Audio Basics

You can create professional-sounding music projects in Soundtrack Pro without any background or training in music. This chapter describes the basic audio and music concepts you need to know to get started.

Basic Audio Concepts

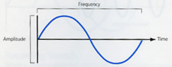
In order to understand how you can work with audio files and use the controls, effects, and envelopes included in Soundtrack Pro, it's helpful to know some basic terms and concepts about audio.

Sound Waves

What we hear as sounds are vibrations traveling through the air as *sound waves*. Sound waves move through the air like ripples in a pond, radiating outward from the sound's source in a regular pattern of compression and rarefaction.

Frequency and Amplitude

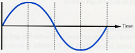
The rate at which a sound wave repeats (the time between two peaks of the wave) is called its *frequency*. Frequency is expressed in cycles per second, or Hertz (Hz). We hear a sound's frequency as being relatively high (like a flute, a child's voice, or a train whistle) or low (like a bass guitar, a man's voice, or the rumble of a train on the tracks). The range of frequencies audible to human beings is roughly 20 Hz–20 kilohertz (kHz).



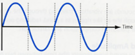
The depth or intensity of a sound is called its *amplitude*, and is expressed in decibels (dB). We hear amplitude as the *volume* or loudness of a sound. The range of audible loudness is roughly 0–130 dB. Higher decibel levels are painful to human hearing.

Musical Sounds

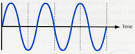
Musical sounds typically have a regular frequency, which we hear as the sound's *pitch*. Pitch is expressed using musical notes, such as C, Eb, and F#. What we hear as the pitch is only the lowest, strongest part of the sound wave, called the *fundamental*. Every musical sound also has higher, softer parts called *overtones* or *harmonics*, which occur at regular multiples of the fundamental frequency. We don't hear the harmonics as distinct pitches, but rather as the tone color (also called the *timbre*) of the sound, which lets us distinguish one instrument or voice from another, even when both are playing the same pitch. When you turn up the treble on your stereo, or adjust an EQ effect, you raise the volume of some of the harmonics in the music, but don't change the fundamental frequencies.



Fundamental



First harmonic



Second harmonic

Envelopes

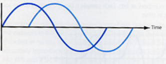
Another aspect of sound that helps us to distinguish between instruments and voices playing the same pitch is a sound's *envelope*. Every note played on a musical instrument has a distinct curve of rising and falling volume over time. Sounds produced by some instruments, particularly drums and other percussion, start at a high volume level but quickly decrease to a much lower level, and die away to silence quickly. Sounds produced by other instruments, for example, a violin or a trumpet, can be sustained at the same volume level, and can be raised or lowered in volume while being sustained. This volume curve is called the sound's envelope, and acts like a signature to help our ears recognize what instrument is producing the sound.

Percussive envelope

Sustained envelope

Phase Relationships

When two instruments or voices are playing the same pitch, the sound waves may have the exact same frequency and amplitude, but the peaks and troughs of the wave reach our ears, or a microphone recording the sound, at slightly different times. This is referred to as a difference in the *phase* of the sound waves. When two sound waves are completely in phase, the volume of the sound is doubled. When two sound waves are completely out of phase, they cancel each other out and we hear silence. Certain effects, such as phase shifters, make use of these properties of phase relationships to alter the sound of an audio signal.

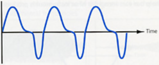


Recording Sound

When a sound is recorded, the sound waves traveling through the air are converted to an electrical signal, using a device called a *transducer*. Sound can be recorded using either *analog* or *digital* recording technology.

Analog Recording

When a sound is recorded using analog technology, the sound waves are recorded as a continuous electrical signal. Typically, the vibrations in the air contact the diaphragm of a microphone, setting the diaphragm in motion. A transducer in the microphone converts the diaphragm's motion into an electric signal. The compressed parts of the sound wave are recorded as positive electrical voltages, and the rarefied parts of the wave are recorded as negative voltages. The voltage of the recorded signal is an *analog* of the wave's frequencies and their relative amplitudes at any point in time.

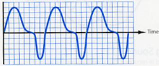


Analog recording technology was originally developed using mechanical means to etch the sound signal directly onto wax cylinders or lacquer disks. Its simplicity, and the rapid development of electronics during the twentieth century, led to its widespread use for recording music and for adding sound to motion pictures.

However, analog audio recording is subject to several problems in achieving high-fidelity reproduction of sound. These include noise, distortion, and loss of quality each time the audio signal is copied or reproduced.

Digital Recording

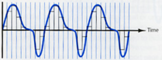
When a sound is digitally recorded, the sound waves are recorded as a series of *samples* onto a hard disk or other digital storage medium. A sample stores the voltages corresponding to the wave's frequencies and their relative amplitudes as a series of binary numbers, or bits. Each sample is like a snapshot of the sound at a particular instant in time.



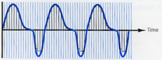
Digital recording technology offers several advantages over analog technology for recording sound, including lower noise, wider frequency response, and less distortion (if the sound is recorded at the proper level). In addition, digital recordings can be reproduced any number of times without any loss of audio quality. These advantages, combined with the popularity of personal computers, have led to the rapid development of digital audio as a leading technology for music production.

Sample Rate and Bit Depth

The audio quality of any digital recording depends on two factors: the sample rate and the bit depth used to record the signal. The *sample rate* is the number of samples recorded per second. The *bit depth* is the number of digital bits each sample contains. Together, these two factors determine the amount of information contained in a digital audio recording. The higher the sample rate and bit depth of a recording, the more accurately the recording reproduces the original sound.



Low sample rate



High sample rate

Recording music digitally requires a very high sample rate and bit depth to reproduce the nuances in the music satisfactorily. The Nyquist theorem states that sounds must be recorded at no less than double the rate of the highest frequency being sampled to accurately reproduce the original sound. Audio CDs are recorded at a sample rate of 44.1 kHz and a bit depth of 16 bits (some CDs use a higher 20- or 24-bit depth). Audio for DVDs is often recorded using a slightly higher sample rate of 48 kHz. Soundtrack Pro lets you record and play back digital audio files at sample rates of up to 96 kHz, and at bit depths of up to 24 bits.

Digital Distortion

To record sound with the widest possible dynamic range, the input level must be set high enough to capture the complete audio signal. When the input level of the signal is set too high, however, the signal exceeds the maximum level that can be sampled or reproduced accurately, resulting in *digital distortion*. Digital distortion is defined for most digital audio applications, including Soundtrack Pro, as any time the signal rises above 0 dB. Even a single sample above 0 dB can produce noticeable distortion, which you hear as a sharp crackling sound in the audio output. Digital distortion is nearly always undesirable, and Soundtrack Pro includes level meters and other controls so that you can identify and remove distortion from your projects.