# Pulse Width Modulation is Not Digital Signal Processing

Digital signal processing (DSP) is a digital representation of an actual signal. It is represented by a discrete discontinuous form, usually a binary electrical signal that is either 'on' or 'off'. These values are usually given as numbers. [4]

This concept is explained by sampling theorem, which states, (in simplified English), that a bandlimited signal can be uniquely represented by sample values that are evenly spaced and sampled at twice the highest frequency to be represented. [1]

Nyquist stated that we must sample at a rate greater than twice the highest frequency in order to avoid two (2) problems. [3], [1]

1. If sampling occurs at the point where the signal crosses zero, then it will produce zero output.
2. It is hard, if not impossible, to build a filter that rolls off perfectly above the highest frequency.

Other problems are a) the accuracy of the sample is limited by the number of data bits in the conversion; b) the highest frequency is limited by the sample rate; c) errors of omission (lost information); and d) errors of commission (extraneous information). [3]

Sounds contain high frequency components, as described by both Fourier and Laplace. [1] These components will not be recreated on the conversion back to analog audio. Although the average listener can not hear sine waves at this frequency, (not to mention those with noise induced hearing loss), the high frequency components add to the sound, i.e. a sizzle on a cymbal. Furthermore, the number of data bits limits the dynamic range.

The Compact Disc format samples at 44.1 kHz, and therefore has an upper limit of 22.05 kHz. This is why some audiophiles and purists still listen to vinyl records or use large open reel tape decks, sometimes with DBX units. Some have even converted VHS VCRs for this use as they have an upper response limit of close to 100 kHz.

Class D is often confused with digital, but it is not. There is no digital representation of the original signal, nor is there any digital processing. The duty cycle of the carrier wave varies on a continuous spectrum. The pulse width of the carrier is a linear function of the input signal. [2] There are no discrete or numeric values. Unlike digital, the carrier frequency limits the dynamic range.

Class D, switch mode, or pulse width modulated, as it is often called, has been around for a while. The positive attribute of Class D is high efficiency. It was first used for regulated power supplies, where efficiency was very important.

Class D amplifiers produce a lot of power inexpensively. Back in the late 1960's, when the SCR (silicon controlled rectifier) first became commonly available, there was discussion of high power public address amplifiers running Class D using these new solid state devices. The combination of high power without requiring high frequency response makes PA systems a good application for this design. I do not know if any were ever commercially produced.

On to some theory. Quantum mechanics aside, energy is neither created nor destroyed. Energy that goes into a system must come out. In the case of an amplifier, electrical energy goes in, and both an electrical signal and heat are produced. Efficiency is the ratio of signal out to power in.

Class D varies the signal energy by changing the duty cycle of the square wave. This means that 'on' time is varied with respect to the 'off' time by the audio signal. This is known as modulating the carrier wave. Typically, a switched integrator, synchronized to a clock, is used to modulate the pulse width. [4] This is a linear function, there are no digital representations.

To calculate the energy at any point in time, we use a mathematical concept called integration. It refers to the area under the curve, and in this case is the amount of time that the output transistors are 'on'. The output represents the energy in a continuous form. This is accomplished electronically with a low pass filter. By definition, a low pass filter does not pass high frequencies. Typically, an inductor is used to perform this integration, and it is a fine line to find one that is capable of reproducing the low frequencies and still passes the high audio frequencies, (not to mention filtering out the carrier frequency). Most recently, some amplifiers use carrier frequencies around 70 Khz and use the voice coil of the speaker as the inductor. This is a very low cost way of producing a high power amplifier.

Again, we must remember that Fourier transforms show that signals are combinations of many frequencies, and a loss of high frequencies will not allow exact reproduction of the original signal. In reality, this is the major weak link in the system.

Additionally, distortion is introduced from core saturation. This is when the magnetic field is as big as it can get for a particular inductor, and any additional current will not linearly increase the size of the field. This can be compared to a glass of water that is full and can not hold any more.

Furthermore, a real inductor is composed of distributed capacitance, (the voltage in one turn is different from the next), resistance (it is wound from real wire), and of course inductance. The result is a tuned second order tank circuit. The bottom line is that it will both heat up and cause distortion. As it is a totally passive filter, there is no active method to control the variations.

Efficiency is related to the heat produced by the amplifier. The heat that is generated, better known as joule heating, can be calculated by multiplying the current through a device by the voltage drop across it. This is mathematically written as $P=EI$ or $P=I^2R$. We commonly think of heat generated by a resistor, but fail to think of all of the other sources.

The current through the inductor times the voltage across it produces heat. Any semiconductor device, be it an FET or bipolar transistor, generates heat, in the same way. For a bipolar transistor in common emitter mode, the heat generated is the collector current ($I_C$) times the collector to emitter voltage ($V_{CE}$). As we all know by the heat sink size that this can be a lot of heat.

The lure of Class D is that the output devices run only in switch mode, that is either 'on' or 'off'. Science has not departed from us here, and it takes a finite amount of time for the device to go from one state to the other. This is known as rise time or fall time. Furthermore, when the device is totally on, there is still a voltage drop across it, (i.e. $V_{CE}$Sat), or saturation voltage. This still produces heat, although not as much as in the other classes of amplifiers.

Class D does have its areas of superiority. Switch mode power supplies, when actively regulated, make excellent energy sources. They have the advantage of only supplying the power that is needed. If they were to be used on a class AB amplifier, for example, it would greatly increase the overall system efficiency, especially at lower power output levels.

To compare one amplifier with an active power supply to another without this feature is not a fair comparison. Properly employed in analog amplifiers, class D power supplies would obsolete, or greatly diminish, the need for "stiffening" capacitors and other such accessories.

A minor point is that darlington bipolar transistors are not parallel devices. The collector current of the first becomes the base current for the second, greatly increasing the gain ($H_{FE}$) and the input impedance. [4] One disadvantage is a higher on turn-on voltage ($V_{BE}$). There are some other disadvantages, but they are too technical to discuss here.

There are several conclusions to be drawn from the above:

1) Pulse width modulation (PWM) is not digital signal processing.
2) At present, class D is not ideally suited for high frequency audio.
3) Since the PWM is not digital, there is no digital to analog converter (DAC).
4) A regulated switch mode power supply, as in the class D unit, would help analog amplifiers improve efficiency.

## References

[1] Gabel, Robert A, & Roberts, Richard A; *Signals and Linear Systems*, John Wiley and Sons, 1973

[2] Tobey, G E, Graeme, J G, & Huelsman, L P; *Operational Amplifiers Design and Application,* McGraw-Hill, 1971

[3] Pippenger, D E, & Tobaben, E J; *Linear and Interface Circuit Applications*, Texas Instruments, 1987

[4] Graf R F, *Modern Dictionary of Electronics, sixth edition,* Howard W Sams & Co, 1984

## About the Author

Brad Morantz holds BS, MS, and PhD degrees in electrical engineering, computer science, computer and information science, and mathematical decision science. He has also taken course work in computational bioscience, project management, and imaging infrared. He has over 25 years experience including military, automotive, and consumer electronics.