Dynamic Range Compressor Design

Technical Report in Analog Signal Processing and Discrete Circuit Design for Applications in Audible Frequency Domain

Shaun Goda

Department of Electrical and Computer Engineering, Rutgers University - New Brunswick

Abstract

This technical report examines the fundamental theory and implementation of dynamic range compressors in the realm of analog signal processing as we embark on a constructional project. Through tests and evaluations, we present methods for achieving efficient and high-performance compressors, where we will start with an overview of the basic building block of the compressor, its representation as a mathematical function, and methods of implementation through analog circuitry. This paper offers an entry point for those looking to understand the fundamentals of how an dynamic range compressor works, while balancing the nuances of the practical methods and challenges associated with developing an electronic hardware device.

Keywords: *Signal Processing, Circuit Design, Dynamic Range Compression, Analog Circuit, Printed Circuit Board, Signal Integrity.*

1 Introduction

A dynamic range compressor (DRC) is an essential tool in audio processing that normalizes the loudness of a signal that has been put through. This is done by using a mechanism that attenuates signals above a set threshold, utilizing a variable gain mechanism governed by a level-detection algorithm. The process effectively lowers the volume of louder segments of the audio while maintaining the level of quieter sections, resulting in a more uniform overall loudness. This technology is widely used in various areas such as music production, live performances, and even in devices like hearing aids to improve sound quality.

With the emergence of high-performance computational platforms and advancement in signal processing technologies, digital signal compression algorithms has been a de-facto standard in the industry. Digital compressors are popular in many applications, from creating music to broadcasting and beyond, as they allow for detailed and precise manipulation of sound. The digital transformation of compressors and other signal processing tools has fundamentally changed audio production. Allowing the average consumer to achieve professional-quality production on a personal laptop without the need for an access to a professional recording studio.

Furthermore, despite the dominance of digital signal processing technologies, hardware compressors, where the signal is processed entirely in the analog domain, widely remain in use. Many audio professionals continue to prefer them for certain tasks as it offers various advantage and benefits that has an edge over software implemented compressors.

In comparison to digital compressors, analog compressors are generally valued for the unique qualities they can add to sound. They are known for introducing warmth[1]and musicality to the audio that is often described as being difficult to achieve with digital methods. [**citations needed**] This effect is due to the slight alterations the analog circuits introduce, including minor distortions and saturations, making the audio sound richer and more engaging to the audience. Analog compressors are also favored for its straightforwardness, avoiding some of the issues that digital formats can introduce, such as digital latency caused by AD/DA conversion and process run time. Furthermore, it is advantageous in avoiding issues that are caused by the inherent nature of digital compressors, such as unwanted sampling artifacts, signal aliasing, and the added complexity of software installation and digital library management. Thus, despite the convenience and precision of digital compressors, analog compressors still hold its own place in audio production for their ability to enhance sound in a distinctive way.

In light of the various advantages seen in analog compressors, this paper will delve deeper into the technical nuances and design considerations that underpins an effective analog compressor. By examining the control parameters, exploring the functional building blocks, and discussing various control mechanisms, we aim to investigate the intricate balance between theory and practical application that lies in designing an analog compressor. Furthermore, the following sections highlights the critical decisions and engineering challenges involved in creating an analog compressors that not only meet the technical specifications, but also stands out as an appealing alternative to the widely used digital compressor.

[1] The term 'warmth' is often used to refer to the pronounced reverberation time at low frequencies relative to that at higher frequencies. [**britanica-warmth**]

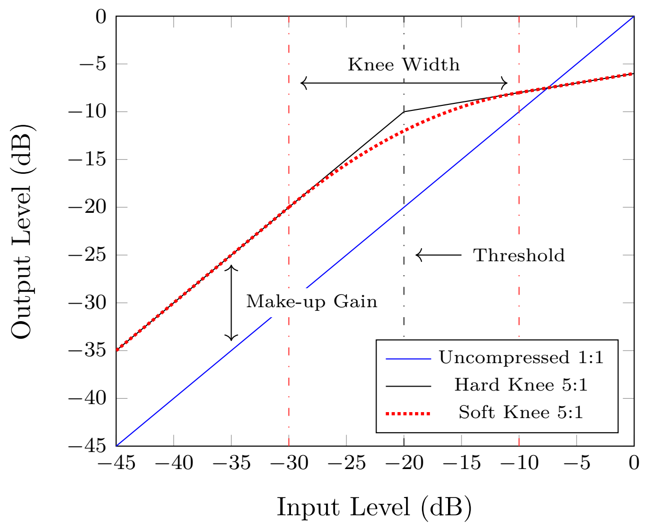


Figure 1: *A typical compressor's transfer characteristics, mapped on a decibel scale.*

2 Compressor Control Parameters

A compressor's functionality is governed by its control parameters, which allow users to configure the effect's characteristics. Key parameters include Threshold, Ratio, Attack, Release, Knee, and Side-chain; each defining a specific aspect of the compressor's behavior and its impact on the input signal.

Threshold The threshold parameter sets the level at which the compressor starts to act. Measured in decibels (dB), it defines the point above which the input signal will be compressed. When the signal level exceeds this threshold, compression is applied.

