Class Overview

1 Introduction

There are typically four steps in producing a CD or movie soundtrack, as shown in Figure 1. In tracking sounds are recorded or synthesized and arranged in tracks. The tracks are then processed in the mixing stage to form a stereo or multichannel mix. The idea is to arrange the sounds spatially and manipulate their character for artistic purposes, and also to fix problems in the tracks. In mastering, subtle adjustments and fixes are made to the mix, and often its dynamic range is limited in preparation for encoding and printing on the target medium.

![Diagram](tracking->mixing->mastering->encoding)

Figure 1: Audio Production Process

This class is about audio effects processing, and, in particular, how to build digital versions of the mainline audio effects used in the mixing and mastering stages of audio production. There are four categories of processing commonly employed in mixing and mastering:

- Dynamic Range Control,
- Reverberation,
- Equalization and
- Spatialization.

Other specialized processing, such as delay effects (including echo, chorus, flanging and phasing), distortion, pitch and time stretching and noise removal, is also used, but less often. In this class we will explore each of these workhorse processor categories, and in homework and laboratory exercises you will build examples of each. You may wish to study one of the other specialized processing effects as your project.

2 Dynamic Range Control

Dynamic range control refers to the manipulation of the level of a signal. This may be desirable, for instance, in the case of a singer who moves closer and further from
the microphone while singing, causing the level to unintentionally rise and fall. It is commonly used to make current pop CDs quite loud, despite their 16-bit integer samples.

As illustrated in Figure 2, dynamic range control may be accomplished using a feed forward architecture in which the signal level is detected and a gain computed and applied to the input. The gain is computed based on a desired output level as a function on input level.

![Diagram of feed forward and feedback compressor architectures](image)

**Figure 2: Feed Forward and Feedback Compressor Architectures.**

Different applications are developed by choice of gain computer and detector. A gain computer which reduces the dynamic range of louder signals results in a compressor, and may be used to make a bass or drum track more even. By designing the gain computer to impose a predefined maximum output level, the input is limited. A noise gate, which eliminates any low-level background noise appearing between notes in the input track, can be implemented by using a gain computer which takes the signal level to zero when the input is small.

If the detector determines level by examining the input signal over a long time period, the output will be “transparent”—that is, it will sound much like the input. If the detector only considers a short period of time in evaluating signal level, it can change the envelope of individual notes and the character of the track. Finally, the detector (or the entire architecture, for that matter) can be applied to selected frequency bands, for instance, to reduce loud sibilence.

### 3 Reverberation

The acoustics of a space can significantly contribute to the feel of a piece: imagine the "sound" of a jazz hall, a dungeon, outdoors. For this reason it is desired to be able to artificially add environmental cues or reverberation to tracks or a mix.
In addition, the tracks of a mix are not often recorded under the same acoustic conditions; consider vocal booths and drum rooms. As a result, artificial reverberation is commonly added to a mix so as to make different tracks feel as if they belong together.

There are two approaches to implementing artificial reverberation in common use today. In one approach, delayed copies of the signal are filtered, mixed and fed back to delay line inputs. The idea is that the process in some ways mimics what happens in actual acoustic spaces, with reflecting surfaces and objects filtering impinging wavefronts and redirecting them to other reflecting surfaces and objects. Loosely speaking, such artificial reverberators are called feedback delay networks.

In the other approach, it is assumed that the acoustic space is approximately linear and time invariant, and can be characterized by its impulse response. Artificial reverberation may be applied to an input signal simply by convolving it with the desired impulse response. In this approach, impulse responses can be measured and manipulated or synthesized to achieve the desired artistic effect. As an example, Figure 3 shows the time evolution of the impulse response of an EMT140 plate reverb commonly applied to vocals.

![Figure 3: EMT140 Impulse Response STFT.](image)

### 4 Equalization and Filtering

*Equalization* and *filtering* are each the manipulation of the frequency content of a track, with filtering commonly referring to the removal of a band of frequencies and equalization
often referring to a more subtle adjustment as a function of frequency.

In many cases equalization and filtering can be used to fix problems with a track. Tracks will sometimes have unwanted frequency components—say, a 60 Hz hum or a rumble from road noise—and filters may be used to remove them. There are other cases where certain frequency components need to be enhanced; for example, a singer’s lisp can be corrected somewhat by enhancing high frequencies.

One of the primary uses for equalization on tracks is to help separate different mix elements by having them occupy somewhat nonoverlapping frequency bands.

Equalization may also be used as an effect. Filtering the waveform to a band between 200 Hz and 3200 Hz (in combination with some other processing) makes the track sound as if it is being played through a telephone. In an architecture similar to that of the feed forward compressor above, the signal level may be detected and used to control a filter. In this way, for instance, the onset of a note can be made to have a very different timbre than its decay.

It turns out that across a given genre, there is surprisingly little variation in overall spectral content. As a result, the mastering engineer will often (and probably subconsciously) use a program equalization which brings the power spectrum of the song in line with the standard.

Audio engineers have a preference for analog equalizers, for their transfer function characteristics, their controls and their signal path. The approach to equalization we present here is based on modeling analog equalizers to fix the transfer function and controls, and to use numerically robust filter structures for signal integrity.

5 Panning and Spatialization

In the presence of a stereo or surround output channel configuration, it is possible to effectively position tracks spatially. This provides an important dimension along which tracks may be separated.

The technique currently used for CD and movie production is called panning, and places different portions of the signal in the different output channels. In this class we will explore techniques for determining the panning weights, and will look at HRTF techniques popular for video games and other interactive environments.