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Comparative study of PWM and PDM in Class D Amplifiers

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Dedication

With all pride and gratitude, I dedicate this master's dissertation to:

My dear parents: who have always been my support and encouragement. Thanks to your encouragement and sacrifices, I have been able to achieve this accomplishment. Thank you for every word of support, every moment of patience, and every drop of sweat you have given for me.

My esteemed teachers: who have not spared any effort or knowledge in guiding and directing me throughout my academic journey. You have always been exemplary role models and have always stood by my side, supporting and guiding me.

To myself: in recognition of the effort and diligent work I have put in to reach this day, and for the challenges I have faced and overcome.

Finally, to everyone who believed in me and supported me with a word or an action. Thank you all, for you have been part of this achievement.

With this, I present to you the fruit of my efforts and hope that this master's dissertation will be a valuable addition to the field of science and knowledge.

Abstract

Pulse width modulation (PWM) and pulse density modulation (PDM) are among the most important modulation techniques in demand today, especially in amplifiers. As amplifiers are divided into several classes, class D is the best one, providing high efficiency and low dissipation.

This master's dissertation is dedicated to a comparative study of PWM and PDM in Class D amplifiers and their effect on their performance through circuit complexity, simplicity, efficiency, and dissipation. PWM outperforms PDM in terms of circuit simplicity, performance, and cost-effectiveness, making it the preferred choice for Class D amplifiers.
Keywords : PWM, PDM, Class-D amplifier, Circuit, THD, Efficiency

الملخص

يعد تعديل عرض النبض (PWM) وتعديل كثافة النبض (PDM) من بين أهم تقنيات التعديل المطلوبة اليوم، خاصة في مضخمات الصوت. وبما أن مضخمات الصوت تنقسم إلى عدة فئات، فإن الفئة D هي الأفضل، حيث توفر كفاءة عالية وتبديدًا منخفضًا.

هذه المذكرة مخصصة لدراسة مقارنة بين PWM وPDM في مضخمات الفئة D وتأثيرها على أدائها من خلال تعقيد الدارة وبساطتها وكفاءتها وتبديدها. تتفوق PWM على PDM من حيث بساطة الدائرة والأداء وفعالية التكلفة، مما يجعلها الخيار المفضل لمضخمات الفئة D.
الكلمات المفتاحية : PWM ، PDM ، مضخم صوت من الفئة D ، دارة ، THD ، الكفاءة

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General Introduction

In the world of audio engineering, amplification stands as a cornerstone technology, pivotal indelivering high-fidelity sound reproduction. Among the myriad types of amplifiers, one finds a diverse array catering to various applications and preferences. In this exploration, we delve into the realm of audio amplifiers, examining their classes and the sophisticated modulation techniques—Pulse Width Modulation (PWM) and Pulse Density Modulation (PDM)—that shape their operation and performance.

Audio amplifiers are electronic devices tasked with increasing the amplitude of audio signals, effectively boosting their strength for transmission through speakers or headphones. These devices come in various classes, each offering distinct advantages and trade-offs in terms of efficiency, linearity, and fidelity. Classes such as Class A, Class B, Class AB, Class C, and Class D have gained prominence, each tailored to specific requirements ranging from high-fidelity audio reproduction to power efficiency in battery-powered devices.

In recent years, advancements in modulation techniques have revolutionized amplifier design, paving the way for more efficient and compact solutions. Pulse Width Modulation (PWM) and Pulse Density Modulation (PDM) are two such techniques that have garnered attention for their ability to deliver high-quality audio with minimal distortion and power consumption.

PWM involves the modulation of a rectangular pulse wave, where the width of the pulse is varied in proportion to the amplitude of the audio signal. This technique offers high efficiency and low distortion, making it ideal for Class D amplifiers, which are known for their energy efficiency and compact size. As well, class D amplifiers are known as “PWM amplifiers”.

On the other hand, Pulse Density Modulation or Delta-Sigma modulation, abbreviated PDM modulation, focuses on encoding audio signals by varying the density of pulses within a fixed time interval. PDM modulation is renowned for its simplicity and resilience to noise, making it a popular choice for digital audio applications, especially in high-resolution audio systems.

A comparative study between PWM and PDM modulation techniques provides valuable insights into their respective strengths and weaknesses. Understanding the trade-offs between these modulation techniques is essential for selecting the most suitable approach based on specific application requirements.

In this master’s dissertation, we embark on a journey to explore the intricacies of audio amplifiers, delving into their classes and the modulation techniques that underpin their operation. Through a comparative analysis of PWM and PDM modulation, we aim to unravel the nuances that shape the landscape of modern audio amplification, shedding light on the path towards optimal audio performance.

Chapter I:Audio amplifier classes and performances

1.1 Introduction

Understanding power amplifiers and their various characteristics is vital in the world of audio and electronics technology. A power amplifier is an essential part of audio and electronics systems as it plays a crucial role in amplifying signals in an efficient and accurate manner. Power amplifier classes and features vary widely, affecting sound quality, energy efficiency, and other important factors

Power amplifier classes include five types: Class A , Class B , Class AB, Class C , Class D. Depending on the technologies used in the signal amplification process, each class works differently and has specific advantages and disadvantages.

THD is important criteria for measuring the performance of a power amplifier. THD reflects the amount of distortion the amplifier adds to the original signal, preferably low to ensure high audio quality.

Efficiency plays a vital role in choosing the right power amplifier, as the efficiency of the amplifier is an important factor in power consumption, heat savings, and limiting its use in limited or portable power applications

This section will present a detailed analysis of power amplifier types and their different characteristics, including THD and power efficiency, to better understand their different applications and improve their performance and use in contemporary audio and electronic systems

1.2 Power amplifier

The power amplifier is the last stage in an amplifier chain. It provides a much higher power than the control signal, while maintaining the same signal shape.

The purpose of amplifiers is to control an actuator (loudspeaker, motor, inductor, resistor, etc.) without distorting the input signal.

In most cases, power amplification is current amplification. This is why we use bipolar transistors, or power MOS transistors [1].

1.3 Audio amplifier

A basic **power amplifier** that is designed to take input as the low strength audio signal and generate the output signal that consists of the high strength value. This process of amplification is utilized in the various domains where an electrical signal is converted to an acoustic signal. This type of amplifiers is known as audio amplifiers. Any circuit which processes the audio signal has the audio amplifier both at the input and also at the output. For example, if a microphone receives a sound wave input signal it need a pre-amplification of the signal before processing it further and similar before sending an electrical signal to a speaker it needs to be amplified [2].

1.4 Amplifier class selection criteria

Many criteria can be taken into account when selecting an amplifier. The most important are:

1.4.1 Power output:

- Power output refers to the capability of an amplifier to deliver power to a load (such as speakers) at a specified sound level. It is measured in watts (W).

- Different amplifier classes vary in their ability to produce output power. For example, Class A amplifiers provide high power but consume more power compared to Class D, which is more efficient[3].

1.4.2 Performance:

- Efficiency reflects how well an amplifier converts electrical power into sound power efficiently.

- Class D amplifiers are more efficient than Class A because they use pulse-width modulation (PWM) techniques to efficiently switch between on and off states, reducing energy loss in the form of heat.

1.4.3 The maximum power the active element can dissipate:

- This refers to the maximum power that the active component (such as transistors or MOSFETs) can withstand in terms of heat dissipation and removal.

- Class A amplifiers are more power-hungry and heat-producing, requiring effective cooling, whereas Class D consumes less power and produces less heat.

1.4.4 Gain (voltage, power):

- Gain represents the amplifier's ability to increase the incoming electrical signal.

- Gains can vary between different amplifier classes depending on the configuration and design. For example, Class A may have high gains but consume more power[3].

1.4.5 The distortion:

- Distortion is any unwanted change in the original audio signal.

- Class A amplifiers are more accurate and have less distortion compared to other classes like Class D, due to their continuous operation nature.

1.4.6 Maximum working frequency:

- This indicates the maximum frequency that the amplifier can efficiently handle.
- This information is crucial in system design, especially if operating at high frequencies or using specific types of speakers.

1.5 Power amplification classes

1.5.1 Class A power amplifiers

The amplifier consists of an output stage with a single transistor. The rest point is approximately in the middle of the load line. Depending on the signal to be amplified, it can therefore move to either side of this point along the load line.

Active components conduct for the entire period of the input signal.

1.5.2 Class B power amplifiers

The amplifier consists of an output stage with two complementary transistors

The rest point is at the blocking limit of each transistor. To amplify the two alternations of a sinusoidal signal, one of the transistors must amplify the positive alternations, while the second amplifies the negative alternations.

During a half-period of the input signal, the active components conduct[3].

1.5.3 Class AB power amplifiers

The amplifier consists of an output stage with two complementary transistors.

This is the basic output structure of a Class B amplifier, modified at the bias level.

The resting point is very close to the transistor blocking limit. In other words, between Class A and Class B, but closer to Class B.

1.5.4 Class C power amplifiers

The output stage consists of a single transistor. The quiescent point lies largely in the blocked region of the transistor's characteristics. Only the peaks of the positive alternations of the input signal will produce an output signal.

Active components conduct less than half a period of the input signal.

1.5.5 Class D power amplifiers

The output stage operates in switching mode, i.e. between two voltage levels. The switching frequency is fixed, but the switching duty cycle is variable. The low-frequency (LF) signal to be amplified is therefore coded in pulse-width modulation (PWM). The switching frequency is at least one order of magnitude higher than the maximum frequency of the LF signal. This signal is reconstituted by low-pass filtering at the output[3].

1.6 Basic structure of power amplifiers

The class of an amplifier is fully determined by the biasing applied to the transistor.

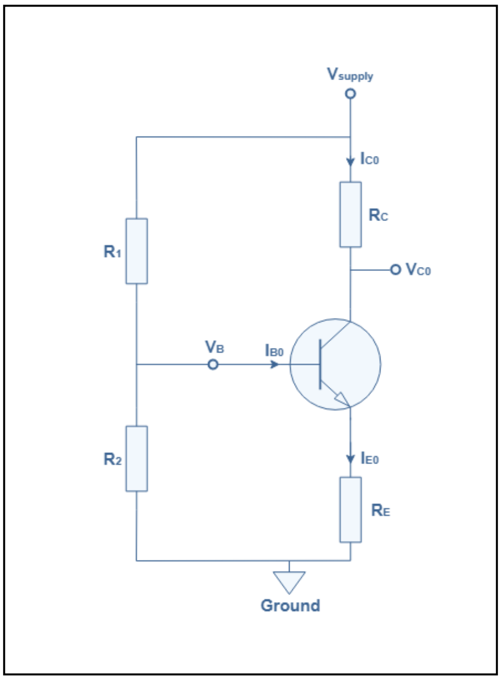

Figure 1.1: Voltage divider network

In **Figure 1.1** a BJT transistor of current gain β is biased by a voltage divider network, which consists of two parallel resistances R_1 and R_2 wired to the base branch. The collector current and voltages when no AC signal is applied (I_{C0},V_{C0}) sets the operating or quiescent point of the amplifier. The quiescent point is very important because its position in the output characteristic dictates the conducting angle value and therefore the class of the amplifier.

The set of values (I_{C0},V_{C0}) can be adjusted with the values of the biasing resistances and emitter resistance. Indeed, the collector current I_{C0} is given by :

$$\mathbf{I_{C0}=\frac{\beta(V_B-0.7V)}{(R_1//\,R_2)+(\beta+1)R_2}} \quad \textbf{1.1}$$

We can clarify two parameters in **Equation 1.1:** 0.7 V corresponds to the voltage V_{BE}, which is the threshold voltage of silicon-based transistors. The resistance $R_1//R_2$ is the parallel equivalent resistance of the biasing circuit and is given by the ratio $(R_1\times R_2)/(R_1+R_2)$.

And the collector voltage V_{C0} satisfies:

$$\mathbf{V_{C0} = V_{Supply} - R_C \times I_{C0}} \quad \textbf{1.2}$$

Note that in the following of this study, we will always consider bipolar transistors, but everything we say applies also to other types of transistors such as MOSFETs. Moreover, as a matter of simplification, we use the Common Emitter Amplifier as the configuration under study, the output signals shown in the figures will therefore be inverted[4].

1.7 Class A amplifier

Aclass A amplifier is characterized by a **360° conduction angle**. To achieve this feature, the quiescent point of a Class A amplifier is chosen to be in the middle of the load line such as shown in Figure 1.2 :

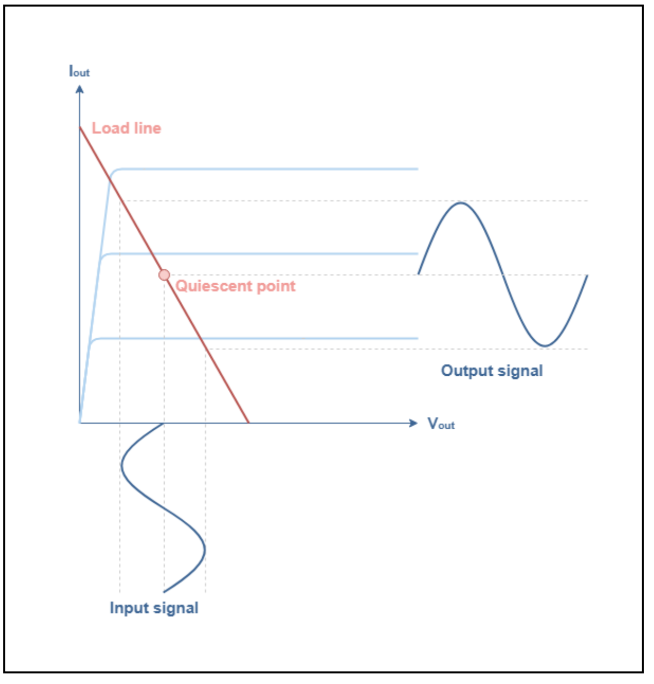

Figure 1.2 : Class A biasing conditions

The quiescent point satisfies $\mathbf{I_{C0}=V_{supply}/2R_C}$ and $\mathbf{V_{C0}=V_{supply}/2}$. These formulas along with **Equation 1.1** and **Equation 1.2** enable to choose proper values for the biasing resistances to get a class A amplifier.

The absorbed power of a class A amplifier is a constant and equal to $\mathbf{P_{abs}=V_{supply}\times I_{C0}}$. The output power is the product of the rms output current and voltage : $\mathbf{P_{out}=V_{out,rms}\times I_{out,rms}}$. The maximal value of P_{out} is given when the output current reaches the upper limit I_{C0} and the output voltage reaches the power supply V_{supply} : $\mathbf{P_{out,max}=(V_{supply}\times I_{C0})/2}$. The maximal efficiency is therefore :

$$\eta_{max} = \frac{P_{out,max}}{P_{abs}} = \frac{1}{2} \qquad \mathbf{1.3}$$

In reality, the efficiency is around 20 to 30 % and 50 % can be achieved with a two transistor configuration. This low efficiency highlights the fact that class A amplifiers use power even when no AC input signals are applied[4].