Ratio The ratio determines the degree of compression applied to an audio signal once it exceeds a predetermined threshold level. This parameter is quantified as a ratio, such as 5:1 or 10:1, extending to infinity:1, at which point the compressor operates as a limiter, equalizing all signal levels above the threshold to a consistent amplitude. For instance, a 5:1 ratio signifies that for every increment of 5 dB by which the input signal surpasses the threshold, the output signal's level will be attenuated by only 1 dB. Consequently, higher ratios yield more pronounced compression.

$$R = \frac{y_{dB} - T}{x_{dB} - T} \tag{1}$$

Equation 1 mathematically models the compression ratio R, where x_{dB} represents the input level gain, y_{dB} the output level gain, and T being the threshold. This formula dictates the change in output level for a given change in input level, relative to the threshold, providing a precise numerical value for the compression effect. The closer the value of R is to 1, the more subtle the compression, while larger values indicate more aggressive compression, leading to the limiting effect.

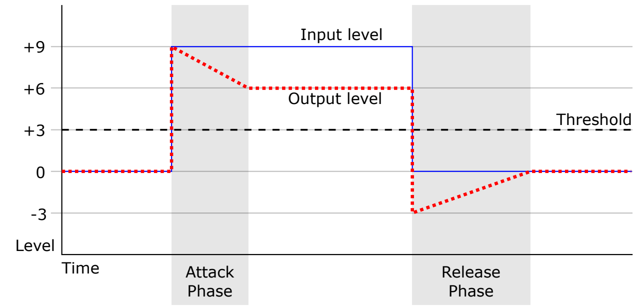


Figure 2: *Transient response of a signal through a compressor.*

Attack The attack time determines the time it takes for the compressor to start acting after the signal exceeds the threshold, as seen in the 'attack phase' section of figure 2. It is usually measured in milliseconds (ms). A fast attack time means the compressor responds quickly to level changes, suitable for controlling sharp, transient sounds. A slower attack allows some of the initial transients through, preserving more of the signal's natural character.

Release Release time refers to the duration needed for the compressor to cease its action once the signal dips below the threshold. (as illustrated in Figure 1) This period, also expressed in milliseconds, influences the sound's dynamics; a brief release time halts the compression swiftly, preserving the natural dynamics but may lead to noticeable fluctuations in volume. Conversely, an extended release time ensures a more seamless and gradual transition back to the uncompressed sound.

Knee The knee parameter (measured in dB in reference to the difference in the lower and upper bound of the knee) precisely controls the compressor's transition from the uncompressed to the compressed state by adjusting its width, which measures how the compression ratio is applied at around the threshold level. A 'hard knee' setting indicates an immediate application of the compression ratio as soon as the signal crosses the threshold, resulting in an abrupt transition. Conversely, a 'soft knee' setting allows for a gradual introduction of compression as the signal nears and surpasses the threshold, due to a wider knee width. The control over the knee width allows the user to achieve a smoother, more natural compression effect, making the transition less noticeable and more musically pleasing.

Make-Up Gain Make-up gain is a control parameter that allows you to increase the output level of the signal after it has gone through compression. This feature helps the user with adjusting the output to a consistent volume level after the signal has undergone attenuation by the compression algorithm. In the context of a song or an audio program, it allows the user to enhances the presence of the material.

Side-chain Side-chain is a control that is established by passing an additional audio signals to trigger the compression on the main input signal. This technique allows the compressor to react not to its own input signal, but to the dynamics of another, offering creative applications such as ducking, where the volume of one sound is reduced by the presence of another (e.g., background music lowering when someone speaks). It's a powerful tool for achieving more complex mixing and mastering effects, enhancing clarity, and creating rhythmic variations in audio production.

3 Compressor Topology

The method of attenuation seen in a compressor could be generally categorized into two topologies including feedback and feed-forward compression. Both topologies can be expressed as a relation described in equation 2, where the output, y_{dB}, is given by the sum of input, x_{dB}, and the gain/attenuation level, g_{dB}, which is determined by the gain computer.

$$y_{dB}(t) = x_{dB}(t) + g_{dB}(t) \tag{2}$$

The topology of an compressor fundamentally influences the performance, sound characteristics, and functionality of dynamic range compressors. Such influences can be seen in factors such as how quickly a compressor reacts to signal changes, the smoothness or aggressiveness of compression. Other topologies in addition to the feedback and feed-forward type is discussed and is examined as well.

3.1 Feedback Compression

In a feedback topology, the output signal is looped back and used as part of the signal processing chain as shown in figure 3. This configuration allows for the compressor to react to the processed signal, enabling a more adaptive response to the audio material. The inherent nature of the feedback system ensures that the compressor's adjustments are directly influenced by its own output, leading to a smoother and more consistent control over dynamic range. However, the design of the feedback mechanism also presents several limitations, including the lack of a look-ahead capability and its ineffectiveness as an ideal limiter, as theoretically, an unlimited negative amplification is required by the gain computer, which is not possible when implemented in physical hardware.

$$g_{dB} = (1 - R)(y_{dB} - T) \tag{3}$$

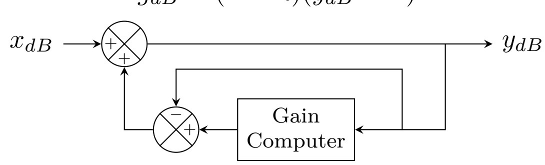


Figure 3: *Control diagram of a feedback compressor topology.*

3.2 Feed-forward Compression

As opposed to the feedback topology, feed-forward topology uses the input signal directly to control the compression process, without the influence of the compressed signal in the control loop. (As seen in figure 4) This allows for more precise and immediate control over the compression characteristics, as the system's response is solely based on the incoming audio signal. Both implementations of feedback and feed-forward compressors involves a common process: subtracting the threshold from the signal level, which is sourced either from the input in feed-forward designs or the output in feedback designs. This is followed by half-wave rectification and then multiplication of the result by a slope variable, which varies between feed-forward and feedback compressor designs.

