FET COMPRESSOR

Formerly “Digitally-Controlled Analog Compressor”

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Abstract Update

Throughout the research and development of the audio compression circuit for this project, the goals and purpose of the project changed significantly. While the original goal of the Digitally-Controlled Analog Compressor project was to design and implement a digitally controlled analog circuit to allow for the inclusion of a comprehensive compression feature-set, it gradually became apparent that designing just the core audio compression sub-circuit would be challenging enough to warrant several months’ commitment—especially for a small development team.

As a result, the Digitally-Controlled Analog Compressor project shifted from the idea of an adequate compression circuit with extensive digital control to instead focus on developing an extremely high quality, original audio compression circuit design: the “FET Compressor”.
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Terms and Keywords

FET – Field Effect Transistor
JFET – Junction Field Effect Transistor
MOSFET – Metal Oxide Semiconductor Field Effect Transistor
Op Amp – Operational Amplifier
AGC – Automatic Gain Control
VGA – Variable Gain Amplifier
ABS – Absolute Value
VCR – Voltage Controlled Resistor
PCB – Printed Circuit Board
Background

Automatic Gain Control

In order to design a professional quality compressor which can be used in studio settings typically occupied by legacy compression equipment, it is crucial to understand the history and purpose of compressor development and implementation. “Compression” is an audio industry term used to describe the squeezing or constricting effect that is frequently applied to audio signals during the recording, mixing, or mastering phase of audio record production. More accurately, compression describes the effects of Automatic Gain Control (AGC). In its simplest form, AGC is implemented by applying a Variable Gain Amplifier (VGA) to an audio signal and allowing the gain to be dynamically controlled by the amplitude of the audio signal.

Commonly, AGC designs are based around the concept of feedback wherein the VGA in the main audio signal path receives some information regarding the amplitude of the output signal that can be used to adjust the gain of the VGA. In the case of negative feedback, some base gain value is applied to the lowest-amplitude components of the audio signal and all other components with higher amplitude cause the gain of the VGA to be reduced. By this method, the dynamic range of the output signal is reduced so that it has a uniform amplitude even as the input signal amplitude varies significantly.

![Figure 1. Negative Feedback Loop, ECE 486](image)

In the context of audio production, the dynamic range processing offered by AGC’s has several important implications.

Initially, AGC’s were developed as a means of protecting other pieces of equipment further down the signal chain from being overloaded at their input stages. This was critically important for both live and recorded audio in the mid-20th century. Both radio broadcasts and analog tape recordings needed to capture loud enough signals to overcome high noise floors and remain intelligible but amplifying these signals too much could result in obvious distortion due to the limited head room available in equipment at the time. By including an AGC in the audio signal path, engineers could increase the overall level of the captured signal without worrying about the level becoming too loud because the AGC would bring any loud transients in the audio signal down to the same level as the rest of the signal.
Following their use in radio broadcasting, AGC’s were quickly adopted by the recording industry as a means of protecting the tape medium used in studios. Due to their high profile and wide availability in studios, audio engineers and musicians soon discovered that the sonic signatures of certain AGC’s could be used in highly musical ways. By adjusting the time constraints and gain variation range that an AGC would operate in, audio engineers could use AGC’s to dynamically affect any part of an audio signal—from entire musical phrases to individual wave cycles. With the advent of AGC’s as a musical tool, their popularity exploded and audio engineers in studios everywhere began to experiment with them and develop ways to completely change the character of a piece of recorded audio using AGC’s.

**Classic Compressor Designs**

During the initial rise of compressors throughout the 1950-1960’s, several influential designs were widely accepted by engineers everywhere and most new compression designs to-date continue to be based on or inspired by these classic designs.

The foremost optical compressor and the first design used in radio broadcast, the Teletronix LA-2A created by Jim Lawrence has remained a studio staple since its inception. The LA-2A’s reputation stems from its simple negative feedback implementation: the output signal amplitude of the LA-2A drives a luminescent panel which is aimed at photoresistors tied to the input stage. As the output amplitude of the LA-2A becomes excessive, the luminous panel grows brighter and the photoresistors at the input react by changing their impedance to effectively reduce the gain available at the input stage.

