masking tone, we have a two-dimensional temporal masking situation, as in Figure 14.7. The closer the frequency to the masking tone and the closer in time to when the masking tone is stopped, the greater likelihood that a test tone cannot be heard. The figure shows the total effect of both frequency and temporal masking.

The phenomenon of saturation also depends on just how long the masking tone has been applied. Figure 14.8 shows that for a masking tone played longer (200 msec) than another (100 msec), it takes longer before a test tone can be heard.

As well as being able to mask other signals that occur just after it sounds (*post-masking*), a particular signal can even mask sounds played just before the stronger signal (*pre-masking*).

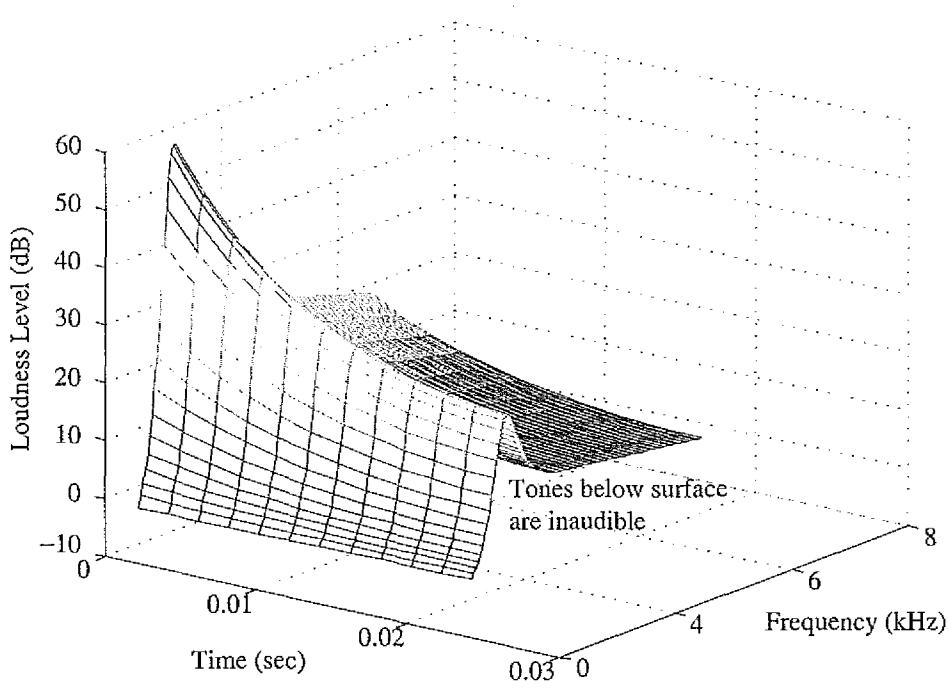

FIGURE 14.7: Effect of temporal masking depends on both time and closeness in frequency.

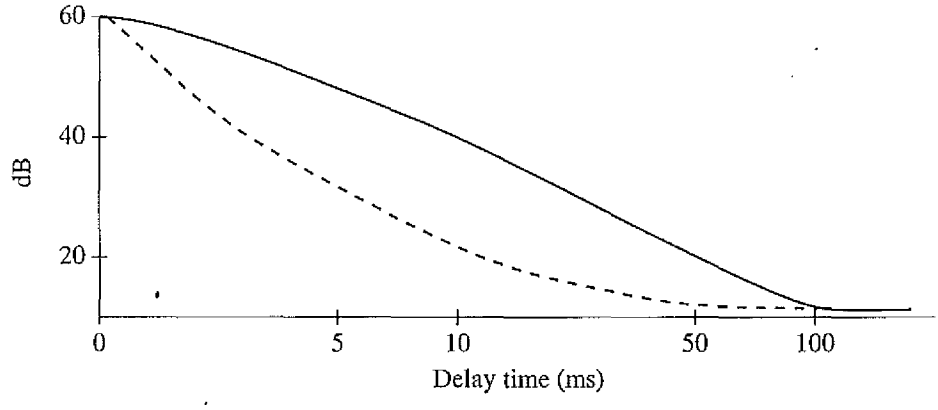

FIGURE 14.8: Effect of temporal masking also depends on the length of time the masking tone is applied. Solid curve: masking tone played for 200 msec; dashed curve: masking tone played for 100 msec.

Pre-masking has a much shorter effective interval (2-5 msec) in which it is operative than does post-masking (usually 50–200 msec).

MPEG audio compression takes advantage of these considerations in basically constructing a large, multidimensional lookup table. It uses this to transmit frequency components that are masked by frequency masking or temporal masking or both, using fewer bits.

14.2 MPEG AUDIO

MPEG Audio proceeds by first applying a filter bank to the input, to break the input into its frequency components. In parallel, it applies a psychoacoustic model to the data, and this model is used in a bit-allocation block. Then the number of bits allocated is used to quantize the information from the filter bank. The overall result is that quantization provides the compression, and bits are allocated where they are most needed to lower the quantization noise below an audible level.

