Institute of Acoustics

# TECHNIQUES FOR DIGITAL POWER AMPLIFICATION

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## Abstract

This paper describes investigations made into the suitability of Pulse Width Modulation as a means of implementing Digital Power Amplifiers.Theoretical and practical investigations are described along with the results obtained and it is concluded that such systems appear to offer a great deal of promise for this application, but that further investigations are required before concrete conclusions can be drawn or a fully viable system capable of competing with conventional amplification methods can be specified.

## Introduction

The use of digital and computer techniques in audio products is now well established in both the professional and consumer markets. However, to date little has been published on the adoption of these techniques to power amplification. There is good reason for this, as will be shown in this paper, which will bring together and extend upon the author's previous publications in this field [1-3]. Others have published articles onthis topic, but in both cases either fallacious assumptions have been made[4] or simple systems presented with no supporting theory [5].

It is important to emphasize at the start just what is meant by the phrase "Digital Power Amplification" as the term has been used commercially to describe significantly different systems. The assumption made in the work in the to be described is that conventional loudspeaker technology is used.

Here the phrase will mean a device for converting digital audio data in a serial or parallel bit stream directly into analogue power to drive a loudspeaker, with no intermediate conversion from digital to analogue voltage. It is in effect a method for power digital-to-analogue conversion.

"Digital Power Amplification" has also been used to describe a method based on naturally sampled Pulse Width Modulation, which has been utilised in a number of commercially available amplifiers of analogue audio signals (such as those from Sony,Peavey and Harrison Information Technology). In such systems, at no point can any signal representation be construed as digital. The phrase is used only to appeal to the consumers' sense of modernity with the excuse that Pulse Width Modulation waveforms switch between two (or sometimes three) possible states. This point will be clarified in the next section.

A different category of PWM has been adopted for this work, namely uniformly sampled PWM. As will be shown, this has a different modulation spectrum, resulting in higher levels of harmonic distortion than the naturally sampled form. It is however quite impossible to make use of the naturally sampled variety due to the uniform sampling inherent in digital audio data streams.

Thus for the purposes of this discussion, digital power amplifiers involve the use of uniformly sampled PWM implemented digitally. This is shown pictorially in figure 1. Simple digital pulse width modulators are available [6] though these are only capable of processing 8 bit data words and are intended for control applications.

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## Pulse Width Modulation

Briefly, PWM can be generalized as a process in which an input signal is modulated onto a high frequency pulse waveform as variations in the mark/space ratio. This is done by applying the message to one input of a comparator and a high frequency triangle or sawtooth waveform to the other: the comparator output toggles in such a way that the PWM waveform is produced. There are more complicated ways of generating more complex varieties involving 2,4 or more comparators, but this is its essence.

This describes naturally sampled PWM. Uniformly sampled PWM is obtained when the input signal is passed through a sample/hold device operating at the comparison waveform frequency, prior to the comparator input. This apparently simple change makes a significant difference to the performance of the system. Because of the sampled nature of digital audio, it is uniformly sampled PWM which has been investigated.

Figure 2 shows the essential difference between hardware for naturally and uniformly sampled PWM taking as an example one-sided (trailing edge) modulation. It can be seen that the two waveforms applied to the comparators are significantly different, although the differences in the output waveforms are harder to discern. These waveforms do differ since the positions of the modulated pulse edges are determined by the sampling instants, which are regularly spaced in time for uniform sampling, but irregularly spaced and signal dependant for natural sampling.

In a digital implementation, the comparator becomes a digital comparator and the comparison waveform is generated as the output of an up-counter (for 1 sided modulation) or an up/down-counter (for 2 sided modulation). However, the circuitry may be simplified by loading each new sample into a counter, counting down and detecting the instant when the count becomes zero. That will work for one-sided (trailing edge ) modulation: for two-sided modulation a circuit has been developed which involves two such counters, one inhibiting the action of the other. The details of this system are not included here as they are of a commercially sensitive nature.

As well as the natural and uniform sampling categories of PWM and the possibility of modulating one or both edges of each pulse, it is also possible to generate a tri-state type of waveform in which the polarity of each pulse reflects the polarity of the relevant sample. After Martin [7] this is known as class BD PWM, whilst the two-state pulse train is known as class AD PWM. Thus for uniform sampling there are the following possible varieties of output waveform:-

i)one sided AD

ii)two sided AD - one sample per pulse

iii)two sided AD - two samples per pulse (one each edge)

iv)one sided BD

v)two sided BD -one sample per pulse

The two-sided BD variety in which two samples modulate each pulse is not considered viable as consecutive samples may be of different polarities. Figure 3 shows this pictorially.

