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DESC9115 DIGITAL AUDIO SYSTEMS.

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WRITTEN REVIEW 2.

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CONTENTS:

1. INTRODUCTION:

The following is a report on the background of a signal processing system and it's working in the digital domain. The effect chosen is a multiband compressor. The report discusses the basics of the chosen signal process in the electronic/ analog domain followed by algorithms to represent this system as a DSP procedure.

2. BASICS OF MULTIBAND COMPRESSION:

Dynamics:

Dynamics of an audio system relates to the amplitude factor and it's changes. Audio processing related to dynamics affects the dynamic range and the headroom of a system.

Dynamic Compression: A compressor is a signal processor that reduces the dynamic range of a signal by limiting (or attenuating) them from crossing a set threshold.

Dynamic processing in the electronic domain was discussed in detail in Written Review 1. For further reference, it is attached to Appendix 1.

Parameters of a compressor:

Threshold (dB): A compressor starts acting on the signal only after it crosses a particular desired level. That level is known as the threshold.

Compression Ratio: The ratio of the change in output level to the input level is known as the ratio. For example, if the compression ratio is 4:1(assuming that the input is above the threshold value), then for an increase of 4db in the input level results in a 1dB increase in the output level.

Attack Time (s): It is the time after which a compressor acts on a signal after the input level crosses the threshold.

Release Time (s): It is the time after which a compressor seizes to act on a signal after the level drops below the threshold.

Attack and release times vary form a few to microseconds to milliseconds depending on the compressor unit.

Limiters are compressors with a compression ration of 8:1 to 20:1 or even higher. Brick wall limiting is associated with units that have a high compression ratio (generally more than 20:1 to 100:1) and extremely fast attack/release times.

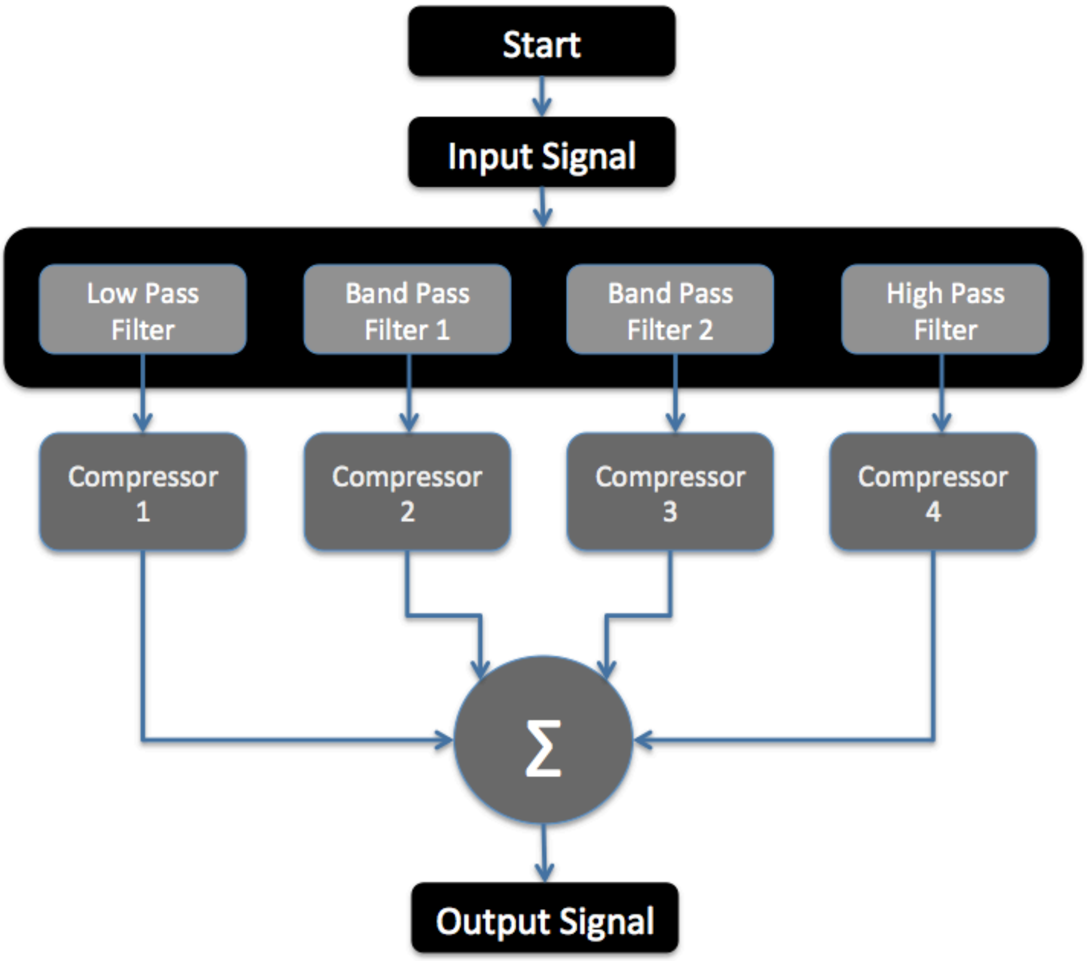
Multiband Compression:

A multi-band compressor comprises a set of filters (similar to a PA system's active crossover) that splits the audio signal into two or more frequency bands. Three- or four-band compressors are perhaps the best compromise between versatility and ease of setting up. After passing through the filters, each frequency band is fed into its own compressor, after which the signals are recombined.

Multiband Compressors have all the parameters of an ordinary compressor but have additional controls to define the band cutoffs. (In some units they are fixed). Different ranges of bands are used for different applications. For example in a audio mixing application, if you restrict heavy processing to above 6kHz, you can significantly enhance the sense of detail and air, without affecting the crucial mid-range and disturbing the natural reproduction of vocals and many acoustic instruments.

Multiband Compressors usually consist of 3 or 4 band crossovers.

The following flowchart shows a summary of the working of a multiband compressor.

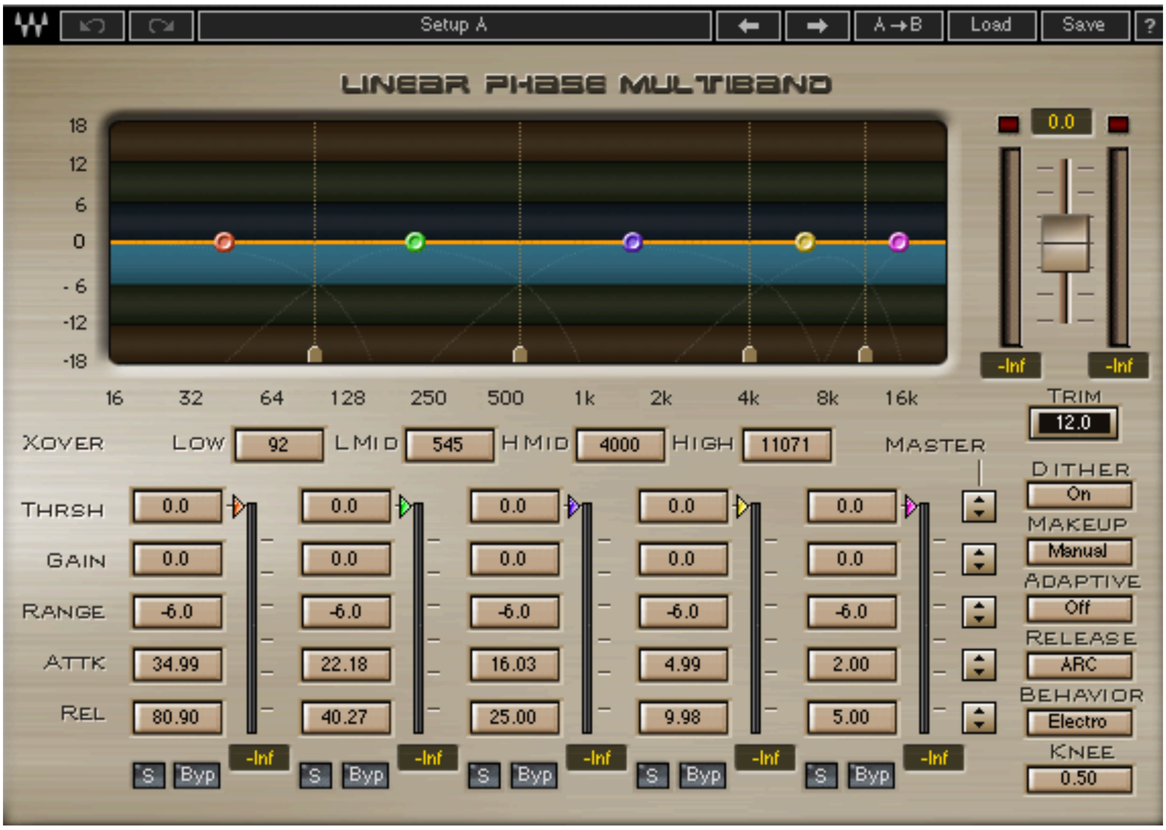


The above procedure shows a 4 band multiband compressor with the low pass section, a low-mid section, a high-mid section and a high pass section.