14.2.1 MPEG Layers

MP3 is a popular audio compression standard. The "3" stands for Layer 3, and "MP" stands for the MPEG-1 standard. Recall that we looked at MPEG video compression in Chapter 11. However, the MPEG standard actually delineates three different aspects of multimedia: audio, video, and systems. MP3 forms part of the audio component of this first phase of MPEG. It was released in 1992 and resulted in the international standard ISO/IEC 11172-3, published in 1993.

MPEG audio sets out three downward-compatible *layers* of audio compression, each able to understand the lower layers. Each offers more complexity in the psychoacoustic model applied and correspondingly better compression for a given level of audio quality. However, an increase in complexity, and concomitantly in compression effectiveness, is accompanied by extra delay.

Layers 1 to 3 in MPEG Audio are compatible, because all layers include the same file header information.

Layer 1 quality can be quite good, provided a comparatively high bitrate is available. Digital Audio Tape typically uses Layer 1. Layer 2 has more complexity and was proposed for use in digital audio broadcasting. Layer 3 (MP3) is most complex and was originally aimed at audio transmission over ISDN lines. Each of the layers also uses a different frequency transform.

Most of the complexity increase is at the encoder rather than at the decoder side, and this accounts for the popularity of MP3 players. Layer 1 incorporates the simplest psychoacoustic model, and Layer 3 uses the most complex. The objective is a good tradeoff between quality and bitrate. "Quality" is defined in terms of listening test scores (the psychologists hold sway here), where a quality measure is defined by:

- 5.0 = "Transparent" — undetectable difference from original signal; equivalent to CD-quality audio at 14- to 16-bit PCM
- 4.0 = Perceptible difference, but not annoying
- 3.0 = Slightly annoying

- 2.0 = Annoying
- 1.0 = Very annoying

(Now that's scientific!) At 64 kbps per channel, Layer 2 scores between 2.1 and 2.6, and Layer 3 scores between 3.6 and 3.8. So Layer 3 provides a substantial improvement but is still not perfect by any means.

14.2.2 MPEG Audio Strategy

Compression is certainly called for, since even audio can take fairly substantial bandwidth: CD audio is sampled at 44.1 kHz and 16 bits/channel, so for two channels needs a bitrate of about 1.4 Mbps. MPEG-1 aims at about 1.5 Mbps overall, with 1.2 Mbps for video and 256 kbps for audio.

The MPEG approach to compression relies on quantization, of course, but also recognizes that the human auditory system is not accurate within the width of a critical band, both in terms of perceived loudness and audibility of a test frequency. The encoder employs a bank of filters that act to first analyze the frequency (*spectral*) components of the audio signal by calculating a frequency transform of a window of signal values. The bank of filters decomposes the signal into subbands. Layer 1 and Layer 2 codecs use a *quadrature-mirror filter* bank, while the Layer 3 codec adds a DCT. For the psychoacoustic model, a Fourier transform is used.

Then frequency masking can be brought to bear by using a psychoacoustic model to estimate the just noticeable noise level. In its quantization and coding stage, the encoder balances the masking behavior and the available number of bits by discarding inaudible frequencies and scaling quantization according to the sound level left over, above masking levels.

A sophisticated model would take into account the actual width of the critical bands centered at different frequencies. Within a critical band, our auditory system cannot finely resolve neighboring frequencies and instead tends to blur them. As mentioned earlier, audible frequencies are usually divided into 25 main critical bands, inspired by the auditory critical bands.

However, in keeping with design simplicity, the model adopts a *uniform width* for all frequency analysis filters, using 32 overlapping subbands [9, 10]. This means that at lower frequencies, each of the frequency analysis "subbands" covers the width of several critical bands of the auditory system, whereas at higher frequencies this is not so, since a critical band's width is less than 100 Hz at the low end and more than 4 kHz at the high end. For each frequency band, the sound level above the masking level dictates how many bits must be assigned to code signal values, so that quantization noise is kept below the masking level and hence cannot be heard.

In Layer 1, the psychoacoustic model uses only frequency masking. Bitrates range from 32 kbps (mono) to 448 kbps (stereo). Near-CD stereo quality is possible with a bitrate of 256–384 kbps. Layer 2 uses some temporal masking by accumulating more samples and examining temporal masking between the current block of samples and the ones just before and just after. Bitrates can be 32–192 kbps (mono) and 64–384 kbps (stereo). Stereo CD-audio quality requires a bitrate of about 192–256 kbps.

