The International Journal Of Engineering And Science (IJES)
ISSN (e): 2319 – 1813 ISSN (p): 2319 – 1805
Pages || 49-65 || March - 2015 ||

Improved low-distrortion three state pulse width modulator for a digital Class-d amplifier

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ABSTRACT

In this project, we investigate the use of indirect approach in Pulse Width Modulator for the design of low-power low-distortion digital Class D amplifier, employing Three State Delta-Sigma Modulation. The amplifier that was built comprised of a three-state Delta-Sigma modulator, an H-bridge amplifier, and a passive filter and was capable of accepting both digital and analog audio inputs with less harmonic distortion (THD) and greater than 100 dB zero-input signal to noise (SNR) ratio. The goal of this project was to create an 80W, 95% efficient Class D audio amplifier with less than 0.5% harmonic distortion and greater than 100 dB zero-input signals to noise ratio that accepts digital inputs. The amplifier that was built comprised of a three-state digital modulator, an H-bridge amplifier, and a passive filter and was capable of accepting both digital and analog audio inputs by means of the SPDIF protocol and an ADC. To allow the modulator design to be quickly altered, it was implemented on a DSP. Because the modulator could be easily changed, several different modulation schemes were simulated, designed and tested in order to achieve optimal audio quality and efficiency results.

KEYWORDS - *Three state delta sigma modulation, H-bridge amplifier, Passive filter, SPDIF Protocol.*

I. INTRODUCTION

Class-D amplifiers have rapidly supplanted linear amplifiers in recent years, both for low-power and high-power devices. Their adoption has been motivated by their high efficiency. Much of the recent research effort on PWM is focused on reducing the distortion introduced by digital implementations of PWM modulators. Audio amplifiers have long been plagued by trade-offs between size, efficiency, and performance. Traditionally, high performance audio amplifiers have come with large footprints to make room for their heavy heat sinks. While efficient, low heat amplifiers have been relegated to portable devices. The reason for this is that in portable devices, the desire for clarity is pushed a side by the need to retain a small package. There are several reasons why higher efficiency amplifiers are, and will continue to be in demand. One reason is the increasing power consumption of home entertainment systems. This is due to increasing screen size, the increasing number of multimedia accessories and the increasing cost of energy. However, for many applications the efficiency of an amplifier alone is not enough to make it competitive. For an audio application, the amplifier must also be able to deliver audio clarity comparable to less efficient competitors. As more multimedia accessories are added, power usage can become a concern, second to the concerns of audio quality and a large number of output channels. At one point stereo outputs were considered luxury, but now it is common to see audio sources like DVD players and game consoles outputting 6 or 8 channels of audio. These devices rely primarily on digital audio standards, where several channels can be transmitted over a single cable.

Class D amplifiers have higher efficiencies when compared to traditional linear amplifiers because of their switching nature. In Class D operation, an incoming signal is converted to a digital signal, amplified, and then filtered.

Amplification of a digital signal is an efficient process because the transistors used are not operated in their wasteful triode region. The efficiency of these amplifiers implies that their heat dissipation is low; heat sinks can be eliminated from the power devices, reducing overall system size and cost. Most Class D amplifiers are designed with pace, heat, and power consumption in mind at the expense of audio quality. Audio quality does not have to suffer if proper care is given in the creation of the amplifier. High audio quality can be achieved by creating an accurate digital representation of the input signal, and also by combating the major sources of audio distortion through the use of closed loop full system feedback.

II. PROPOSED METHOD

The amplifier that was built comprised of a three-state Delta-Sigma modulator, an H-bridge amplifier, and a passive filter and was capable of accepting both digital and analog audio inputs with less harmonic distortion (THD) and greater than 100 dB zero-input signal to noise (SNR) ratio.

III. SYSTEM FUNCTION

3.1 CLASS D AMPLIFIER: A CONCEPTUAL DESCRIPTION

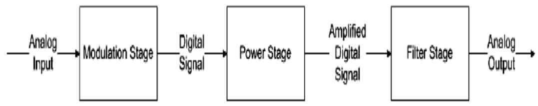

Fig: 3.1 functional block diagram of the system

The first stage is the modulation stage. The input signal must be converted to a digital signal before being amplified. To accomplish this here Delta-Sigma Modulation is used. After the signal is modulated, it must be amplified. The amplification stage uses several Metal Oxide Semiconductor Field Effect Transistors (MOSFETs).A H-Bridge configuration of MOSFETs are employed to form the amplification stage. After the modulated signal is amplified, it must be filtered before it can be sent to the output stage. The last stage is the filtering, or demodulation, stage, which consists of a low pass filter.

Class D-Amplifier: Audio Amplifier

The goal of audio amplifiers is to reproduce input audio signals at sound-producing output elements, with desired volume and power levels—faithfully, efficiently, and at low distortion. Audio frequencies range from about 20 Hz to 20 kHz, so the amplifier must have good frequency response over this range (less when driving a band-limited speaker, such as a *woofer* or a *tweeter*). Power capabilities vary widely depending on the application, from milliwatts in headphones, to a few watts in TV or PC audio, to tens of watts for “mini” home stereos and automotive audio, to hundreds of watts and beyond for more powerful home and Commercial sound systems and to fill theaters or auditoriums with sound. A straightforward analog implementation of an audio amplifier uses transistors in linear mode to create an output voltage that is a scaled copy of the input voltage. The forward voltage gain is usually high (at least 40 dB). If the forward gain is part of a feedback loop, the overall *loop gain* will also be high. Feedback is often used because high loop gain improves performance suppressing distortion caused by nonlinearities in the forward path and reducing power supply noise by increasing the power-supply rejection (PSR).

The Class D Amplifier Advantage

In a conventional transistor amplifier, the *output stage* contains transistors that supply the instantaneous continuous output current. The many possible implementations for audio systems include Classes A, AB, and B. Compared with *Class D* designs; the output-stage power dissipation is large in even the most efficient *linear* output stages. This difference gives Class D significant advantages in many applications because the lower power dissipation produces less heat, saves circuit board space and cost, and extends battery life in portable systems. Linear Amplifiers, Class D Amplifiers, and Power Dissipation Linear-amplifier output stages are directly connected to the speaker (in some cases via capacitors). If bipolar junction transistors (BJTs) are used in the output stage, they generally operate in the linear mode, with large collector-emitter voltages. The output stage could also be implemented with MOS transistors, as shown in Figure 3.2.

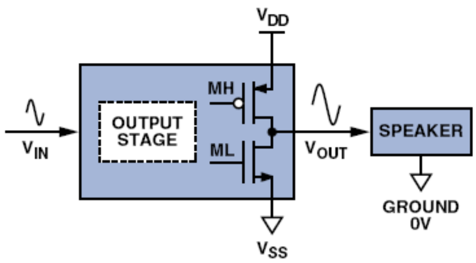

Fig: 3.2 CMOS linear output stage.