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![Figure 2. LA-2A Opto-Electrical Cell, Universal Audio](image)

The LA-2A’s ‘musical’ qualities come from the time it takes the circuit to react to changes in the audio signal. The LA-2A is considered a slow compressor as it typically acts on a time scale comparable to musical notes and phrases rather than individual wave cycles. Additionally, the optical feedback method employed by the LA-2A results in the gain of the device varying logarithmically. Since human hearing is based around a logarithmic scaling of perceived loudness, the amplitude modulation employed by the LA-2A sounds very natural and pleasing even if the amplitudes of the audio signal have been dramatically altered.

As the world of solid state electronics expanded to replace valve-based amplifiers like those used in the LA-2A, new compressor designs utilizing modern transistors began to see regular use in studios. The most notable of these types of compressor is the UREI 1176 designed by Bill Putnam. The 1176 features
the same negative feedback topology as most classic compressors but uses a MOSFET-based variable attenuator at the input stage to control the dynamic gain of the device.

Since solid state electronics such as FET’s can react to changes in applied voltage and current with extreme speed, the 1176 allowed compression to be applied to transients in an audio signal almost instantaneously with feedback times as low as 20µs. With this quick reaction time a FET-based compressor such as the 1176 can operate at a timescale equivalent to individual wave cycles in the audible frequency spectrum and recorded audio can be manipulated with such precision as to completely alter the perceived character of the signal.

Compressor Challenges

Although the use of analog compression circuits is often considered preferential over digital audio compression methods due to the availability of harmonic distortion, such distortion from analog audio circuits is generally acceptable only in very small amounts and in some cases distortion of any kind is completely undesirable. As a result, the design of analog audio equipment can quickly become a balancing act between attempting to create a circuit that is both harmonically pleasing enough to be interesting, but not so fraught with imperfections and audio ‘coloration’ that it is not useful in more than a handful of situations.

In the context of compression circuits, harmonic distortion is primarily the result of non-linear gain elements in the signal path. As the audio signal is modulated by the compressor, each amplitude level is affected by a distinct gain coefficient. When these gain coefficients follow a linear trend, entire wave cycles are reduced proportionally and the original harmonic content of the signal is preserved with no added distortion. However, when these gain coefficients do not follow a significantly linear trend, different sections of an individual wave cycle can be reduced by disproportionate amounts. As a result, the signal becomes a superposition of the original harmonic content as well as mathematically related frequencies which comprise the harmonic distortion.

For instance, consider a Sine wave audio signal: as the magnitude of the signal increases during each wave cycle, a non-linear gain element will reduce the peaks of the wave more severely than the rest of the wave cycle. The resulting signal would then be expressed as a version of the original wave superposed with a related square wave. This particular type of harmonic distortion is equivalent to adding perfect-5th or perfect-8th versions of the original frequency to the overall signal.
While this harmony can add a sense of excitement and interest to an audio signal when used subtly, any more than 0.5-1.0% harmonic distortion can be very distracting and in many instances the audio signal would be considered ruined. In order to keep harmonic distortion as low as possible without completely eliminating it, careful consideration must be paid to the compression circuit to ensure that all elements in the signal path will operate within a known linear response region with only slight deviations.

**Development**

**Initial Designs**

In order to understand the fundamental principles of building a FET compression circuit without the complex feature sets that frequently obscure the design intentions of commercially available compressors, it was necessary to start investigating as basic a compression circuit as possible. Sourced from an AGC project tutorial at Electro Schematics, the following circuit diagram seen in figure 5 represents the most reduced form of a FET based compressor. In this design, the P-Channel JFET device forms a variable attenuator at the input of the circuit. As the amplitude of the audio signal at the output exceeds the threshold of the BJT detector the BJT allows a charge-up capacitor—which would normally hold the JFET gate at a high voltage—to discharge to ground. As the voltage applied to the gate of the P-Channel JFET is reduced, the impedance across the JFET is also reduced. As a result, the bottom half of the voltage-divider formed by the JFET and the input resistor is reduced and the attenuation provided by the voltage-divider is increased. As the output amplitude of the signal falls back down below the threshold for the BJT, the capacitor attached to the JFET gate is charged back up so that JFET is returned to its original high impedance and the voltage-divider returns to its nominal attenuation.
At this point it should be noted that the use of a JFET in this design is a departure from the traditional inclusion of a MOSFET in compressors such as the 1176. This is due to the JFET’s significantly later introduction in the 1970’s when the 1176 had already been optimized for use with MOSFET components available in the 1960’s. In this case, MOSFET components in the 1176 can be considered in part as legacy inclusions. While MOSFET’s display a similar gate-voltage-impedance relationship and can still be used to great effect, they are more susceptible to interference in their impedance behavior due to current considerations as well as the applied voltage. Conversely, JFET’s can be controlled exclusively using the voltage applied to the gate terminal with little consideration for interference in impedance behavior due to other factors. For this reason, JFET’s are significantly easier to understand and incorporate in a compression circuit.