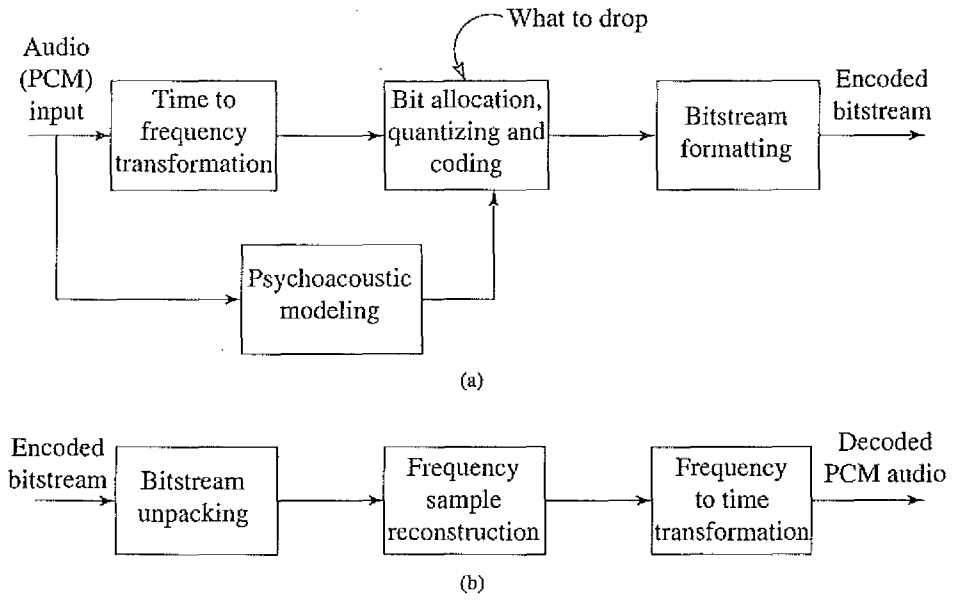

FIGURE 14.9: (a) Basic MPEG Audio encoder; and (b) decoder.

However, temporal masking is less important for compression than is frequency masking, which is why it is sometimes disregarded entirely in lower-complexity coders. Layer 3 is directed toward lower bitrate applications and uses a more sophisticated subband analysis, with nonuniform subband widths. It also adds nonuniform quantization and entropy coding. Bitrates are standardized at 32–320 kbps.

14.2.3 MPEG Audio Compression Algorithm

Basic Algorithm. Figure 14.9 shows the basic MPEG audio compression algorithm. It proceeds by dividing the input into 32 frequency subbands, via a filter bank. This is a linear operation that takes as its input a set of 32 PCM samples, sampled in time, and produces as its output 32 frequency coefficients. If the sampling rate is f_s, say $f_s = 48$ ksps (kilosamples per second; i.e., 48 kHz), then by the Nyquist theorem, the maximum frequency mapped will be $f_s/2$. Thus the mapped bandwidth is divided into 32 equal-width segments, each of width $f_s/64$ (these segments overlap somewhat).

In the Layer 1 encoder, the sets of 32 PCM values are first assembled into a set of 12 groups of 32s. Hence, the coder has an inherent time lag, equal to the time to accumulate 384 (i.e., 12×32) samples. For example, if sampling proceeds at 32 kbps, then a time duration of 12 msec is required since each set of 32 samples is transmitted each millisecond. These sets of 12 samples, each of size 32, are called *segments*. The point of assembling them is to examine 12 sets of values at once in each of the 32 subbands, after frequency analysis has been carried out, then base quantization on just a summary figure for all 12 values.

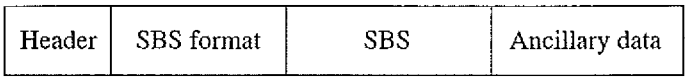

FIGURE 14.10: Example MPEG Audio frame.

The delay is actually somewhat longer than that required to accumulate 384 samples, since header information is also required. As well, *ancillary data*, such as multilingual data and surround-sound data, is allowed. Higher layers also allow more than 384 samples to be analyzed, so the format of the subband-samples (SBS) is also added, with a resulting *frame* of data, as in Figure 14.10. The header contains a synchronization code (twelve 1s — 111111111111), the sampling rate used, the bitrate, and stereo information. The frame format also contains room for so-called "ancillary" (extra) information. (In fact, an MPEG-1 audio decoder can at least partially decode an MPEG-2 audio bitstream, since the file header begins with an MPEG-1 header and places the MPEG-2 datastream into the MPEG-1 Ancillary Data location.)

MPEG Audio is set up to be able to handle stereo or mono channels, of course. A special *joint-stereo* mode produces a single stream by taking into account the redundancy between the two channels in stereo. This is the audio version of a composite video signal. It can also deal with *dual-monophonic* — two channels coded independently. This is useful for parallel treatment of audio — for example, two speech streams, one in English and one in Spanish.

Consider the 32 × 12 segment as a 32 × 12 matrix. The next stage of the algorithm is concerned with scale, so that proper quantization levels can be set. For each of the 32 subbands, the maximum amplitude of the 12 samples in that row of the array is found, which is the *scaling factor* for that subband. This maximum is then passed to the bit-allocation block of the algorithm, along with the SBS (subband samples). The key point of the bit-allocation block is to determine how to apportion the total number of code bits available for the quantization of subband signals to minimize the audibility of the quantization noise.

As we know, the psychoacoustic model is fairly complex — more than just a set of lookup tables (and in fact this model is not standardized in the specification — it forms part of the "art" content of an audio encoder and is one major reason all encoders are not the same). In Layer 1, a decision step is included to decide whether each frequency band is basically like a tone or like noise. From that decision and the scaling factor, a masking threshold is calculated for each band and compared with the threshold of hearing.

The model's output consists of a set of what are known as *signal-to-mask ratios* (*SMRs*) that flag frequency components with amplitude below the masking level. The SMR is the ratio of the short-term signal power within each frequency band to the minimum masking threshold for the subband. The SMR gives the amplitude resolution needed and therefore also controls the bit allocations that should be given to the subband. After determination of the SMR, the scaling factors discussed above are used to set quantization levels such that quantization error itself falls below the masking level. This ensures that more bits are used in regions where hearing is most sensitive. In sum, the coder uses fewer bits in critical bands when fewer can be used without making quantization noise audible.