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In order to emphasize the difference between uniform and natural sampling, one sided class AD modulation will now be examined in terms of its tone modulation performance. The following legend will be used:-

| | |
|---|---|
| $\Omega_c$ | - pulse repetition frequency |
| $\Omega^m$ | - signal frequency |
| q | $= \Omega_c/\Omega_m$ |
| p | - harmonic of $\Omega_m$ |
| s | - harmonic of $\Omega_c$ |
| m | - modulation index |
| $J_n(x)$ | - Bessel function, 1st kind, order n, argument x |

Equation (1) below is the Fourier series expansion of the expression for the naturally sampled variety [8]

$$f(t) = H(1+m\sin\Omega_m t)/2 + \sum_{s=1}^{\infty} H\mathrm{sink}(\Omega_c t)/\pi s + \sum_{s=1}^{\infty}\sum_{p=-\infty}^{\infty} \frac{H(-1)^s(-1)^{p+1}\, J_n(\pi m s)\sin(s\Omega_c+p\Omega_m)t}{\pi s} \quad \text{-- (1)}$$

which contains an amplified message signal, carrier harmonics and message sidebands around carrier harmonics. It is the message sidebands that are of greatest concern, since for *n* negative and large enough, these will manifest themselves as distortion components falling within the audio passband. Notice that their level will be quite low relative to the wanted component (as indicated by the Bessel function order) but they will in general be inharmonically related to the signal and may therefore manifest themselves in an undesirable manner.

The corresponding expression for uniform sampling, derived by the author and presented in [1] is

$$F(p) = \left.\frac{1 - J_0(\pi m p/q)}{j\pi p/q}\right|_{\substack{p=sq \\ p\neq 0}} + (q/2\pi p)e^{j\pi/q(p+q/2)}\left\{J_p(\pi m p/q) + \sum_{s=1}^{\infty} J_{|sq-p|}(\pi m p/q) + J_{|sq+p|}(\pi m p/q)\right\} \quad \text{-- (2)}$$

This is presented in the frequency domain, indexed by message harmonics, $p$. The first term represents carrier harmonics and the second represents message sidebands around the carrier harmonics and around dc. Thus the wanted component is the $p=1$ term and all other values of $p$ represent distortion. The term under the summation may be disregarded, as for realistic carrier to message ratios, $q$ the Bessel coefficients become insignificant. That this assumption is justified will be demonstrated later. Further disregarding the phase term, the individual components in the tone modulation spectrum can thus be approximated by

$$F(p) = (q/2\pi p)\ J_p(\pi mp/q) \qquad \text{-- (3)}$$

Using an approximation for Bessel coefficients [9], the following expressions were derived in [1] for the relative levels of the wanted $p=1$ component to the first few distortion terms:-

$$F_1/F_2 \approx 0.66q/m \qquad \text{-- (4a)}$$

$$F_1/F_3 \approx 0.29q^2/m^2 \qquad \text{-- (4b)}$$

$$F_1/F_4 \approx 0.1q^3/m^3 \qquad \text{-- (4c)}$$

Thus it can be seen that the distortion falls both as $q$ increases (i.e. sampling frequency increase or message frequency decrease) and as the modulation index, $m$ decreases, and that this effect increases for higher distortion components as powers of $q$ and $m$.

This implies that for $q$ fixed, if $m$ is halved (the signal level is halved) then the level of the first distortion term will fall by 6dB relative to the fundamental and by 6dB absolute. The level of the second distortion term will fall by 12dB absolute and 12 dB relative, and so on.This effect has been confirmed both by simulation and empirically.

Since $q$ falls as message frequency increases (for fixed sampling or pulse repetition rate) this would imply that distortion will increase with signal frequency.However the effect of the modulation index is particularly important for audio applications, in which signal power falls with increasing frequency, so that these distortion effects will tend to be offset.This has been consistently ignored by other workers who have rejected PWM without paying attention to the kinds of signals encountered in practice.

Expressions of a form similar to (2) have been derived for cases (ii) and (iv) and a similar simplification to (4) has been made for case (ii).These are presented in [1] which shows that for two-sided AD modulation there will only be odd order harmonics of the message present in the tone modulation spectrum and that the level of the third order component is the same as in (4b) above.This is in agreement with the expressions derived in [10].

