Dynamic Range Compressor Design

Technical Report in Analog Signal Processing and Discrete Circuit Design for Applications in the 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 through a project involving the audible domain frequencies. 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. 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 have 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. [24] 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 their 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 their 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 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.

[1] The term "warmth" is often used to refer to the pronounced reverberation time at low frequencies relative to that at higher frequencies. [23]

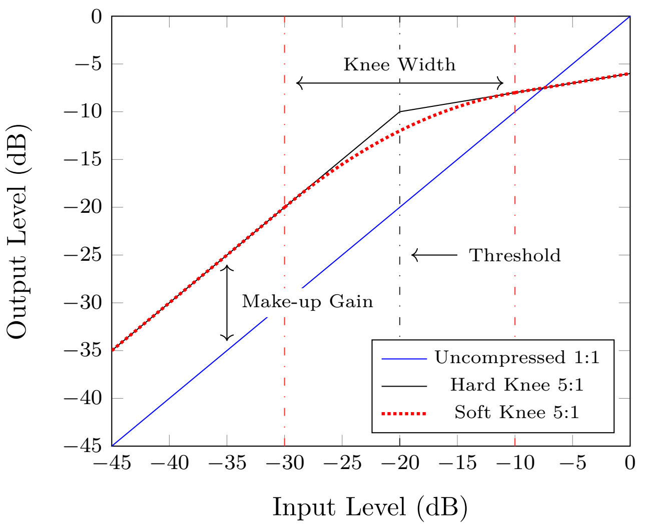


Figure 1: *A typical compressor's transfer characteristics, mapped on a decibel scale.*

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 make-up gain; each defines 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.

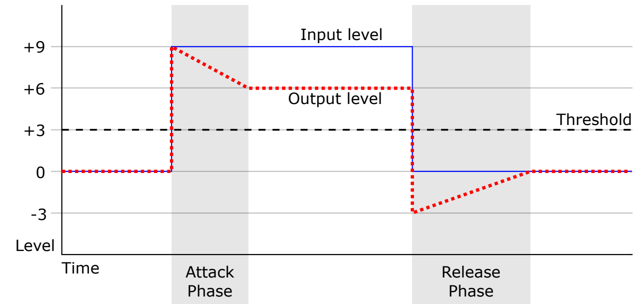


Figure 2: *Transient response of a signal through a compressor.*

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 characteristics.

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 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. The width is measured in dB in reference to the difference in the lower and upper bound of the knee. 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.

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.

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.

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

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.

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. Assuming a hard knee with no attack or release, equation 3 models the output of the gain computer in such topology.

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

In general, the feedback topology allows for the compressor to react to the processed signal, enabling a more adaptive response to the audio material. Furthermore, 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 it would require an infinite negative amplification applied at the gain computer stage to achieve equal infinite negative attenuation.

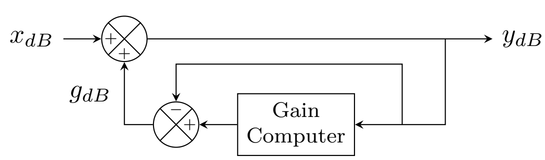


Figure 3: *Control diagram of a feedback compressor topology.*

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.

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

The gain computer's output, g_{dB} can be modeled by equation 4, and when in comparison to equation 3, it can be shown that the inherent issue seen in requiring an infinite amplification is no longer a problem in feed-forward topology as the slope for the limiter simply reaches a slope of -1. For this reason, many modern implementation of the both digital and analog compressor rely on the feed-forward variant.

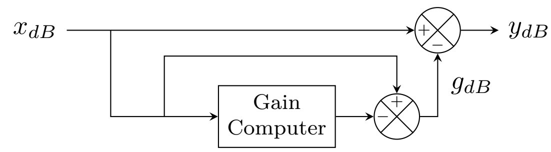


Figure 4: *Control diagram of a feed-forward compressor topology.*

3.3 Additional Topologies

In addition to feed-forward and feedback topologies, there are other methods in where additional features in a compressor can be implemented. This section discuss features commonly seen in commercial compressors, such as external side-chaining, multiband compression, and upward compression.

3.3.1 External 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.

3.3.2 Upward compression

Upward compression, in contrast to traditional downward compression described in previous sections, operates by selectively amplifying signals below the threshold, effectively minimizing dynamic range by elevating softer sounds closer to the level of louder ones. This process is critical for enhancing audibility of low-level details without affecting the peaks, making it valuable for audio applications requiring nuanced sound enhancement, such as in music production and sound design.

Key to its application is the careful management of the noise floor, as amplification of quiet signals inherently raises background noise levels. Thus, upward compression requires a delicate balance between enhancing desired signals and controlling unwanted noise, utilizing advanced digital algorithms for RMS or peak detection to maintain audio fidelity.

3.3.3 Broadband, Multiband, and Spectral Compression

Broadband compressors are the most standard type of compressor discussed in the previous sections where its operates 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.

4 Functional Building Blocks of a Compressor

The inner components that form the functionality of a compressor can be mainly separated into three sections, including the compression calculator, level detector, and the output amplifier. Where the output amplifier takes the role of directly adjusting the amplitude of the compressor, the compression calculator and the level detector (consisting the gain computer) is responsible for generating the control signal for the output amplifier that varies the attenuation level. Within this section, various implementations of output amplifier, compression calculator, and the level detector are examined.