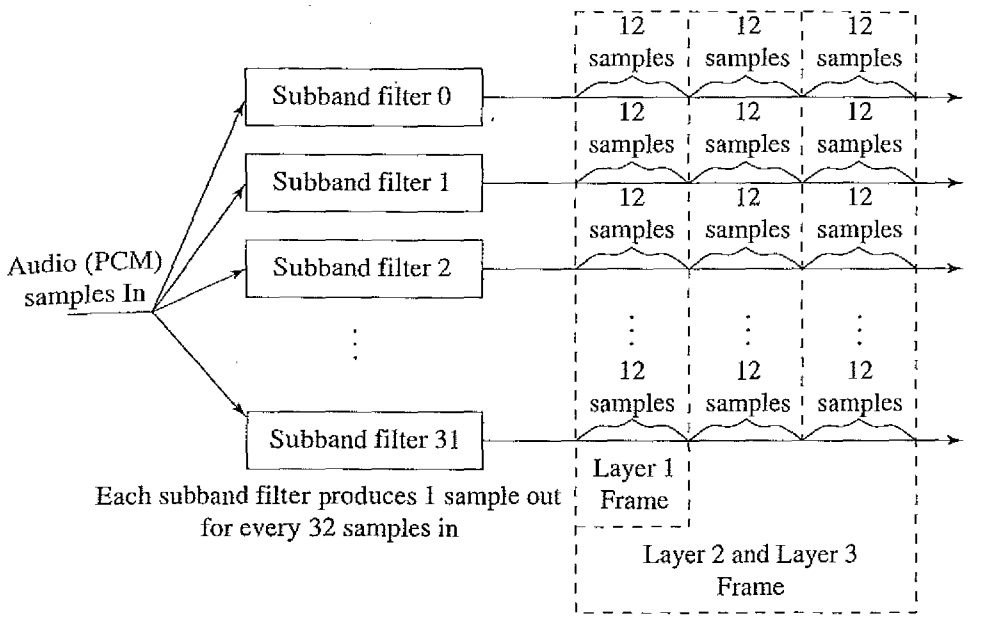

FIGURE 14.11: MPEG Audio frame sizes.

The scaling factor is first quantized, using 6 bits. The 12 values in each subband are then quantized. Using 4 bits, the bit allocations for each subband are transmitted, after an iterative bit allocation scheme is used. Then the data is transmitted, with appropriate bit depths for each subband. Altogether, the data consisting of the quantized scaling factor and the 12 codewords are grouped into a collection known as the Subband-Sample format.

On the decoder side, the values are de-quantized, and magnitudes of the 32 samples are reestablished. These are passed to a bank of *synthesis filters*, which reconstitute a set of 32 PCM samples. Note that the psychoacoustic model is not needed in the decoder.

Figure 14.11 shows how samples are organized. A Layer 2 or Layer 3 frame actually accumulates more than 12 samples for each subband: instead of 384 samples, a frame includes 1,152 samples.

Bit Allocation. The bit-allocation algorithm is not part of the standard, and it can therefore be done in many possible ways. The aim is to ensure that all the quantization noise is below the masking thresholds. However, this is usually not the case for low bitrates. The psychoacoustic model is brought into play for such cases, to allocate more bits, from the number available, to the subbands where increased resolution will be most beneficial. One common scheme is as follows.

For each subband, the psychoacoustic model calculates the *Signal-to-Mask Ratio*, in dB. A lookup table in the MPEG Audio standard also provides an estimate of the SNR (signal-to-*noise* ratio), assuming quantization to a given number of quantizer levels.

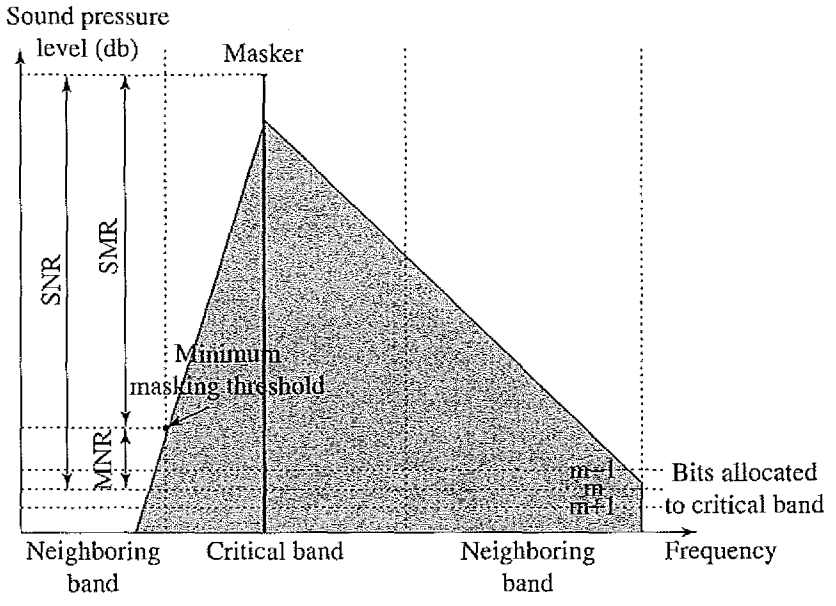

FIGURE 14.12: Mask-to-noise ratio and signal-to-mask ratio. A qualitative view of SNR, SMR and MNR, with one dominant masker and m bits allocated to a particular critical band.