Power is dissipated in all linear output stages, because the process of generating *VOUT* unavoidably causes nonzero I_{DS} and V_{DS} in at least one output transistor. The amount of power dissipation strongly depends on the method used to bias the output transistors.

The *Class A* topology uses one of the transistors as a dc current source, capable of supplying the maximum audio current required by the speaker. Good sound quality is possible with the Class A output stage, but power dissipation is excessive because a large dc bias current usually flows in the output-stage *transistors* (where we do not want it), without being delivered to the *speaker* (where we do want it). The *Class B* topology eliminates the dc bias current and dissipates significantly less power. Its output transistors are individually controlled in a push-pull manner, allowing the MH device to supply positive currents to the speaker, and ML to sink negative currents. This reduces output stage power dissipation, with only signal current conducted through the transistors. The Class B circuit has inferior sound quality, however, due to nonlinear behavior (*crossover distortion*) when the output current passes through 0 and the transistors are changing between the on and off conditions. *Class AB*, a hybrid compromise of Classes A and B, uses some dc bias current, but much less than a pure Class A design. The small dc bias current is sufficient to prevent crossover distortion, enabling good sound quality. Power dissipation, although between Class A and Class B limits, is typically closer to Class B. Some control, similar to that of the Class B circuit, is needed to allow the Class AB circuit to supply or sink large output currents. Unfortunately, even a well-designed class AB amplifier has significant power dissipation, because its midrange output voltages are generally far from either the positive or negative supply rails. The large drain-source voltage drops thus produce significant I_{DS} X V_{DS} instantaneous power dissipation.

Thanks to a different topology (Figure 3.3), the *Class D* amplifier dissipates much less power than any of the above. Its output stage switches between the positive and negative power supplies so as to produce a train of voltage pulses. This waveform is benign for power dissipation, because the output transistors have zero current when not switching, and have low *VDS* when they are conducting current, thus giving smaller *IDS* X *VDS*.

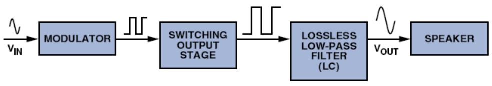

Fig: 3.3. Class D open-loop-amplifier block diagram.

Since most audio signals are not pulse trains, a modulator must be included to convert the audio input into pulses. The frequency content of the pulses includes both the desired audio signal and significant high-frequency energy related to the modulation process. A low-pass filter is often inserted between the output stage and the speaker to minimize electromagnetic interference (EMI) and avoid driving the speaker with too much high frequency energy. The filter (Figure 3.4) needs to be lossless (or nearly so) in order to retain the power-dissipation advantage of the switching output stage. The filter normally uses capacitors and inductors, with the only intentionally dissipative element being the speaker.

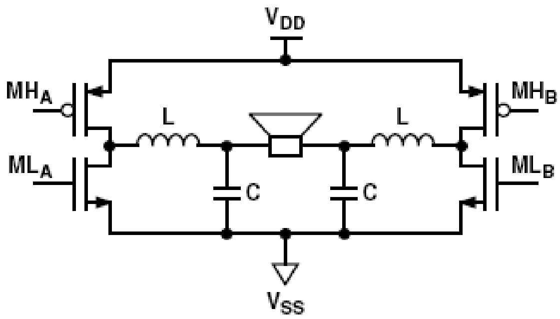

Fig: 3.4 Differential switching output stage and LC low-pass filter.

Figure 3.5 compares ideal output-stage power dissipation (*PDISS*) for Class A and Class B amplifiers with measured dissipation for the AD1994 Class D amplifier, plotted against power delivered to the speaker (*PLOAD*), given an audio-frequency sine wave signal.

The power numbers are normalized to the power level, *PLOAD max*, at which the sine is clipped enough to cause 10% total harmonic distortion (THD). The vertical line indicates the *PLOAD* at which clipping begins. Significant differences in power dissipation are visible for a wide range of loads, especially at high and moderate values. At the onset of clipping, dissipation in the Class D output stage is about 2.5 times less than Class B, and 27 times less than Class A. Note that more power is consumed in the Class A output stage than is delivered to the speaker—a consequence of using the large dc bias current. Output-stage power efficiency, *Eff*, is defined as

$$Eff = \frac{P_{LOAD}}{P_{LOAD} + P_{DIS}}$$

At the onset of clipping, *Eff* = 25% for the Class A amplifier, 78.5% for the Class B amplifier, and 90% for the Class D amplifier (see Figure 5). These best-case values for Class A and Class B are the ones often cited in textbooks.

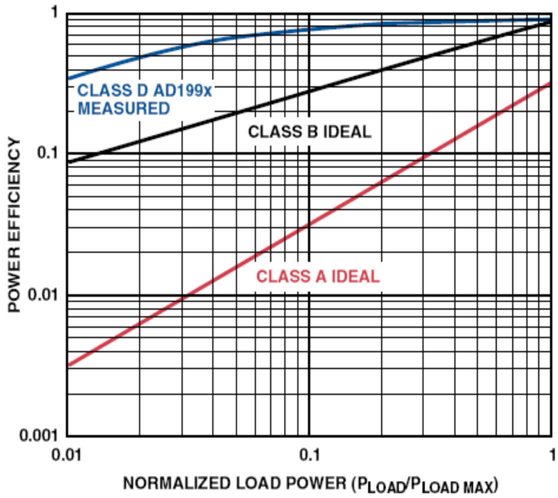

Fig: 3.6. Power efficiency of Class A, Class B, and Class D output stages.

The differences in power dissipation and efficiency widen at moderate power levels. This is important for audio, because long term average levels for loud music are much lower (by factors of five to 20, depending on the type of music) than the instantaneous peak levels, which approach *PLOAD max*. Thus, for audio amplifiers, [*PLOAD* = 0.1 3 *PLOAD max*] is a reasonable average power level at which to evaluate *PDISS*. At this level, the Class D output-stage dissipation is nine times less than Class B, and 107 times less than Class A.

For an audio amplifier with 10-W *PLOAD max*, an average *PLOAD* of 1 W can be considered a realistic listening level. Under this condition, 282 mW is dissipated inside the Class D output stage, vs. 2.53 W for Class B and *30.2 W* for Class A. In this case, the Class D efficiency is reduced to 78%—from 90% at higher power. But even 78% is much better than the Class B and Class A efficiencies—28% and 3%, respectively. These differences have important consequences for system design. For power levels above 1 W, the excessive dissipation of linear output stages requires significant cooling measures to avoid unacceptable heating—typically by using large slabs of metal as heat sinks, or fans to blow air over the amplifier. If the amplifier is implemented as an integrated circuit, a bulky and expensive thermally enhanced package may be needed to facilitate heat transfer. These considerations are onerous in consumer products such as flat-screen TVs, where space is at a premium—or automotive audio, where the trend is toward cramming higher channel counts into a fixed pace. For power levels below 1 W, wasted power can be more of a difficulty than heat generation. If powered from a battery, a linear output stage would drain battery charge faster than a Class D design. In the above example, the Class D output stage consumes 2.8 times less supply current than Class B and 23.6 times less than Class A—resulting in a big difference in the life of batteries used in products like cell phones, PDAs, and MP3 players.