4.1 Compression Calculator

The gain computer is responsible for generating the control voltage, which dictates the gain reduction level applied to the signal. This stage is based on determining amplitude-based parameters of the control signal generated by the gain computer, 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 equation 5.

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

Furthermore, due to the inherent nature of how knee compression is implemented, when the knee width is set to zero, the smooth knee is essentially identical to the hard knee.

4.2 Level Detector

The level detector is responsible for providing a representation of the loudness of the input signal, where its output function is mainly based on operating time-varying (often introducing delay) functions on the input. Two main implementations of the level detectors are commonly seen in compressors including the peak detector and the RMS detector. Such detectors are employed to ensure a consistent depiction of the signal's intensity and can be implemented at multiple points within the side-chain based on the topology.

4.2.1 Peak Reading Detector

The peak reading detector is crucial for capturing the maximum amplitude of the input signal within a specified time-frame, ensuring the dynamic range of audio material is maintained without clipping. Furthermore, a variant on the standard peak detector, called a quasi-peak detector, are often used in the context of compressors, where attack and release time can be incorporated by incorporating a charge-storing component. (as shown in figure 5)

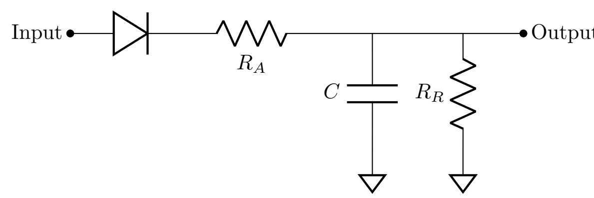


Figure 5: *Circuit diagram of a lossy peak detector.*

In the configuration shown above, the diode allows current to flow in one direction, charging the capacitor to the peak voltage level of the input signal. The resistor provides a discharge path for the capacitor, setting the time it takes for the detector to respond to changes in signal level. This arrangement captures and holds the maximum amplitude of the signal momentarily, allowing the compressor circuitry to adjust the gain based on this peak value.

While highly effective at preserving signal transients, 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

The average reading detector is designed to capture the mean amplitude of an input signal over a defined time period, providing a measure that reflects the general level of the signal rather than its peak values. This approach is particularly useful in applications where the overall energy of the signal, rather than its instantaneous peaks is of interest.

The operation of an average reading detector can be implemented by simply removing the rectification stage

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.*

(diode) where the signal is directly passed through a low-pass filter to obtain a DC level proportional to the average amplitude of the input waveform.

4.2.3 RMS Reading Detector

Utilizing the root mean square (RMS) value of the input signal is particularly helpful when you would like to reference the average power of the input signal, which is more aligned with how our ears perceive loudness. Unlike peak detection, the output of the RMS reading detector has an affect on the both transient and amplitude of the output, where its continuous form expression is defined by equation [6].

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

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. Additionally, the RMS detector offers an edge over the average reading detector particularly in their ability to provide true measurements of power across a variety of waveform types as highlighted in table 1.

Furthermore, utilizing the RMS value may fall short compared to peak reading detectors as it may not react quickly enough to fast input transient, which can lead to unwanted distortion or clipping if the peaks are not adequately managed. Additionally, due to the function of RMS value being based on an indeterminate time ranging from the $t = 0$ to ∞, it comes at a cost of having less control over parameters such as attack and release in the context of implementing compressors.

4.3 Output Amplifier

The gain control in a dynamic range compressor can be achieved by using a components such as 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). Furthermore, counterparts to the VCA that can be used in analog circuits are later discussed in section 6.5.

5 Optimizing Compressor Performance

5.1 Compressor Topology Selection

Both feedback and feed-forward topologies offer unique benefits and are chosen based on the specific requirements of the application. However, due to the inherent setback seen in feedback topology where it requires an infinite level of amplification to achieve perfect limiting, (as discussed in section 3) the feed-forward variant remains a more precise and popular option in the modern day.

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 the gain computer, 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, 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 7.

$$V(t) = V_0 e^{-\frac{t}{RC}}\ . \tag{7}$$

This equation shows how the voltage across a capacitor decays exponentially over time. The exponential decay that is exhibited by this characteristics can result in a per-

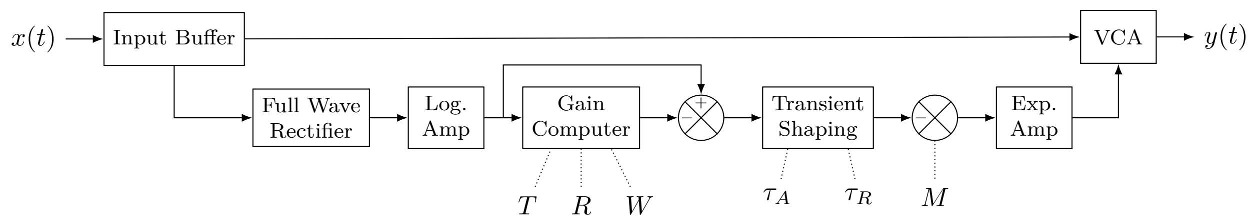


Figure 6: *System diagram of a proposed analog DRC system.*

ceptually unnatural volume decrease, and 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 8.

$$\ln(V(t)) = \ln(V_0) - \frac{t}{RC} \ . \tag{8}$$

In this logarithmic representation, the exponential decay of voltage and consequently, of perceived volume is 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. [21]

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.

