Dynamic Range Compressor Design

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

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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, Dynamic Range Compression, Circuit Design, Analog, Printed Circuit Boards, 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 approach to compressors and various other signal processing tools‘’ has revolutionized the way we work with audio, making it easier to achieve high-quality production on your personal laptop without having access to a professional recording studios.

However, despite the dominance of digital signal processing technologies, hardware compressors, where the signal is processed entirely in the analog domain, remain in use. Many audio professionals continue to prefer them for certain tasks as it has the advantage of being straightforward, avoiding some of the issues that digital formats can introduce, such as digital latency caused by AD/DA conversion or digital processing delay. Furthermore, it is advantageous in avoiding issues such as unwanted sampling artifacts and the added complexity due to its digital nature.

Analog compressors are also valued for the unique qualities they can add to sound. They are known for introducing a warmth and musicality to audio that is often described as difficult to achieve with digital methods. This effect is due to the slight alterations they make to the sound, including adding minor distortions and saturation, which can make the audio feel richer and more engaging. 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.

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. The following sections will offer a comprehensive analysis of the compressor topology, discussing various optimization strategies in designing an ideal compressor. Through this exploration, the paper aims to highlight 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.

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 audio signal.

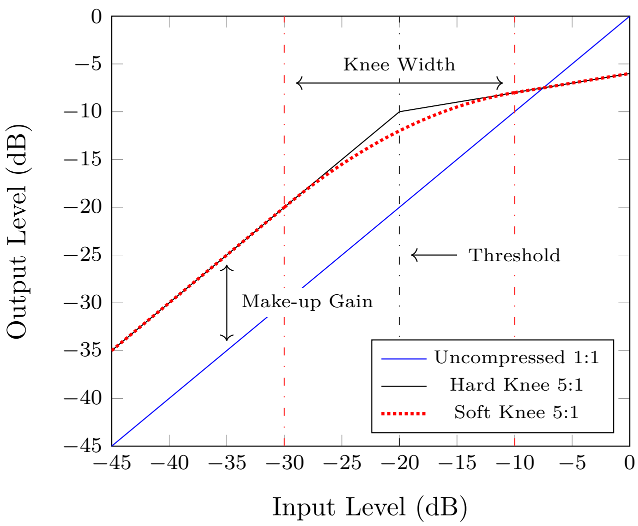


Figure 1: *A typical compressor's transfer characteristics.*

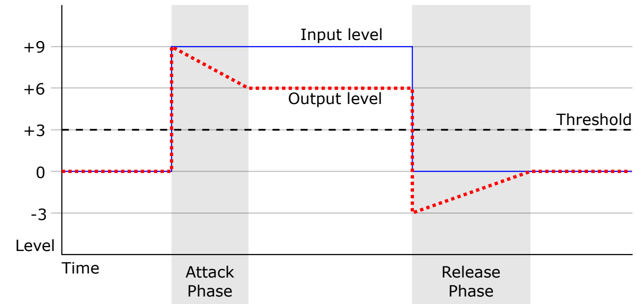


Figure 2: *Transient response of a signal through a compressor.*

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 the signal once it surpasses the threshold. It is expressed as a ratio (e.g., 5:1, 10:1), and could scale up to ∞:1 where the compressor basically function as an limitter, where all signal levels above the set threshold level is limited to equal amplitude. In simple terms, a 5:1 ratio means that for every 5 dB, the input signal exceeds the threshold, the output signal level will be increased by only 1 dB. Higher ratios result in more aggressive compression.

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 is the time it takes for the compressor to stop acting after the signal falls below the threshold. Also measured in milliseconds, a shorter release time stops the compression effect more quickly, which can help maintain natural dynamics but might result in a 'pumping' sound. A longer release time provides a smoother, more gradual return to the uncompressed state.

Knee The knee parameter adjusts how the compressor transitions from the non-compressed to the compressed state. A 'hard knee' setting means the compression ratio is applied abruptly as soon as the signal crosses the threshold. In contrast, a 'soft knee' setting introduces compression gradually as the signal approaches and then exceeds the threshold. This results in a more natural, less noticeable compression effect.

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 compressor control that enables external audio signals to trigger compression on another audio track. 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.

Broadband, Multiband, and Spectral Compression Broadband compressors works on the dynamics of an input signal across the entire frequency range. When a set threshold is exceeded, the entire signal is affected, regardless of which frequencies the energy of the signal is composed of.

In a multiband compressor, the entire frequency spectrum is split up into multiple bands that can be parametrized independently from each other. Hence, you are able to compress a single track or instrumental group much more flexible and heavier, without having to fear that bigger changes end up being audible too much.

With a spectral compressor, the spectral distribution of energy of a signal is analyzed continuously and compared with target values that have been calculated by intelligent algorithms. When the system recognizes that certain frequency areas are currently being overemphasized and therefore affect the compression disproportionately, these areas are automatically compressed more.

3 Compressor Topology

The method of attenuation seen in a compressor could be generally categorized into two topologies including feedback and feed-forward compression. 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, and how transparently it applies gain reduction.