$$g_{dB} = (\frac{1}{R} - 1)(x_{dB} - T) \tag{4}$$

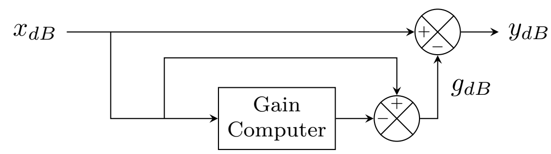


Figure 4: *Control diagram of a feed-forward compressor topology.*

3.3 Additional Topologies

3.3.1 Broadband, Multiband, and Spectral Compression

Broadband compressors operate by adjusting the dynamics of an input signal across the entire frequency range; when a set threshold is exceeded, the entire signal is affected, regardless of the specific frequencies composing the signal's energy. Differing from this approach, a multiband compressor divides the entire frequency spectrum into multiple bands that can be independently adjusted. This allows for more flexible and significant compression of a single track or instrumental group without the risk of making extensive changes too audibly noticeable. On another front, spectral compressors continuously analyze the spectral distribution of a signal's energy and compare it with target values determined by intelligent algorithms. When the system identifies that certain frequency areas are being overemphasized, thus disproportionately affecting the compression, those areas are automatically subjected to more intense compression. This nuanced handling of frequency-specific dynamics facilitates a more tailored and effective compression strategy.

4 Functional Building Blocks of a Compressor

The inner components that consist the functionality of a compressor can be mainly separated into three building blocks, where the gain controller takes the role of applying the attenuation to the input, and the threshold level detector and the input volume detector consisting the gain computer is responsible of controlling the gain controller by generating the signal to control it.

4.1 Gain Detector

The gain computer is responsible for generating the control voltage, which dictates the gain reduction level applied to the signal. This stage encompasses parameters including Threshold (T), Ratio (R), and Knee Width (W), where its relation in regards to the output of the gain computer is described by a set of bounded equations shown in 5.

$$y = \begin{cases} \frac{x-T}{R} + T & \text{for } 2(x-T) > W \\ \frac{(x-T+\frac{W}{2})^2(\frac{1}{R}-1)}{2W} + x & \text{for } 2\,|x-T| \leq W \\ x & \text{for } 2(x-T) < -W \end{cases} \tag{5}$$

When it comes to implementing this function on an analog equation, the non-linear relation makes it harder.

4.2 Level Detector

The level detector is responsible for providing a representation of the loudness of the input signal. There are a two main ways including the peak detector and the RMS detector to generate the control voltage based on the input. The level detector is employed to ensure a consistent depiction of the signal's intensity and can be implemented at multiple points within the side-chain. This smooth adjustment in gain results from the specified attack and release durations. Fine-tuning these durations is essential for the compressor's effectiveness, as the selection of these parameters frequently correlates with the occurrence of undesirable distortions.

4.2.1 Peak Reading Detector

The peak reading detector is crucial for capturing the maximum amplitude of the input signal within a specified timeframe, ensuring the dynamic range of audio material is maintained without clipping. The peak detector operates by identifying the highest level of the signal, which is essential for applications like limiting and peak normalization to prevent distortion. While highly effective at preserving signal integrity, its primary disadvantage lies in its potential to cause overly conservative gain reduction in compressors, as it responds to transient peaks that may not reflect the overall perceived loudness. This can sometimes result in less natural compression effects, as it disregards the human ear's time-based perception of loudness.

4.2.2 Average Reading Detector

4.2.3 RMS Reading Detector

Referencing the Root Mean Squared (RMS) value of the input signal may be useful when we would like to set the compression reference based on the smoothed average of the input signal. Unlike peak detection,

The RMS value in its continuous form expression is defined by the following function. [0]

$$V_{RMS} = \lim_{T\to\infty} \sqrt{\frac{1}{T}\int_{-\infty}^{T} v_{in}^2(t)\cdot dt} \tag{6}$$

Utilizing the RMS value of the input signal is particularly helpful when you would like to utilize a reference that is more aligned with how our ears perceive loudness. RMS detection ensures more consistent levels throughout the audio material, as it's less influenced by short transient peaks, and is suited for providing a more musical and natural-sounding compression.

4.2.4 Crest Factor

The crest factor, representing the ratio of a signal's peak amplitude to its root mean square (RMS) value, plays a important role in the control of dynamic range compression. By quantifying the extent of peakiness in audio signals, it offers a valuable metric for adjusting the gain in dynamic range compressors, aligning more closely with the intricacies of human auditory perception. This parameter is particularly crucial in scenarios where maintaining audio fidelity is paramount, as it helps in tailoring the compression process to preserve the natural dynamics of the source material while preventing auditory fatigue in listeners.

In practical terms, the crest factor informs the design of gain control circuitry by facilitating a more discerning response to varying signal characteristics. It enables the compression algorithm to dynamically adjust its parameters, such as threshold and ratio, based on the signal's momentary peakiness rather than its average level or instantaneous peaks alone. This approach ensures a more balanced and transparent output, minimizing the risk of overcompression, which can lead to a loss of dynamic range and a 'flattened' audio experience.