5.4 Mixed Use of Peak Detection and RMS Level

To mitigate the inherent drawbacks identified in peak and RMS (Root Mean Square) level detectors, as discussed in section 4.2, a unique and a previously un-explored method can be adopted by allowing the user to dynamically mix the two signals. While there are commercial hardware compressors that allow the user to toggle its level detector either between peak and RMS detectors, we can further modify this feature by incorporating a cross-fading mixer. The cross-fading mixer enables a fluid transition between RMS level detection and peak level detection, letting the user determine its respective mixing ratio.

Furthermore, by implementing a feature that automatically adjusts the mix between the peak and RMS levels, governed by a algorithm (such as a PID controller) that maintains a constant crest factor over a moving average signal, the compressor is able to automatically scale its attack and release values based on the input audio material. This dynamic adjustment is made possible by continuously analyzing the crest factor, defined as the ratio of the peak level to the RMS level of a signal, represented by the equation:

$$\text{Crest Factor} = \frac{\text{Peak Level}}{\text{RMS Level}} \ . \tag{9}$$

Furthermore, employing the crest factor enables users to assess the equilibrium between transient and sustained components in a mix, offering insights on compression strategy.

6 Discrete Circuit Implementation of Compressor Components

The following section continues with an in-depth exploration of compressor components, further extending the discussion to transition into their implementation through the use of analog circuits.

6.1 Precision Full-Wave Rectifier

A precision full-wave rectifier seen in figure 18 can be implemented to output the absolute value of the input signal. By converting a signal that ranges over the negative and positive ranges into a signal that is represented only in the positive domain, it can safely be fed into the log amplifier.

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 Mhz. This design is often utilized with sensors in control systems due to its frequency limitations.

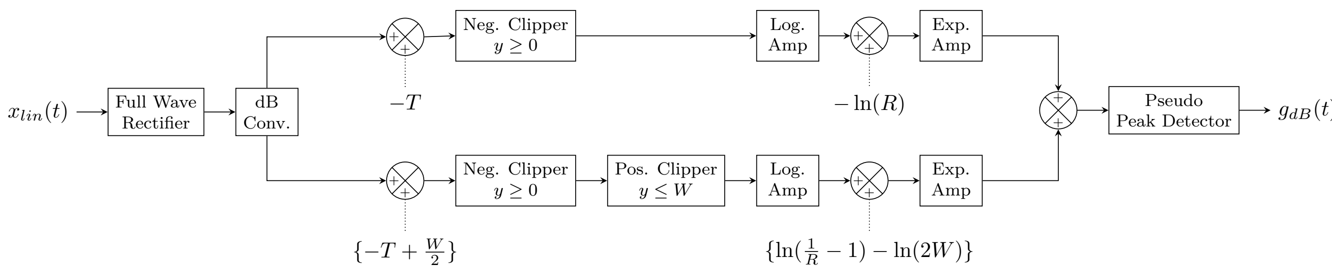


Figure 7: *System diagram showing inner components of the gain computer being proposed.*

6.2.1 Transimpedance Log Amplifier

The voltage drop across a bipolar junction transistor 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.

$$V_{out} = \frac{kT}{q} ln(\frac{I_{in}}{I_{es}}) \ . \tag{10}$$

The configuration as shown in figure 8 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.

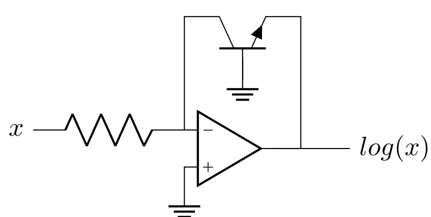


Figure 8: *Transimpedance log amplifier*

While this configuration of log amplifier can achieve an dynamic range of around -120dB, [15] 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 antilogarithmic 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. [15]

6.2.2 Pseudo-Logarithmic Approximator

A multistage logarithmic amplifier utilizes a cascaded arrangement of amplifier stages as shown in figure 9, 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.[16]

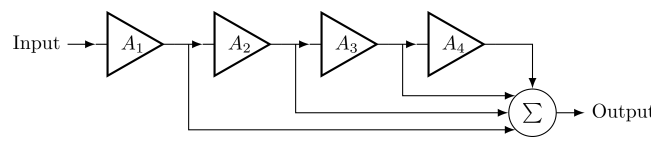


Figure 9: *Multistage log amplifier architecture*

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

A unique approach has been taken to implement the knee characteristics of an gain computer within the analog domain. The proposed design shown in figure 7 takes on the implementation of the feed-forward topology discussed in previous sections, and expands on it by sub-dividing the gain computer topology based on the each domain of operation shown in equation 5. To implement the segmentation of the compression function applied by the gain computer, the copy of the signal entering the side-chain is split into two, where an offset is applied, get its amplitude clipped according to the range of function bounded by parameters such as threshold and knee width.

Furthermore, the design diverges from the architecture of the feed-forward discussed in section 3.2, by removing the need to feed the input signal after the gain computer. While this was required in the original description of the feed-forward topology to ensure that the control signal that gets fed into the VCA is based on the difference between the result of the gain compute and the input signal, this will no longer be required as the output of the gain compute will clip signal that is below $2(x - T) \leq -W$, which represents an level of input amplitude at which no compression is applied.