For simplicity, the analysis thus far has focused exclusively on the amplifier *output* stages. However, when all sources of power dissipation in the amplifier system are considered, linear amplifiers can compare more favorably to Class D amplifiers at low output-power levels. The reason is that the power needed to generate and modulate the switching waveform can be significant at low levels. Thus, the system-wide quiescent dissipation of well-designed low-to-moderate-power Class AB amplifiers can make them competitive with Class D amplifiers. Class D power dissipation is unquestionably superior for the higher output power ranges, though. Class D Amplifier Terminology and Differential vs. Single-Ended Versions Figure 3.4 shows a differential implementation of the output transistors and LC filter in a Class D amplifier.

The *H-bridge* has two *half-bridge* switching circuits that supply pulses of opposite polarity to the filter, which comprises two inductors, two capacitors, and the speaker. Each half-bridge contains two output transistors—a high-side transistor (MH) connected to the positive power supply, and a low-side transistor (ML) connected to the negative supply. The diagrams here show high-side *p*MOS transistors. High-side *n*MOS transistors are often used to reduce size and capacitance, but special gate-drive techniques are required to control them.

Full H-bridge circuits generally run from a single supply (VDD), with ground used for the negative supply terminal (VSS). For a given VDD and VSS, the differential nature of the bridge means that it can deliver twice the output signal and four times the output power of single-ended implementations. Half-bridge circuits can be powered from bipolar power supplies or a single supply, but the single-supply version imposes a potentially harmful dc bias voltage, VDD/2, across the speaker, unless a blocking capacitor is added. The power supply voltage buses of half-bridge circuits can be "pumped" beyond their nominal values by large inductor currents from the LC filter. The dV/dt of the pumping transient can be limited by adding large decoupling capacitors between VDD and VSS. Full-bridge circuits do not suffer from bus pumping, because inductor current flowing into one of the half-bridges flows out of the other one, creating a local current loop that minimally disturbs the power supplies. Factors in Audio Class D Amplifier Design The lower power dissipation provide a strong motivation to use Class D for audio applications, but there are important challenges for the designer.

Choice of Output Transistor Size The output transistor size is chosen to optimize power dissipation over a wide range of signal conditions. Ensuring that *VDS* stays small when conducting large *IDS* requires the on resistance (*RON*) of the output transistors to be small (typically 0.1 V to 0.2 V). But this requires large transistors with significant gate capacitance (*CG*). The gate-drive circuitry that switches the capacitance consumes power—*CV 2f*, where *C* is the capacitance, *V* is the voltage change during charging, and *f* is the switching frequency. This "switching loss" becomes excessive if the capacitance or frequency is too high, so practical upper limits exist.

The choice of transistor size is therefore a trade-off between minimizing I_{DS} x V_{DS} losses during conduction vs. minimizing *switching* losses. Conductive losses will dominate power dissipation and efficiency at high output power levels, while dissipation is dominated by switching losses at low output levels. Power transistor manufacturers try to minimize the *RON* x *CG* product of their devices to reduce overall power dissipation in switching applications, and to provide flexibility in the choice of switching frequency. Protecting the Output Stage The output stage must be protected from a number of potentially hazardous conditions:

Sound Quality:

Several issues must be addressed to achieve good overall sound quality in Class D amplifiers.***Clicks and pops***, which occur when the amplifier is turning on or off can be very annoying. Unfortunately, however, they are easy to introduce into a Class D amplifier unless careful attention is paid to modulator state, output-stage timing, and LC filter state when the amplifier is muted or unmuted.

Inductor Design Factors:

Important factors in designing or selecting the inductor include the core's current rating and shape, and the winding resistance. *Current rating*: The core that is chosen should have a current rating above the highest expected amplifier current. The reason is that many inductor cores will magnetically saturate if current exceeds the current-rating threshold and flux density becomes too high—resulting in unwanted drastic reduction of inductance. The inductance is formed by wrapping a wire around the core. If there are many turns, the resistance associated with the total wire length is significant. Since this resistance is in series between the half-bridge and the speaker, some of the output power will be dissipated in it. If the resistance is too high, use thicker wire or change the core to a different material that requires fewer turns of wire to give the desired inductance.

Finally, it should not be forgotten that the form of inductor used can affect EMI, as noted above. System Cost What are the important factors in the overall cost of an audio system that uses Class D amplifiers? How can we minimize the cost? The *active* components of the Class D amplifier are the switching output stage and modulator. This circuitry can be built for roughly the same cost as an analog linear amplifier. The real trade-offs occur when considering other components of the system. The lower dissipation of Class D saves the cost (and space) of cooling apparatus like heat sinks or fans. A Class D integrated circuit amplifier may be able to use a smaller and cheaper package than is possible for the linear one. When driven from a digital audio source, analog linear amplifiers require D/A converters (DACs) to convert the audio into analog form. This is also true for analog-input Class D amplifiers, but digital-input types effectively integrate the DAC function. On the other hand, the principal cost disadvantage of Class D is the LC filter. The components—especially the inductors—occupy board space and add expense. In high-power amplifiers, the overall system cost is still competitive, because LC filter cost is offset by large savings in cooling apparatus. But in cost-sensitive, low-power applications, the inductor expense becomes onerous. In extreme cases, such as cheap amplifiers for cell phones, an amplifier IC can be cheaper than the total LC filter cost. Also, even if the monetary cost is ignored, the board space occupied by the LC filter can be an issue in small form-factor applications.

3.2: SIGMA–DELTA MODULATION:

Sigma–delta modulation is the most popular form of analog-to-digital conversion used in audio applications. It is also commonly used in D/A converters, sample-rate converters, and digital power amplifiers. In this tutorial the theory behind the operation of sigma–delta modulation is introduced and explained. We explain how performance is assessed and resolve some discrepancies between theoretical and experimental results. We discuss the issues of usage, such as limit cycles, idle tones, harmonic distortion, noise modulation, dead zones, and stability. We characterize the current state of knowledge concerning these issues and look at what are the most significant problems that still need to be resolved. Finally, practical examples are given to illustrate the concepts presented.

3.2.1 INTRODUCTION:

Sigma–delta modulation (SDM) is perhaps best understood by comparison with traditional pulse-code modulation (PCM). A PCM converter typically samples an input signal at the Nyquist frequency and produces an *N*-bit representation of the original signal. This technique, however, requires quantization to 2*N* levels. Whether implemented using successive approximation registers, pipelined converters, or other techniques, high resolution is difficult to obtain in PCM conversion due to the need to accurately represent many quantization levels and the subsequent circuit complexity. This is the motivation for sigma–delta modulation, a form of pulse-density modulation, which exploits oversampling and sophisticated filter design in order to employ a low-bit quantizer with high effective resolution.