4.3 Gain Level Controller

The gain control in a dynamic range compressor can be achieved by using a voltage controlled amplifier (VCA) where its gain factor is modulated by a control voltage generated by the level detection circuitry. The VCA adjusts the signal's gain based on this control voltage, which reflects the signal's amplitude relative to the compressor's parameters (threshold, ratio, attack, release).

5 Optimizing Compressor Performance

5.1 Compressor Topology Selection

Taking in account of various components that consists the gain computer, which has been discussed in the previous section, different types of variation can be further implemented to...

Both feedback and feedforward topologies offer unique benefits and are chosen based on the specific requirements of the application. The selection between feed-

Type of Waveform (1V Peak Amplitude)	Crest Factor (V_{Peak}/V_{RMS})	RMS value	Average Reading Circuit	Error (%)
Undistorted Sine Wave	1.414	0.707	0.707	0
Gaussian Noise	3	0.333	0.295	-11.4
Undistorted Triangle Wave	1.73	0.577	0.555	-3.8
Gaussian Noise (98% of Peaks<1V)	3	0.333	0.295	-11.4
Rectangular	2	0.5	0.278	-44
Pulse Train	10	0.1	0.011	-89
SCR Waveform (50% Duty)	2	0.354	0.354	-28
SCR Waveform (25% Duty)	4.7	0.212	0.150	-30

Table 1: *Error introduced by an average responding circuit when measuring common waveforms.*

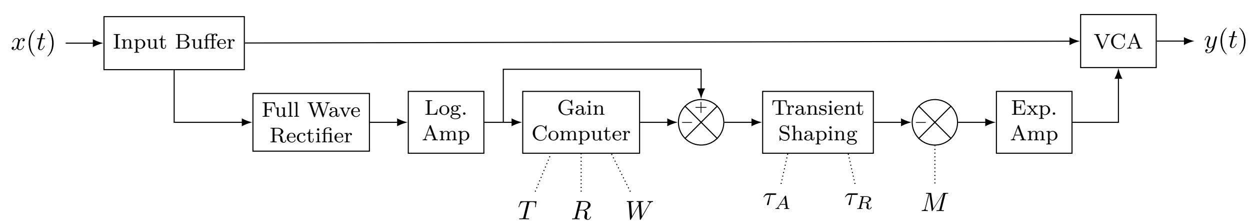


Figure 5: *System diagram of a proposed analog DRC system.*

back and feedforward topology affects aspects such as the compressor's responsiveness, the ease of implementation, and the overall sound character. Integrating these topologies within the broader system design, taking into account control systems theory and practical implementation considerations, is essential for optimizing the compressor's performance and achieving the desired dynamic range control.

According to a paper by Giannoulis et al. [0], "the detector directly smooths the control voltage instead of the input signal." Since the control voltage automatically returns back to zero when the compressor does not attenuate, we do not depend on a fixed threshold, and a smooth release envelope is guaranteed. The trajectory now behaves exponentially in the decibel domain, which means that the release time is independent of the actual amount of compression. This behavior seems smoother to the ear since the human sense of hearing is roughly logarithmic.

$$x_G(t) = 20log_{10}|x(t)| \tag{7}$$
$$x_L(t) = x_G(t) - y_G(t) \tag{8}$$
$$x_{dB}(t) = -y_L(t) \tag{9}$$

5.2 Logarithmic Domain Signal Processing

The perception of loudness by the human ear exhibits a relation that can be mapped to the logarithmic scale. This characteristic implies that as sound levels increase, our sensitivity to changes in volume diminishes. Leveraging this understanding, by converting the input signal into a logarithmic scale before it is processed by a compressor, we enable the compressor to modulate functions such as knee width compression in alignment with the perceived loudness rather than the signal's linear amplitude. This approach ensures that compression adjustments correlate more closely with human auditory perception. Furthermore, the benefits of logarithmic domain signal processing is not limited to only this.

In the realm of transient shaping, circuits like the peak detection circuit utilize the natural delay inherent in the charging and discharging cycles of a capacitor. The behavior of this cycle, especially its amplitude within a RC network, is mathematically described by equation 10.

$$V(t) = V_0 e^{-\frac{t}{RC}} \tag{10}$$

This equation delineates how the voltage across a capacitor decays exponentially over time. However, this exponential decay can result in a perceptually unnatural volume decrease, an artifact that could potentially detract from the desired sound quality. By transitioning the signal processing to a logarithmic domain, we can mitigate this issue. Transforming the original equation into a logarithmic form yields equation 11.

$$\ln(V(t)) = \ln(V_0) - \frac{t}{RC} \tag{11}$$

In this logarithmic representation, the exponential decay of voltageand consequently, of perceived volumeis linearized. This adjustment facilitates a more natural and smoother compression effect when deploying peak-detection based circuits, aligning the signal processing more closely with the nuances of human auditory perception.

5.3 Side-Chain Filtering

Side-chain filtering can be introduced to limit the range of frequency the gain computer perceives. This functionality is particularly useful in emphasizing or de-emphasizing, which tells the compressor to be more sensitive or less sensitive to a specific range of frequencies. As an example, a high-pass filter might be applied to prevent low-frequency content from triggering the compression, which is particularly useful for avoiding unnecessary compression due to bass-heavy elements in a mix. Pre-detection filtering allows for more targeted and musically relevant compression by controlling which parts of the frequency spectrum most influence the compressor's action, leading to a more controlled dynamic response.[0]

5.4 Mixed Use of Peak Detection and RMS Level

While this is not a feature seen in commercial DRC hardware, introducing a mixer that allows a smooth adjustment between RMS level and peak detected level may offer additional flexibility for dynamic control, enabling more tailored responses to different audio materials. This proposed feature allows the user to leverage the advantage of both RMS level and peak detected level, combining a smoothed average of the input signal with the fast transients.