Then the *Mask-to-Noise Ratio* (MNR) is defined as the difference

$$\text{MNR}_{\text{dB}} = \text{SNR}_{\text{dB}} - \text{SMR}_{\text{dB}} \tag{14.6}$$

as Figure 14.12 shows. The lowest MNR is determined, over all the subbands, and the number of code-bits allocated to this subband is incremented. Then a new estimate of the SNR is made, and the process iterates until no more bits are left to allocate.

Mask calculations are performed in parallel with subband filtering, as in Figure 14.13. The masking curve calculation requires an accurate frequency decomposition of the input signal, using a Discrete Fourier Transform (DFT). The frequency spectrum is usually calculated with a 1,024-point Fast Fourier Transform (FFT).

In Layer 1, 16 uniform quantizers are pre-calculated, and for each subband the quantizer giving the lowest distortion is chosen. The index of the quantizer is sent as 4 bits of side information for each subband. The maximum resolution of each quantizer is 15 bits.

Layer 2. Layer 2 of the MPEG-1 Audio codec includes small changes to effect bitrate reduction and quality improvement, at the price of an increase in complexity. The main difference in Layer 2 is that three groups of 12 samples are encoded in each frame, and temporal masking is brought into play, as well as frequency masking. One advantage is that if the scaling factor is similar for each of the three groups, a single scaling factor can be used for all three. But using three frames in the filter (before, current, and next), for a total of 1,152 samples per channel, approximates taking temporal masking into account.

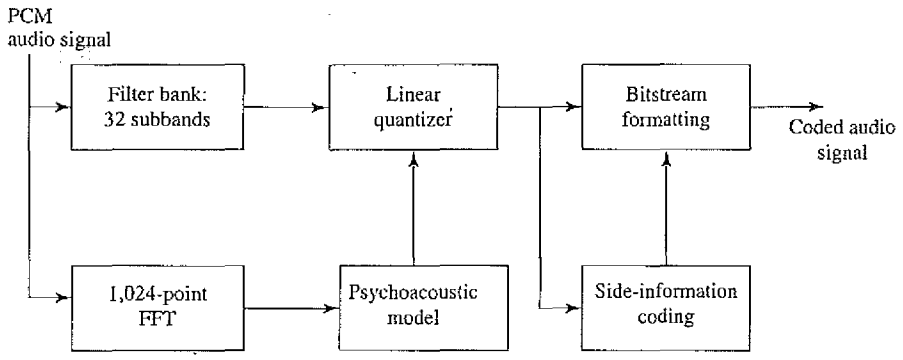

FIGURE 14.13: MPEG-1 Audio Layers 1 and 2.

As well, the psychoacoustic model does better at modeling slowly-changing sound if the time window used is longer. Bit allocation is applied to window lengths of 36 samples instead of 12, and resolution of the quantizers is increased from 15 bits to 16. To ensure that this greater accuracy does not mean poorer compression, the number of quantizers to choose from decreases for higher subbands.

Layer 3. Layer 3, or MP3, uses a bitrate similar to Layers 1 and 2 but produces substantially better audio quality, again at the price of increased complexity.

A filter bank similar to that used in Layer 2 is employed, except that now perceptual critical bands are more closely adhered to by using a set of filters with nonequal frequencies. This layer also takes into account stereo redundancy. It also uses a refinement of the Fourier transform: the *Modified Discrete Cosine Transform* (*MDCT*) addresses problems the DCT has at boundaries of the window used. The Discrete Fourier Transform can produce block edge effects. When such data is quantized and then transformed back to the time domain, the beginning and ending samples of a block may not be coordinated with the preceding and subsequent blocks, causing audible periodic noise.

The MDCT shown in Equation (14.7), removes such effects by overlapping frames by 50%.

$$F(u) = 2 \sum_{i=0}^{N-1} f(i) \cos \left[\frac{2\pi}{N} \left(i + \frac{N/2+1}{2} \right) (u + 1/2) \right], u = 0, \ldots, N/2 - 1 \quad (14.7)$$

The MDCT also gives better frequency resolution for the masking and bit allocation operations. Optionally, the window size can be reduced back to 12 samples from 36. Even so, since the window is 50% overlapped, a 12-sample window still includes an extra 6 samples. A size-36 window includes an extra 18 points. Since lower frequencies are more often tonelike rather than noiselike, they need not be analyzed as carefully, so a mixed mode is also available, with 36-point windows used for the lowest two frequency subbands and 12-point windows used for the rest.