The following chart shows general values of frequency bands used in a 4 band Multiband Compressor.

Band	**Frequency Ranges**
Low Pass	40Hz – 350 Hz.
Low Mid Band pass	150Hz – 3000Hz.
High Mid Band pass	1kHz – 8kHz.
High pass	4kHz – 16kHz.

The following is an example of a popular software multiband compressor called LinMB developed by Waves Audio. This particular plugin gives control over 5 different bands for surgical multiband compression useful for mastering applications.



3. DIGITAL IMPLEMENTATION:

Digital audio effect algorithms for dynamics processing fall into the category of nonlinear processing.

Dynamic range compression can be expressed symbolically as a filter of the form y(n) = g_n(x).x(n), where x(n) is the input signal, y(n) is the output signal and g_n(x) is the gain which depends on the input x(n).

However,

$$g(x_1 + x_2) \cdot [x_1(n) + x_2(n)] \neq g(x_1) \cdot x_1(n) + g(x_2) \cdot x_2(n).$$

where x1, x2 are two different input signals.

That is, the compression of the sum of two signals is not generally the same as the addition of the two signals compressed individually. Therefore, the superposition condition of linearity fails establishing it as a non-linear system. (Smith III 2007).

Digital implementation of a non-multiband compressor was discussed in written review 1. The information is added to appendix 2 for further reference.

Digital Implementation of a multiband compressor in Matlab:

The following are steps briefly explaining the implementation.

Step 1: The audio file is input to the system.

Step 2: The audio file is separated into 4 bands by a crossover section, which would store them in separate files for individual processing. These are specified by three user specefied, which serve as the crossover frequencies between a lowpass, 2 bandpass, and a highpass section. The filters are generated in Matlab using built-in functions to generate 3rd order Butterworth filters. Care is taken to avoid phase distortion between the sections. These filter coefficients are then fed to the zero-phase filter function which generates the four filtered bands of the signal. The filtered signals are with zero phase distortion but also in a filter order that is double the order of the one specified. This results in steep crossover roll-offs resulting in little overlap between the bands being processed by the multiband compression algorithm.

Step 3: Separate identical compression algorithms are executed to treat the four bands individually.

i: All samples of the signal are analyzed for amplitude (peak) information. This information is translated to level (dB).
ii: The R.M.S. (Root Mean Square) value of each of the samples is calculated from the formula R.M.S. = 0.707 x Peak Value. (Some compressors (like on the SSL 9000K console) allow for selection of level detection between RMS and Peak. This can be used for different aspects and it is a common practice to use peak compression in the processing of drums sounds, i.e. signals with fast transient information).

iii: The parameters for compression are fed to the system. The attack time and release time are converted to sample lengths. These sample lengths are used as a buffer to trigger and release the respective times of the parameters.
iv: The ratio of the compressor is read and the gain reduction multiplier is calculated. A filter is prepared to detect sample data crossing the threshold limit. The attack buffer is activated and gain reduction takes place by processing it by the gain reduction multiplier.

A set of additional mathematical instructions has to be implemented to allow the system to determine if the compressor is in the attack or release state. The following mathematical argument helps implement the attack/release relationship of the nth sample.

$[xd(n) = xd(n-1)*(1-RT)+AT*a]$

where $x(n)$ is the input signal, $xd(n)$ is the n^{th} sample of the vector xd (which is initially a vector of all zeros and equal to the length of the input signal x) , RT is the release time, AT is the attack time and a = absolute value of $[x(n,1)) – xd(n-1)]$.
The non linear gain reduction multiplier can then be calculated by checking if the value of $xd(n)$ crosses the compressor's threshold. If it does not, then the multiplier remains as 1. If the threshold is exceeded, then the multiplier $[f(n)]$ can be expressed as:
$f (n) = 10\text{^}(-slope*(\log10(xd(n))-\log10(threshold))$

where $0<slope<1$. Slope determines the onset of the compression.

The output $y(n) = x(n) * f(n)$.

The above process is repeated for the n samples.

(Zolzer 2002).

Step 4: The four bands are added back together to create the affected output signal, which spans across the entire frequency spectrum the original input wave does. An additional overall makeup gain option can be added to make changes in the overall loudness of the resultant wave.

Step 5: The audio is played back.