6 Implementation of Discrete Circuit Compressor Components

6.1 Precision Full-Wave Rectifier

A precision full-wave rectifier seen in figure 15 [in addendum] has been implemented to feed the absolute value of the input signal into the level detection circuitry. This is necessary as the level detection circuitry could detect signals only over positive ranges. By converting a signal that ranges over the negative and positive ranges into a signal that is represented only in the positive domain, the level detector can output a value that represents the energy of the entire signal instead of half of it.

6.2 Logarithmic Amplifiers

Two primary log amplifier designs exist: the multistage log amplifier and the transimpedance log amplifier. The multistage variant relies on a series of amplifiers limiting the signal in sequence, a technique commonly employed for processing high-frequency signals up to several gigahertz, especially in radar and communications applications. Conversely, the DC log amplifier incorporates either a diode or a diode-connected transistor within the feedback loop of an op-amp, limiting its frequency range to below 20 megahertz. This design is often utilized with sensors in control systems due to its frequency limitations.

6.2.1 Transimpedance Log Amplifier

The voltage drop across a silicon diode exhibits a relationship that is proportional to the logarithm of the current flowing through when it's integrated into the feedback loop of an inverting operational amplifier.

$$V_{out} = \frac{kT}{q} ln(\frac{I_{in}}{I_{es}}) \tag{12}$$

The configuration as shown in figure **??** enables the output voltage of the op-amp to be directly proportional to the logarithm of the input current, where it effectively transforms the op-amp into a logarithmic amplifier.

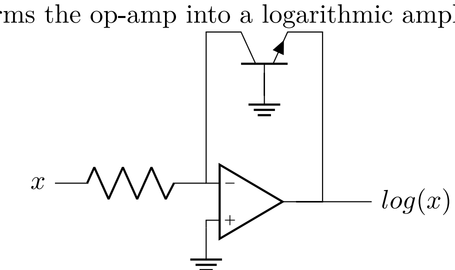


Figure 6: *Transimpedance log amplifier*

However, the transimpedance logarithmic amplifier faces challenges such as temperature-dependent performance, limitation to unipolar signals, and variable/limited bandwidth, which affects precision and efficiency. Furthermore, in a situation combining multiple amplifiers for both logarithmic and anti-logarithmic computations on a single chip, temperature-induced inconsistencies are mitigated. Such cases can be emulated when working with discrete circuit components where opamps, transistors, and resistors that are part of the log amp consists from a single package, or in close proximity to match its ambient temperature. [**ad-log-amp-basics**]

6.2.2 Pseudo-Logarithmic Approximator

A multistage logarithmic amplifier utilizes a cascaded arrangement of amplifier stages as shown in figure 7, where each stage is designed to operate over a specific portion of the input signal's dynamic range. The principle behind this configuration is to divide the wide input dynamic range into smaller segments, with each segment being processed by a different amplifier stage. Each stage typically consists of a linear amplifier followed by a compression mechanism, which together produce an output proportional to the logarithm of the input signal within its designated segment.[0]

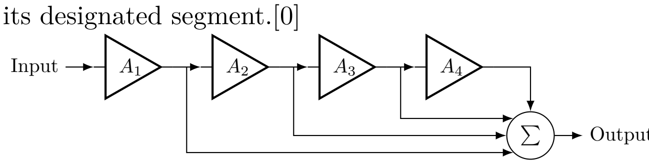


Figure 7: *Multistage log amplifier architecture*

The overall output is a piecewise linear approximation of the logarithmic function across the input range, achieved by summing the outputs of all stages. This approach allows for a broader dynamic range and better approximation accuracy than a single-stage logarithmic amplifier. The multistage design compensates for the non-idealities of individual components and enables the circuit to emulate a more accurate logarithmic response by effectively stitching together the outputs of each stage. Calibration and careful design considerations are essential to ensure continuity and minimize discontinuities between the segments handled by different stages.

6.3 Gain Computer

6.3.1 Precision Clipper

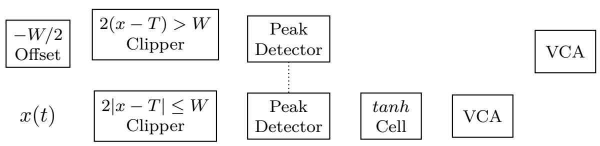


Figure 8: *Knee control scheme*

6.3.2 Analog Multiplier and Dividers

Leveraging the log and anti-log amplifiers explored in section 6.2, we possess the ability to execute both multiplication and division operations within the realm of analog circuitry. This unique capability stems from the fundamental principles of logarithmic computations, where the addition of two numbers in the logarithmic domain equates to their multiplication in the standard numerical domain once the anti-logarithm (exponential function) is applied to the resultant value as shown in equation 13.

$$10^{(\log_{10} A+\log_{10} B)} = A \times B \tag{13}$$

Similarly, a subtraction in the logarithmic domain corresponds to division in the conventional numerical domain as shown in equation 14.

$$10^{(\log_{10} A-\log_{10} B)} = \frac{A}{B} \tag{14}$$

Furthermore, such set of operations using log and anti-log amplifiers could be implement in the following manner.