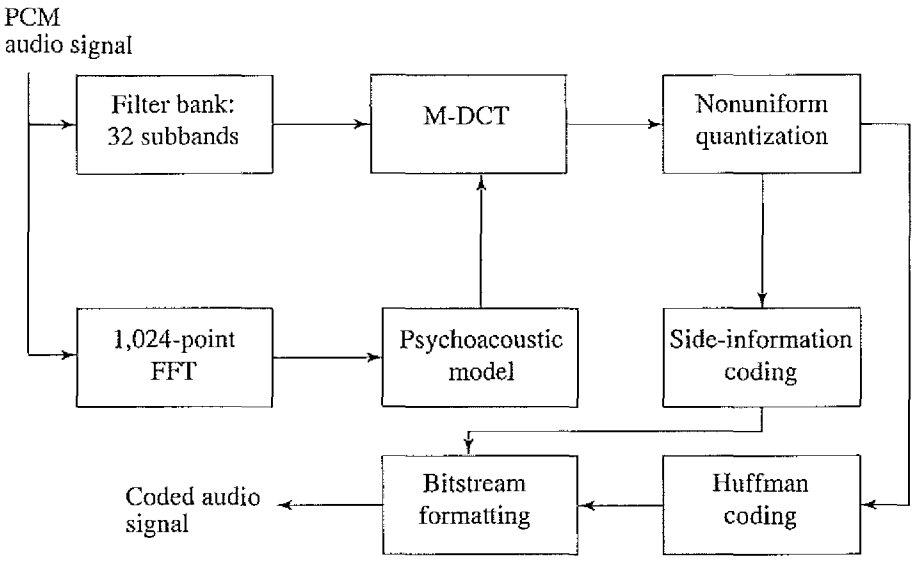

FIGURE 14.14: MPEG-1 Audio Layer 3.

As well, instead of assigning scaling factors to uniform-width subbands, MDCT coefficients are grouped in terms of the auditory system's actual critical bands, and scaling factors, called *scale factor bands*, are calculated from these.

More bits are saved by carrying out entropy coding and making use of nonuniform quantizers. And, finally, a different bit allocation scheme is used, with two parts. Firstly, a nested loop is used, with an inner loop that adjusts the shape of the quantizer, and an outer loop that then evaluates the distortion from that bit configuration. If the error ("distortion") is too high, the scale factor band is amplified. Second, a *bit reservoir* banks bits from frames that don't need them and allocates them to frames that do. Figure 14.14 shows a summary of MPEG Audio Layer 3 coding.

Table 14.2 shows various achievable MP3 compression ratios. In particular, CD-quality audio is achieved with compression ratios in the range of 12:1 to 8:1 (i.e., bitrates of 128 to 192 kbps).

14.2.4 MPEG-2 AAC (Advanced Audio Coding)

The MPEG-2 standard is widely employed, since it is the standard vehicle for DVDs, and it, too, has an audio component. The *MPEG-2 Advanced Audio Coding* (*AAC*) standard [11] was aimed at transparent sound reproduction for theaters. It can deliver this at 320 kbps for five channels, so that sound can be played from five directions: left, right, center, left-surround, and right-surround. So-called 5.1 channel systems also include a *low-frequency enhancement* (*LFE*) channel (a "woofer"). On the other hand, MPEG-2 AAC is also capable of delivering high-quality stereo sound at bitrates below 128 kbps. It is the audio coding

TABLE 14.2: MP3 compression performance.

Sound quality	Bandwidth	Mode	Compression ratio
Telephony	3.0 kHz	Mono	96:1
Better than shortwave	4.5 kHz	Mono	48:1
Better than AM radio	7.5 kHz	Mono	24:1
Similar to FM radio	11 kHz	Stereo	26:1 to 24:1
Near-CD	15 kHz	Stereo	16:1
CD	> 15 kHz	Stereo	14:1 to 12:1

technology for the *DVD-Audio Recordable* (*DVD-AR*) format and is also adopted by XM Radio, one of the two satellite radio services in North America.

MPEG-2 audio can support up to 48 channels, sampling rates between 8 kHz and 96 kHz, and bitrates up to 576 kbps per channel. Like MPEG-1, MPEG-2 supports three different "profiles", but with a different purpose. These are the *Main*, *Low Complexity* (*LC*), and the *Scalable Sampling Rate* (*SSR*). The LC profile requires less computation than the Main profile, but the SSR profile breaks up the signal so that different bitrates and sampling rates can be used by different decoders.

The three profiles follow mostly the same scheme, with a few modifications. First, an MDCT transform is carried out, either on a "long" window with 2,048 samples or a "short" window with 256 samples. The MDCT coefficients are then filtered by a *Temporal Noise Shaping* (*TNS*) tool, with the objective of reducing pre-masking effects and better encoding signals with stable pitch.

The MDCT coefficients are then grouped into 49 scale factor bands, approximately equivalent to a good-resolution version of the human acoustic system's critical bands. In parallel with the frequency transform, a psychoacoustic model similar to the one in MPEG-1 is carried out, to find masking thresholds.

The Main profile uses a predictor. Based on the previous two frames, and only for frequency coefficients up to 16 kHz, MPEG-2 subtracts a prediction from the frequency coefficients, provided this step will indeed reduce distortion. Quantization is governed by two rules: keep distortion below the masking threshold, and keep the average number of bits used per frame controlled, using a bit reservoir. Quantization uses scaling factors — which can be used to amplify some of the scale factor bands — and nonuniform quantization. MPEG-2 AAC also uses entropy coding for both scale factors and frequency coefficients.