Figure 1.3 : Class A power amplifier

1.8 Class B amplifier

Class B amplifiers have been developed as an answer to the low efficiency of class A amplifiers. This class of amplifier is characterized by a 180° conduction angle, that is to say that they use only half of the input signal to realize the amplification process. To achieve a class B amplification, one needs to bias the circuit accordingly with Figure 1.4 :

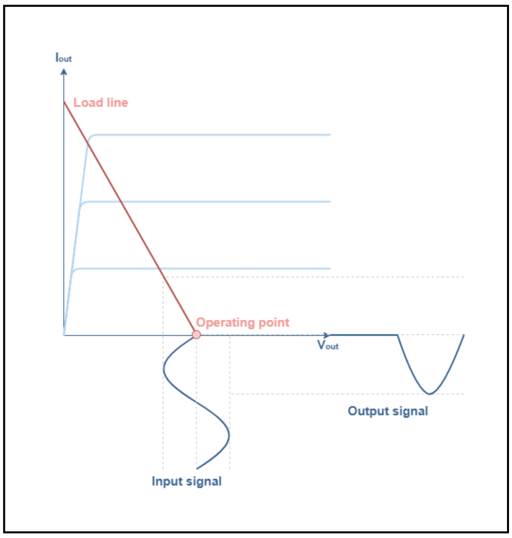

Figure 1.4 : Class B biasing conditions

The operating point here is located at the cutoff point and satisfies $\mathbf{I_{C0}=0}$ and $\mathbf{V_{C0}=V_{supply}}$.

It is pretty obvious that a faithful amplification cannot be achieved with a class B amplifier. To solve this problem, one of the most common solution is to use two transistors (one NPN and one PNP) in a so called “push-pull” configuration:

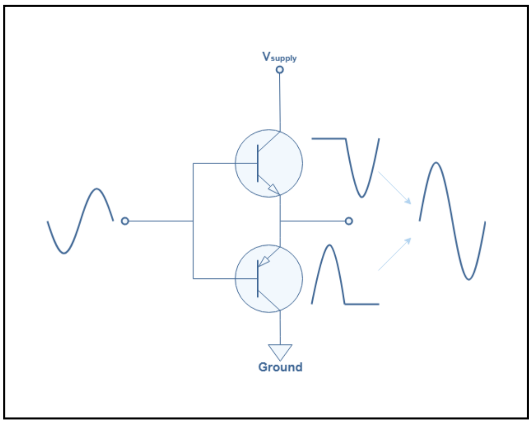

Figure 1.5 : Class B push-pull configuration

The NPN transistor takes care of amplifying the positive signal of the input and the PNP amplifies the negative signal. The combination results in an addition of the two independent amplifications that reproduce the shape of the input signal.

However, there is a phenomenon called the **crossover distortion** that prevents class B amplifiers, even in a push-pull configuration to give a 100 % faithful amplification. The cause is due to the threshold voltage of transistors (+0.7 V for NPN and -0.7 V for PNP) that creates an interval of 1.4 V where no amplification at all is performed from neither the NPN nor the PNP transistor. The consequence is a distortion of the signal around the 0 V point of the output signal, well known by audiophiles.

Nonetheless, class B amplifier presents the advantage over the class A to be more efficient with a theoretical **maximum efficiency of η_{max}=78.5 %**. The efficiency observed on real configurations however does not exceed 70 %[4].

Figure 1.6 : Class B power amplifier

1.9 Class AB amplifier

As the name refers to, class AB amplifier behaves as a combination of class A and class B amplifiers. It has been developed in order to overcome the low efficiency of the class A and the distortion of the class B. Class AB amplifiers are characterized by a conduction angle in the interval]180°;360°[. The operating point given by the bias circuit is located between a class A quiescent point and the cutoff point:

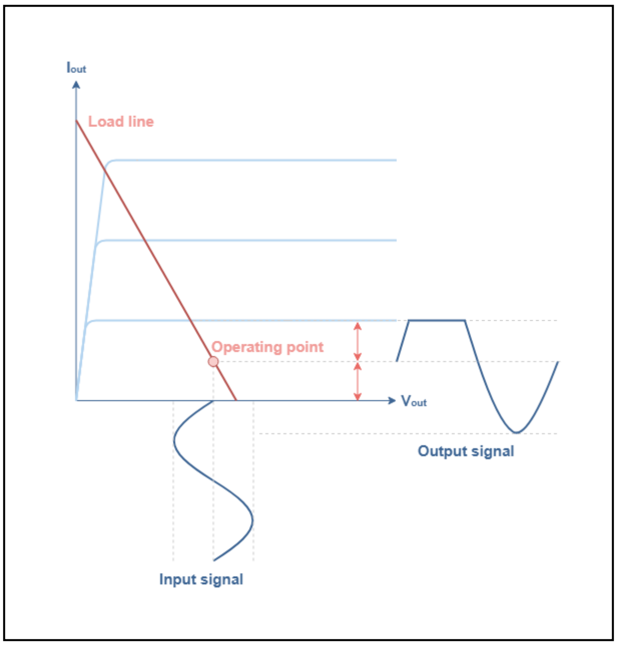

Figure 1.7 : Class AB biasing conditions

The operating point of a class AB amplifier satisfies:

$$0<I_{C0}<V_{supply}/2R_C \textbf{ and } V_{supply}/2<V_{C0}<V_{supply}.$$

When the operating point is closer to the cutoff point, the amplifier "becomes" more as a class B than a class A : the signal gets more distorted but the efficiency increases. At the opposite, when the operating point approaches the quiescent point in the middle of the load line, the amplifier behaves more as a class A than a class B : the output is more faithfully reproduced but the efficiency decreases.

Since class AB amplifiers offer a good compromise between the advantages of linearity of class A and the good efficiency of class B, they are commonly used today in many applications. They are usually found in a push-pull configuration, such as presented in **Figure 1.5** and they even eliminate the crossover distortion during the addition of the two amplified outputs from the NPN and PNP transistors[4].

Figure 1.8: Class AB power amplifiers

1.10 Class C amplifier

It is characterized by a small conduction angle that is in the interval]0°;90°[. The operating point of class C is beyond the cutoff point, aligned with the load line but in the region of negative biasing currents :

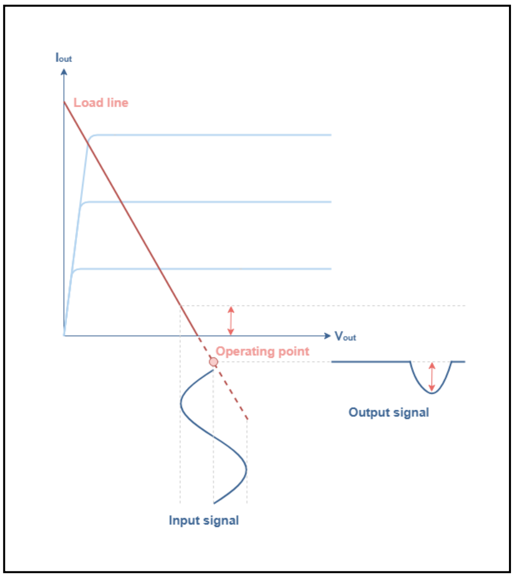

Figure 1.9 : Class C biasing conditions

As a matter of fact, the operating point of a class C satisfies :

$\mathbf{I_{C0}<0}$ and $\mathbf{V_{C0}>V_{supply}}$ (which makes sense from **Equation 1.2** if $I_{C0}<0$)

The high distortion produced by class C amplifiers can be processed by a parallel resonance circuit L//C that consists of an inductance (L) and a capacitance (C). This circuit can indeed convert the output pulses into complete sine waves. For this reason, class C amplifiers are used in **high frequency** applications.

The biggest advantage of a class C amplifier is its efficiency that is above 78.5 % and can approach 100 % depending how far the operating point is from the cutoff point[4].

Figure 1.10 : Class C power amplifier

1.11 Class D amplifier

Class D amplifiers generally consists of three different modules : a modulator, a switching stage and a low-pass filter.This is the basic architecture, but we can add more modules to enhancing perfection. The signal path along with the succession of these different modules is presented in Figure 1.11 below :

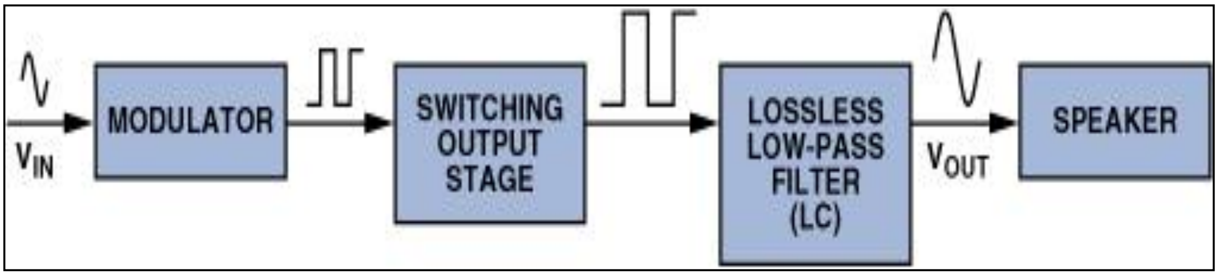

Figure 1.11 : Class D open-loop-amplifier block diagram

While classic amplifiers accept as an input a sinusoidal signal, Class D amplifiers previously transform it through a modulator into a rectangular signal. We will see in the dedicated section that the view proposed in **Figure 1.11** concerning the modulation is oversimplified.

The switching stage is where the amplification takes places thanks to transistors. We present in detail in the section dealing with this stage that the transistors work in a particular regime and a complementary configuration in order to amplify correctly the rectangle signal.

Finally, a low-pass filter is used in order to restore the sinusoidal shape of the signal. Moreover, this final stage eliminates undesirable harmonics than may have been generated during the amplification process[5].

Figure 1.12 : Class D power amplifier

1.11.1 Class D amplifier modulation techniques

Pulse width modulation (PWM)

Bang-bang control

Delta-sigma modulation (ΔΣ Modulation)

Self-oscillating

1.11.1.1 Pulse width modulation (PWM)

Many modulations techniques exist, however the most common and widely used for many applications is the **Pulse Width Modulation (PWM)**. A simple graph representing the PWM is shown in Figure 1.13 below :

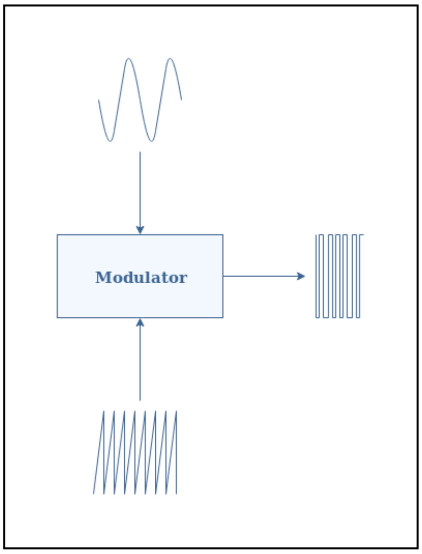

Figure 1.13: Principle of a PWM modulator

This technique consists in comparing the input sinusoidal signal with a high frequency triangular signal commonly called carrier obtained from an independent generator. In order to be consistent with Shannon's theo rem, the frequency of the carrier signal must be at least twice as high as the sinusoidal signal frequency.

The outputof the modulator is obtained by doing the following comparison between these two signals:

- If the sine is above the carrier signal, the output is equal to 1
- Otherwise, the output is equal to 0. In the **Figure 1.14** below, an input signal of frequency 2 Hz is plotted along with a carrier signal of frequency 20 Hz. Moreover, the PWM output is plotted by doing the comparison explained previously.

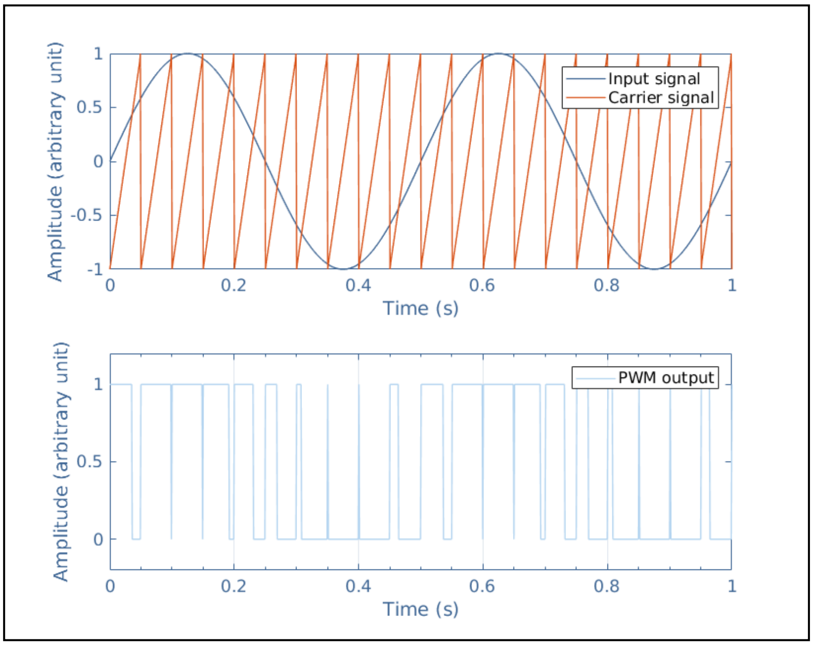

Figure 1.14: PWM input and output. Plotted with MatLab

It is important to note that the frequency of the PWM output is the same as the carrier frequency. The **duty cycle** is the number characterizing the proportion where the signal value is 1 during a period. For example, if the pulse is symmetrical, half of the signal will be 1 and half 0, the duty cycle is therefore 50 % or 0.5. In the case of a PWM, while the frequency is constant, the duty cycle varies.