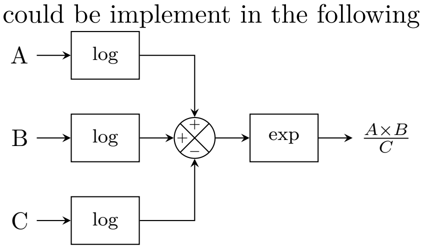


Figure 9: *Multiplication and division using log and anti-log amplifiers*

Using this feature of mathematical operations, the circuit can be arranged in a way so that we can implement the output of an gain detector described in equation 5.

6.3.3 Hyperbolic Tangent Function Cell

A hyperbolic tangent function can be achieved in the analog signal domain by implementing a differential transistor pair coupled with an input of an operational amplifier. As the input range of the function cell is determined by the voltage range of the active region given by the base input of the bipolar junction transistor, it is crucial to normalize the amplitude of the signal entering the cell beforehand. As seen in figure 10, the gated nature of BJT input allows us to create unique characteristics in the definition of how the knee control is applied.

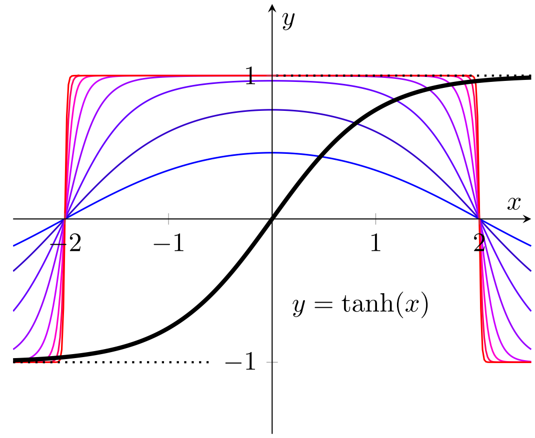


Figure 10: *The black line shows the function plot of $y = tanh(x)$. The red line on the other hand represents the transient output of the tanh function where amplitude of the sinusoidal input is varied.*

6.4 Transient Shaping

6.4.1 Precision Peak Detector

This is an alternative to the RMS detector. However, it excels at fast attack signals compared to the RMS detector. This allows the compressor to process signals at higher speeds instead of simply normalizing the volume.

A simple peak detection circuit can be implemented using a diode, a capacitor, and a resistor.

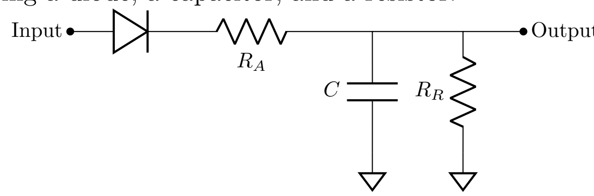


Figure 11: *Circuit diagram of a lossy peak detector.*

6.4.2 True RMS Detector

A traditional approach to reading the average of an input signal was to implement a

6.4.3 Crossfader

To implement the mechanism discussed in section 5.4, a crossfader can be implemented to allow an continuous

6.5 Voltage Controlled Amplifier

A voltage controlled amplifier allows an

7 Physical Compressor Hardware

This section diverges from the topics discussed in previous sections and focuses more on the technicalities involved in implementing the circuit that has been designed. Specific challenges associated with audio domain signals will be introduced to make the reader aware of anything to be cautious. I plan to use Altium Designer for designing the PCB. Describe steps taken during the PCB design process and any important points noticed.

7.1 Miscellaneous Circuitry

7.1.1 Power Delivery

Audio domain applications gain advantages from utilizing a bipolar bias power supply. Such a supply enhances the effective utilization of IC's full dynamic range, facilitates rail-to-rail amplification, shields the analog signal from ground noise, and delivers numerous additional benefits. [0]

To receive the benefits of a split rail power supply while reducing unwanted noise and ripples seen in common topologies such as a simple switching mode power supply, a topology where a inverting charge pump is combined with an linear & low-dropout (LDO) regulator. (shown in figure)

Discuss the difference in LDO over traditional power regulation sources. The DRC circuit will require a DC power source that will be converted to several voltage domains (+12V, -12V, 5V).

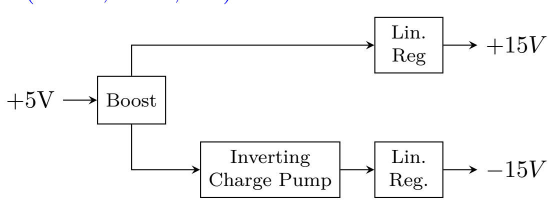


Figure 12: *Power Managemenet Schema*

7.1.2 Balanced Line Input/Output

The use of a balanced line driver/receiver are commonly seen in professional audio hardware, and it aids in minimizing noise and interference. The used of balanced line input and output employs differential signaling to effectively cancel out noise and electromagnetic interference picked up along cable runs, which allows for preserving the dynamic range and subtleties of the audio signal. This setup is particularly advantageous for long cable runs and in environments with high electronic noise, ensuring that analog compressors can accurately control dynamics without the detrimental effects of added noise or signal degradation.

While differential line receivers and drivers can be designed with ease using a set of precision resistors and precision resistors, specialized ICs such as

7.1.3 Precision Current Sink Source

A current source is required to generate a constant ink of current across the collector and the emitter of the BJT being used in the tanh function cell. We utilize the design shown in [0].