Although this particular project design teaches the significant advantages of JFET gain elements—which were chosen for the eventual FET Compressor design—it does little to tackle the problems of harmonic distortion as a result of JFET non-linearity. Furthermore, by disconnecting the JFET control voltage from the audio signal itself, this compression circuit fails to deliver compression behavior which is consistently related to the audio content. Instead, the amount of gain reduction applied to the signal in this design is dependent on the overall **amount of time the signal amplitude spends in excess of the BJT threshold voltage** rather than the **extent to which the signal amplitude exceeds** the threshold. While it is logical that signal peaks with a higher amplitude will typically exceed the BJT threshold for a longer time, this design does not consider any case in which a limited signal excursion lasts longer than the discharge.
time of the relevant capacitor. For these reasons, this design was determined to be insufficient as a basis for the purposes of the FET Compressor.

Core Compression Module

In order to retain the best parts of legacy compressors while including modern components, the core module of the FET Compressor was created by designing a modernized version of the 1176 high-level topology using op amps and JFET’s rather than the custom amplifiers and MOSFET’s of the original design.

Control Signal Detector

The 1176 feedback loop features a classic variable-attenuation voltage divider at its input stage using the FET gain element, but rather than controlling the impedance of the FET using an unrelated bias voltage—as per the previously investigated design—the 1176 routes a highly specialized, rectified version of the audio signal to the gate terminal of the FET.

As per the original 1176 topology, the FET Compressor is designed around a voltage-divider at the input stage formed by the input resistor and a JFET element. The voltage-divider provides variable attenuation to the audio signal by altering the impedance of the JFET element according to the amplitude of the audio signal. The JFET must be controlled by a rectified version of the audio signal—for this purpose a full-wave rectifier or Absolute Value (ABS) detector was chosen. The ABS detector is connected to the output of the compressor, where it takes readings of the current AC signal magnitude and converts this to a corresponding DC value. Many equivalent ABS circuits are commonly available although different designs vary in performance. It is best to choose a rectifying circuit.
which will not allow any AC components of the audio signal to pass to the FET element. The ABS detector utilized in the FET Compressor was based on a simulated design available through Analog Devices’ LTSpice program, as seen in figure 7.

**Time Constant Control**

From this point in the FET Compressor circuit, the rectified control signal must be passed through a time-constant control section. This sub-circuit allows the crucial ‘attack’ and ‘release’ times of the compressor to be adjusted by the user depending on the type of audio signal being processed. The FET Compressor utilizes the most basic form of the time-constant control section, as described in figure 8 from Cranesong.

This section operates by introducing a delay between the input and output of the sub-circuit. As the DC control signal from the ABS detector rises, the time-constant resulting from R1 and C1 determines how long it will take for C1 to charge up to and pass along the DC control signal. As the signal from the ABS detector falls, the time-constant resulting from R2 and C1 determines how long it will take C1 to discharge and follow the DC control signal back to nominal levels. In this example, D1 ensures that no negative DC leaks from the ABS detector will pass into the time-constant control.

**JFET Variable Attenuator**

Although using a JFET as a Voltage-Controlled-Resistor (VCR) is a well known solution to AGC implementation in many compressor designs, significant attention must still be paid to the particular placement and operation of the JFET with regard to the reduction of harmonic distortion. JFET devices have an accessible linear response region known as the Ohmic Region, however the JFET will only stay in this region for a particular range of terminal parameters. As the input signal applied to the JFET increases, the JFET begins to exit the Ohmic Region and the device’s response will become increasingly non-linear. This will result in added harmonic distortion and even hard clipping if the input signal is so large as to push the JFET into Saturation. It is possible to keep a JFET operating within the Ohmic Region simply by reducing the input signal to stay within a certain range—for most JFET devices this is around 0.05V to 0.1V—but this solution presents a new set of problems by severely limiting the available signal headroom of the design and bringing the audio signal dangerously close to the noise floor of the circuit.
An alternative option to keep a JFET operating largely within the Ohmic Region is to apply high-impedance resistors around the JFET’s drain and gate terminals, as seen in figure 9.

