MULTIBAND COMPRESSION PROPOSAL

McLean Pierce 311267165

Digital Audio Systems, DESC9115, Semester 1 2012

Graduate Program in Audio and Acoustics

Faculty of Architecture, Design and Planning, The University of Sydney

PROBLEM DESCRIPTION

Dynamic range compression is an audio tool whose name succinctly describes the purpose and results of the tool itself. Often simply called compression, it is a tool, effect or process that decreases the dynamic range of an audio signal by reducing peaks in the signal. After this, the overall signal level is typically normalized to the original maximum level and the resulting output is a louder, less dynamic version of the original input signal. Compression has been used in the audio world for a long time, first being implemented through analog means for broadcast applications, allowing dynamic recorded music to fit within the more limited dynamic range of AM radio broadcast. In this early era compression also became popular with radio DJs, allowing them to talk over music broadcasts and still be heard. Compression quickly moved beyond these practical applications and became a tool used for artistic purposes as well, increasing the "punch" and loudness of a signal without changing its basic sound or feeling. Since then it has become a ubiquitous and indispensible tool for audio recording and is used in almost every style of music. At the same time, it continues to be used in more practical applications, increasing the volume of audio signals where they need to sound louder or overcome background noise.

Compression has been implemented through a variety of means in the analog domain, resulting in a number of effects with the same purpose, but somewhat different audio characteristics. Because of the non-linear nature of most analog components, different types of analog compressors vary in the way compression is applied and the types of distortions introduced into the system. In fact, a pure compressor that operates to the theoretical ideal may not always be preferred to one that behaves in a less predictable, non-linear fashion.

Digital implementations of compression have existed for a long time as well and are used widely in the audio world. They are more controllable and predictable than analog compressors and can overcome some of the drawbacks of non-linear, analog compressors such as unwanted distortion. At the same time, they may have less "character" than analog compressors unless they are designed to model the idiosyncrasies of analog solutions.

One popular type of compressor is the multi-band compressor, in which compression is applied differently to different frequency bands of the input signal. This allows the user to be selective about how compression is applied to a signal and only add power to certain parts of the frequency spectrum. Multi-band compression can be useful for compressing final mixes of music where, for instance, one might want to increase the power of the low end without effecting the high end of the mix. Many other uses of multi-band compression exist where a user might want to be selective about the application of compression to the signal.

The purpose of this project was to develop a digital multi-band compressor that would be useful across a range of settings, and practical for a number of different applications. The resulting software is a three band dynamic range compressor using 6^{th} order Butterworth crossovers and allowing the user to set crossover frequencies and the typical compressor settings of attack time, release time, compression threshold and compression slope.