6.3.1 Precision Clipper

A clipper is a type of electronic circuit that generates an output by eliminating segments of the input that exceed or fall below a specified reference value. This results in the output of a clipper mirroring the input except for the section that has been clipped. Consequently, the peak-to-peak amplitude of a clipper's output will invariably be smaller than that of its input.

In context of the compressor that is being implemented, a clipper is being implemented to restrict the signal entering the gain computer based on equation 5.

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 11.

$$10^{(\log_{10} A + \log_{10} B)} = A \times B \ . \tag{11}$$

Similarly, a subtraction in the logarithmic domain corresponds to division in the conventional numerical domain as shown in equation 12.

$$10^{(\log_{10} A - \log_{10} B)} = \frac{A}{B} \ . \tag{12}$$

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

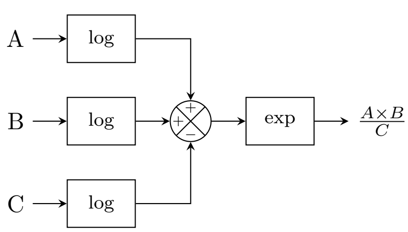


Figure 10: *Multiplication and division using log and anti-log amplifiers*

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

6.3.3 Hyperbolic Tangent Function Cell

In place of using log dividers and multipliers to achieve the quadratic knee compression characteristics described by equation 5, a unique method of utilizing a hyperbolic tangent function can be implemented to model the behavior.

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.

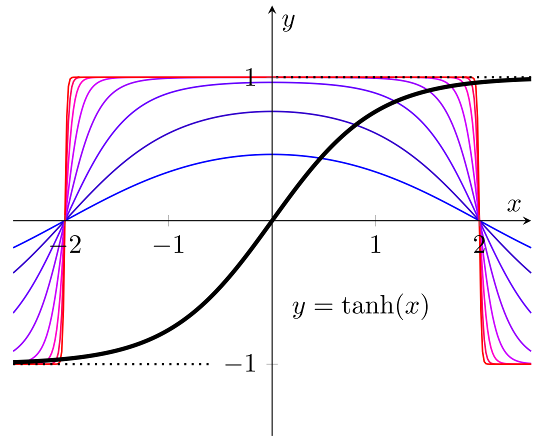


Figure 11: *The black line shows the function plot of $y = tanh(x)$. The red line on the other hand represents the transient output of the tanh function where amplitude of the sinusoidal input is varied.*

6.4 Transient Shaping

6.4.1 Precision Peak Detector

A simple peak detection circuit can be implemented using a diode, a capacitor, and a resistor as shown in figure 5. Furthermore, an alternative peak detector with decoupled attack and release can be implemented as seen in figure 17, where the input signal undergoes immediate peak detection for swift peak estimation, followed by buffering and smoothing via a first-order low-pass filter. This approach ensures the detector remains unaffected by the level discrepancies arising from the varying time constants typical of standard peak detector circuits.

6.4.2 True RMS Detector

When the first electronic RMS detectors were introduced to the industry, they were designed to strictly adhere to the RMS definition, produced linear, not logarithmic, outputs by squaring the input signal, averaging it over time with a low-pass filter or integrator, and then taking the square root. This analog computation involved the use of costly variable-transconductance analog multipliers. [25]

In the modern day, as RMS values can be determined through implementation of digital processing technologies, most analog IC offering of an RMS detector is based on application related to the RF domain, where it remains difficult to perform digital processing due to its extreme high frequency. However, companies such as THAT corporation™ produces RMS detectors for audio domain applications developed during the 1980 1990s, based on the Blackmer RMS detector, where its architecture relies on nonlinear filters implemented in the log domain.

6.5 Gain Controller Alternatives

Gain control methods in audio processing are essential for managing the amplitude of audio signals, and they come in various forms, each with its unique characteristics and applications. These methods include variable-mu, optical, FET-based, diode-bridge, and Voltage Controlled Amplifier (VCA) techniques, each offering different sound qualities and technical advantages.

Variable-mu Variable-mu compressors operate on the principle of variable transconductance in vacuum tubes. The "mu" refers to the tube's gain factor, which changes with the input signal level. By altering the bias voltage, these devices adjust the gain dynamically, offering a compression effect that's inherently program-dependent. This results in a compression characteristic that's musically pleasing, providing a smooth leveling effect that's highly sought after for its warmth and ability to "glue" mixes together.

Optical Compression Optical compressors use a light-emitting element and a photoconductive cell to modulate gain. The input signal's amplitude affects the brightness of the light source, which in turn varies the resistance of the photoconductive cell, achieving gain reduction. This opto-electrical approach allows for a smooth, gradual compression response due to the inherent inertia of the light element's intensity changes, making it ideal for applications requiring transparent dynamic control.

FET FET-based compressors leverage Field Effect Transistors to mimic the variable impedance characteristics of tube circuits but with potentially faster response times. The FET's conductivity is directly proportional to the input signal's voltage, allowing for precise control over the gain reduction with a characteristic that can be fine-tuned for aggressiveness or subtlety. This design is favored for its ability to impart a distinct presence and energy to signals, particularly useful for emphasizing transients in percussive and dynamic sources.

Diode-Bridge Diode-bridge compressors utilize a network of diodes to create a nonlinear resistance path for the audio signal, facilitating compression. This arrangement allows for unique distortion characteristics and a distinct coloration due to the asymmetrical clipping and harmonic generation, often described as aggressive and rich. The diode bridge's response to signal level changes contributes to a compression quality that can significantly enhance the perceived weight and texture of the audio.