![Figure 9. JFET Linearization, Vishay](image)

These resistors allow a portion of the input signal to be applied to the JFET gate in order to reduce the potential difference across the device. As a result, signals with a much larger amplitude can be applied to the JFET with little to no added harmonic distortion. As seen in figure 10, the response of the JFET with high impedance resistors added can be linear for a much larger signal range.

![Figure 10. JFET Linear Response, Vishay](image)

This method of reducing the harmonic distortion from a JFET is instrumental in utilizing the full signal handling capability of any JFET device and it is paramount to the design of low-noise, low-distortion audio equipment design.

**PCB Design**

Following extensive breadboard-based testing of the core compression module for the **FET Compressor** design, a two-channel stereo version of the **FET Compressor** was able to be produced using Printed
Circuit Board (PCB) prototyping. As seen in figure 11, the layout of the **FET Compressor** required an unusually large PCB. This is due to the intended standard 1U rackmount enclosure of the **FET Compressor**. By increasing the size of the PCB layout to match the size of this standard enclosure, manufacturing complexity of the **FET Compressor** is significantly reduced. The large PCB size allows for direct mounting of the PCB onto the enclosure by the peripheral components of the circuit. This design choice eliminates the need for any electrical intervention when assembling the **FET Compressor**. The entire circuitry for the compressor can be mounted inside the enclosure without any jumper cables or additional connections, leaving calibration and final testing of each unit as the only remaining tasks following basic assembly. This design choice results in significantly more costly PCB prototypes but has very little impact on the overall cost of a production-run unit and increases both the speed and reliability of manufacturing.

![Figure 11. FET Compressor PCB Layout, Ricky Mannion 2018](image)

Furthermore, sourcing the optimal components for the PCB incarnation of the **FET Compressor** proved to be very challenging but necessary. Due to their reliable performance and well regarded audio signal handling, all capacitors for the **FET Compressor** were selected to be film-based materials where possible. Since film capacitors cannot be manufactured above moderate capacitance values, all capacitors up to 1.5\(\mu\)F were able to be sourced from film materials while all higher value capacitors were chosen from premium-grade, audio-optimized electrolytic capacitor manufacturers. Electrolytic capacitors are typically avoided in audio equipment design wherever possible due to the unpleasant, high-frequency interference they can cause in audio signals. For this reason, large capacitance values were removed from the **FET Compressor** design wherever reasonable and all remaining capacitance in the design were satisfied with high quality electrolytic capacitors to mitigate the negative impact of these components on the audio signal quality as much as possible.

In order to reduce noise and provide industry standard connections expected by professional producers and audio engineers, all inputs and outputs to the **FET Compressor** were fitted with balanced line
receivers and drivers. These specialized differential amplifiers allow the **FET Compressor** to display the same robust and harmonically rich audio content as classic transformer-based audio equipment at a much lower cost and with more reasonable design constraints.

Finally, all user-accessible potentiometer and toggle switch controls were sourced from premium-grade, audio-specific manufacturers. Although it may seem trivial in prototyping, the construction and tactile feel of any piece of analog audio equipment plays a significant role in the overall success with or acceptance of the device by the professional audio community. Investing in quality components is critical to producing a prototype that not only sounds great but feels great.

Sound samples included with this report demonstrate the superior audio quality of a carefully designed and implemented piece of analog audio equipment such as this **FET Compressor**.

The prototype of the **FET Compressor** resides in ECEB 2076.
Figure 13. FET Compressor Undergoes Edits, Ricky Mannion 2018
Future Work

PCB Revision

Although the first FET Compressor PCB prototype has been very successful, there is still much work to do to achieve a stable prototype version. Following complete testing and characterization of the initial PCB prototype during the Summer of 2018, future FET Compressor PCB revisions will incorporate fixes to all significant bugs, adjustments to the feature parameters, and improved layout and routing based around the lessons learned during the creation of the first PCB version.

Enclosure Design

Also planned for the Summer of 2018, the FET Compressor revisions will be accompanied by a custom rackmount enclosure. The FET Compressor PCB’s have been and will continue to be designed for use in a standard 1U size rackmount enclosure. The custom enclosure is planned to be developed in conjunction with the ECEB in-house machine shop so that structure of the enclosure may be tailor-made to enhance the user experience with the FET Compressor.
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