SPECIFICATION

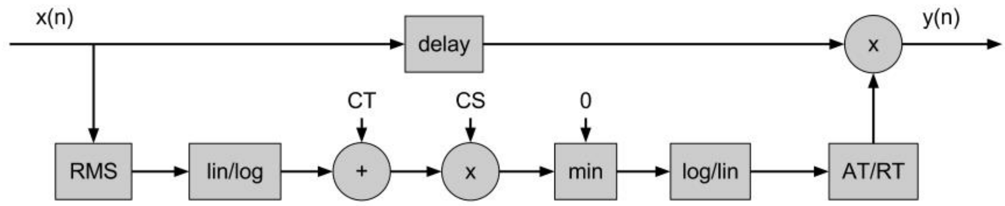

Figure 1: Basic Compressor Signal Flow Diagram

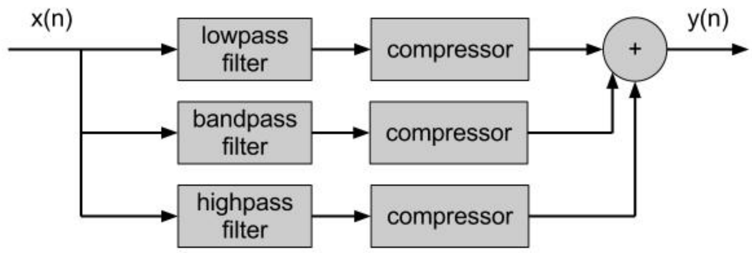

Figure 2: Multi-Band Compressor Signal Flow Diagram

The typical signal flow diagram for a basic compressor is shown in Figure 1. Figure 2 depicts the signal flow diagram for our multiband compressor, with the compressor from Figure 1 reduced to a single "compressor" element. As such, the bulk of the processing occurs in the "compressor" elements, with the multiple bands being implemented by 3 filters (lowpass, bandpass and highpass) which split the signal into three parts which are then separately compressed and re-combined into a single output signal.

The details of Figure 1, i.e. the process of compression, are described as follows. First, the input signal is split into two paths, one of these is used for level detection and gain calculations, the other path is delayed slightly to account for calculation time, then has its level controlled by the results of the detection path. The first two elements in the detection path detect the RMS amplitude of the signal, then convert this into a log value. "CT" stands for "compression threshold" which is a user defined variable in the range of 0 to $-\infty$ dB. This negative value is added to the positive value coming from the log converter. "CS" stands for "compression slope" and defines the amount of compression (gain reduction) applied to the signal above the threshold by multiplication. The "min" element assures the application of gain reduction or nothing if the compressor tries to increase the gain. The values are converted back into linear values by the "log/lin" element and then the "AT/RT" element (attack and release) applies attack and release times to the calculated gain reductions. Finally, these values are applied to the delayed original signal by multiplication and result in the compressed output signal.

IMPLEMENTATION

This solution is implemented in the digital domain through the combination of a number of common digital audio processes. The Matlab function “butter” is first used to generate filter coefficients for three 3^{rd} order Butterworth filters, a lowpass, bandpass and highpass. These are specified by two user-input frequencies, which serve as the crossover frequencies between the lowpass and bandpass, and the bandpass and highpass. These filter coefficients are then fed to the zero-phase filter function “filtfilt” which generates the three filtered bands of the signal. “Filtfilt” generates filtered signals with zero phase distortion but results in a filter order that is double the order of the one specified. As such, the resulting filtered signals have steep 6^{th} order crossovers which result in very little duplication of frequencies between bands but do result in a small loss in output around the crossover frequencies. After the filtering process, each of the three filtered outputs is separately fed through the “holtcomp” compression function. The “holtcomp” outputs are then normalized to the maximum of the uncompressed filtered signal and then added back together. The final signal is then normalized again and presented as the output.

The “holtcomp” function does most of the “heavy lifting” in the application, performing the compression on each band of the signal. As inputs, “holtcomp” takes a signal, a compression threshold, a compression ratio, and an attack and release time. The first step of this function is to perform RMS detection on the input signal which is done by squaring the input then using a moving average filter to determine a time-averaged RMS. This value is compared to the user defined “threshold” value and gain reduction (based on user setting) is applied to the signal that exceeds this level. Finally, “attack” and “release” values are applied to the signal by a time averaging method and the new signal is presented to the function output. The resulting signal is a compressed audio signal with increased loudness in the bands of the user’s choosing.

EVALUATION

To evaluate the compression function we call it as follows: **`[comp orig fs]=mcomp('dasdrum.wav', 1000, 10000, -35, .03, .3, .8, 0, .03, .3, 0, 0, .03, .3, 0).`**

This calls the function and applies compression to the lowpass band below 1000Hz with a slope setting of 0.8 and a threshold of -35. The other two bands have no compression applied as their thresholds are set to 0 dB, a level the input signal will never reach. To assess the results of this function, we analyze the signal after it has been filtered into separate bands to ensure that compression is only being applied to the specified band. The following figures show the uncompressed and compressed signal for each of the 3 bands.