We can note that when the input signal is maximum, the PWM duty cycle tends to **1** and at the opposite, tends to **0** when the input signal is minimal. Therefore, the duty cycle of the PWM is directly related to the original shape of the sine signal. This affirmation can indeed be confirmed with a simple algorithm that averages the PWM output independently for each cycle, the result is plotted and shown in **Figure 1.15** :

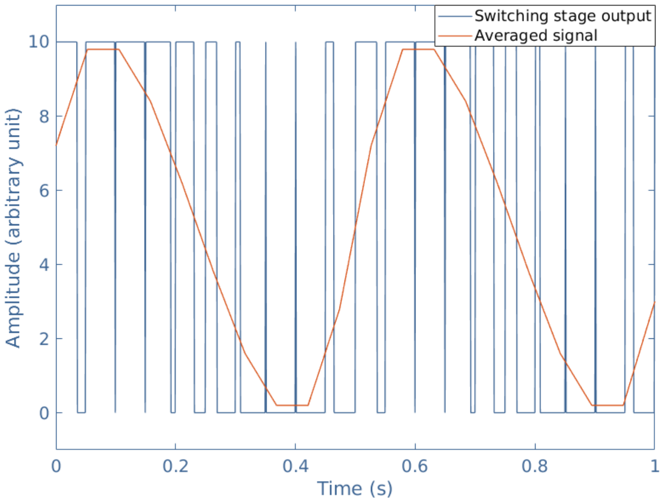

Figure 1.15 : Average operation on the PWM signal

It is clear in this figure that when averaging the PWM signal, the sine shape of the original signal appears again. In real circuits, this operation is done by a filter as we will see in the section “Filtering”[5].

1.11.1.2 Bang-bang control

The bang-bang CDA, as shown in Figure 1.16, is less popular since the advantages already offered by PWM and ΣΔ modulations. However, the CDA bang-bang presents the simplest hardware and is the most energy-efficient structure.

The only quiescent power dissipation block in a bang-bang CDA is the hysteresis comparator. Compared with PWM and ΣΔ CDAs, both of which have integrating power consumption, the bang-bang CDA is more energy-efficient.

In addition, the switching frequency of a typical bang-bang controller is lower than MLI and ΣΔ CDAs, which lead to lower switching power dissipation [1].

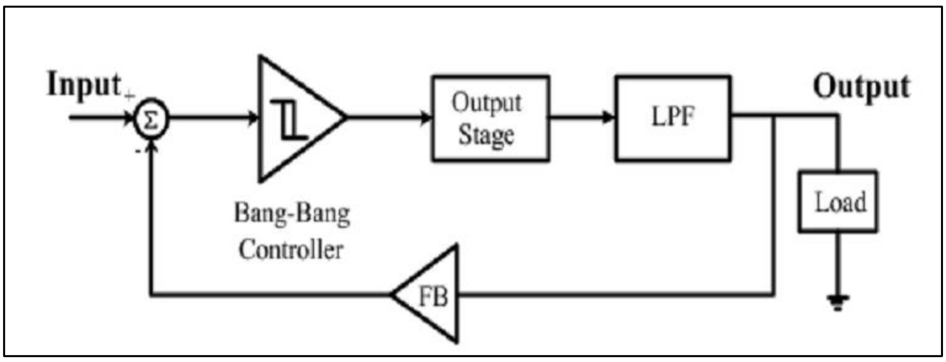

Figure 1.16: Basic architecture of the Bang-Bang Control Class D Amplifier

1.11.1.3 Delta-sigma modulation (ΔΣ Modulation)

The Delta-Sigma (Δ-Σ) modulation is a modulation method that provides lower output noise and non-linearities to the Class D amplifier. **Figure 1.17a** illustrates a topology of a Δ-Σ modulator, which runs according to the modulator clock, which identifies the sampling interval of the input. The modulation cycle starts by integrating the difference between the input sample and the 1-bit DAC. Each modulator clock pulse produces another modulator output pulse. The resulting output bitstream becomes an alternate representation of the input voltage proportional to the reference voltage. **Figure 1.17b** shows the input sine wave and the resulting modulator output bitstream, assuming a reference voltage of 1 V. As the input approaches 1 V, the 1 s density of the modulator's bitstream approaches 100%.

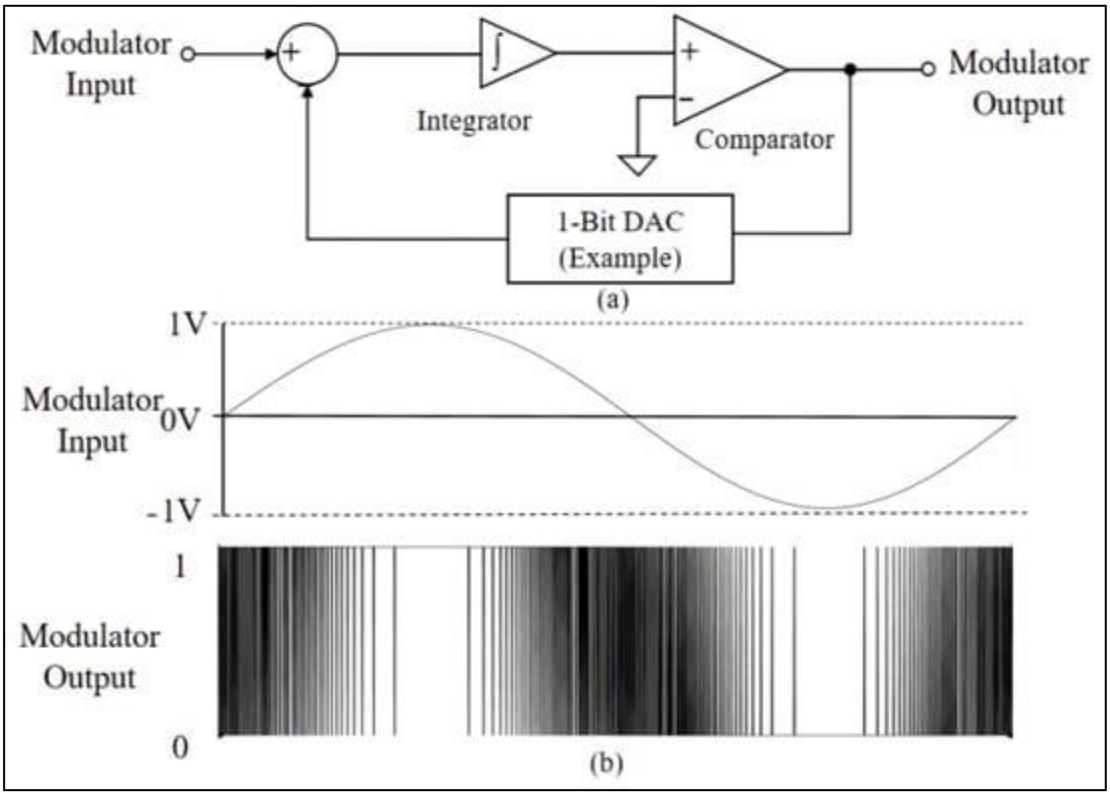

Figure 1.17 : The structure of Δ-Σ modulator (a) and the comparison of input and output signal (b).

The distortion caused by this type of modulation makes the method imperfect for Class D amplifiers. A series of improved designs have been proposed; for example, A class D GaN audio amplifier using Delta-Sigma modulation is introduced in and the improvement is noticeable. And presented a low-power class-D audio amplifier using a first-order sigma-delta modulator as a PDM modulator; this design can effectively improve efficiency.

Depending on the quantizer design, Delta-Sigma Class D amplifiers can be classified as synchronous or asynchronous. **Figure 1.18** shows a synchronous δ Sigma Class D amplifier with a quantizer of a comparator and a Class D flip-flop with a fixed clock frequency. Higher order modulators and higher sampling frequencies can significantly reduce nonlinearity, especially in-band quantization noise, and lead to complex hardware and higher costs.

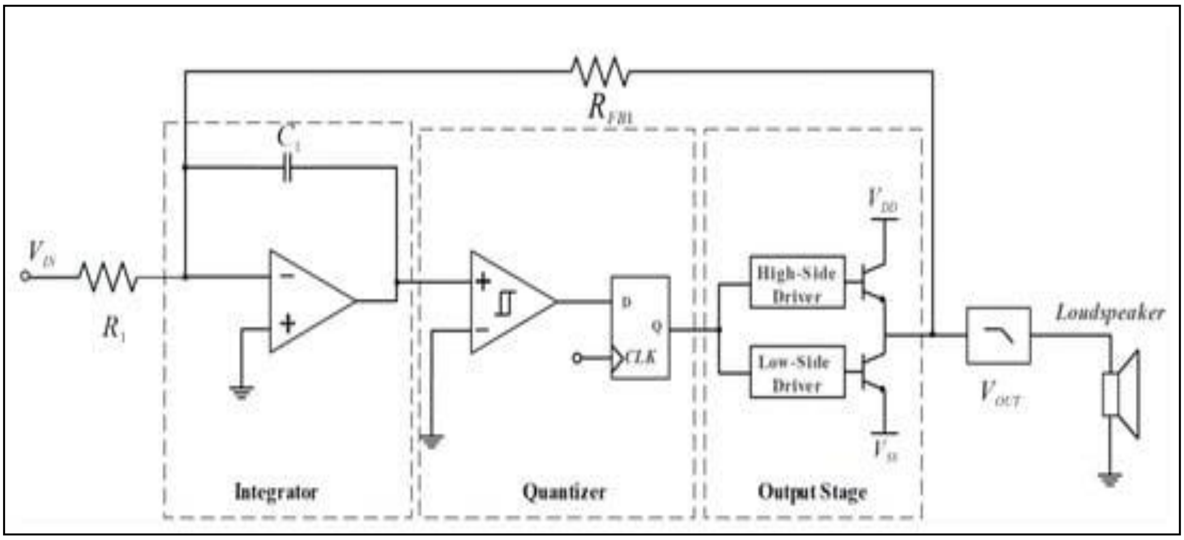

Figure 1.18 : First-order synchronous Δ-Σ Class D amplifier block diagram.

An asynchronous Class D amplifier with a hysteresis comparator as a quantizer is shown in **Figure 1.19** . Its quantization error is zero because the switch depends on input rather than a fixed clock, making the switch continuous.The disadvantages of asynchronous Delta-Sigma Class D amplifiers are the variability of the switching frequency and the difficulty of multi-channel design coupling on a single IC[6].

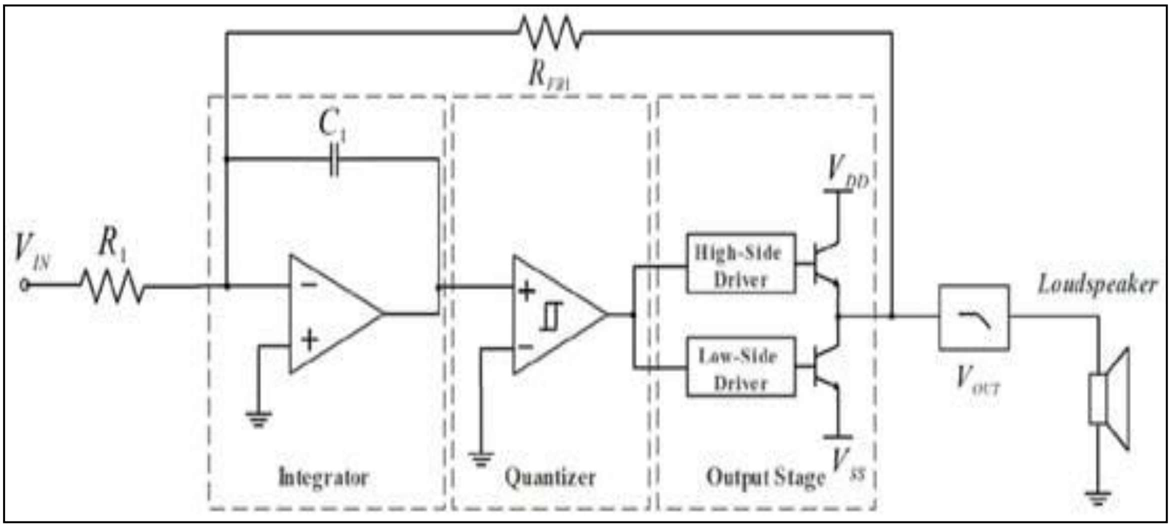

Figure 1.19 : Block diagram of a first-order asynchronous Δ-Σ Class D amplifier.

1.11.1.4 Self-oscillating

The operation of a self-oscillating Class D amplifier **Figure 1.20** can be viewed as a "high-power" Sigma-Delta converter, which is not PWM based, and the power-stage amplifies a PDM STREAM. The self-oscillating Class D is not founded on an external clock and does not use any flip-flop behind the comparator (compared to a classical Sigma-Delta).The frequency of the self-oscillating Class D is influenced by the power supply voltage, integrator behavior, and propagation delays of the comparator and the power-stage; practically, the designer sets the "idle" frequency by adjusting the integrator circuit.

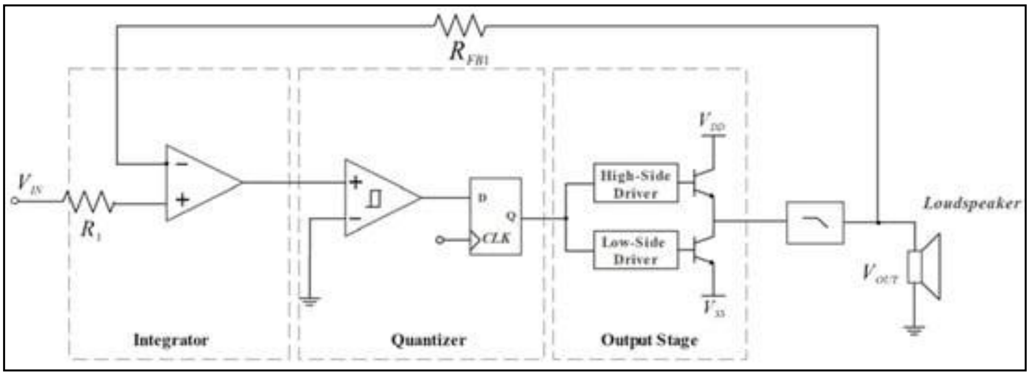

Figure 1.20 : Self-oscillating Class D amplifiers block diagram.

The advantages and disadvantages: the first advantage of self-oscillation is eliminating the need for a low-jitter clock source, which is required for high-performance analog and digital input fixed-frequency amplifiers. Switching frequency is subsequently set by an oscillation condition and varies nonlinearly as a function of the input signal. The one-cycle control topology is notable for considerable switching frequency variation.In addition, since the low-pass filter precedes the feedback, the low-pass filter of the feedback loop can suppress the nonlinearity. Disadvantages of the self-oscillating feature include the intermodulation products that are generated when using multiple amplifiers in a system. Moreover, a large-signal analysis only becomes feasible with simulation, as the small-signal approximations typically only hold for modulation indices of D.

There are two specifications of the comparator that should be considered. The first important specification is the input-to-output voltage amplitude range. These values determine the amplitude of the square wave produced by the comparator. If the comparator cannot achieve the full rail output signal, the amplitude of the square wave will be limited.

The second specification is the propagation delay. This is the delay for the signal to pass through the internal circuit of the comparator. When signal A exceeds signal B, the comparator produces an output, but the output does not change immediately. The amplifier circuit will not be able to function correctly if the comparator has too large of a propagation delay[6].