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 ?. This configuration allows for the compressor to react to the processed signal, enabling a more adaptive response to the audio material. Feedback topology is often praised for its musicality, as it tends to produce a more natural compression effect. 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. This topology excels in applications where preserving the natural dynamics and timbre of the audio is important.

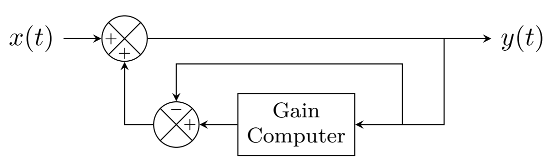


Figure 3: *Control diagram of a feedback compressor topology.*

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. 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. Feed-forward compressors are known for their accuracy and fast response, making them ideal for applications requiring precise dynamic control, such as in limiting scenarios where preventing signal peaks is crucial.

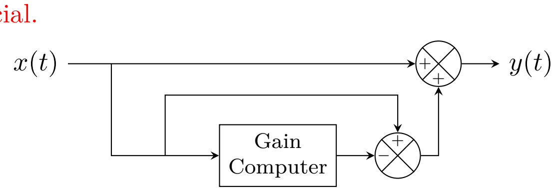


Figure 4: *Control diagram of a feed-forward compressor topology.*

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 1.

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

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.

4.2.1 Average Reading Level

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

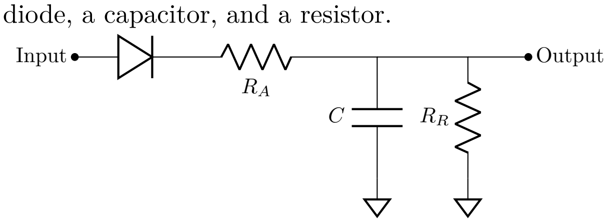


Figure 5: *Circuit diagram of a lossy peak detector.*

4.2.2 RMS Level Referencing

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. [6]

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

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.

4.2.3 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 feedback 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. [1], "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.

$$x_G(t) = 20log_{10}|x(t)| \quad (3)$$

$$x_L(t) = x_G(t) - y_G(t) \quad (4)$$

$$x_{dB}(t) = -y_L(t) \quad (5)$$

5.2 Logarithmic Domain Signal Processing

The intensity of loudness perceived by the human ear scales logarithmically. This means that as the volume of the sound increases, our ears become less sensitive to the change in that volume. By scaling the signal that gets fed into the compressor on a logarithmic domain, we can operate functions such as knee width compression based on the perceived loudness of the sound instead of the linear amplitude of the signal that the compressor sees. Additionally, processing the signal in the logarithmic domain can aid in achieving a better transient response when dealing with analog transient shaping circuits.

A typical transient shaping circuit such as the peak detection circuit operates on the signal by using the inherent delay seen in the charge and discharge of a capacitor, where its amplitude in relation to the surrounding RC network is expressed by equation 6.

$$V(t) = V_0 e^{-\frac{t}{RC}} \quad (6)$$

The above equation shows that the decay in the voltage of a capacitor is represented as an exponential function, however, this could introduce an unwanted artifact as an exponentially decreasing level in volume is perceived to sound unnatural. Furthermore, this defect could be compensated by simply processing the signal in the log domain. The expression shown in equation 6 can be easily manipulated to show its behavior in the logarithmic domain, where we will get the expression shown in equation 7.

$$\ln(V(t)) = \ln(V_0) - \frac{t}{RC} \quad (7)$$

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.*

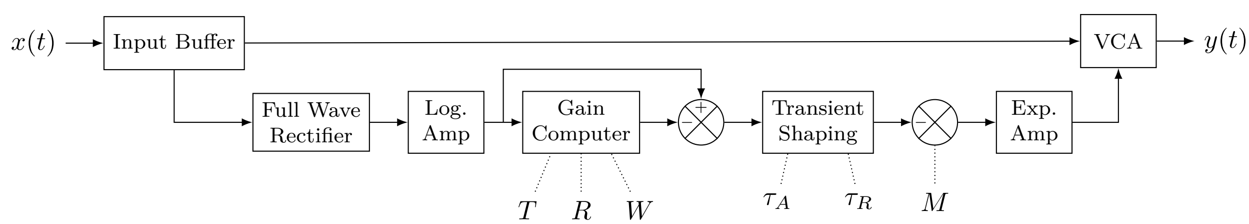


Figure 6: *System diagram of a proposed analog DRC system.*

This expression essentially linearizes the decay in the perceived volume level and allows us to achieve a smoother compression when using peak-detection based circuits.

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.[20]

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 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.

In the context of analog compressors,

6.2.1 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.[15]

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.