In this system, we will consider common designs of sigma–delta modulators as used for analog-to-digital conversion. The basic principle is the same for SDMs employed in D/A or sample-rate conversion. We will restrict the analysis to asynchronous, discrete-time designs. However, these are by far the most common designs and include most feed forward, feedback, and multistage implementations. We will explain the theory of operation, emphasizing signal-to-noise ratio estimation and comparison with PCM conversion.

We will also introduce the linear model, which assists in understanding filter design and noise shaping principles. Since sigma–delta modulation is highly nonlinear, there are various phenomena that cannot be explained using this technique, such as instability and limit cycles. The literature on these phenomena can be confusing, so we attempt to give a clear definition of the terms and clarify the current state of understanding. Finally, we introduce several state-of the- art techniques that can be used to deal with these unwanted phenomena.

3.2.2 THE LINEAR MODEL AND PULSE-CODE MODULATION:

The allowed values in the output signal, after quantization, are called quantization levels, whereas the distance between two successive levels, is called the quantization step size, *q*. For a quantizer with *b* bits covering the range from +1 to -1, there are 2*b* quantization levels, and the width of each quantization step is q=2 / (2b -1). This is depicted in Fig. 3.9 for a 3-bit quantizer.

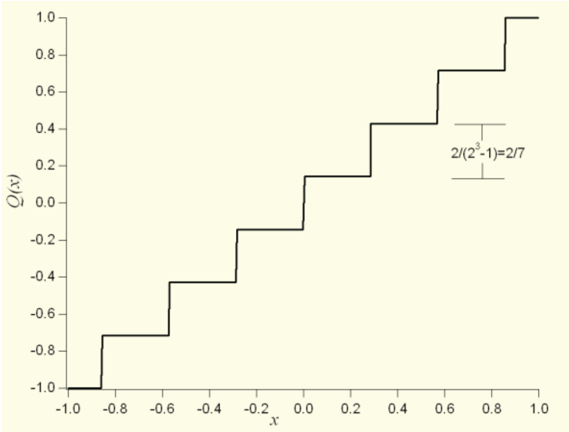

Fig: 3.9. Transfer characteristics for a 3-bit quantizer and V=1

The rounding, or midriser, quantizer assigns each input sample $x(n)$ to the nearest quantization level. The quantization error is simply the difference between the input and output to the quantizer, $eq=Q(x)-x$. It can easily be seen that the quantization error $eq(n)$ is always bounded by $-q \leq e\ n \leq q/2$. Since quantization is a highly nonlinear process, the exact effect of quantization on the signal content and the nature of quantization noise may be difficult to measure. For this reason, several assumptions are often made.

1. The error sequence, $eq(n)$, is a stationary, random process.
2. The error sequence is uncorrelated with itself and with the input sequence $x(n)$.
3. The probability-density function of the error is uniform over the range of quantization error.

Delta Sigma modulation uses a technique known as pulse density modulation to encode high resolution data into a low resolution bit stream (the modulated signal). Pulse density modulation represents the amplitude of an analog signal with the relative density of the output pulses. In a continuous time representation of a delta sigma modulator, an input signal is first integrated. At the sampling frequency, Fs, the integration is compared to a predetermined threshold, which either flips or resets the output. Because the output can only be changed at interval 1/Fs the width of the digital pulses are always the same. Even in the most basic Delta Sigma modulators, closed loop feedback is used to monitor the accuracy of the output. The continuous and discrete time models of a first order Delta Sigma modulator are depicted below in Figure 3.10.

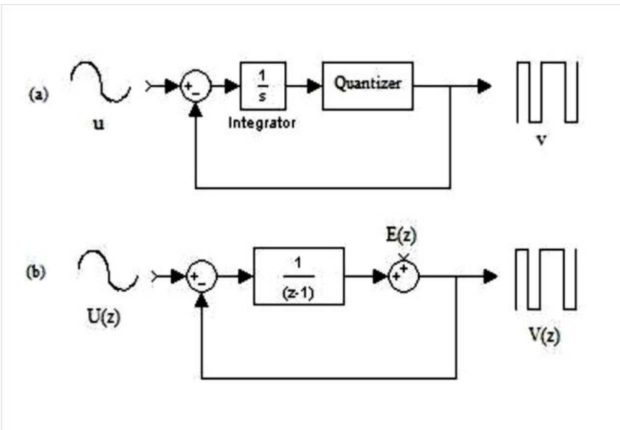

Fig: 3.10 - (a) Continuous Time $\Delta\sum$ and (b) Linear Z-Domain Model.

When higher order delta sigma modulators are built, many scale factors can be added in order to manipulate the modulator's performance. A Sample modulator structure, called Cascade of integrators, feedback form (CIFB), is shown in Figure 5. In this diagram, z is the discrete time variable z, n is the order of the modulator, and all a's, b's, c's, and g's are constant gains. The "integrators" seen in this model are actually integrators with inherent delays. Basically, each 1/(z-1) block contains a discrete time integrator preceded by a one unit delay. This form was one of the ones used in the process of designing our final modulator.

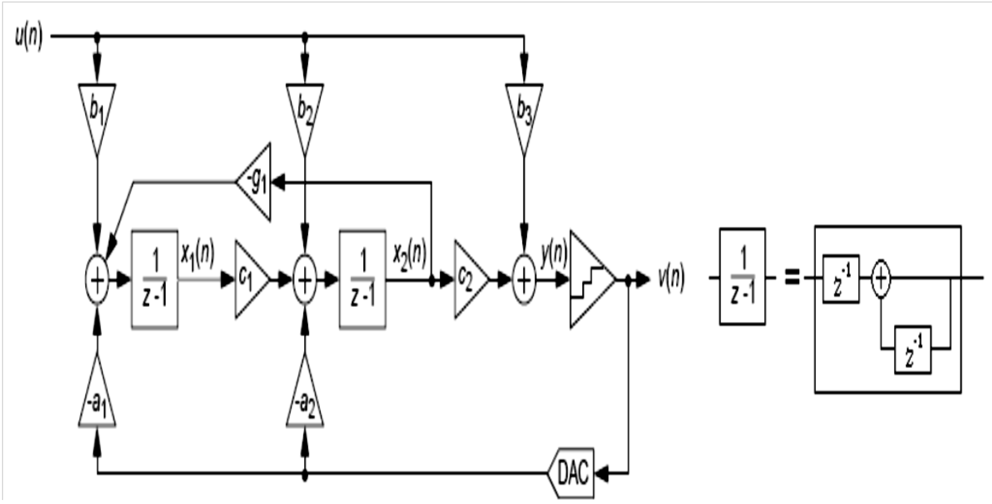

Fig: 3.11 - A Second Order CIFB Modulator Schematic

Delta sigma modulation modeling is made easier by replacing the non-linear quantizer with an additive error signal E(z). This error signal is the difference between the integrated signal and the modulated signal. From this model we can obtain the formula V(z) = [z-1 • U(z)] + [(1-z-1) • E(z)]. This equation can be written in terms of a noise transfer function (NTF) and a signal transfer function (STF). The STF of the modulator defines the frequency region which the modulator will pass from input to output. A modulator used for audio will have an STF of 1 or it will be a low pass filter surrounding the audio band. The NTF defines the region that quantization noise will exist. For an audio modulator the NTF will be a high pass filter. An effective modulator will separate the NTF and STF as far as possible to ensure quantization noise does not over run the signal band. A plot of the NTF of a first order modulator is depicted in Figure 3.12.