7.2 Printed Circuit Board Design

7.2.1 Layer Stackup

With the emergence of low-cost off-shore PCB manufacturing services, PCBs are easily in the reach of an average consumer who are looking to design and produce a small quantity of electronic hardware. Due to the low-cost of 2 layer and 4 layer PCBs, a 4 layer PCB can be selected as its flexible in the routing of traces and offer better signal integrity.

The three PCB stackups depicted in figure 14, each offers distinct advantages for power delivery and signal integrity across dual power domains. The outer GND stackup provides superior EMI shielding by enclosing signal and power layers within ground planes, ensuring stable signal propagation but potentially increasing PDN inductance due to distant power planes. The one-sided hybrid stackup allows for a compact design, situating power close to its loads on one side but risking signal integrity due to the lack of adjacent ground planes. Finally, the inner GND stackup optimizes for both power integrity, with closely coupled power and ground planes minimizing PDN impedance, and signal integrity, as signals are tightly sandwiched between grounds, albeit at the potential cost of less effective EMI shielding compared to the outer GND configuration. Each stackup requires a tailored approach to manage the complexities of positive and negative power rails, thermal performance, and mechanical robustness.

7.2.2 Component Placement

7.2.3 Design Rules and Consraints

The majority of design rule and constraints have been determined by referencing the tolerance specification set by the PCB manufacturer.

7.2.4 Signal Integrity and EMI Precautions

7.3 Component Selection

Go over components used in the circuit and why that specific part was picked. Discuss how component selection could affect thermal characteristics, linearity, bandwidth etc.

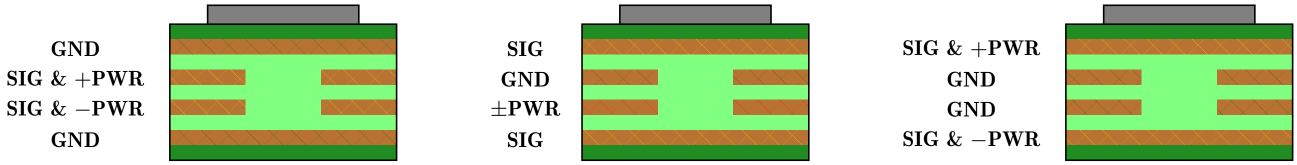
8 Discussion

The discussion goes here.

Acknowledgments

Put acknowledgments here.

Attribution



(a) *Outer GND stackup.* ***(b)*** *One-sided hybrid stackup.* ***(c)*** *Inner GND stackup.*

Figure 14: *Various PCB stackup configurations.*

Figures 15, 18: *Schematic diagrams are generated using Altium Designer.*

Resources

All resources developed upon the completion of this project including the schematic, simulation, and CAD files are available for download at the following **GitHub repository**.

- Latest updated schematic of the implemented analog compressor.

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Addendum

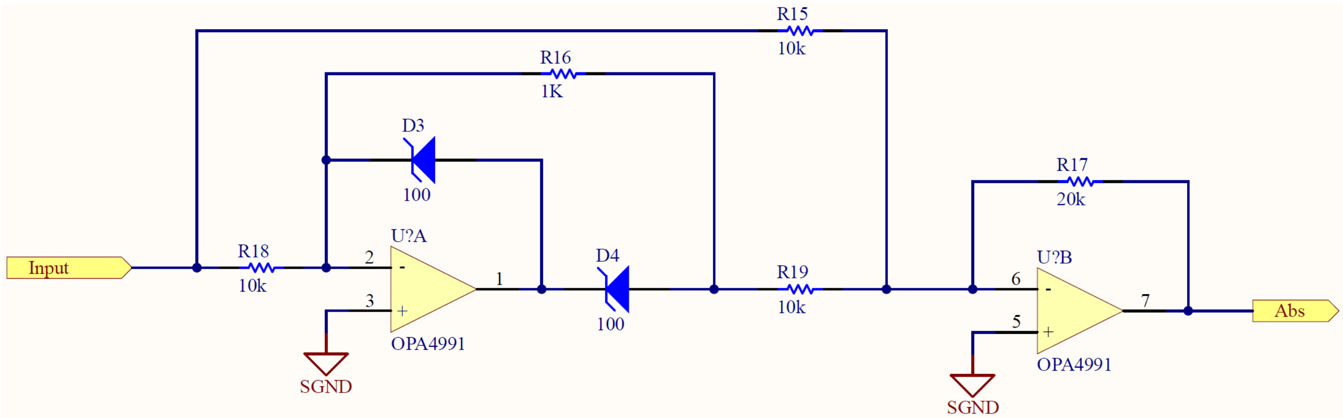

Figure 15: *Circuit diagram of a precision full bridge rectifier implemented in the design.*