1.11.2 Class D audio amplifier main stages

1.11.2.1 Amplification

Since the carrier signal is usually chosen such as its frequency is much higher than the input signal, the PWM output to amplify can be above the high cutoff frequency of a BJT-based amplifier.This is the reason why high frequency MOS transistors are preferred over the classical bipolar-based amplifiers for class D amplification.

In a class D amplifiers, one NMOS and one PMOS are connected in a push-pull configuration as shown in Figure 1.21 :

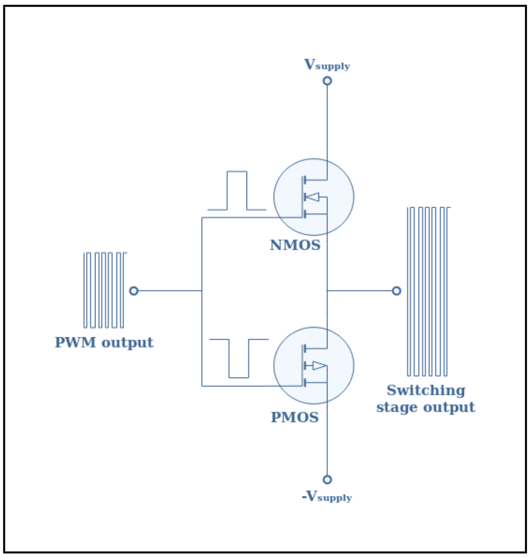

Figure 1.21 : Push-pull configuration of the amplification stage

Like for a class B amplifier, the complementary transistors are biased in such a way that the NMOS amplifies only positive half-waves and the PMOS only the negative half-waves. This amplification stage is also called the **switching stage** because the transistors behave precisely as switches : they are either fully ON (short circuit) or OFF (open circuit)[5].

1.11.2.2 Filtering

In order to recover the original sine shape of the signal, the amplified pulse signal must be processed by a filter. This filter should respect some conditions :

- Suppress the high frequencies above the normal bandwidth (midrange frequencies) of the amplifier, especially the carrier frequency and its harmonics.

- Reproduce the midrange frequencies of the amplifier with a good level of gain. For example 20 Hz – 20 kHz for an audio amplifier.
- Achieve a maximally flat band for the midrange frequencies.

This kind of filter is commonly known as a Butterworth filter. The typical filter used to fulfill these requirements is a parallel LC circuit. When connected in parallel to a load R_L, it actually can be seen as an RLC filter.

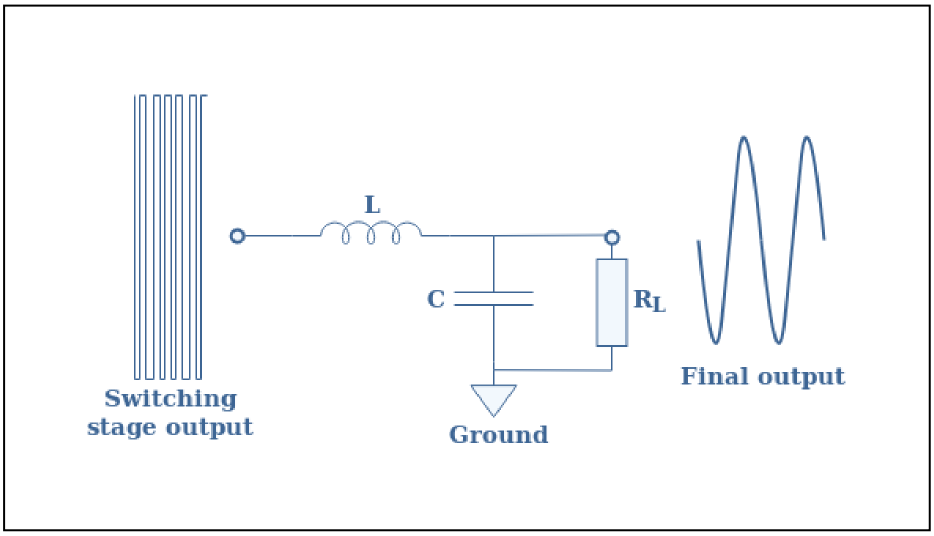

Figure 1.22: Low-pass L//C filter

The bandwidth of this filter is characterized by its cutoff frequency f_c at -3 dB that satisfies the **Equation 1.4** :

$$\boldsymbol{F_C = \frac{1}{2\pi\sqrt{LC}}} \qquad \textbf{1.4}$$

Moreover, since the RLC circuit is a second-order filter, a strong roll-off of -40 dB/dec is observed above f_c. An asymptotic diagram of the frequency response of this filter is given is Figure 1.23 :

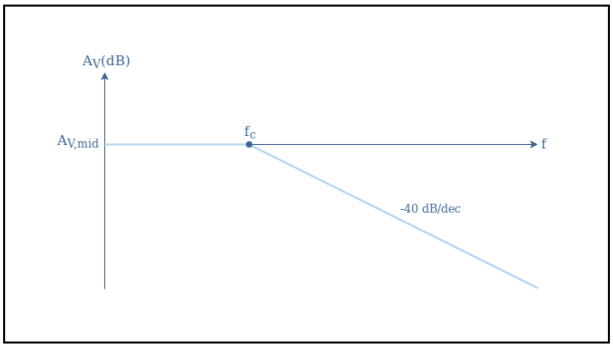

Figure 1.23 : Second order Butterworth filter frequency response

Several levels of parallel LC configurations are appreciated since each level increases the order of the filter and therefore the quality of filtration. In Figure 1.24, we can note the difference between the output resulting of a first or second order Butterworth filter applied to our example :

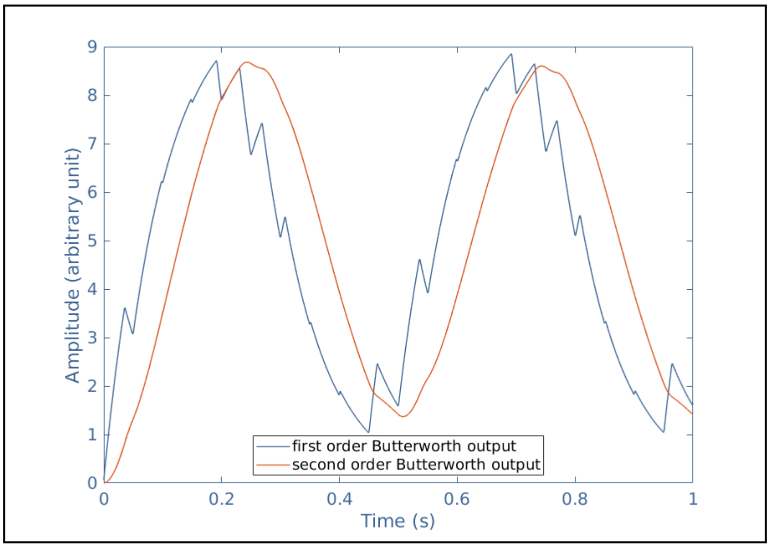

Figure 1.24 : Difference of output between a first and second order Butterworth filter

Since the input signal has a frequency of 2 Hz and the carrier frequency is 20 Hz, a cutoff frequency of 4 Hz has been chosen for this filter. We can highlight the fact that a first order filter is not appropriate since it does not attenuate the carrier frequency enough while the second order filter output is much more sinusoidal[5].

1.12 More performed stages

1.12.1 Integrator

An integrator is a circuit that performs a mathematical operation called integration. The most common application of an integrator is to produce an output voltage ramp, which is a linearly increasing or decreasing voltage. The integrator is sometimes called the Miller integrator, after its inventor.

An op-amp based integrator produces an output, which is an integral of the input voltage applied to its inverting terminal. The circuit diagram of an op-amp based integrator is shown in the following figure

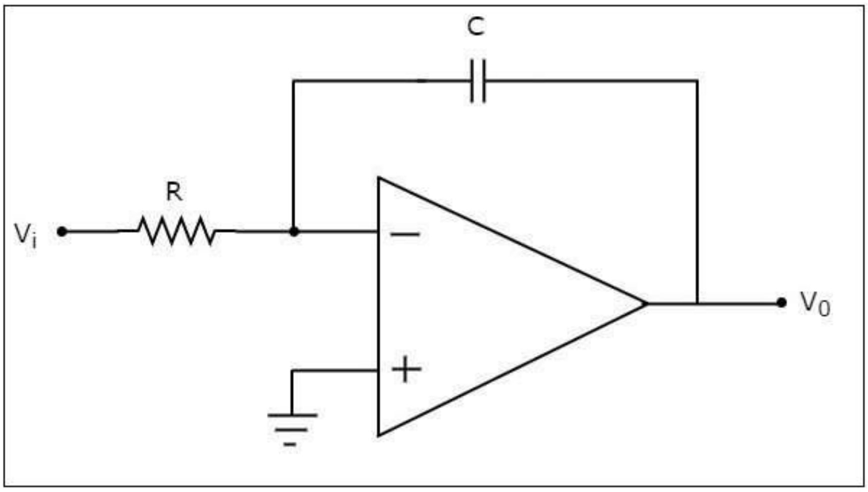

Figure 1.25 : circuit diagram of an op-amp based integrator

In the circuit shown above, the non-inverting input terminal of the op-amp is connected to ground. That means zero volts is applied to its non-inverting input terminal.

According to virtual short concept, the voltage at the inverting input terminal of op-amp will be equal to the voltage present at its non-inverting input terminal. So, the voltage at the inverting input terminal of op-amp will be zero volts.

The nodal equation at the inverting input terminal is

$$\mathbf{dV_0} = (-\frac{\mathbf{V_i}}{\mathbf{RC}})\mathbf{dt} \qquad \mathbf{1.5}$$

Integrating both sides of the equation shown above, we get

$$\mathbf{V_0} = -\frac{\mathbf{1}}{\mathbf{RC}}\int \mathbf{V_t dt} \qquad \mathbf{1.6}$$

So, the op-amp based integrator circuit discussed above will produce an output, which is the integral of input voltage Vi, when the magnitude of impedances of resistor and capacitor are reciprocal to each other[7].

1.12.2 Feedback

The feedback-amplifier can be defined as an amplifier which has feedback lane that exists between o/p to input. In this type of amplifier, feedback is the limitation which calculates the sum of feedback given in the following amplifier. The feedback factor is the ratio of the feedback signal and the input signal.

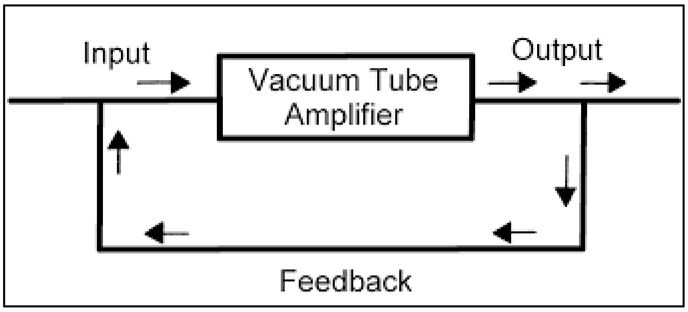

Figure 1.26 : Feedback Amplifier

1.12.2.1 Types of feedback amplifier

The procedure of introducing some device's output energy fraction from back to the i/p is termed as Feedback. This is mainly used to reduce the noise as well as to make the operation of an amplifier is constant. This amplifier can be classified into two types based on the feedback signal helps such as positive and negative feedback amplifier.

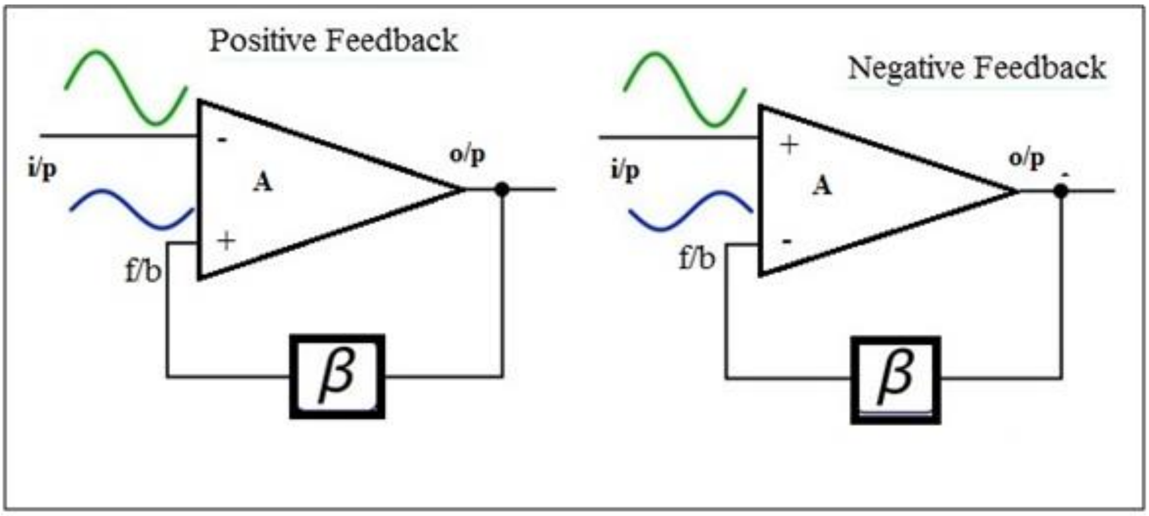

Figure 1.27: Positive and Negative Amplifiers

1.12.2.1.1 Positive feedback amplifier

The positive feedback can be defined as when the feedback current otherwise voltage is applied for increasing the i/p voltage, then it is named as positive feedback. Direct feedback is another name of this positive feedback. Because positive feedback generates unnecessary distortion; it is not often used in amplifiers. But, it amplifies the original signal power and can be used in oscillator circuits.

1.12.2.1.2 Negative feedback amplifier

The negative feedback can be defined as if the feedback current otherwise voltage can be applied for reducing the amplifier i/p, then it is called as negative feedback. Inverse feedback is another name of this negative feedback. This kind of feedback is regularly used in amplifier circuits[8].

1.13 An amplifier performance parameters

1.13.1 Efficiency

The original way of functioning of the class D amplifier brings its efficiency to very high levels. This high efficiency is explained by the fact that the transistors behave nearly as ideal switches :

- When they are OFF, no current I_{DS} flows between the drain and the source.
- When they are ON, no voltage V_{DS} is observed across the drain and the source.