If the NTF and STF of a modulator are known, the modulator can be turned into a single loop filter with two filters G and H, as shown below in Figure 3.13. In this diagram, G = STF/NTF And H = 1-NTF/STF which can be confirmed by solving the output in terms of the input and the error. Both G and H filters are the same order as the order of the delta sigma modulator that they produce.

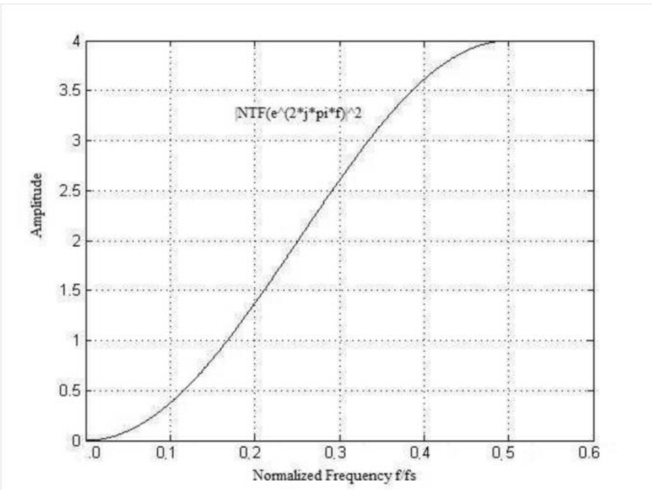

Fig: 3.12 - The Noise Transfer Function of a Delta Sigma Modulator

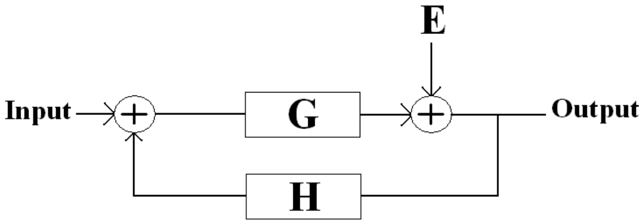

Fig: 3.13 - Loop Filter

Two parameters govern the effectiveness of a Delta Sigma modulator; those are the OSR (over sampling ratio) and the order of delta sigma modulator. In continuous time models, the order of the modulator is directly related to the number of integrators in the signal path. In a discrete time modulator, the order of the system is the order of the NTF. A higher order NTF increased the slope of the cutoff of the filter, decreasing the noise in the signal band. The over-sampling ratio (OSR) is OSR = FS/2*Fb , where Fb is the bandwidth of the signal and Fs is the sampling frequency of the modulator. Increasing the OSR shifts the NTF out of the signal band. From figure 3.13 we see in a discrete time system that the noise filter is a function of the normalized frequency.

When the OSR is high the signal band moves lower in the normalized frequency. The expected SNR of these systems can be estimated from the order of the system and the OSR (assuming the OSR >> 1, typically 22 to 210). In order to achieve a specific SNR, there is a trade-off between the order of the modulator and the OSR needed. Figure 8 below shows the trade-off between OSR and maximum SNR for a 2-bit Nth order delta sigma modulator. In this figure, SNR is shown as SQNR or Signal to Quantization Noise Ratio. For our purposes, these terms are synonymous.

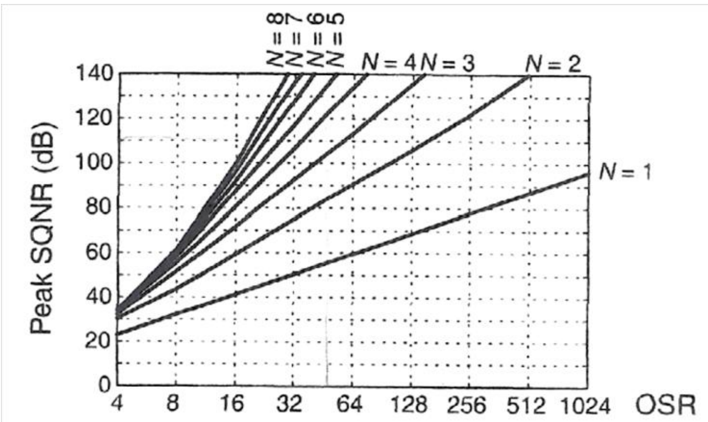

Fig: 3.14 - OSR vs. Maximum SQNR for a 2-bit Delta Sigma Modulator

As stated earlier, delta sigma modulation can offer superior SNR because of its internal feedback and its ability to shape the noise out of the signal band. This fact makes the delta sigma modulator a good choice for an amplifier where high audio quality is required. Since the quality of the delta sigma modulator is dependent on both OSR and the order of the system being high, and a feedback loop required for the system, the stability of the modulator must be considered. Increasing the order of the modulator increases the phase delay of the system, which could potentially destabilize the modulator for some or all input amplitudes.

3.2.3 Signal to Noise Ratio

Signal to Noise Ratio is defined as shown in Equation 1. This means that the SNR of the system can only be improved by increasing the power of the desired signal or by reducing the power of the noise.

$$SNR = 10log10\left(\frac{signal\ power}{noise\ power}\right)$$

Equation 1 - Definition of SNR

In the previous sections we introduced the NTF of a modulator. We explained that increasing the order of the modulator increases the order of the NTF. Increasing the modulation frequency moves the NTF further out of the signal band. Moving the NTF out of the signal band increases the SNR of the system. An alternate way to categorize the modulator's SNR is by using its effective number of bits (ENOB). As mentioned in previous sections, the number of bits used in an analog to digital converter is directly related to its SNR. With a delta sigma modulator we are limited to a low number of output states to remain compatible with a Class D power stage. In a delta sigma modulator we can compensate for less output bits with a higher modulation frequency. By keeping the modulation frequency high we can maintain a high ENOB and therefore maintain a high SNR even with a small number of output bits.

3.3 FEEDBACK SYSTEM STABILITY:

Having full system feedback in a Class D amplifier system has the potential to greatly decrease both noise and distortion. Like in any other feedback system, the output can be monitored and checked against the input in order to eliminate noise or distortion that is added by the system itself. One of the main challenges involved in building a system with feedback is making sure it is stable for all expected inputs. According to Dorf (1989), Feedback systems are considered stable if for any bounded input, they produce a bounded output. One way to determine system stability is by referencing the phase margin of a system. There are also many circuits which can improve the stability of a system, three of which are named phase-lead, phase-lag, and lead-lag networks.