VCA Voltage Controlled Amplifiers (VCAs) in compression circuits modulate gain based on a control voltage, which is derived from the input signal level. This design affords precise control over the attack, release, and ratio parameters, enabling a wide range of dynamic effects from subtle volume leveling to aggressive limiting. VCAs are celebrated for their versatility and clarity, capable of providing transparent dynamic control without introducing unwanted coloration or artifacts.

7 Physical Hardware

This section diverges from the topics discussed in previous sections and focuses more on the technicalities involved in implementing the compressor circuit. Specific challenges associated with audio domain signals will be introduced, as well as certain features seen in commonly used compressor hardware.

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, and shields the analog signal from ground noise. [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 can be combined with an linear & low-dropout (LDO) regulator. (shown in figure)

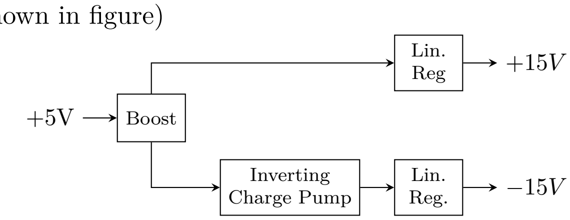


Figure 12: *Power Managemenet Schema*

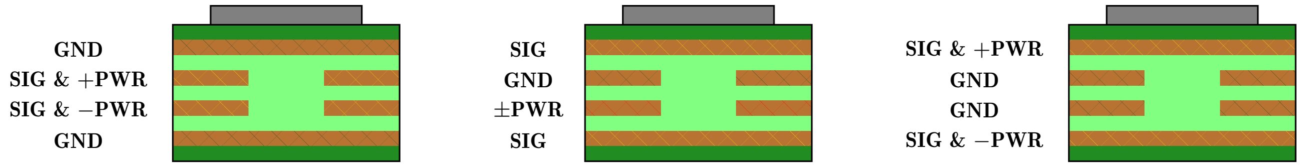
7.1.2 Balanced Line Input/Output

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.

7.2 Component Selection

This section provides any recommendation for component selection that are crucial for the implementation of circuits discussed in the previous sections.

Op-Amp For high-precision applications within the audio domain, specifications such as low input offset voltage, input bias current, low input voltage noise density, and



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

Figure 14: *Various PCB stackup configurations.*

a high slew rate is preferable. Furthermore, a package where multiple op-amps are included should be selected for the log/anti-log computation.

VCA THAT corp offerings of audio specific VCAs are ideal for analog compressors in their product lineup such as the THAT2181 and THAT2180

BJT For log/anti-log computation and the tanh function cells, a transistor pair that exhibit a close value of β is important so that the current gain characteristics are also matched. Analog Devices offers a line of monolithic dual and quad matched transistor arrays such as the SSM2212 and the MAT14.

7.3 Printed Circuit Board Design

7.3.1 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.

7.3.2 Component Placement

While mechanical requirements determine the initial requirements, general guidelines on good floor-planing should be implemented to reduce trace length while spacing out components that may produce external EMI. In lieu of implementing log-based multipliers discussed in section 6.3.2, the components consisting the log and the anti-log section should either be thermally coupled to each other or be placed in proximity to reduce error seen in its operation.

Discussion

In our analysis of dynamic range compression in the realm of analog audio signal processing, we've delved into the details seen in between theory and practice, revealing the depth and complexity inherent in designing effective compressors. The choice between feedback and feed-forward topologies is not merely technical; it's an artistic decision that reflects the intended sonic character and functionality of the compressor. Feedback designs, with their smooth and musical compression, contrast with the precision and control offered by feed-forward configurations, showcasing the diverse palette of sounds that engineers and musicians can draw from.

The discussion around the relevance of analog techniques in a predominantly digital era underscores a significant finding: the unique character and warmth that analog compressors impart to audio signals are irreplaceable and highly valued. This characteristic warmth, a result of complex analog non-linearities, remains a sought-after quality, emphasizing the importance of analog technology in achieving certain sonic aesthetics.

Looking ahead, the potential for integrating analog and digital methodologies opens new avenues for innovation. The development of hybrid systems that combine the tactile richness of analog with the versatility and precision of digital control points to a future where the boundaries of audio processing are continually expanded. This evolving landscape not only challenges us to rethink our approaches to dynamic range compression but also to consider how these technologies can enhance musical expression and creativity.

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:

https://github.com/ShaunG-RU/DRC-Project

Attribution

Figure 2: The attack and release phases in a compressor, Iainf (Own work) via Wikipedia, License: CC BY-SA 3.0 DEED

Figures 15, 18, 19, 17: *Schematic diagrams are generated using Altium Designer.*

References

[1] Giannoulis et al. "Digital Dynamic Range Compressor Design A Tutorial and Analysis". In: *The Journal of the Audio Engineering Society* 60.6 (June 2012), pp. 399–408. URL: https://www.eecs.qmul.ac.uk/~josh/documents/2012/GiannoulisMassbergReiss-dynamicrangecompression-JAES2012.pdf.

[2] Rod Elliott. *Peak, RMS And Averaging Circuits*. 2016. URL: https://sound-au.com/appnotes/an012.htm.

[3] *LTC1966 Precision Micropower ΔΣ RMS-to-DC Converter*. Version LT0511 Rev.B. Linear Technology. 2001. URL: https://www.analog.com/media/en/technical-documentation/data-sheets/1966fb.pdf.