Therefore, no power $V_{DS} \times I_{DS}$ is dissipated in form of losses (heat). Typically the efficiency of class D amplifiers is above **90 %**[5].

$$\eta_D = \frac{R_L}{R_L + R_{DSon} + R_{in} + \frac{2 I_r^2 R_L^2 (R_{DSon} + R_{in})}{\hat{V}_0^2}} \tag{1.7}$$

1.13.2 Total Harmonic Distortion Plus Noise THD+N

The THD+N technique is the most common method of measuring harmonic distortion. In its basic form, it is implemented with a sharp notch filter tuned to the fundamental sine frequency, a bandwidth limiting filter, and an rms level meter **Figure 1.28** . The notch filter and bandwidth limiting filter can also be realized with FFT analysis. The notch filter removes the fundamental sine signal, leaving a residual signal which consists of the harmonics and noise. The THD+N Ratio is calculated as the bandwidth limited rms level of the residual divided by the rms level of the entire signal.

THD+N can be expressed as follows:

$$\mathrm{THD+N} = \sqrt{\frac{\sum_{i=2}^{N} V_i^2 + V_n^2}{V_1}} \tag{1.8}$$

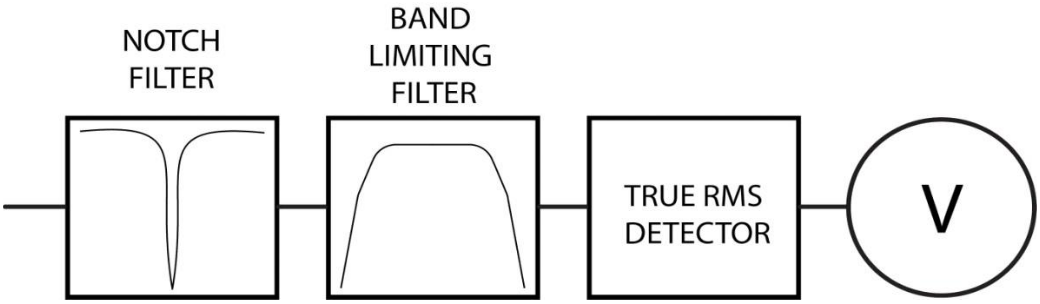

Figure 1.28 : Block diagram of a THD+N analyzer.

THD+N ratio is usually expressed either in percent or dB. Percent is commonly used for THD+N levels of 0.01% or higher. For very low distortion, dB units are easier to work with. For example, the typical residual distortion of an APx555 audio analyzer's analog I/O **Figure 1.29** is -120 dB or 0.0001%.

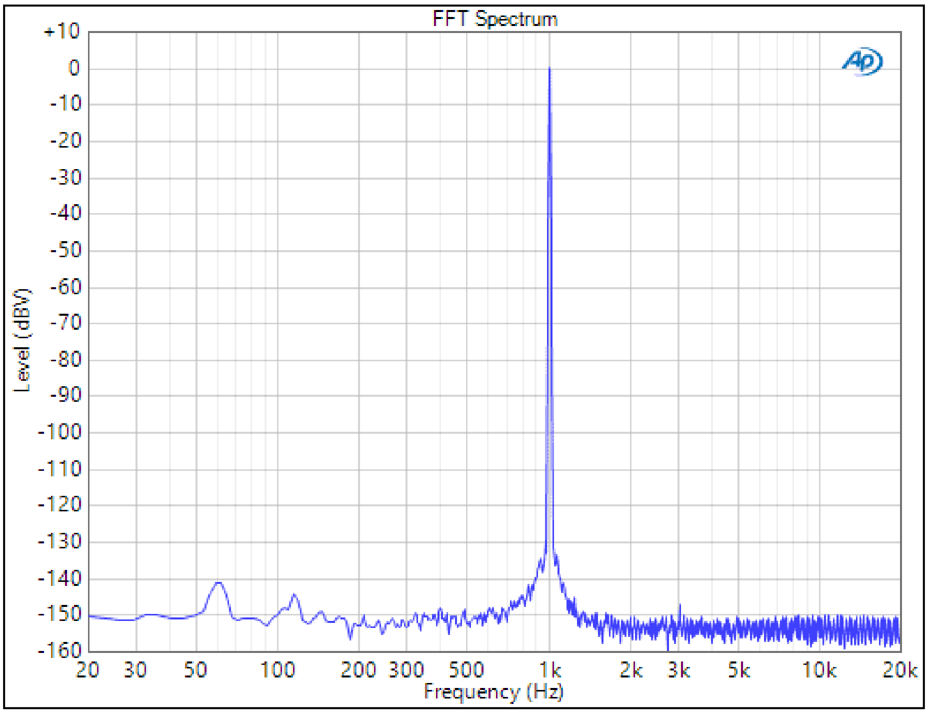

Figure 1.29 : FFT of a stimulus signal from the APx555 (High Performance Sine Generator, 1 Vrms sine at 1 kHz

Sometimes a weighting filter is applied to the residual, which will lower the measured THD+N ratio. For example, an A-weighting filter might be used to hide the presence of hum artifacts at multiples of the AC mains voltage frequency (50 or 60 Hz). If a weighting filter is used, this should be clearly stated with the reported THD+N result.

Conceptually, the THD+N measurement technique is appealing because anything in the DUT output signal beside the pure stimulus signal will degrade the measurement result. A low THD+N result indicates not only that harmonic distortion is low, but also that hum, interference signals and broadband noise introduced by the DUT are at or below the low measured value.

When THD+N measurements results are stated or specified, to be meaningful, details such as the stimulus level and frequency (or frequency range), the measurement bandwidth and the DUT gain, should be included[9].

1.13.3 Total Harmonic Distortion THD

A THD measurement is similar to a THD+N measurement, except that the analysis considers only the harmonic distortion components and ignores everything else. Hence, the test setup is inherently more complicated than the THD+N technique. In the early days of audio test, this would require an analyzer with a tunable narrow bandpass filter, that could be tuned to each harmonic in serial fashion. Currently, THD is derived from FFT spectrum analysis **Figure 1.30** below.

Knowing the relative strength of harmonic components can be a useful design tool, since different non-linearity mechanisms can give rise to different patterns of harmonic distribution (odd versus even numbered harmonics, etc.). The THD+N measurement in APx has a convenient feature that clearly displays the fundamental and the first 10 harmonics in a result called the Distortion Product Ratio **Figure 1.31** below, while simultaneously measuring both THD+N and THD[9].

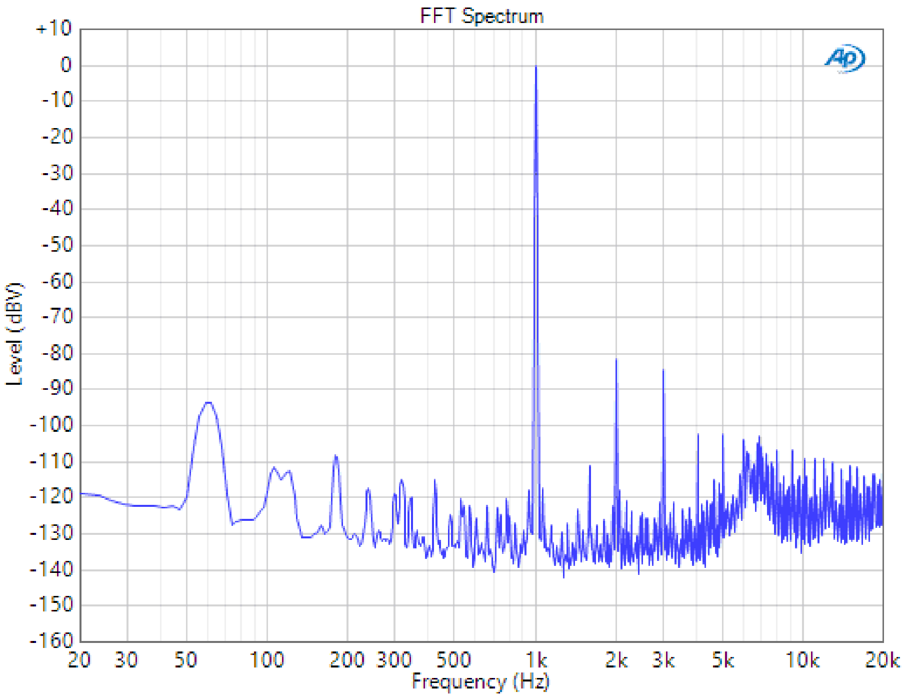

Figure 1.30 : FFT of DUT output signal (APx1701, 20 dB gain, 8-ohm load).

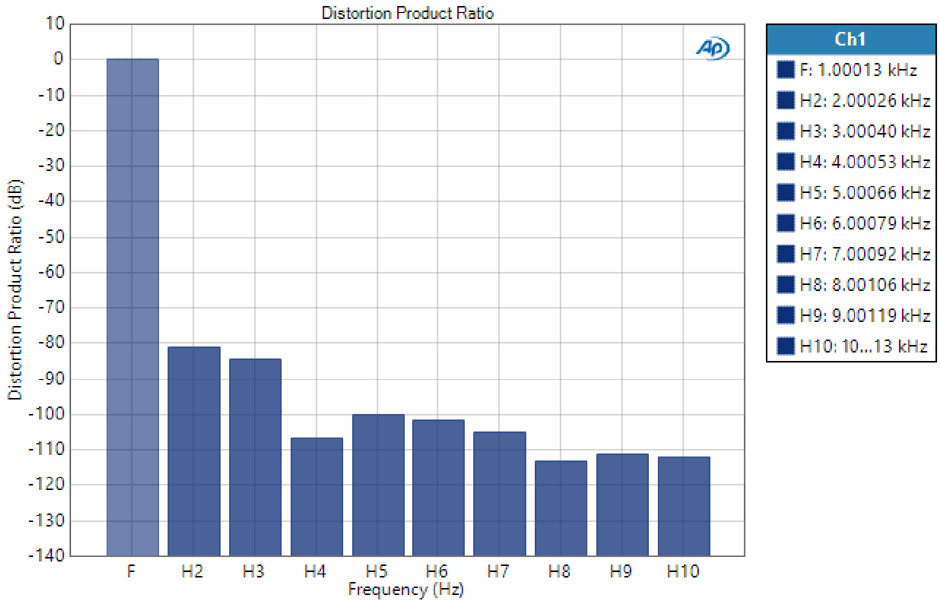

Figure 1.31: Distortion Product Ratio (right) of DUT output signal (APx1701, 20 dB gain, 8-ohm load).

The THD Ratio is calculated as the RSS summation (square root of the sum of squares) of the harmonic components divided by the rms level of the fundamental signal. Like THD+N, it is typically expressed in percent or dB.

As noted above, many publications incorrectly use the term THD when referring to a THD+N test (even the recently published IEC standard on audio amplifiers does so). It's important to make the distinction between the two, because the THD ratio will always be less than the THD+N ratio.

Note that the number of harmonics included in the analysis is important, and the analysis bandwidth of the audio analyzer comes into play in this regard. For example, if the input bandwidth is, say 22 kHz, the third harmonic of the signal will only be included up to fundamental frequencies of about 7 kHz and the second harmonic up to about 11 kHz.

When THD results are reported, the same measurement details as THD+N should be included, except that the number of harmonics included replaces the measurement bandwidth.

1.14 Conclusion

At the conclusion of this section, it can be concluded that power amplifiers are an essential element in audio and electronics technologies, offering effective signal amplification with various features that affect sound quality and energy efficiency. Power amplifier classes offer different options to suit the needs of different applications.

THD and power efficiency are critical determinants of power amplifier performance, and analyzing these parameters helps select the right amplifier to ensure high audio quality and efficient power consumption. Based on these characteristics, power amplifier designs and applications can be optimized to meet the needs of today's evolving audio and electronic systems.

Chapter II :PWM and PDM modulation

2.1 Introduction

Pulse Density Modulation (PDM) and Pulse Width Modulation (PWM) are two critical techniques used in signal processing, particularly in the fields of digital audio, power electronics, and telecommunications. Both methods convert analog signals into digital forms, enabling efficient transmission, storage, and manipulation. PDM encodes the amplitude of the signal into the density of pulses over time, resulting in a stream where the pulse density is proportional to the signal's amplitude. This technique is particularly prevalent in digital microphones and audio applications. On the other hand, PWM encodes the signal's amplitude by varying the width of pulses within a constant frequency signal, making it widely used in motor control, power delivery systems, and LED dimming. Understanding the nuances of PDM and PWM is essential for engineers and technologists who design systems that rely on precise signal modulation and control.

2.2 Pulse Width Modulation (PWM)

Pulse width modulation, often referred to as PWM, is a modulation method that modulates the average power delivered to a load by varying the width of a pulse signal in an electrical system. PWM is particularly helpful for efficiently regulating the power of audio amplifiers, the speed of motors, and the brightness of light. They are commonly used in ASICs for microcontrollers and PWM controllers.

2.3 The Generation Of PWM

A pulse width modulating signal is generated using a comparator. The modulating signal forms one part of the input to the comparator, while the non-sinusoidal wave or triangle signal forms the other part of the input. The comparator compares two signals and generates a PWM signal as its output waveform.

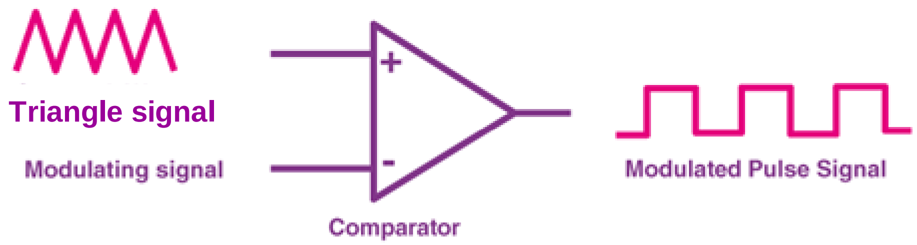

Figure 2.1 : PWM Signal Generator

If the triangle signal is more than the modulating signal, then the output signal is in a "High" state. The value of the magnitude determines the comparator output which defines the width of the pulse generated at the output[10].

2.4 Important Parameters Associated with PWM Signal

An analog circuit can be controlled using digital pulses produced by a PWM signal. The behavior of a PWM signal is determined by these factors:

2.4.1 Duty Cycle of PWM

The fraction of a second that a signal or system is operational is called a duty cycle. A duty cycle is usually expressed as a percentage or ratio. The amount of time a signal takes to complete an ON-OFF cycle is called a period. The proportion of time a digital signal is on throughout a period of time or interval is precisely described by the percentage duty cycle. The waveform's time is equal to its inverse frequency.