3.3.1 Phase Margin:

Phase margin is a widely used measure of system stability when working in the frequency domain. Phase margin is defined as the distance of the system transfer function’s phase shift from 180º when at the system’s zero dB point or the frequency at which the system’s amplification factor is 1. The phase margin's positive distance from zero can determine the stability of a system. This makes sense, because a system inherently oscillates when its gain is one and its phase shift is 180º. When a system's phase margin is below zero, the system is considered unstable, because its output could approach infinity. Phase margin can be determined using Nyquist plots such as the ones explained in the previous section, but it can also be measured using Bode plots. On a system's Bode plots, the phase margin represents the frequency response's distance from -180˚ at the point where the system has zero gain. This point is known as the zero dB point.

3.3.2 Increasing System Stability

There are several approaches to making a system more stable. Many of them are rooted in changing the design characteristics of the functional blocks of the system. If this sort of change is not possible, or is not enough to make the system stable, another approach can be taken. This approach is to add another block to the feedback loop, GC(s). This function is called a compensator, because it compensates for the system's previous inadequacies. One type of compensator is the phase-lead network. This block is made simply from a resistor which is in parallel with a resistor and capacitor and has the transfer function given below in Equation 2. In this equation, R1 is the resistor in parallel with the parallel combination of R2 and C.

$$G(s) = \frac{R_2}{R_2 + \left\{\frac{R_1\left(\frac{1}{Cs}\right)}{\left[R_1 + \left(\frac{1}{Cs}\right)\right]}\right\}}$$

This block also has a positive phase angle of almost 90 degrees. Since blocks in series simply add their phase responses, this block's positive phase response can directly increase the phase margin of the entire system. Another side effect of the phase lead function is that it is an amplifier for frequencies greater than the location of its own zero. This could distort the audio signal in our project if the necessary frequency of this block's zero is lower than 20 kHz. Another type of compensator is the phase-lag network. This block is made from the same components as the phase-lead, with them arranged in a different way. For the phase-lag network, there is a resistor in parallel with a series combination of another resistor and a capacitor. The transfer function for this block is given below in Equation 3. For this equation, the R1 is the resistor in parallel with the series combination of C and R2.

$$G(s) = \frac{R_2 + \left(\frac{1}{Cs}\right)}{R_1 + R_2 + \left(\frac{1}{Cs}\right)}$$

Equation 3 - Transfer Function of Phase-Lag Network

This block actually has a negative phase response at a very low frequency. This should not affect the phase margin of the system, because the frequencies at which the phase-lag network's phase response is negative should be much lower than the crossover frequency of the system. Instead of directly affecting the phase margin of the system, the phase-lag network adds attenuation, moving the crossover frequency to a much smaller frequency. At this new frequency the phase margin will be greater and therefore the system will be more stable. Because this block attenuates some parts of the audio band of frequencies more than others, no analog audio signal can be passed through it without being distorted.

3.4 NOISE, DISTORTION, AND FILTERING

The goal of a Class D amplifier is to convert a continuous time waveform into a series of binary pulses so they may be more efficiently amplified. This conversion adds unwanted harmonics as well as other noises to the output. Some of these are audible, and therefore undesired for noise quality reasons. Other noises are inaudible, but still undesired for efficiency reasons. A Class D modulator runs at speeds significantly higher than the highest desired output frequency, as dictated by the over sampling ratio. The noise introduced by the modulator is intentionally "shaped" out of the audio band. Despite the audio quality benefits of this "shaping", if the constant switching of the modulation stage was allowed to propagate to the speaker the overall efficiency of the system would be decreased. Although the noise added by the modulation stage is in audible, it still requires power to push the speaker cone. To increase the efficiency of the system, and reduce the possibility of cone damage because of high frequency oscillation, the output is filtered to remove this high frequency noise.

Not all noise added by the modulator will be outside of the audio band. Low fundamental frequencies will result in odd order harmonics that may lie within the audio band, increasing THD. In band noise cannot be filtered at the output. Low pass filters can be implemented either actively or passively. Active filters exhibit exceptional frequency roll offs, and have the ability to have extremely minimal pass band rippling. Active filters do not rely on non-ideal components like inductors and capacitors, whose non-linearity will distort the filter output.

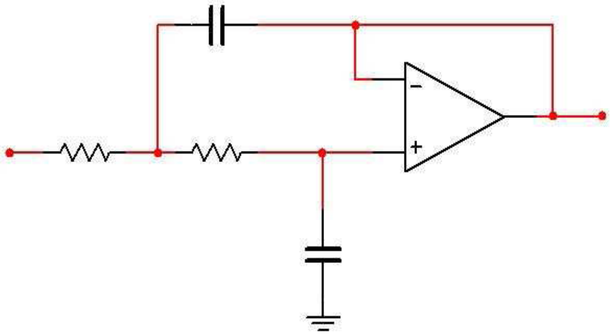

Fig: 3.15 – Example of Low Pass Active Filter

Unfortunately the basic design of an active filter makes it impractical for Class D amplifiers. Active filters are based on amplifiers themselves. Class D amplifiers are efficient because of their switching power supplies; the addition of a linear amplifier would defeat the purposes of making a high efficiency amplifier. A passive filter is a more practical choice for a Class D amplifier.

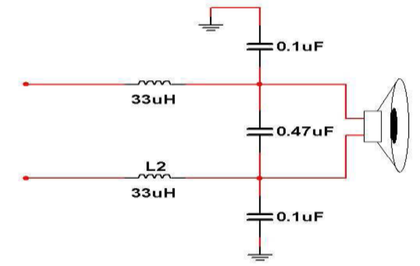

Fig: 3.16 - 2008 Class D Audio MQP Filter Design

A passive filter can, at best, have unity gain. A passive Butterworth filter has a very flat pass-band, although its frequency cutoff is not as sharp as some other filter configurations. Passive filters are built with non-linear components, producing output distortion. Last year's team created a passive Butterworth output filter that performed as required, and this design was reused for this year. The following figure shows the balanced low pass filter configuration and component values used by last year’s team. The component values used result in a 30 kHz cut off.

3.4.1 Passive filters

Passive implementations of linear filters are based on combinations of resistors (R), inductors (L) and capacitors (C). These types are collectively known as *passive filters*, because they do not depend upon an external power supply and/or they do not contain active components such as transistors. Inductors block high-frequency signals and conduct low-frequency signals, while capacitors do the reverse.

A filter in which the signal passes through an inductor, or in which a capacitor provides a path to ground, presents less attenuation to low-frequency signals than high-frequency signals and is therefore a *low-pass filter*. If the signal passes through a capacitor, or has a path to ground through an inductor, then the filter presents less attenuation to high-frequency signals than low-frequency signals and therefore is a *high-pass filter*. Resistors on their own have no frequency-selective properties, but are added to inductors and capacitors to determine the *time-constants* of the circuit, and therefore the frequencies to which it responds.