[4] *THAT2181 Blackmer őTrimmable IC Voltage Controlled Amplifiers*. Version 600030 Rev.02. THAT Corporation. 2008. URL: https://www.thatcorp.com/datashts/2181data.pdf.

[5] *THAT4305 Pre-trimmed Analog Engine Dynamics Processor IC*. Version 600067 Rev.02. THAT Corporation. 2015. URL: https://www.thatcorp.com/datashts/THAT_4305_Datasheet.pdf.

[6] Fred Floru. *Attack and Release Time Constants in RMS-Based Feedback Compressors*. Technical Report. THAT Corporation, 1998. URL: https://thatcorp.com/datashts/AES4703_Attack_and_Release_Time_Constants_I.pdf.

[7] *Log Converters*. Application Report AN-30. Texas Instruments, 2013. URL: https://www.ti.com/lit/an/snoa641b/snoa641b.pdf?ts=1706185688957.

[8] Greg Lubarsky. *The Forgotten Converter*. Application Report. Texas Instruments, 2015. URL: https://www.ti.com/lit/wp/slpy005/slpy005.pdf?ts=1707685687935&ref_url=https%253A%252F%252Fwww.google.com%252F.

[9] Antony Pierre Carvajales. *The Top Three Ways to Split a Voltage Rail to a Bipolar Supply*. Technical Article SSZTAI6. Texas Instruments, 2016. URL: https://www.ti.com/lit/ta/ssztai6/ssztai6.pdf?ts=1707663378187&ref_url=https%253A%252F%252Fwww.bing.com%252F.

[10] *Theory and Applications of Logarithmic Amplifiers*. Application Report AN-311. Texas Instruments, 2013. URL: https://www.kennethkuhn.com/students/ee431/mfg_data/theory_and_applications_of_log_amplfiers.pdf.

[11] *Precision Current Sources and Sinks Using Voltage References*. Application Report SNOAA46. Texas Instruments, 2020. URL: https://www.ti.com/lit/an/snoaa46/snoaa46.pdf?ts=1708064878742&ref_url=https%253A%252F%252Fwww.google.com%252F.

[12] Ting Ye. "Precision Full-Wave Rectifier". In: *TI Precision Designs* (2013). URL: https://www.ti.com/lit/ug/tidu030/tidu030.pdf?ts=1703507626027&ref_url=https%253A%252F%252Fwww.google.com%252F.

[13] Glenn Morita. *Noise Sources in Low Dropout (LDO) Regulators*. Application Note AN-1120. Analog Devices, 2011. URL: https://www.analog.com/media/en/technical-documentation/application-notes/AN-1120.pdf.

[14] *Nonlinear Circuits Handbook*. 1976. URL: https://www.analog.com/en/resources/technical-books/nonlinear-circuits-handbook.html.

[15] *Log Amp Basics*. MT-077. 2009. URL: https://www.analog.com/media/en/training-seminars/tutorials/MT-077.pdf.

[16] *High Frequency Log Amps*. Tutorial MT-078. Analog Devices, 2009. URL: https://www.analog.com/media/en/training-seminars/tutorials/MT-078.pdf.

[17] *Basic Compressor / Limiter Design with the THAT4305*. Design Brief 203. THAT Corporation, 2019. URL: https://www.thatcorp.com/datashts/DB203%20Basic%20Comp-Limiter%20Design%20THAT4305.pdf.

[18] Ashok Bindra. "Design Tips for Generating Split-Rail Power Supplies". In: *DigiKey* (2013). URL: https://www.digikey.com.mx/es/articles/design-tips-for-generating-split-rail-power-supplies.

[19] Art Pini. "The Fundamentals of Logarithmic Amplifiers and How They Handle Wide Dynamic Range Signals". In: *Digikey* (2019). URL: https://www.digikey.com/en/articles/the-fundamentals-of-logarithmic-amplifiers#:~:text=The%20multistage%20log%20amp,summing%20circuit%20(Figure%202)..

[20] Barrie Gilbert. "The Multi-tanh Principle: A tutorial Overview". In: (1998). URL: https://web.archive.org/web/20060913234401id_/http://ee.sharif.edu/~comcir/readings/mixers/mixer-gilbert.pdf.

[21] Mark Bassett. "Side-Chain Filtering". In: *Audio Technology* (2014). URL: https://www.audiotechnology.com/tutorials/side-chain-filtering.

[22] Nick Messitte. "4 Types of Analog Compressionand Why They Matter in a Digital World". In: (2023). URL: https://www.izotope.com/en/learn/4-types-of-analog-compression-and-why-they-matter-in-a-digital-world.html.

[23] Inc. Encyclopædia Britannica. *Acoustic criteria*. 2024. URL: https://www.britannica.com/science/acoustics/Acoustic-criteria.

[24] Hugh Robjohns. *Analog Warmth*. 2010. URL: https://www.soundonsound.com/techniques/analogue-warmth.

[25] Gordon W. Roberts and Vincent W. Leung. *Design and Analysis of Integrator-Based Log-Domain Filter Circuits*. Springer, 2006.

[26] Max McAllister. *Types of Audio Compressors and Their Uses*. 2023. URL: https://producelikeapro.com/blog/types-of-audio-compressors/.