- **Duty Cycle:** On Time / (On Time + Off Time)

We would say a digital signal has a 50% duty cycle and looks like a perfect square wave if it is on for half of the time and off for the other half. The digital signal spends more time in the high state than the low state if the percentage is greater than 50%, and vice versa if the duty cycle is lower than 50%. Here is a graph depicting the three scenarios.:

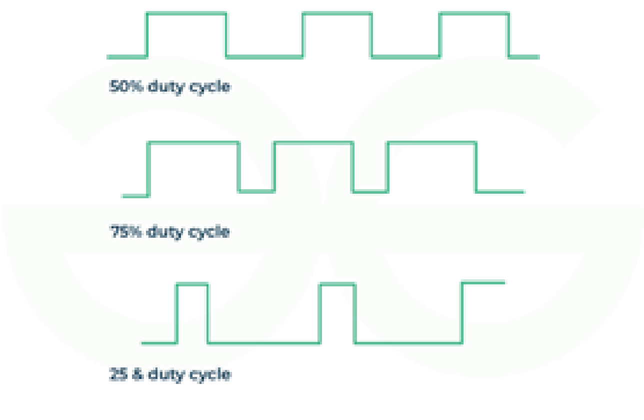

Figure 2.2 : Duty Cycle of PWM

A 100% duty cycle is equivalent to a 5 volt (high) voltage setting. Grounding the signal would be equivalent to 0% duty cycle[11].

2.4.2 Frequency of PWM

The Speed at which something occurs over a specific time period is known as its frequency. In another way, the rate at which a vibration occurs that results in a wave, such as radio, light, or sound waves; this rate is usually measured in seconds.It is easy to define a frequency or period for regulating a particular servo.

Data is currently transmitted over communication channels using PWM's duty cycle in an unambiguous communication system. PWM serves as a technique for converting high-frequency pulses into low-frequency output signals[11].

Frequency : 1 / Time Period

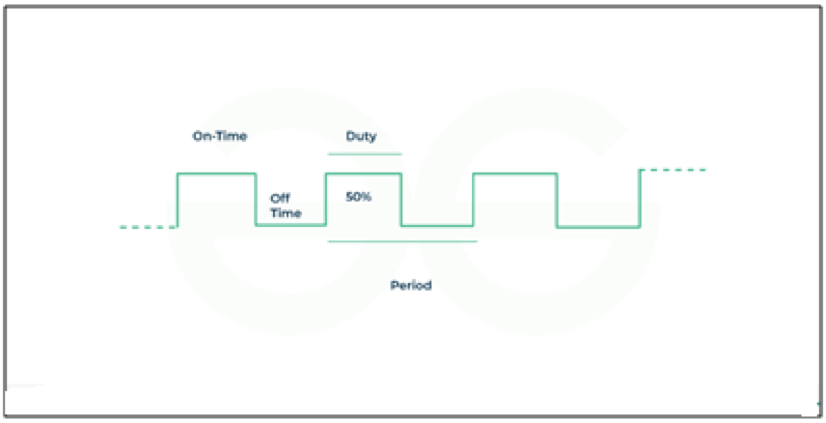

Figure 2.3 : Frequency of PWM

2.4.3 Output Voltage of PWM signal

It is the percentage of the duty cycle and can be calculated in that way only by calculating its percentage also. Let's say the duty cycle is 100% then the output voltage will be 5V[11].

2.5 Types of PWM

There are 4 types of Pulse Width Modulation, such as

- Single-pulse width modulation
- Multiple-pulse width modulation
- Sinusoidal pulse width modulation
- Hysteresis band pulse width modulation

2.5.1 Single-Pulse Width Modulation (PWM)

A single pulse is produced at each switching cycle in single-pulse width modulation. The average power applied to the load is controlled by varying the pulse's width. Single-

PWM is straightforward and simple to use, although it could have a larger harmonic content and be unsuitable for applications that need precision control at low power levels[11].

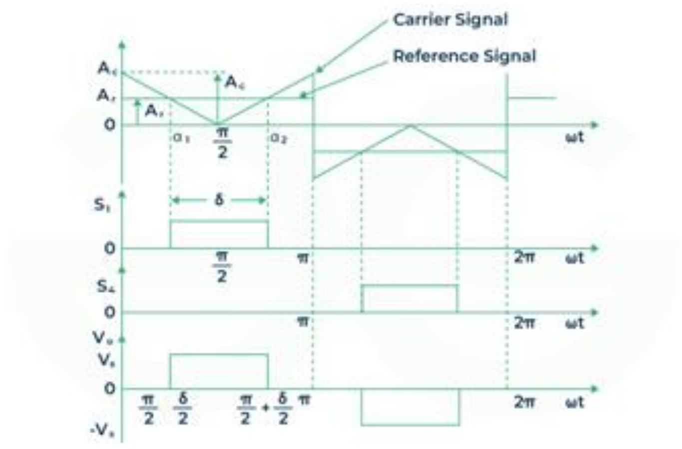

Figure 2.4 : Single-pulse width modulation

2.5.2 Multiple-Pulse Width Modulation (MPWM)

Multiple pulses are generated during each switching cycle in multiple-pulse width modulation. This method seeks to lower harmonic distortion while raising the output waveform’s general quality.

Two-level and three-level MPWM are frequently used; in these implementations, the number of pulses in each cycle is increased to improve waveform fidelity, which makes MPWM especially helpful in high-power applications [11].

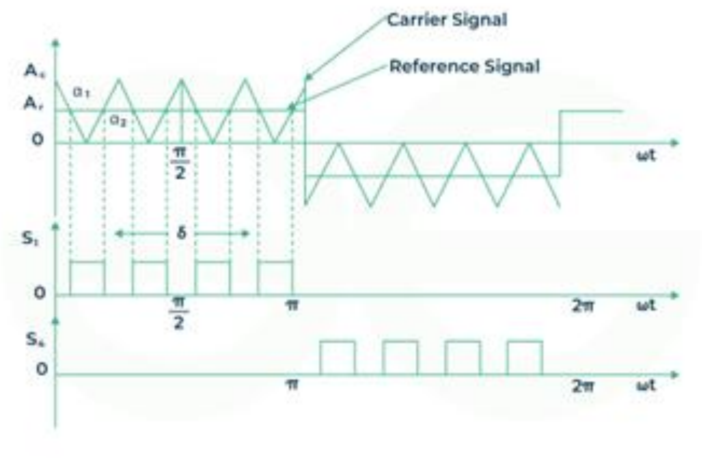

Figure 2.5 : Multiple-pulse width modulation

2.5.3 Sinusoidal Pulse Width Modulation (SPWM)

The process of altering pulse width to resemble a sinusoidal waveform is known as sinusoidal pulse width modulation. Through the use of a sine wave reference to alter the pulse widths, SPWM minimizes harmonic distortion and yields a smoother output.

Inverters and motor control systems, where a high-quality output waveform is crucial for lowering harmonic interference and raising system efficiency, are two common applications for this technology[11].

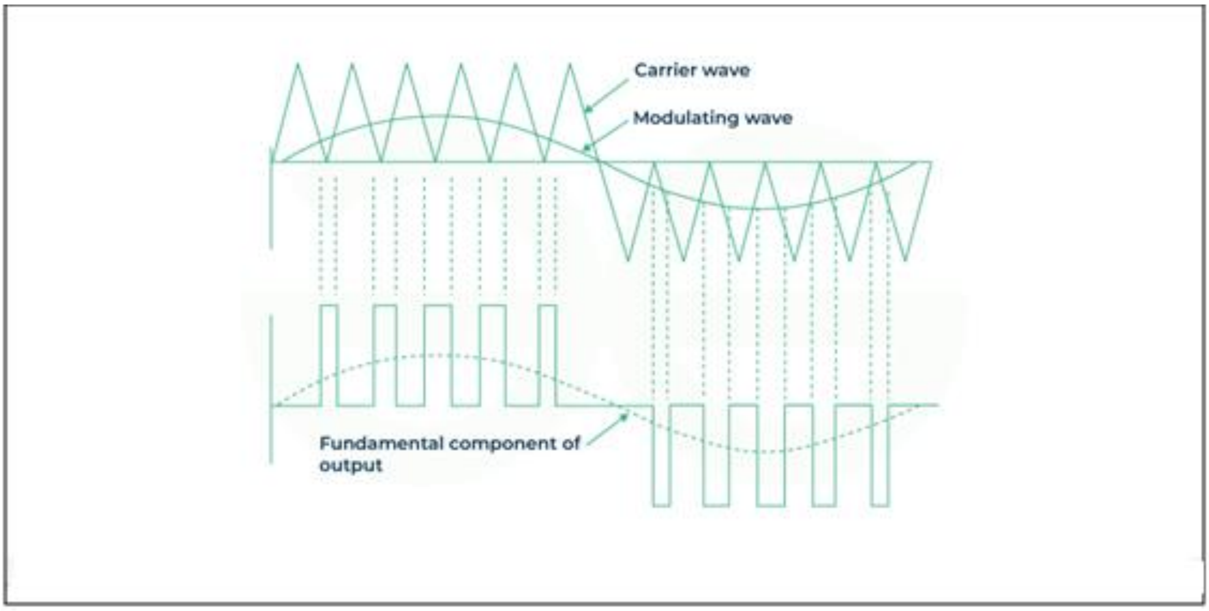

Figure 2.6 : Sinusoidal pulse width modulation

2.5.4 Hysteresis Band Pulse Width Modulation

Hysteresis Band Pulse Width Modulation is the comparison of the erroneous signal with a predetermined band, called hysteresis. The pulse width is changed to put the system back into balance when the erroneous signal rises above the hysteresis range.

This approach provides a straightforward and reliable control strategy, especially in applications where sudden variations in load or disturbances are frequent. Power converters and voltage regulators frequently use hysteresis band PWM because to its built-in noise immunity and simplicity of use[11].

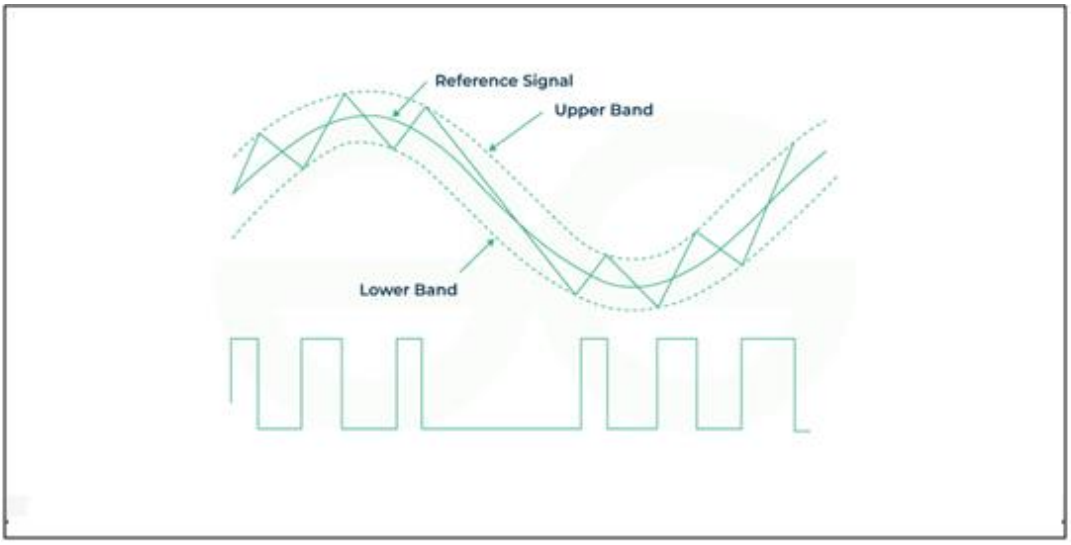

Figure 2.7: Hysteresis band pulse width modulation

Applications of PWM

Due to the high efficiency, low power loss, and the PWM technique's ability to precisely control the power, the technique is used in a variety of power applications. Some of the applications of PWM are as follows:

- The pulse width modulation technique is used in telecommunication for encoding purposes.
- The PWM helps in voltage regulation and therefore is used to control the speed of motors.
- The PWM technique controls the fan inside a CPU of the computer, thereby successfully dissipating the heat.

PWM is used in Audio/Video Amplifiers[10].

2.7 Advantages and Disadvantages of PWM

2.7.1 Advantages

- PWM technique prevents overheating of LED while maintaining its brightness.
- PWM technique is accurate and has a fast response time.
- PWM technique provides a high input power factor.
- PWM technique helps motors generate maximum torque even when they run at lower speeds[10].

2.7.2 Disadvantages

- The significant switching losses are a result of the high PWM frequency.

– Radio Frequency Interference is caused by it (RFI)[11].

2.8 PWM-BASED CLASS-D AUDIO AMPLIFIER TECHNIQUE EXPLANATION

While there are a variety of modulator topologies used in modern Class D amplifiers, the most basic topology utilizes pulse-width modulation (PWM) with a triangle-wave (or sawtooth) oscillator. Figure 2.8 shows a simplified block diagram of a PWM-based, half-bridge Class D amplifier. It consists of a pulse-width modulator, two output MOSFETs, and an external lowpass filter (L_F and C_F) to recover the amplified audio signal. As shown in the figure, the p-channel and n-channel MOSFETs operate as current-steering switches by alternately connecting the output node to V_{DD} and ground. Because the output transistors switch the output to either V_{DD} or ground, the resulting output of a Class D amplifier is a high-frequency square wave. The switching frequency (f_{SW}) for most Class D amplifiers is typically between 250kHz to 1.5MHz. The output square wave is pulse-width modulated by the input audio signal. PWM is accomplished by comparing the input audio signal to an internally generated triangle-wave (or sawtooth) oscillator. This type of modulation is also often referred to as "natural sampling" where the triangle-wave oscillator acts as the sampling clock. The resulting duty cycle of the square wave is proportional to the level of the input signal. When no input signal is present, the duty cycle of the output waveform is equal to 50%. Figure 2 illustrates the resulting PWM output waveform due to the varying input-signal level[12].

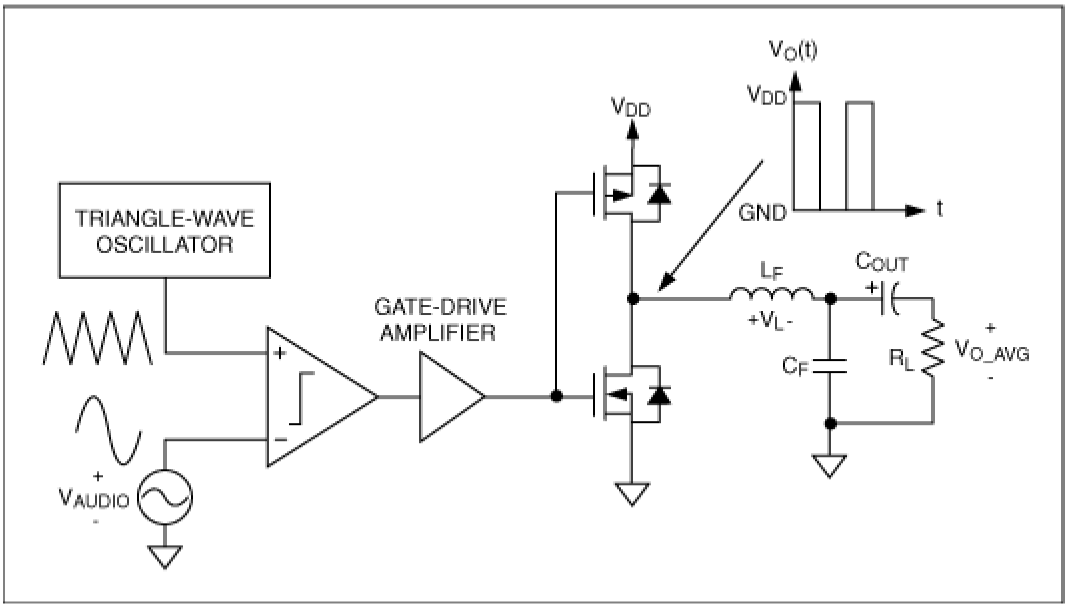

Figure 2.8 : This simplified functional block diagram illustrates a basic half-bridge Class D amplifier.