The expression for one-sided BD modulation in [1] indicates that all distortion terms have their amplitudes given as values of the second order Bessel function $J_2(x)$ rather than as $p$th order Bessel functions and therefore it is to be expected that there are many more distortion terms with significant energy: this has been confirmed in simulation.

PWM may be summarized as an inherently non-linear process which will inevitably produce distortion, but that as signal level falls, or as pulse repetition rate increases so the process becomes progressively more linear.

## Investigations

These fall into three categories:- theoretical, the results of which have been summarized above; simulation and practical.

The simulations were performed on a mainframe computer offering hardcopy graphical output of results.The simulator allowed one-sided AD and BD modulators to be evaluated when driven by sinusoidal, intermodulation and noise test signals, among others. The results obtained concurred with those predicted by the previously presented theory. It was also possible to perform system identification tests which demonstrated that PWM is indeed a reasonably linear process whose linearity improves as the signal power is reduced.

The simulation suite also allowed a hybrid Pulse Amplitude/Pulse Width modulation scheme to be investigated. This turned up some interesting results which will be discussed in the section dealing with linearizing the PWM process.

The practical experiments involved both naturally and uniformly sampled modulators, though the natural sampling modulator was investigated only to confirm that these two sampling schemes give substantially different performance.

The uniform modulator may be simply reconfigured as either one- or two-sided and is constructed from fast TTL logic circuitry. It operates on 9 bit data generated by a microcomputer. There is a master clock frequency of 18MHz which drives the counters so that the pulse repetition rate is 18MHz/512 = 36kHz. This is not intended to be a realistic rate in terms of audio reproduction, but is sufficient to investigate the properties of DPWM and the concurrence of theory and practice.

The signals provided by the microcomputer at the 36kHz sampling rate are digital sine waves whose amplitude may be reduced in steps of 1/2: thus the modulation index $m$ may be set to 1,1/2,1/4 and so on, but with an increase in the signal to quantization noise ratio in the input signal as the amplitude falls.

## Results

Two sets of results are presented:- from the simulation of 1-sided AD and BD modulation and from the experiments performed on 1 and 2 sided AD modulation. These may be compared with the theoretical predictions, but the only direct comparison possible among the results is for AD 1-sided modulation.

Dealing first with the simulation data as presented in graph 1, it can be seen that the distortion level falls with increasing pulse repetition frequency at approximately 6dB per octave. Note that all these experiments used a 1kHz test tone, although others using different frequencies were performed and these corresponded with the theory in exactly the same way as the results presented here. The readings were taken from spectral plots.

These results have been compared with the theoretical predictions based on the approximate expressions (4) in [2] which showed that for 1-sided AD modulation they agree to within 1 dB which is close enough for the difference to be explained as graph reading error plus perhaps some small influence from the sidebands around the carrier harmonics which were ignored in deriving expressions (4).

No comparable simplifications for BD 1-sided modulation has yet been derived, but the general conclusion that there would be many significant terms is borne out in practice.

There were insufficient simulation experiments on lower modulation indices than 0.95 for similar plots to be made, but for those that were performed, the approximations (4) were again upheld as valid.

Turning now to the experimental data as presented in graphs 2-5, the major difference between these and the simulations is that the pulse repetition rate is now fixed and the signal frequency varied. These results have been compared with the appropriate approximations (4) in [2] and it was shown that they concur for case of 1-sided modulation for a variety of values of modulation index as long as the predicted distortion level is above the level of the quantization distortion in the 9 bit quantized input signal.

There is some deviation from the theoretical predictions in the case of 2-sided modulation and this is manifested in two ways:- firstly that there are even order distortion components present which the theory does not predict, and secondly that the distortion levels for 3rd order components differ form the predictions.

It is probable that the even order distortions arise because the waveforms are not symmetric with respect to the regular timing markers due to the quantizations of the pulse durations: this manifests itself for pulses which are odd multiples of the minimum pulse duration (ie that corresponding to the least significant bit only being set) in length. This is shown in figure 6. There are however techniques which are currently under examination which should alleviate this problem.

As to the differences between the actual and predicted distortion levels for 3rd order components in 2-sided modulation and also for 3rd and 4th order components with 1-sided modulation, it is believed that this is largely due the poor quantization in the input which is resolved to only 9 bits. Once again this point is being addressed in current work and it is hoped to be able to provide further explanation in later publications.