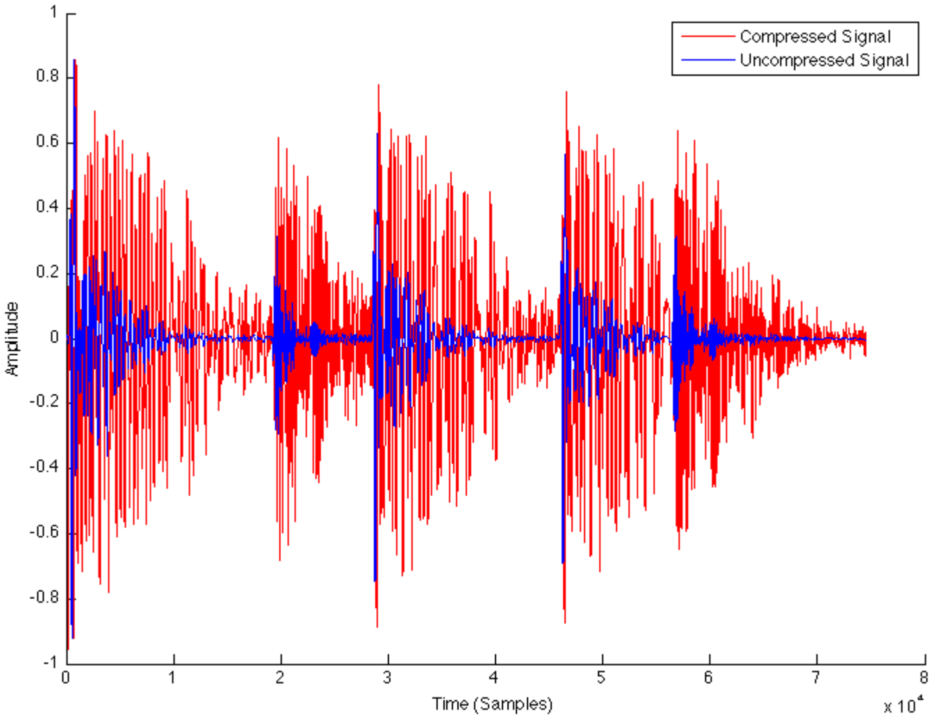

Figure 3: Lowpass (low band) Uncompressed and Compressed Signal

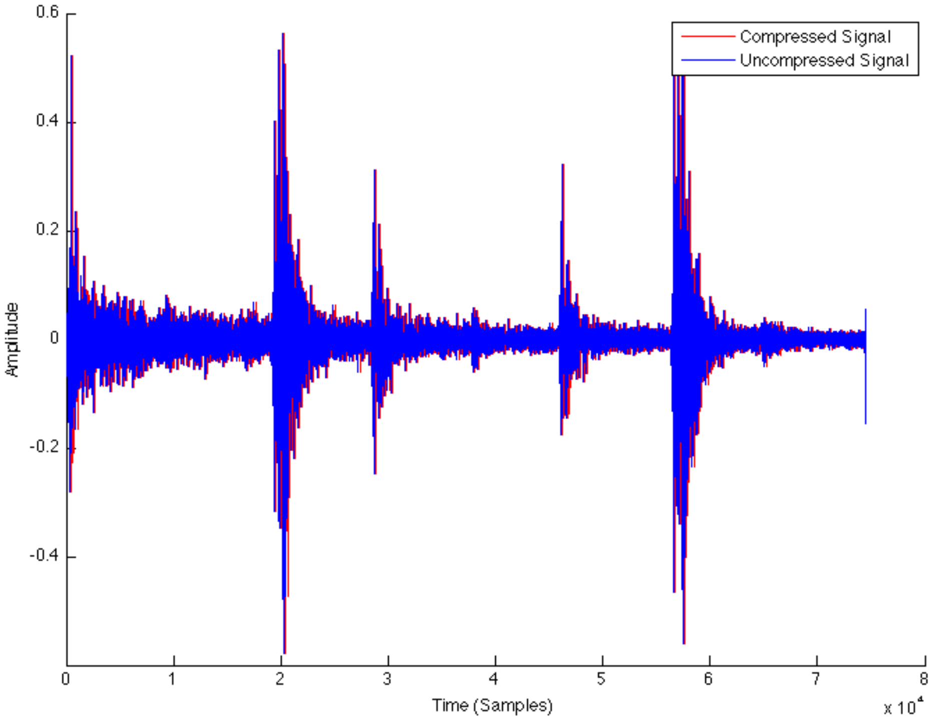

Figure 4: Bandpass (mid band) Uncompressed and Compressed Signal

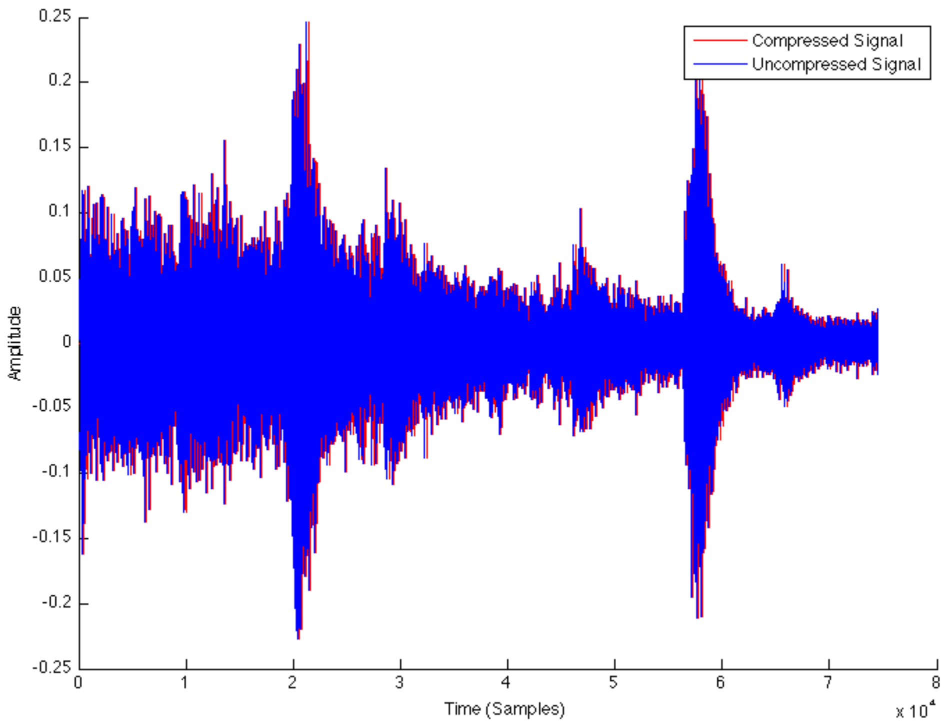

Figure 5: Highpass (high band) Uncompressed and Compressed Signal

Analysis of these figures shows a considerable amount of extra energy in frequencies below 1000Hz with a peak level that remains the same as the original uncompressed signal. As such, this demonstrates that this dynamic range compressor is an effective tool for increasing energy in targeted bands of a sound's frequency spectrum. No change is made in the bands where compression is not specified.

HUMAN USER ASSESSMENT

While the multi-band compressor clearly functions as it is intended, it would be useful to know whether listeners like the resulting sounds and what settings might be preferred when using the compressor in certain situations. Since preferred settings for such a subtle effect as compression are rather subjective, it would be useful to compare the digital compressor with the same settings on an analog compressor processing the same material. If an ABX test is performed that compares the digital and analog compressor and listeners cannot discern between the digital and analog compression, this could be a strong selling point for the digital effect in a world where many people prefer analog effects because they perceived superiority in sound. An ABX discernment test could convince users that the digital effect sounds just like the "real" thing.