The inductors and capacitors are the reactive elements of the filter. The number of elements determines the order of the filter. In this context, an LC tuned circuit being used in a band-pass or band-stop filter is considered a single element even though it consists of two components. At high frequencies (above about 100 megahertz), sometimes the inductors consist of single loops or strips of sheet metal, and the capacitors consist of adjacent strips of metal. These inductive or capacitive pieces of metal are called stubs.

Single element types

The simplest passive filters, RC and RL filters, include only one reactive element, except hybrid LC filter which is characterized by inductance and capacitance integrated in one element.

L filter

An L filter consists of two reactive elements, one in series and one in parallel.

T and π filters: Low-pass π filter

Three-element filters can have a 'T' or 'π' topology and in geometries, a low-pass, high-pass, band-pass, or band-stop characteristic is possible. The components can be chosen symmetric or not, depending on the required frequency characteristics.

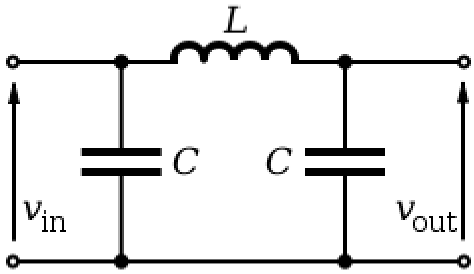

Fig: 3.18 LC low-pass passive filter

The high-pass T filter in the illustration has very low impedance at high frequencies, and very high impedance at low frequencies. That means that it can be inserted in a transmission line, resulting in the high frequencies being passed and low frequencies being reflected. Likewise, for the illustrated low-pass π filter, the circuit can be connected to a transmission line, transmitting low frequencies and reflecting high frequencies. Using m-derived filter sections with correct termination impedances, the input impedance can be reasonably constant in the pass band.

Multiple element types

Multiple element filters are usually constructed as a ladder network. These can be seen as a continuation of the L,T and π designs of filters. More elements are needed when it is desired to improve some parameter of the filter such as stop-band rejection or slope of transition from pass-band to stop-band.

The transfer function

The transfer function H(S) of a filter is the ratio of the output signal Y(S) to that of the input signals X(S) as a function of the complex frequency s:

$$H(s) = \frac{Y(s)}{X(s)}$$

With $s = \sigma + j\omega$.

The transfer function of all linear time-invariant filters, when constructed of discrete components, will be the ratio of two polynomials in s, i.e. a rational function of s. The order of the transfer function will be the highest power of $s$encountered in either the numerator or the denominator.

3.5 MOSFET H-BRIDGE DRIVER:

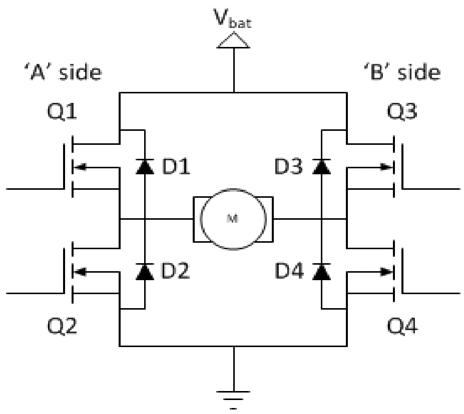

Fig: 3.19 schematic of MOSFET H-Bridge circuit

Drive circuitry

The drive circuitry for an H-Bridge is basically the electronics that sits between the PWM (and potentially other) digital control inputs and the MOSFET gates. It has two major purposes: Translate the input voltages to suitable levels to drive the gates Provide enough current to charge and discharge the gates fast enough On top of that, many drive circuits include additional functionality: Translate the input command into gate-drive signals according to the drive mode Provide shoot-through protection Generate voltages for the high-side gate-drive circuitry (for N-channel drivers) Provide additional safety functions, like over-current protection Control the turn-on and turn-off times of the FETs Drive circuits can come in many shape or form.

IV. HARDWARE REQUIREMENTS AND SPECIFICATIONS

4.1 MOSFET

The **metal–oxide–semiconductor field-effect transistor** (**MOSFET**, **MOS-FET**, or **MOS FET**) is a transistor used for amplifying or switching electronic signals. Although the MOSFET is a four-terminal device with source (S), gate (G), drain (D), and body (B) terminals, the body (or substrate) of the MOSFET often is connected to the source terminal, making it a three-terminal device like other field-effect transistors. Because these two terminals are normally connected to each other (short-circuited) internally, only three terminals appear in electrical diagrams. The MOSFET is by far the most common transistor in both digital and analog circuits, though the bipolar junction transistor was at one time much more common.

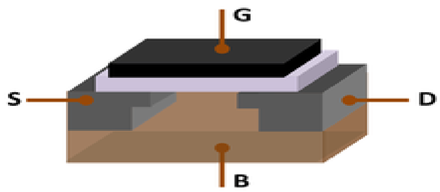

Fig: 4.1 MOSFET structure

In *enhancement mode* MOSFETs, a voltage drop across the oxide induces a conducting channel between the source and drain contacts *via* the field effect. The term "enhancement mode" refers to the increase of conductivity with increase in oxide field that adds carriers to the channel, also referred to as the *inversion layer*. The channel can contain electrons (called an nMOSFET or nMOS), or holes (called a pMOSFET or pMOS), opposite in type to the substrate, so nMOS is made with a p-type substrate, and pMOS with an n-type substrate (see article on semiconductor devices). In the less common *depletion mode* MOSFET, detailed later on, the channel consists of carriers in a surface impurity layer of opposite type to the substrate, and conductivity is decreased by application of a field that depletes carriers from this surface layer.

The 'metal' in the name MOSFET is now often a misnomer because the previously metal gate material is now often a layer of polysilicon (polycrystalline silicon). Aluminium had been the gate material until the mid-1970s, when polysilicon became dominant, due to its capability to form self-aligned gates. Metallic gates are regaining popularity, since it is difficult to increase the speed of operation of transistors without metal gates. Likewise, the 'oxide' in the name can be a misnomer, as different dielectric materials are used with the aim of obtaining strong channels with smaller applied voltages. An insulated-gate field-effect transistor or **IGFET** is a related term almost synonymous with MOSFET. The term may be more inclusive, since many "MOSFETs" use a gate that is not metal, and a gate insulator that is not oxide. Another synonym is MISFET for metal–insulator–semiconductor FET.

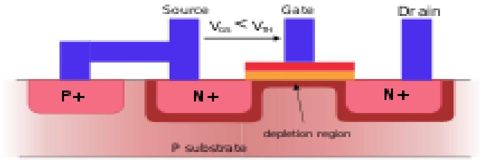

Fig: 4.2 Cross sectional view of MOSFET

4.1.1 Composition

Usually the semiconductor of choice is silicon, but some chip manufacturers, most notably IBM and Intel, recently started using a chemical compound of silicon and germanium (SiGe) in MOSFET channels. Unfortunately, many semiconductors with better electrical properties than silicon, such as gallium arsenide, do not form good semiconductor-to-insulator interfaces, thus are not suitable for MOSFETs. Research continues on creating insulators with acceptable electrical characteristics on other semiconductor material. In order to overcome the increase in power consumption due to gate current leakage, a high-κ dielectric is used instead of silicon dioxide for the gate insulator, while polysilicon is replaced by metal gates (see Intel announcement).