4.REFERENCES:

Davis, Gary, Jones, Ralph (1990). The Yamaha Sound Reinforcement Handbook, p.268, 2nd edition, Hal Leonard Publishing Corporation, Milwaukee.

Zolzer, U(ed.) 2002, *DAFX – Digital Audio Effects*, John Wiley & Sons, New York, NY.

Smith III, Julius (2007). Introduction to Digital Filters, Audio Center for Computer Research in Music and Acoustics (CCRMA), Stanford University.

Pierce, Allan D. *Acoustics: an Introduction to its Physical Principles and Applications,* New York: Acoustical Society of America 1989.

S. Smith, The Scientist and Engineer's Guide to Digital Signal Processing, California Technical Publishing, 1999.

5. APPENDICES:

APPENDIX 1: BASICS OF COMPRESSION:

Dynamics:
Dynamics of an audio system relates to the amplitude factor and it's changes. Audio processing related to dynamics effects the dynamic range and the headroom of a system.

The Dynamic range of a system is defined as the difference in the peak output level of the system and the electro-acoustic noise floor.

Dynamic Range (dB) = [Peak Level] – [Noise Floor Level].

The dynamic range of the system is where the signal resides and can be heard without artifacts (ideally) such as clipping etc.

Bit depth determines the dynamic range of a digital audio system. In digital audio, the bit depth describes the number of bits of information recorded for each sample.

Dynamic Range (dB) of a digital audio system = Bit depth x 6.

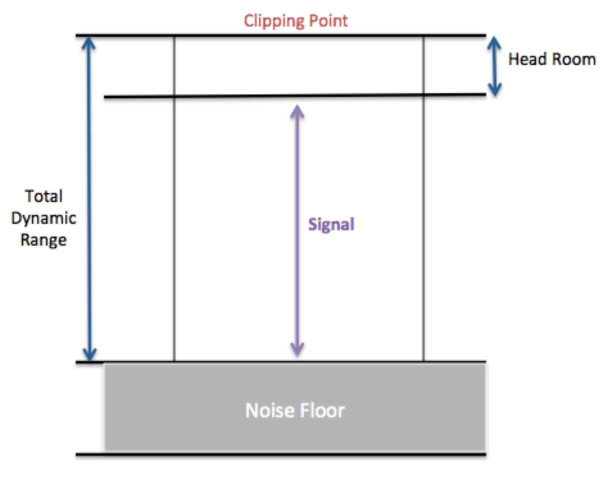
Examples: For a 16-bit system, the dynamic range is 96db and for a 24-bit system, the dynamic range is 144dB.

The headroom of an acoustic system is defined as the difference between the nominal level and peak level and the nominal level (average level).

Headroom (dB) = [Peak Level] – [Nominal Level].

Average Level (R.M.S.) is usually measured as 0.707 times the Peak Level.

Headroom, as a specification, tells us something about the ability of the sound system to handle loud program peaks. Given two sound systems that both operate at the same nominal level, the system with the greater headroom will be able to handle louder peaks before distorting or destroying itself. ((Davis, Jones 1990).



<u>Dynamics Processing</u>:

There are many types of dynamics processing that can be performed on a signal depending on the result that is needed. One of the most common dynamics processor is a Compressor.

Dynamic Compression: A compressor is a signal processor that reduces the dynamic range of a signal by limiting (or attenuating) them from crossing a set threshold.

Parameters of a compressor:

<u>Threshold (dB):</u> A compressor starts acting on the signal only after it crosses a particular desired level. That level is known as the threshold.

<u>Compression Ratio</u>: The ratio of the change in output level to the input level is known as the ratio. For example, if the compression ratio is 4:1(assuming that the input is above the threshold value), then for an increase of 4db in the input level results in a 1dB increase in the output level.

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<u>Release Time (s):</u> It is the time after which a compressor seizes to act on a signal after the level drops below the threshold.

Attack and release times vary form a few to microseconds to milliseconds depending on the compressor unit.

Limiters are compressors with a compression ration of 8:1 to 20:1 or even higher. Brick wall limiting is associated with units that have a high compression ratio (generally more than 20:1 to 100:1) and extremely fast attack/release times.

To sum it up, a basic compressor works as shown in the flowchart below. In this example, the parameters are as follows. Threshold: -15dB, Ratio: 3:1, Attack: 5ms, Release: 10ms.

Figure 1: Representation of compression.