Addendum

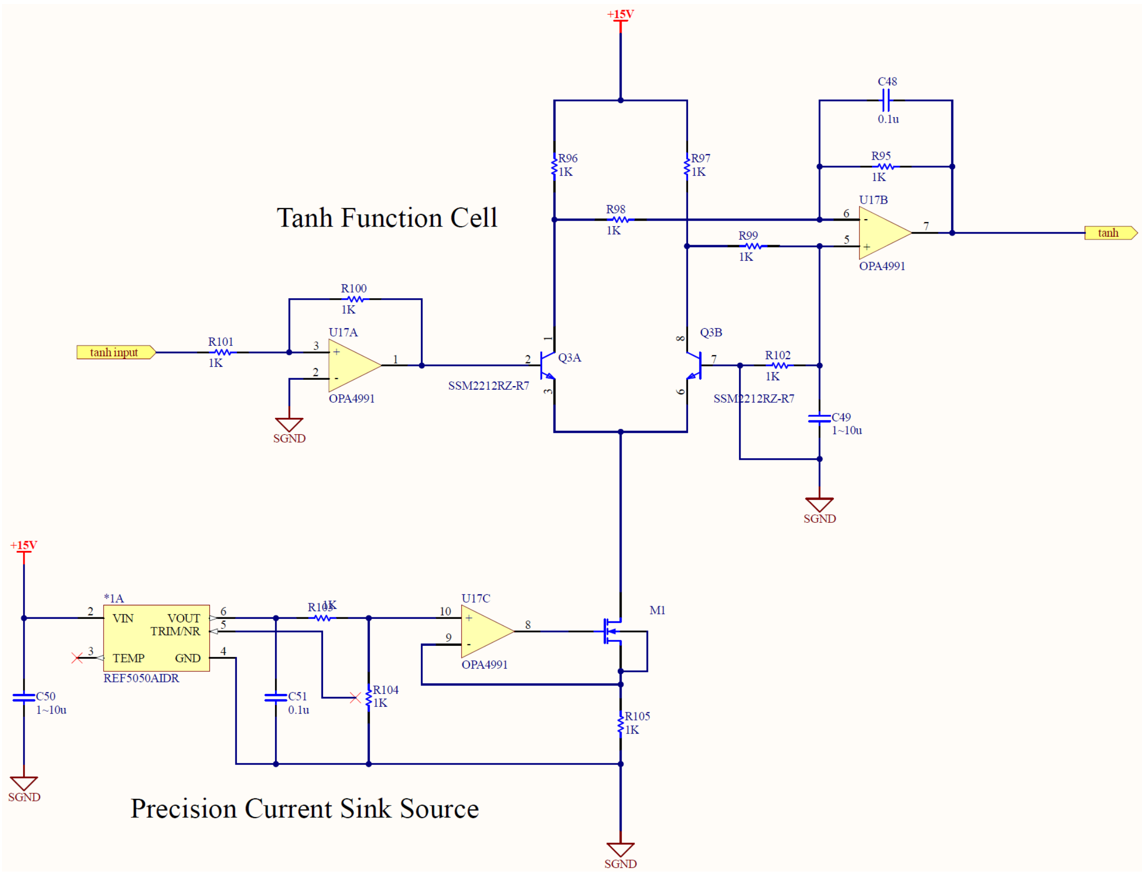


Figure 15: *Circuit implementation of the hyperbolic tangent function cell.*

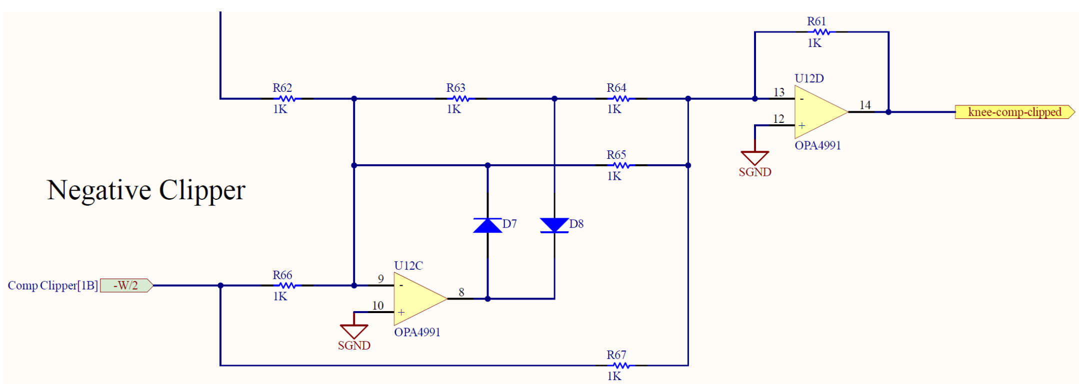


Figure 16: *Circuit diagram of a precision clipper.*

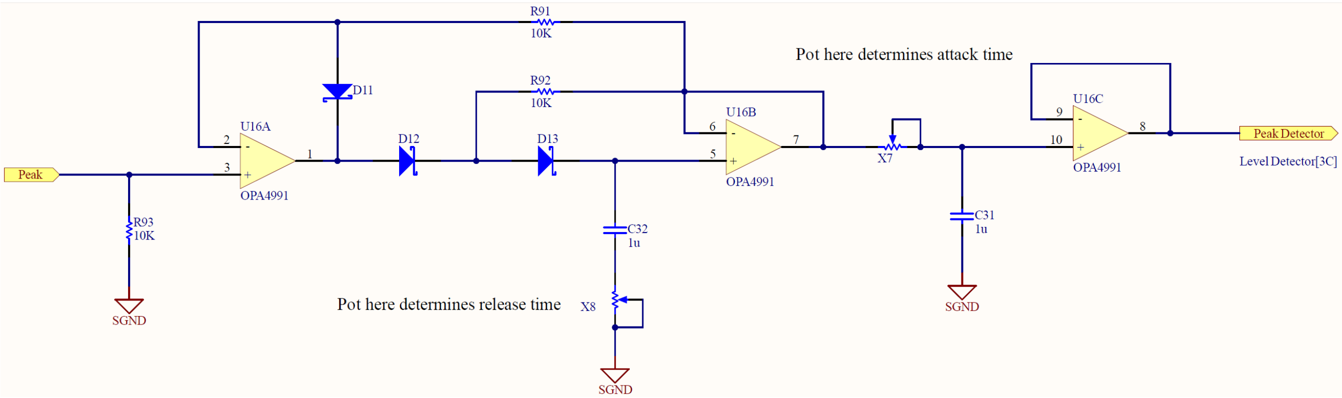