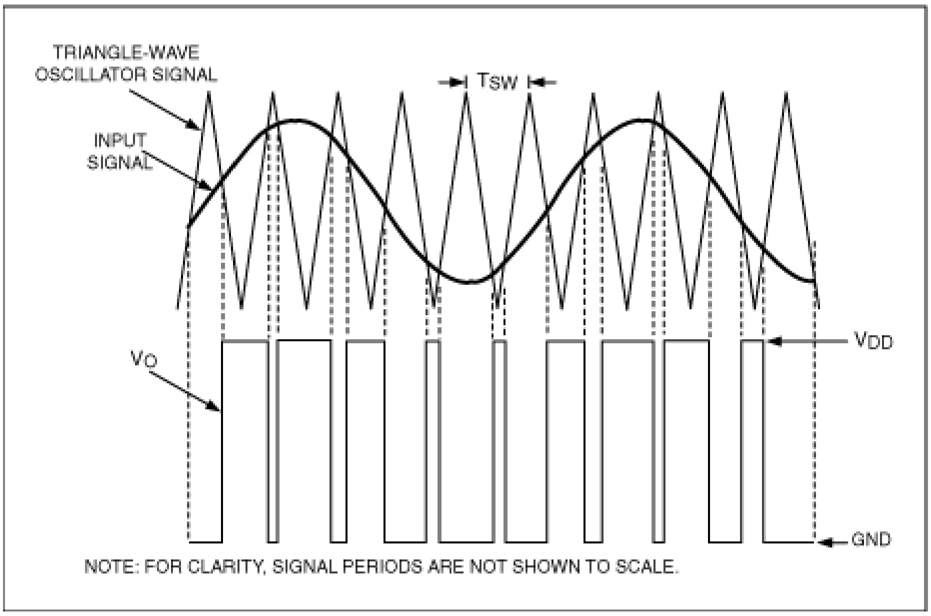

Figure 2.9 : The output-signal pulse widths vary proportionally with the input-signal magnitude.

2.9 Pulse Density Modulation (PDM)

Pulse-density modulation, or PDM, is a form of modulation used to represent an analog signal with a binary signal. In a PDM signal, specific amplitude values are not encoded into codewords of pulses of different weight as they would be in pulse-code modulation (PCM); rather, the relative density of the pulses corresponds to the analog signal's amplitude. The output of a 1-bit DAC is the same as the PDM encoding of the signal[13].

2.10 Basic Principles of PDM

PDM represents an analog signal by modulating the density of pulses. A higher density of pulses corresponds to a higher amplitude of the analog signal, while a lower density indicates a lower amplitude. Each bit in a PDM signal is either a '1' (pulse) or a '0' (no pulse), with the average density of '1's over time corresponding to the analog signal's amplitude.

2.11 Applications of PDM

2.11.1 Digital Microphones

PDM is commonly used in MEMS (Micro-Electro-Mechanical Systems) microphones due to its ability to provide high-quality audio while maintaining low power consumption.

2.11.2 Audio Processing

PDM signals are often converted to PCM for further audio processing and playback in digital audio systems.

2.11.3 Communication Systems

In certain communication systems, PDM can be employed for encoding and transmitting data with robustness against noise.

2.12 Advantages and Disadvantages of PDM

2.12.1 Advantages

Simple Encoding and Decoding

Efficient Digital Representation

Low-Latency Processing

Noise Shaping

Direct Compatibility with Delta-Sigma ADCs and DACs

Power Efficiency

2.12.2 Disadvantages

High Data Rate

Complex Filtering Requirements

Sensitivity to Noise

Reduced Compatibility with Standard Digital Systems

Higher Power Consumption for High Data Rates

2.13 Pdm-based class-d audio amplifier technique explanation

2.13.1 First-order sigma-delta modulation

We picked the first order as it is the focus of our comparative study.

Sigma-Delta modulation: The voltage change (Delta) at the analog input is integrated into a pulse train (Sigma) and subsequently forwarded to digital signal processing. This modulation is also known as PDM "pulse density modulation", where the density of the pulses determines the strength and frequency of the digital signal due to the integrating function of the analog-to-digital converter

The principle of PDM is to apply a high frequency clock signal to sample and hold the compensator output. The quantizer block is used to convert the voltage information into a pulse density code. The quantizer output is amplified in the class-D output stage and applied to the output filter. More recently, the switching output has been used as a feedback signal and fed back to the compensator input. Authors in [14] demonstrated the pdm waveform from their proposed first-order sigma-delta circuit as illustrated in figure 2.10.

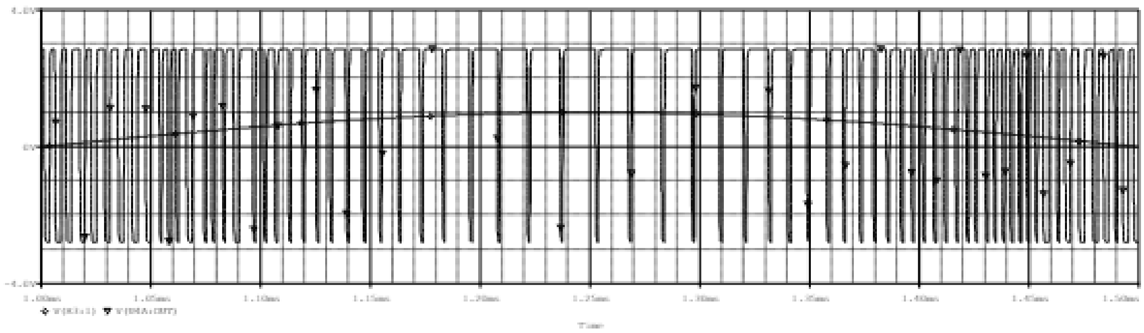

Figure 2.10 : PDM waveform from [14] proposed circuit

Figure 2.11 shows the functional sigma-delta modulator block diagram. It generates at the output, a digital pulse train signal, where the width of the impulses, is sampled as multiple the period clock.

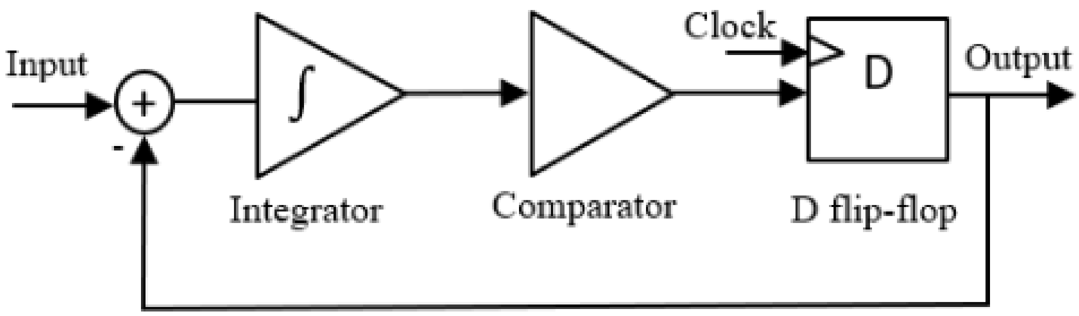

Figure 2.11 : Functional block diagram of a first-order Sigma-delta modulator

The sigma-delta design consists of three consecutive components: an integrator, a comparator, and a buffer (D flip-flop). Research has found that nonlinearity can be reduced by using higher order modulators or increasing the sampling frequency. These technologies also come at a high cost as hardware complexity and power consumption increased[15].

2.14 Pdm-Based Class-D Audio Amplifier Stages Explanation

2.14.1 Quantizer

It is composite of a D flip-flop and a comparator. The clock frequency is steady and high. The comparator compares the integrator output with grounded signal resulting pulses in the output. These pulses are with constant width and inconstant period. The D latch lifts the comparator output and clears glitches.

2.14.2 Power stage (switching stage)

The half bridge circuit consists of an upper and lower MOSFETs connected in a cascade arranging PMOS/NMOS as the circuit presented in this paper. This configuration is also called a “totem pole” since the MOSFETs are stacked on top of each other. The high side MOSFET is utilized to switch the positive supply to the load for forward direction while the low side MOSFET is utilized to switch the negative supply to the load for reverse direction.presented on [14] this stage amplifies signal, dissipating less power since it is switches, negligible current circulates in “Off’’ state and negligible voltage is present in “On’’ state (very small $Rdson$ resistors). The MOSFETs are polarized using a gate drive.

2.14.3 Compensator

To conquer the PDM drawbacks, mainly total harmonic distortion, we use negative feedback loop. A couple of resistors as a negative feedback loop added to a second-order integrator consists of two capacitors, and one resistor, are used to build the compensator.

2.14.4 Low-pass filter

Since the signal comprises high frequency carrier and its harmonics, a low-pass filter is required to block and eliminate any frequencies higher than 20 kHz. Second-order Butterworth LC filter is the one used as it is simple and highly efficacious. As more, it is designed with the flat response in the audio spectrum pass-band. It is performed by using an inductor and a capacitor in parallel with the load[14].

2.15 Conclusion

In summary, both Pulse Density Modulation (PDM) and Pulse Width Modulation (PWM) serve crucial roles in modern electronic systems by enabling efficient and accurate signal processing and control. PDM, with its density-based encoding, is particularly suited for high-fidelity audio applications and digital microphones, offering a streamlined approach to capturing and transmitting sound. Conversely, PWM's method of varying pulse widths at a consistent frequency provides robust solutions for controlling power in applications ranging from motor drives to lighting systems. Each technique has its unique advantages and optimal use cases, and the choice between PDM and PWM depends on the specific requirements of the application, such as signal fidelity, power efficiency, and system complexity. As technology advances, the integration and optimization of these modulation methods continue to drive innovation in various fields, underscoring their enduring significance in the digital age.

Chapter III: Simulation , Results and Discussions

3.1 Introduction

Class D audio amplifiers have emerged as a preferred choice for various audio applications due to their high efficiency and compact design. Among the key components influencing the performance of Class D amplifiers are the modulation techniques employed to encode audio signals into digital pulses. Pulse Width Modulation (PWM) and Pulse Density Modulation (PDM) are two widely used modulation schemes in Class D amplifiers, each offering distinct advantages and challenges.

This chapter aims to comparison between PWM and PDM modulation techniques in Class D audio amplifiers , and analyze various performance metrics, including efficiency and distortion characteristics.

3.2 PWM-BASED CLASS-D AUDIO AMPLIFIER CIRCUIT

The circuit shown in **Figure 3.1**. It consists of: An input sine wave representing the audio signal, A second-order integrator with increased bandwidth to optimize THD,Comparator with low propagation delay,High-performance triangular carrier wave ,Two gate drivers based on bipolar technology: level-shifted for P-channel MOSFET and totem pole driver for N-channel MOSFET,The power stage consists of two MOSFETs with low operating resistance (R_{DSon}) compared to the input impedance of the output filter, Butterworth LC Low-pass second-order filter , A negative RC network proposed as a feedback loop and as a high-frequency cutoff filter,Resistive load as a low-power speaker[15].

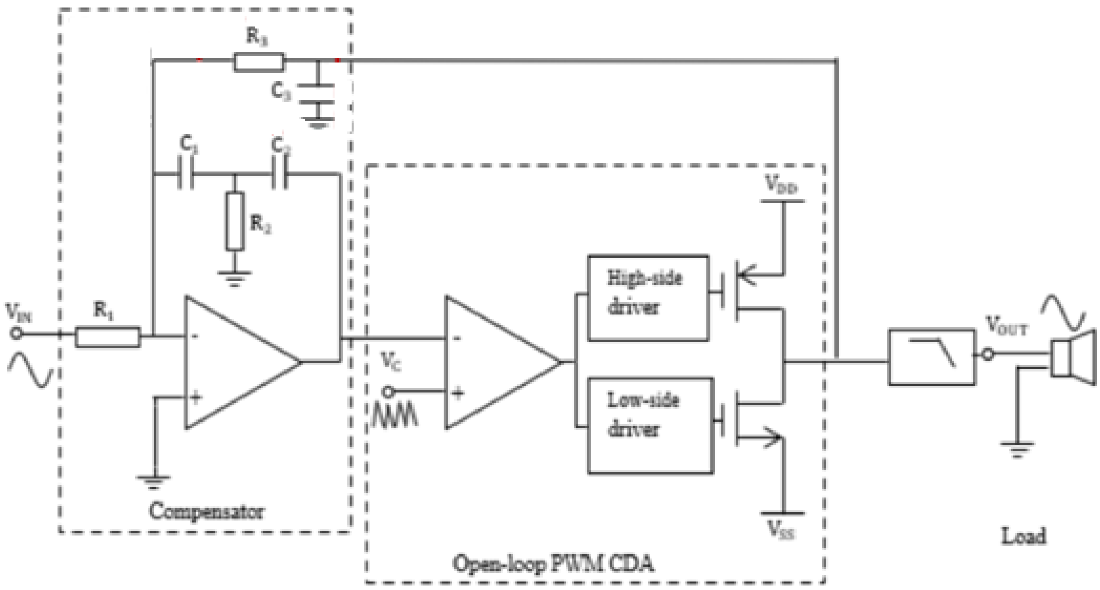

Figure 3.1: Block diagram of PWM class-D amplifier

the feedback signal is compared with the input signal, using a second-order integrator, to produce an error signal. The comparator input terminals are connected to the error signal and to the carrier (high-frequency triangular signal) to compare together and produce a switch manipulator signal to drive the switching stage which drives load and LC filter .

This audio amplifier operates with power supplies $\pm$3 V and gives 90 mW RMS output power. The comparator MAX942 combines high speed, low power and rail-to-rail input. A low power amplifier TL061 used to build second-order integrator. The power stage composes fast-switching NMOS and PMOS MOSFETs, IRF520 and IRF9530. Bipolar transistors: 2N3904 and 2N3906 used to build gate drivers. The architecture reduces overlapping of MOSFETs inputs, guarantees power, and voltage-level adaptation and decreasing switching time by conveniently discharging different MOSFETs capacitors. The picked MOSFET transistors have very small “on” output resistance and small gate capacitance for low dissipation in portable applications .