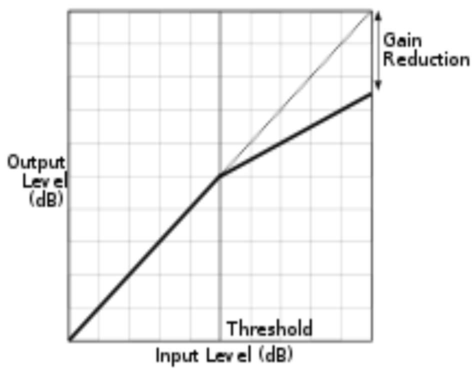
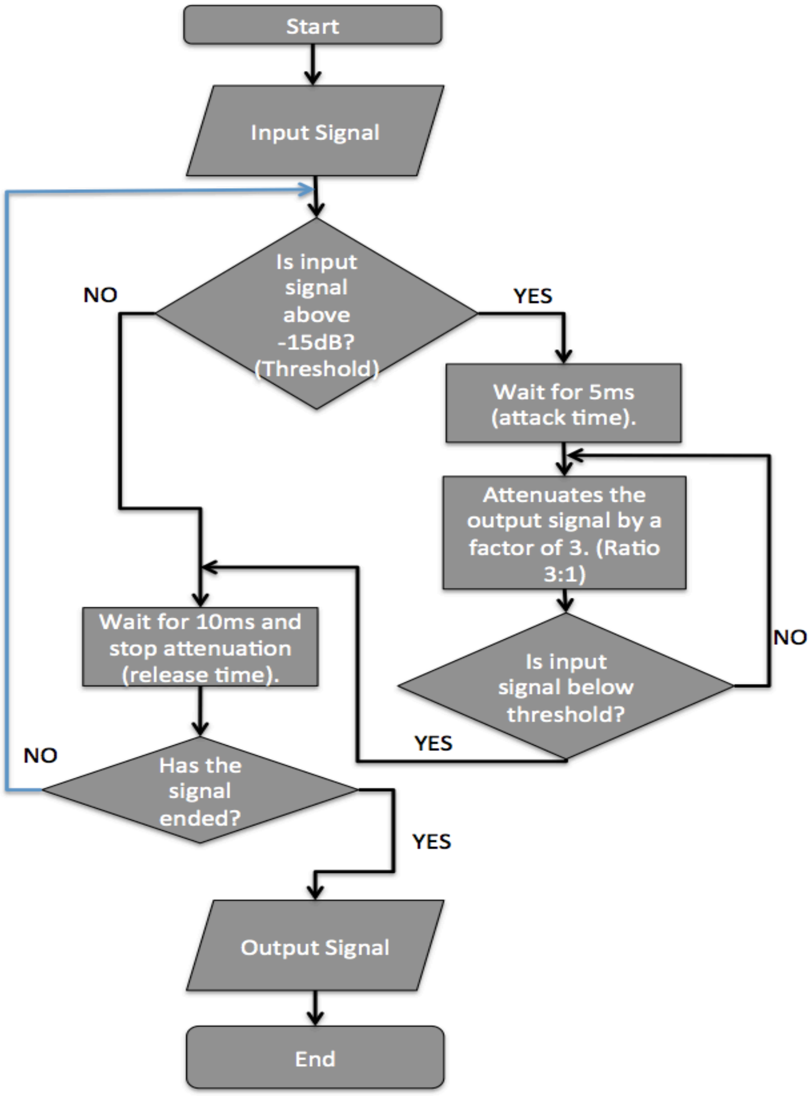


Figure 2: Flowchart describing the process of compression.



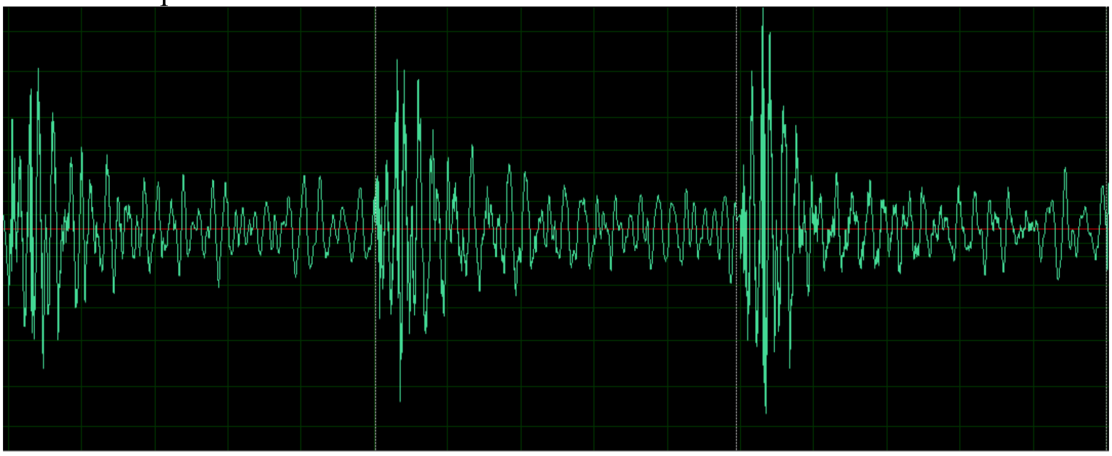
The process is however more complex as the state of the system (attack or release) is determined by more complex procedures.