4.1.2 Circuit symbols

Varieties of symbols are used for the MOSFET. The basic design is generally a line for the channel with the source and drain leaving it at right angles and then bending back at right angles into the same direction as the channel. Sometimes three line segments are used for enhancement mode and a solid line for depletion mode. Another line is drawn parallel to the channel for the gate. The bulk connection, if shown, is shown connected to the back of the channel with an arrow indicating PMOS or NMOS. Arrows always point from P to N, so an NMOS (N-channel in P-well or P-substrate) has the arrow pointing in (from the bulk to the channel). If the bulk is connected to the source (as is generally the case with discrete devices) it is sometimes angled to meet up with the source leaving the transistor. If the bulk is not shown (as is often the case in IC design as they are generally common bulk) an inversion symbol is sometimes used to indicate PMOS, alternatively an arrow on the source may be used in the same way as for bipolar transistors (out for nMOS, in for pMOS).

Comparison of enhancement-mode and depletion-mode MOSFET symbols, along with JFET symbols (drawn with source and drain ordered such that higher voltages appear higher on the page than lower voltages):

4.1.3 MOSFET operation

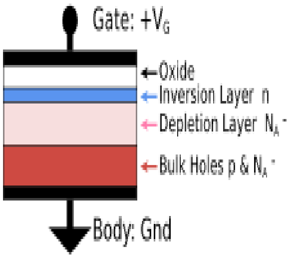

Fig: 4.4MOSFET layers

Metal–oxide–semiconductor structure

The traditional metal–oxide–semiconductor (MOS) structure is obtained by growing a layer of silicon dioxide (SiO_2) on top of a silicon substrate and depositing a layer of metal or polycrystalline silicon (the latter is commonly used). As the silicon dioxide is a dielectric material, its structure is equivalent to a planar capacitor, with one of the electrodes replaced by a semiconductor. When a voltage is applied across a MOS structure, it modifies the distribution of charges in the semiconductor. If we consider a p-type semiconductor (with $N_A$the density of acceptors, p the density of holes; $p = N_A$ in neutral bulk), a positive voltage, V_{GB}, from gate to body (see figure) creates a depletion layer by forcing the positively charged holes away from the gate-insulator/semiconductor interface, leaving exposed a carrier-free region of immobile, negatively charged acceptor ions (see doping (semiconductor)). If $V_{GB}$is high enough, a high concentration of negative charge carriers forms in an **inversion layer** located in a thin layer next to the interface between the semiconductor and the insulator. Unlike the MOSFET, where the inversion layer electrons are supplied rapidly from the source/drain electrodes, in the MOS capacitor they are produced much more slowly by thermal generation through carrier generation and recombination centers in the depletion region. Conventionally, the gate voltage at which the volume density of electrons in the inversion layer is the same as the volume density of holes in the body is called the threshold voltage. When the voltage between transistor gate and source (V_{GS}) exceeds the threshold voltage (V_{th}), it is known as overdrive voltage.

This structure with p-type body is the basis of the n-type MOSFET, which requires the addition of an n-type source and drain regions.

4.1.4 MOSFET structure and channel formation

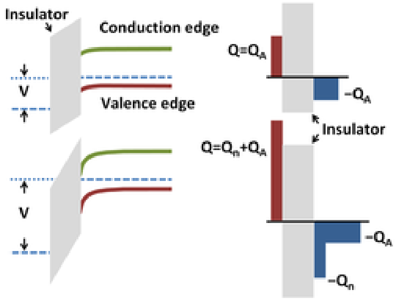

Fig: 4.5 Channel formations in nMOS

4.1.5 Modes of operation

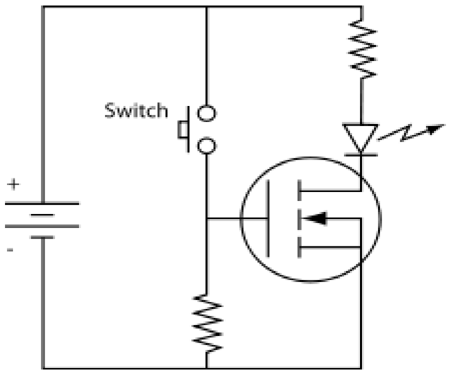

Fig: 4.6 Example of an N-Channel MOSFET.

The operation of a MOSFET can be separated into three different modes, depending on the voltages at the terminals. In the following discussion, a simplified algebraic model is used. Modern MOSFET characteristics are more complex than the algebraic model presented here. For an **enhancement-mode, n-channel MOSFET**, the three operational modes are: Cutoff, sub threshold, or weak-inversion mode

When $V_{GS} < V_{th}$:

Where V_{GS}gate-to-source is bias and $V_{th}$is the threshold voltage of the device.

4.1.6 Body effect

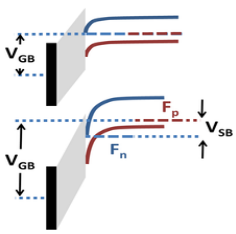

Fig: 4.8 Band diagram showing body effect

The occupancy of the energy bands in a semiconductor is set by the position of the Fermi level relative to the semiconductor energy-band edges. Application of a source-to-substrate reverse bias of the source-body pn-junction introduces a split between the Fermi levels for electrons and holes, moving the Fermi level for the channel further from the band edge, lowering the occupancy of the channel. The effect is to increase the gate voltage necessary to establish the channel, as seen in the figure. This change in channel strength by application of reverse bias is called the *body effect*.

V. CONCLUSION

While this project was successful in designing and implementing a fully functional Class D audio amplifier there are many additional areas that can be explored to complement the completed Class D audio amplifier. The focus of this project was to develop an amplifier to drive an 8Ω load. However, a working amplifier is a component that can be used in many different products, from guitar amplifiers to home theater receivers. This section will discuss several of the ideas for future work concerning Class D audio amplifiers. While an amplifier can be thought of as simply a transistor that amplifies an input signal, a finished product is much more than that. While the major accomplishment of this MQP was a completed Class D audio amplifier, it was tied to the lab bench due to the need for a power supply. To be considered a finished commercial product an amplifier must have its own dedicated power supply, ideally one that can be plugged into a standard wall outlet or a high power battery. The power supply would have to provide several different voltage rails, ranging from 5 Volts to 40 Volts, for different parts of the modulator and power stage. Since the power supply would be for a Class D audio amplifier it would need to be an extremely high-efficiency power supply. Otherwise the efficiency of the entire system could drop to unacceptable levels. Any work undertaken to develop such a power supply should also consider the power requirements of running multiple amplifiers in parallel. Many professional quality amplifiers offer 2 or 4 channels per amplifier unit, usually for stereo or surround sound systems.

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