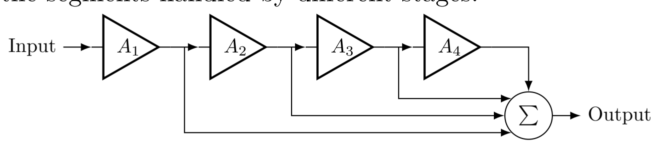


Figure 7: *Multistage log amplifier architecture*

6.2.2 Transimpedance Log Amplifier

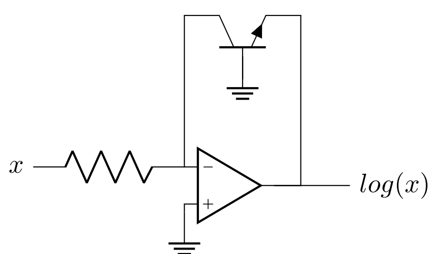


Figure 8: *Transimpedance log amplifier*

$$V_{out} = \frac{kT}{q} ln(\frac{I_{in}}{I_{es}}) \tag{8}$$

Explain the difference in exponential decay vs linear decay and its significance. I plan to implement the circuit that appears in an application report [7] by Texas Instruments. This circuit reverts the signal that has converted to dB scale back into the linear domain signal.

6.3 Gain Computer

6.3.1 Precision Clipper

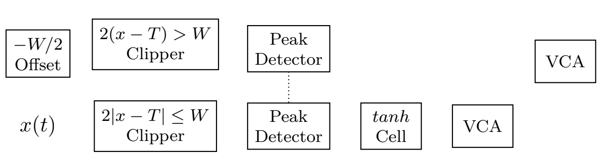


Figure 9: *Knee control scheme*

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 9.

$$10^{(\log_{10} A + \log_{10} B)} = A \times B \tag{9}$$

Similarly, a subtraction in the logarithmic domain corresponds to division in the conventional numerical domain as shown in equation 10.

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

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

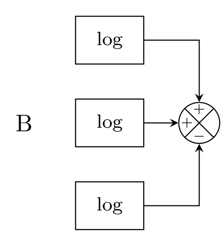


Figure 10: *Multiplication and division using log and anti-log amplifiers*

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 11, the gated nature of BJT input allows us to create unique characteristics in the definition of how the knee control is applied.

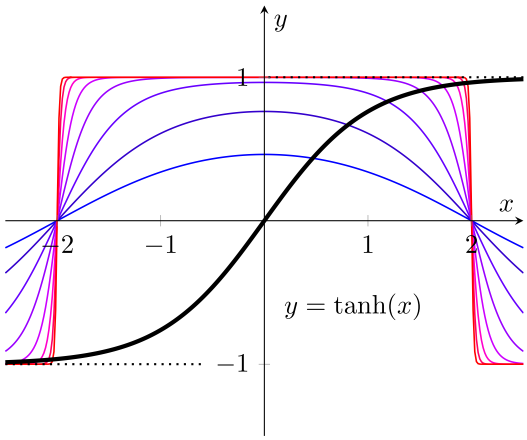


Figure 11: *The black line shows the function plot of $y = tanh(x)$. The red line on the other hand represents the transient output of the tanh function where amplitude of the sinusoidal input is varied.*

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.

6.4.2 True RMS Detector

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

For the RMS detector, an LTC1966 true-RMS detector by Analog Devices will be used. The LTC1966 utilizes a $\Delta\Sigma$ computational technology patented by Analog Devices, which makes it simpler to use, more accurate, requires less power, and is dramatically more flexible than the conventional log anti-log RMS-to-DC converters.

6.4.3 Crossfader

6.5 Voltage Controlled Amplifier

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. [9]

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).

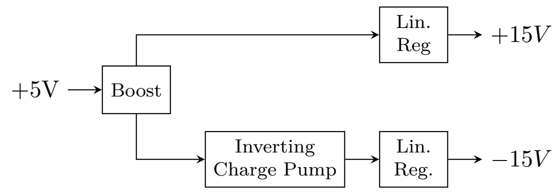


Figure 12: *Power Managemenet Schema*

7.1.2 Balanced Line Input/Output

7.1.3 Precision Current Sink Source

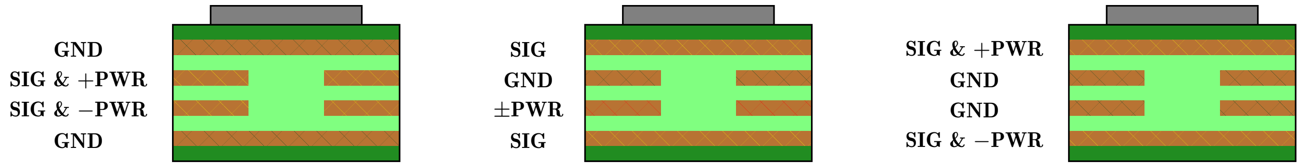
A current source is required to generate a constant ink of current across the collector and the emitter of the BJT being used in the tanh function cell. We utilize the design shown in [11].

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.



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

***Figure 14:** Various PCB stackup configurations.*

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 Precausions

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 Performance Evaluation

8.1 Simulation

Talk about simulation methodology and results here. Note if changes are made to the circuit retroactively at this stage of the design process.

8.2 Measurements

Here, I discuss the performance of the physical compressor hardware and draw attention between the real-life and simulated values. This section will be redacted in case the hardware can't be put together before the deadline of this paper, which is March 8th, 2024.

9 Discussion

The discussion goes here.

Acknowledgments

Put acknowledgments here.

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

Put pictures of completed project, PCB layout, etc. over here.