Result of compression to an audio signal:

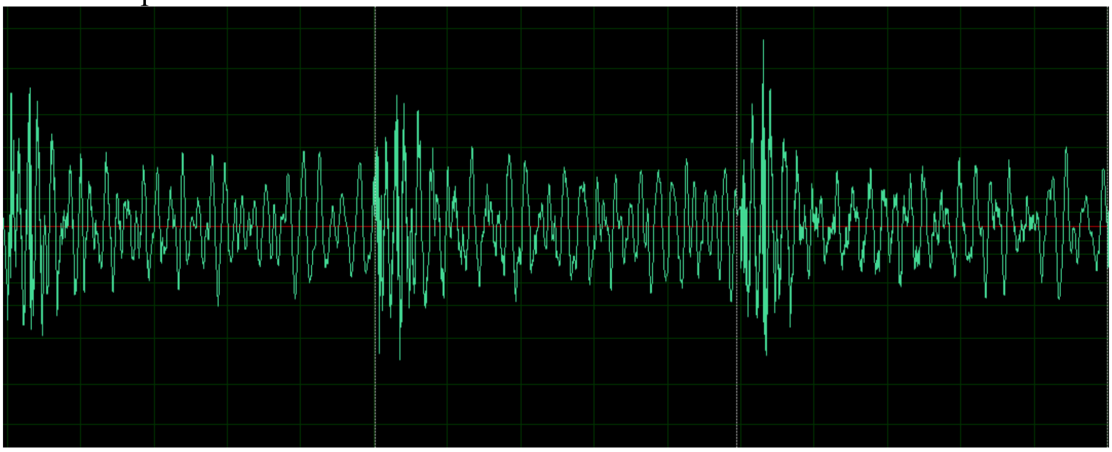
- The most obvious result of compression is the reduction in the dynamic range of the signal.
- The difference between the peak level and average level of the output signal is lesser than the input signal (when normalized). This means that compression helps in increasing the average level of the signal thus increasing loudness without clipping the system.
- More audio information can reside in the peak headroom range as explained in the previous statement.
- Compression distorts the signal to an extent as it adds harmonic content to the signal.

Below is an example of a signal before and after compression. No makeup gain was used on the second diagram. The RMS values of both the signals were taken after normalizing the peak to -1dB. It can be seen how applying compression on the audio signal raised the RMS level of the signal from -17.64dB to -14.23dB.

Before Compression: RMS -17.46dB.



After Compression: RMS -14.23dB.



(Processed in Adobe Audition CS6).

APPENDIX 2: DIGITAL IMPLEMENTAION OF A COMPRESSOR:

The steps are analogous to the flow chart in figure 2. They involve an algorithm for non real time compression and the system is meant to work on a sample-by-sample basis.

Step 1: All samples of the signal are analyzed for amplitude (peak) information. This information is translated to level (dB).

Step 2: The R.M.S. (Root Mean Square) value of each of the samples is calculated from the formula R.M.S. = 0.707 x Peak Value. (Some compressors (like on the SSL 9000K console) allow for selection of level detection between RMS and Peak. This can be used for different aspects and it is a common practice to use peak compression in the processing of drums sounds, i.e. signals with fast transient information).

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where x(n) is the input signal, xd(n) is the n^{th} sample of the vector xd (which is initially a vector of all zeros and equal to the length of the input signal x) , RT is the release time, AT is the attack time and a = absolute value of [x(n,1)) – xd(n-1)].

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$$f(n) = 10\text{^}(-slope*(\log10(xd(n))-\log10(threshold))$$

where $0<slope<1$. Slope determines the onset of the compression.

The output y(n) = x(n) * f(n).

The above process is repeated for the n samples.

(Zolzer 2002).