Again, a nested loop is used for bit allocation. The inner loop adapts the nonlinear quantizer, then applies entropy coding to the quantized data. If the bit limit is reached for the current frame, the quantizer step size is increased to use fewer bits. The outer loop

decides whether for each scale factor band the distortion is below the masking threshold. If a band is too distorted, it is amplified to increase the SNR of that band, at the price of using more bits.

In the SSR profile, a *Polyphase Quadrature Filter* (*PQF*) bank is used. The meaning of this phrase is that the signal is first split into four frequency bands of equal width, then an MDCT is applied. The point of the first step is that the decoder can decide to ignore one of the four frequency parts if the bitrate must be reduced.

14.2.5 MPEG-4 Audio

MPEG-4 audio integrates several different audio components into one standard: speech compression, perceptually based coders, text-to-speech, and MIDI. The primary general audio coder, MPEG-4 AAC [12], is similar to the MPEG-2 AAC standard, with some minor changes.

Perceptual Coders. One change is to incorporate a *Perceptual Noise Substitution* module, which looks at scale factor bands above 4 kHz and includes a decision as to whether they are noiselike or tonelike. A noiselike scale factor band itself is not transmitted; instead, just its energy is transmitted, and the frequency coefficient is set to zero. The decoder then inserts noise with that energy.

Another modification is to include a *Bit-Sliced Arithmetic Coding* (*BSAC*) module. This is an algorithm for increasing bitrate scalability, by allowing the decoder side to be able to decode a 64 kbps stream using only a 16 kbps baseline output (and steps of 1 kbps from that minimum).

MPEG-4 audio also includes a second perceptual audio coder, a vector-quantization method entitled *Transform-domain Weighted Interleave Vector Quantization* (*TwinVQ*). This is aimed at low bitrates and allows the decoder to discard portions of the bitstream to implement both adjustable bitrate and sampling rate. The basic strategy of MPEG-4 audio is to allow decoders to apply as many or as few audio tools as bandwidth allows.

Structured Coders. To have a low bitrate delivery option, MPEG-4 takes what is termed a *Synthetic/Natural Hybrid Coding* (*SNHC*) approach. The objective is to integrate both "natural" multimedia sequences, both video and audio, with those arising synthetically. In audio, the latter are termed *structured* audio. The idea is that for low bitrate operation, we can simply send a pointer to the audio model we are working with and then send audio model parameters.

In video, such a *model-based* approach might involve sending face-animation data rather than natural video frames of faces. In audio, we could send the information that English is being modeled, then send codes for the basesounds (phonemes) of English, along with other assembler-like codes specifying duration and pitch.

MPEG-4 takes a *toolbox* approach and allows specification of many such models. For example, *Text-To-Speech* (*TTS*) is an ultra-low bitrate method and actually works, provided we need not care what the speaker actually sounds like. Assuming we went on to derive Face Animation Parameters from such low bitrate information, we arrive directly at a very low bitrate videoconferencing system.

TABLE 14.3: Comparison of audio coding systems.

Codec	Bitrate kbps/channel	.Complexity	Main application
Dolby AC-2	128–192	Low (encoder/decoder)	Point-to-point, cable
Dolby AC-3	32–640	Low (decoder)	HDTV, cable, DVD
Sony ATRAC	140	Low (encoder/decoder)	Minidisc

Another "tool" in structured audio is called *Structured Audio Orchestra Language* (*SAOL*, pronounced "sail"), which allows simple specification of sound synthesis, including special effects such as reverberation.

Overall, structured audio takes advantage of redundancies in music to greatly compress sound descriptions.

14.3 OTHER COMMERCIAL AUDIO CODECS

Table 14.3 summarizes the target bitrate range and main features of other modern general audio codecs. They bear many similarities to MPEG-2 audio codecs.

14.4 THE FUTURE: MPEG-7 AND MPEG-21

Recall that MPEG-4 is aimed at compression using objects. MPEG-4 audio has several interesting features, such as 3D localization of sound, integration of MIDI, text-to-speech, different codecs for different bitrates, and use of the sophisticated MPEG-2 AAC codec. However, newer MPEG standards are mainly aimed at "search": how can we find objects, assuming that multimedia is indeed coded in terms of objects?

The formulation of MPEG-21 [13] is an ongoing effort, aimed at driving a standardization effort for a *Multimedia Framework* from a consumer's perspective, particularly addressing interoperability. However, we can say something more specific about how MPEG-7 means to describe a structured model of audio [14], so as to promote ease of search for audio objects.

Officially called a method for *Multimedia Content Description Interface*, MPEG-7 provides a means of standardizing metadata for audiovisual multimedia sequences. MPEG-7 is meant to represent information about multimedia information.

The objective, in terms of audio, is to facilitate the representation and search for sound content, perhaps through the tune or other descriptors. Therefore, researchers are laboring to develop descriptors that efficiently describe, and can help find, specific audio in files. These might require human or automatic content analysis and might be aimed not just at low-level structures, such as melody, but at actually grasping information regarding structural and semantic content [15].