The feedback loop consists of a resistor and a capacitor. It is used as a negative feedback loop added to second-order integrator provided with two capacitors and a resistor. It operates as a low-pass filter for undesirable high frequencies. This closed-loop attenuates extremely the total harmonic distortion plus noise produced by power stage , reduces nonlinearities and non-idealities, minimizes pulse-height errors, and mitigates ripples[15].

3.3 PDM-BASED CLASS-D AUDIO AMPLIFIER CIRCUIT

The circuit shown in **figure 3.2**. The circuit is powered by $\pm$3 V DC voltage.They used three ICs: a MAX942 comparator, a 74HC74 D latch, and a TL061 with two capacitors and a resistor to build a second order integrator. They connected the negative feedback combination of two resistors as voltage driver to the integrator. The additional resistor $R4$ shown in Figure 3.2 plays an important role in removing the nonlinearity of the output signal as it controls the loop gain, thereby reducing THD. As power switches they used IRF520 and IRF9530 MOSFETs. The output filter is a second order LC low pass filter, Speaker are ohmic loads[14].

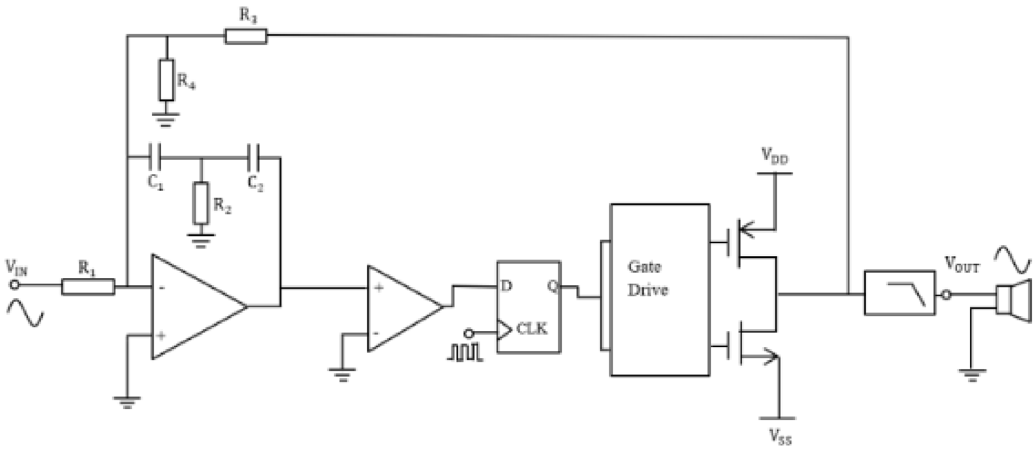

Figure 3.2: Block diagram of PDM class-D amplifier

As mentioned above, thiscircuit is powered by ±3 VDC voltage sources. ICs Used: MAX942 Comparator used for sort of comparison or signal conditioning. 74HC74 D latch used as a data storage element, typically in digital circuits. TL061 Op-Amp used to build a second-order integrator. With the capacitors and resistors, it forms part of a filter or signal processing stage.Second-order Integrator used for filtering or signal processing. With negative feedback, it can help control the frequency response of the amplifier. Comprising two resistors used for negative feedback in the integrator stage, it helps in stabilizing and controlling the output signal.MOSFETs (IRF520 and IRF9530) used as power stage switches. They control the amplification or switching of the signal.Output Filter (LC Second-order Low-pass Filter) used to filter out high-frequency noise and maintain the desired frequency response. It's designed as a low-pass filter.Speaker is the final output device where the amplified or processed signal is delivered. It's described as an 8 Ω resistive load[14].

3.4 Results and discussions

3.4.1 PWM-BASED CLASS-D AUDIO AMPLIFIER

The output signal is 1.2 V V_{max} and the input signal is 0.9 V V_{max} at 1 kHz frequency

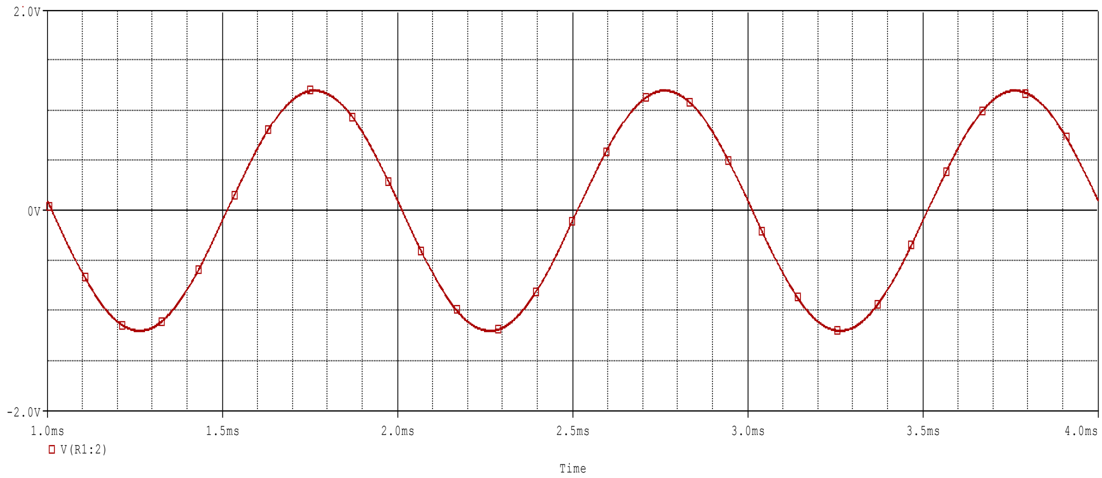

Figure 3.3: Output filtered PWM signal

In **Figure 3.3** : The drop in this voltage in comparison to the full dynamic range is due the output impedance of the LP filter which also includes the effect of the R_{DSon}resistance and mainly the closed-loop

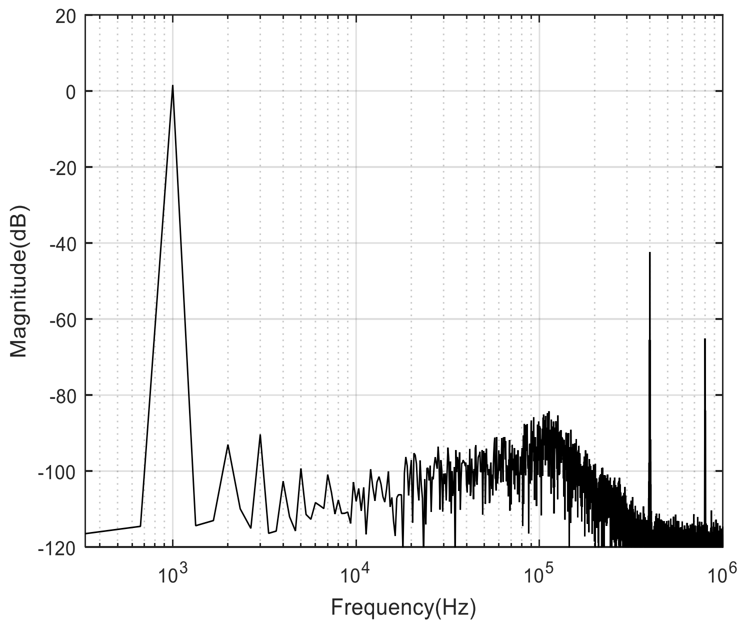

Figure 3.4: Output signal spectrum

The lowlest THD is 0.003%, obtained at 1 kHz and attains its maximum at 10 kHz with 0.03%.

We can see from **Figures 3.3** and **3.4** that the proposed circuit fulfills an efficiency of 90%. The obtained THD is 0.003 % with 400 kHz switching frequency.

3.4.2 PDM-BASED CLASS-D AUDIO AMPLIFIER

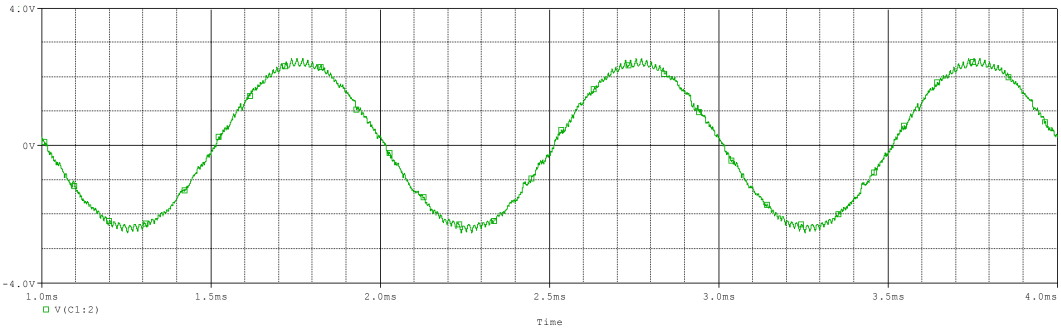

Figure 3.5: Output filtered PDM signal

The output signal is 2.5 V $Vmax$ and the input signal is 1 V $Vmax$ at 1 kHz frequency

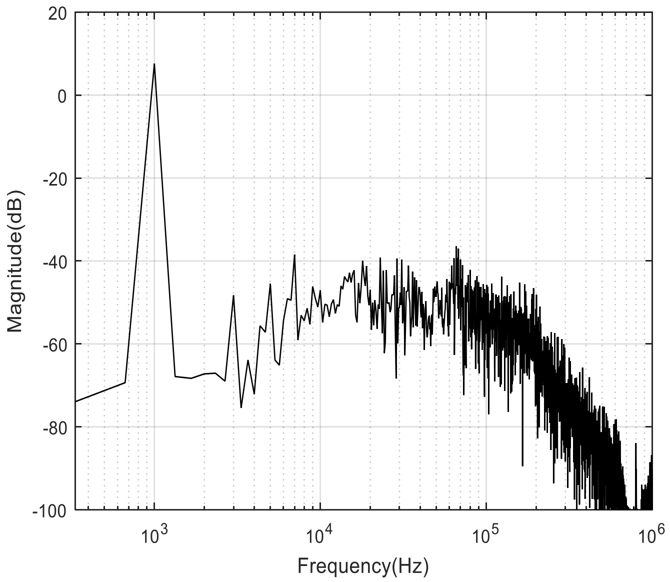

Figure 3.6 : Output spectrum

In **Figure 3.6**, the highest harmonic is around -50 dB below the fundamental frequency level (+10 dB).

We can see from **Figures 3.5** and **3.6** that the proposed circuit fulfills an efficiency of 92%. The obtained THD is 0.2 % with 800 kHz switching frequency.

3.5 Difference between PWM and PDM

We can distinguish the difference between PWM and PDM through the shape of the circuit and the performance (THD and efficiency)

3.5.1 Circuit :

3.5.1.1 Complexity:

PWM has Moderate complexity due to the inclusion of both high-side and low-side drivers along with a compensator and PWM controller.While PDM has higher complexity owing to the PDM control loop and the use of a comparator and gate driver.

3.5.1.2 Simplicity :

PWM is simple in terms of design and feedback mechanism, the use of PWM is a well-established method for Class D amplifiers. While PDM less simple due to the additional stages and more intricate modulation process compared to PWM.

3.5.1.3 Switching Frequency :

PWM with a top pick switching frequency of 400 kHz offers a well-established technique with a moderate frequency, and give us a precise control of pulse width.WhilePDM with atop pick switching frequency of 800 kHz might provide better efficiencybut come with increased complexity and potential for energy losses and trade-offs, it can increase power consumption, introduce more EMI (Electromagnetic Interference), and demand more sophisticated filtering and processing techniques to maintain signal integrity, briefly, it requires more complicated materials.

3.5.1.4 Components and different stages :

PWM utilizes a second-order integrator and a low-pass filter for signal processing in the feedback loop, comparator compares the error signal with a high-frequency triangular signal. While PDM utilizes a second-order integrator but includes additional feedback elements such as a latch stage (74HC74 D latch) and a negative feedback composite of resistors, emphasizes the role of an additional resistor ($R4$) in reducing nonlinearities from the output signal and controlling loop gain to reduce THD.

PWM utilizes feedback loop includes a second-order integrator and a low-pass filter. While PDM utilizes feedback loop includes a second-order integrator, latch stage, and negative feedback composite of resistors, emphasizes the role of additional feedback elements in reducing distortion and improving signal quality.

3.5.2 Total Harmonic Distortion (THD) :

THD of PWM is lower (0.003%) due to precise control over pulse width, resulting in cleaner signal reproduction. While THD of PDM is higher due to the shape of PDM circuit and the higher switching frequency, which can introduce more noise and distortions.

3.5.3 Efficiency:

PWM efficiency is slightly lower (90%) because the lower switching frequency requires more energy to maintain signal integrity and control. While PDM efficiency is higher due to the higher switching frequency, which allows for better energy management.

3.6 Conclusion

Through our comprehensive comparison and detailed analysis, the results conclusively demonstrate that Pulse Width Modulation (PWM) outperforms Pulse Density Modulation (PDM) in Class D audio amplifiers. This superiority is evident in both performance and simplicity, making PWM the preferred choice for applications.

Conclusion general :

the comparative study between Pulse Width Modulation (PWM) and Pulse Density Modulation (PDM) in Class D audio amplifiers has provided valuable insights into their respective strengths and applications. Through meticulous analysis, it has become evident that PWM stands out as the superior choice due to its simplicity of circuitry and superior performance.

The simplicity of PWM circuits offers practical advantages in terms of design, implementation, and maintenance. Its straightforward approach to varying pulse widths at a consistent frequency streamlines the amplifier's construction and operation, reducing complexity and potential points of failure. This simplicity translates into cost-effectiveness and reliability, factors crucial in mass production and deployment of audio amplifier systems.

Moreover, PWM's performance superiority further solidifies its position as the preferred modulation technique for Class D audio amplifiers. Its ability to deliver high-fidelity audio reproduction while maintaining efficiency and minimizing distortion underscores its suitability for demanding audio applications. By efficiently controlling power delivery to the speaker, PWM ensures optimal utilization of resources while preserving signal integrity, resulting in an enhanced listening experience.

PDM demands complicated materials and operates at very high switching frequencies. This requirement for advanced materials and elevated frequencies significantly impacts the design and manufacturing processes. The complexity of materials needed for PDM circuits often leads to higher production costs and more intricate fabrication techniques. PDM is very sensitive to high switching frequencies, making it more susceptible to noise and requiring precise control and high order filtering to maintain signal integrity.

In summary, the comparative study unequivocally concludes that PWM excels over PDM in Class D audio amplifiers, offering a compelling combination of circuit simplicity and superior performance. This finding not only informs the design and engineering of audio amplifier systems but also underscores the importance of selecting the most appropriate modulation technique to meet specific application requirements effectively.

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