Figure 17: *Circuit diagram of an peak detector circuit with adjustable attack and release*

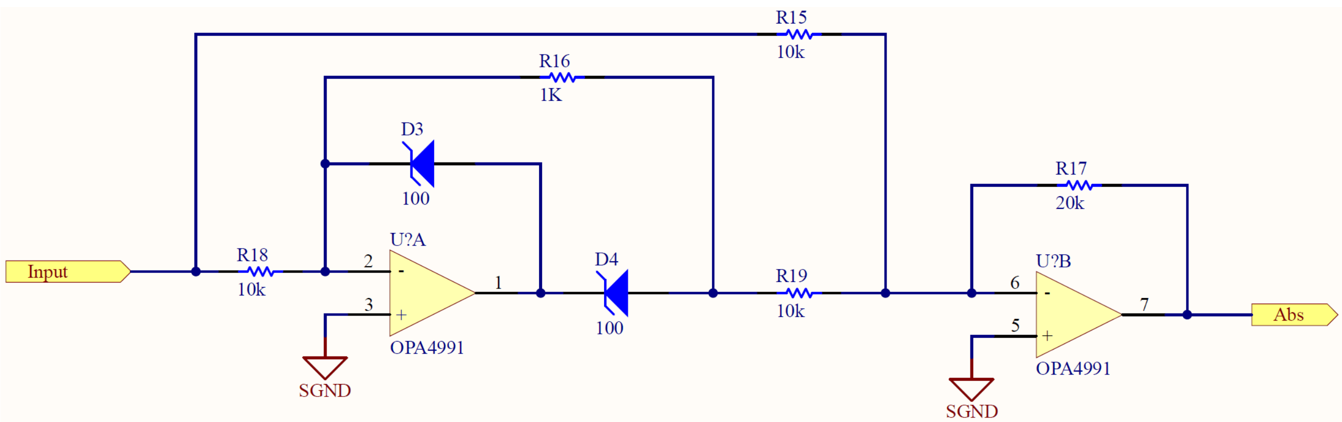


Figure 18: *Circuit diagram of a precision full bridge rectifier implemented in the design.*

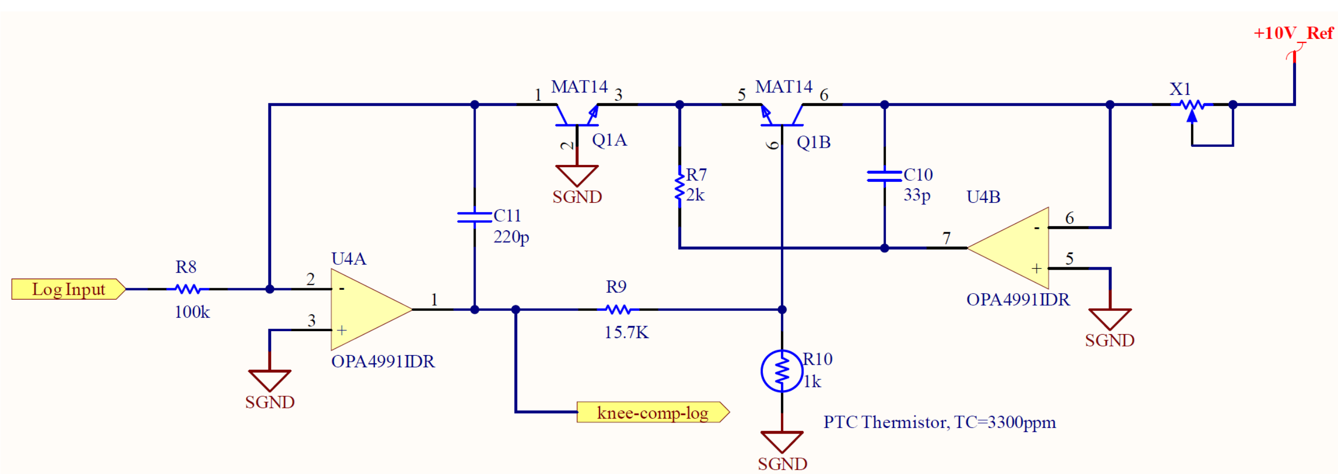


Figure 19: *Circuit diagram of a temperature compensated precision log amplifier.*