An example application supported by MPEG-7 is *automatic speech recognition* (*ASR*). Language understanding is also an objective for MPEG-7 "content". In theory, MPEG-7 would allow searching on spoken and visual events: "Find me the part where Hamlet says,

'To be or not to be.'" However, the objective of delineating a complete, structured audio model for MPEG-7 is by no means complete.

Nevertheless, low-level features are important. A recent summary of such work [16] sets out one set of such descriptors.

14.5 FURTHER EXPLORATION

Good reviews of MPEG Audio are contained in the articles [9, 17]. A comprehensive explication of natural audio coding in MPEG-4 appears in [18]. Structured audio is introduced in [19], and exhaustive articles on natural and structured audio in MPEG-4 appear in [20] and [21].

The Further Exploration section of the text web site for this chapter contains a number of useful links:

- Excellent collections of MPEG audio and MP3 links
- The MPEG audio FAQ
- An excellent reference by the Fraunhofer-Gesellschaft research institute, "MPEG 4 Audio Scalable Profile," on the subject of Tools for Large Step Scalability. This allows the decoder to decide how many tools to apply and at what complexity, based on available bandwidth.

14.6 EXERCISES

1. (a) What is the threshold of quiet, according to Equation (14.1), at 1,000 Hz? (Recall that this equation uses 2 kHz as the reference for the 0 dB level.)
   (b) Take the derivative of Equation (14.1) and set it equal to zero, to determine the frequency at which the curve is minimum. What frequency are we most sensitive to? Hint: One has to solve this numerically.
2. Loudness versus amplitude. Which is louder: a 1,000 Hz sound at 60 dB or a 100 Hz sound at 60 dB?
3. For the (newer versions of the) Fletcher-Munson curves, in Figure 14.1, the way this data is actually observed is by setting the y-axis value, the sound pressure level, and measuring a human's estimation of the effective perceived loudness. Given the set of observations, what must we do to turn these into the set of perceived loudness curves shown in the figure?
4. Two tones are played together. Suppose tone 1 is fixed, but tone 2 has a frequency that can vary. The *critical bandwidth* for tone 1 is the frequency range for tone 2 over which we hear *beats*, and a roughness in the sound. Beats are overtones at a lower frequency than the two close tones; they arise from the difference in frequencies of the two tones. The critical bandwidth is bounded by frequencies beyond which the two tones sound with two distinct pitches.
   (a) What would be a rough estimate of the critical bandwidth at 220 Hz?
   (b) Explain in words how you would set up an experiment to measure the critical bandwidth.

5. Search the web to discover what is meant by the following psychoacoustic phenomena:

   (a) Virtual pitch
   (b) Auditory scene analysis
   (c) Octave-related complex tones
   (d) Tri-tone paradox
   (e) Inharmonic complex tones

6. If the sampling rate f_s is 32 ksps, in MPEG Audio Layer 1, what is the width in frequency of each of the 32 subbands?
7. Given that the level of a *masking tone* at the 8th band is 60 dB, and 10 msec after it stops, the masking effect to the 9th band is 25 dB.

   (a) What would MP3 do if the original signal at the 9th band is at 40 dB?
   (b) What if the original signal is at 20 dB?
   (c) How many bits should be allocated to the 9th band in (a) and (b) above?

8. What does MPEG Layer 3 (MP3) audio do differently from Layer 1 to incorporate temporal masking?
9. Explain MP3 in a few paragraphs, for an audience of consumer-audio-equipment salespeople.
10. Implement MDCT, just for a single 36-sample signal, and compare the frequency results to those from DCT. For low-frequency sound, which does better at concentrating the energy in the first few coefficients?
11. Convert a CD-audio cut to MP3. Compare the audio quality of the original and the compressed version — can you hear the difference? (Many people cannot.)
12. For two stereo channels, we would like to be able to use the fact that the second channel behaves, usually, in a parallel fashion to the first, and apply information gleaned from the first channel to compression of the second. Discuss how you think this might proceed.

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PART THREE

MULTIMEDIA COMMUNICATION AND RETRIEVAL

Multimedia places great demands on networks and systems. This part examines several important multimedia networks and applications that are essential and challenging.

Multimedia Networks

With the ever-increasing bandwidth made available by breakthroughs in fiber optics, we are witnessing a convergence of telecommunication networks and computer and multimedia networks and a surge in mixed traffic types (Internet telephony, video-on-demand, etc.) through them. The technologies of multiplexing and scheduling are being constantly reexamined. Moreover, we are also witnessing an emergence of wireless networks (think about our cell phones and PDAs).

In Chapter 15, we look at basic issues and technologies for computer and multimedia networks, and in Chapter 16 we go on to consider multimedia network communications and applications. Chapter 17 provides a quick introduction to the basics of wireless networks and issues related to multimedia communication over these networks.

Content-Based Retrieval in Digital Libraries

Automated retrieval of syntactically and semantically useful contents from multimedia databases is crucial, especially when the contents have become so rich and the size of

the databases has grown so rapidly. Chapter 18 looks at a particular application of multimedia database systems, examining the issues involved in content-based retrieval, storage, and browsing in digital libraries.