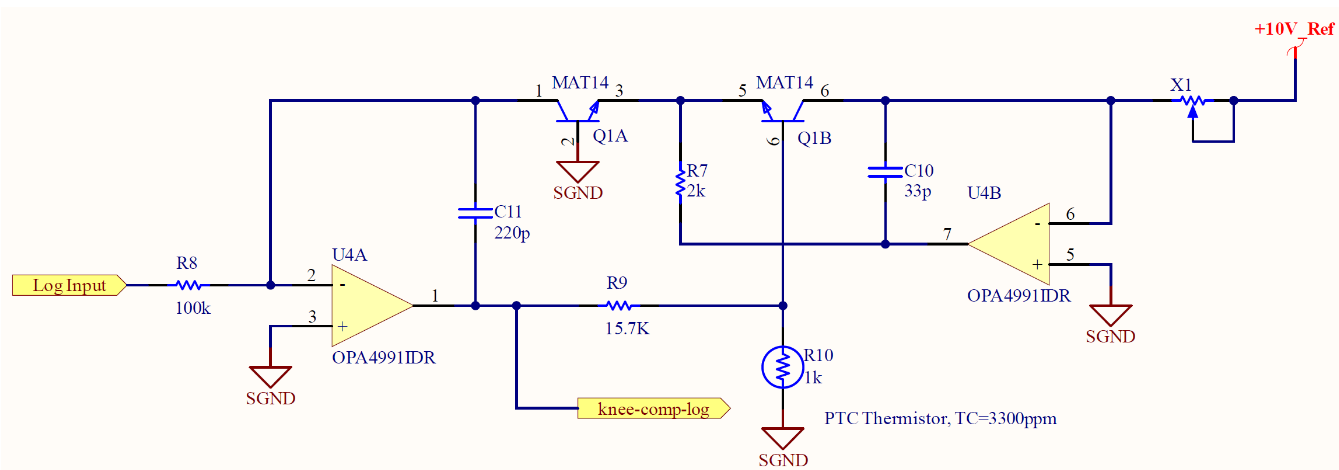

Figure 16: *Circuit diagram of a temperature compensated precision log amplifier.*

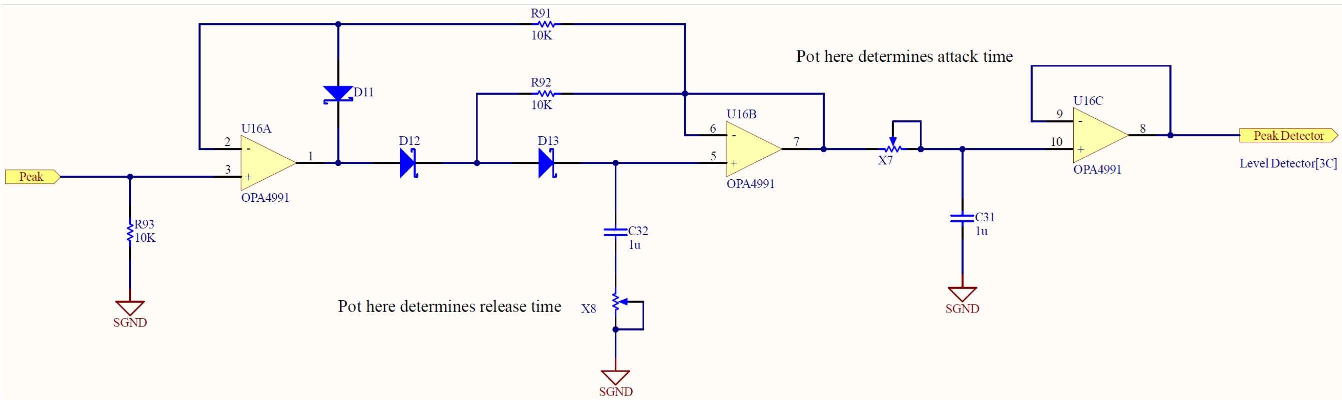

Figure 17: *Circuit diagram of an peak detector circuit with adjustable attack and release*

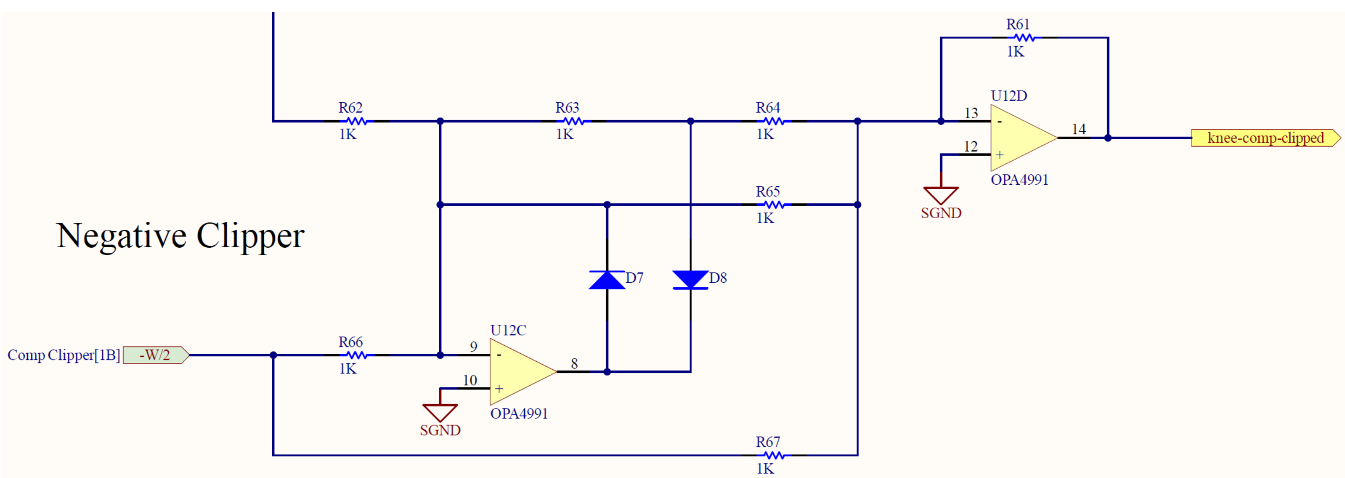

Figure 18: *Circuit diagram of a negative clipper*