It must be emphasized the the pulse repetition frequency used in the experiments is rather low and would in practice be perhaps 4 times higher. Based on the assumption that expressions (4) are correct, it would be expected that for the various test signal frequencies chosen, the distortion levels would be 12dB better for 2nd order components, 24dB better for 3rd order and 36dB better for 4th order. Extending this assumption to the premise that AD 2-sided modulation, implemented in such a way as to alleviate the problems outlined above, this would mean a highest distortion component for a 1 kHz sinewave at some 90dB or more below the level of the fundamental for a full amplitude signal, which as noted earlier, is the most severe test amplitude to choose for PWM systems.

## Implementation

The simplest implementation for a PWM amplifier (either analogue or digital) is to follow the modulator with a power switch and LC low pass filter as shown in figure 7. The maximum power output of the amplifier is determined by the voltage rails to power switch. As shown in the figure a dc block will be needed in the filter as the pulse train will have an average level of V/2.

This is alleviated by the arrangement of figure 8, for which the pulse train will switch between +V and -V. For this arrangement, care must be taken over selection of the switching transistors (particularly with regard to their turn-on/turn-off) times so that both switches will never be closed at the same time allowing a large current transient to pass between the supply rails. This arrangement is often used in the analogue PWM amplifiers that have been produced commercially.

The arrangement for class BD modulators is shown in figure 9, in which the upper and lower switches are independently controlled and there is a third switch to ground which is activated when neither of the other power switches are activated, to prevent the output stage going into a high impedance state and so keep a constant input impedance to the LC filter. Here the relative timings of the various switches is even more crucial and indeed to the author's knowledge no manufacturer has yet managed to produce such an output stage.

Power mosfets are often used in these output stages as they are essentially current controlled devices, the turn-on/turn-off times being controlled by the rate of current charge and discharge to the gate, and these times may be as low as 5nsecs or so under correct drive conditions.

Care must be taken in the design of the output smoothing filter. The purpose of this is to remove all the unwanted components in the complex PWM modulation waveform. The filter requirements are quite stringent, but for maximum linearity, the inductors used should be air-cored or at least ferrite cores with air gaps. Air cored inductors are somewhat bulky, thereby reducing some of the size reductions that can be achieved through VLSI construction of the modulator stage. This may be offset to some extent if amplifiers of this type were to be adopted for a new form of digitally controlled active loudspeaker in which the crossover and smoothing filter are combined.

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Some simple experiments have been performed using SPICE, a well known circuit simulation package, to determine which type of filter should be used. Butterworth, Bessel, Chebyshev and various other designs were compared on the basis of a trade-off between delay and amplitude characteristics, and it was concluded that Butterworth was the best choice. This is in spite of the superior delay distortion properties of the Bessel polynomial. This is because a higher order of Bessel filter is needed to achieve the same order of amplitude characteristics as a Butterworth (roughly a 9th order Bessel compares with a 5th order Butterworth) and all the extra components needed not only greatly increase the cost and bulk of the filter but negate almost all the delay advantages associated with Bessel filters - a low order Butterworth filter has reasonably linear phase over most of the passband. Chebyshev and other fast amplitude roll-off designs were rejected because of their unacceptable phase performance.

Techniques for building class D output stages are fairly well established, even if not straight-forward because the medium power levels being switched at medium-high speeds (100-500kHz). What is not well established is just how to build the digital pulse width modulator, which will operate on 16 bit words and produce pulses at a rate of 100kHz or more. Simple calculations highlight the difficulties.

Assume:

p.r.f = 32kHz
bits = 16
AD 1-sided modulation

then:

a pulse will be produced every 31 msecs
and will have $2^{16}$ possible lengths

thus

the minimum possible duration will be
1/(32000*65536) secs
= 0.477 msecs

this implies

a master clock speed of 2.1 GHz needed to drive the down counter

Although this may be halved by using BD modulation (one bit is used to select the pulse polarity) this is clearly not achievable with current Silicon logic technologies. Thus it is necessary to resort to Gallium Arsenide (GaAs) technologies, which are still very much the subject of research, even though some integrated circuit logic devices are available (from Harris Semiconductors and Gigabit Logic amongst others).

If a resolution of 13 bits can be accommodated, and this does give quite acceptable results, then a clock speed of around 500MHz can be used, this being possible with ECL, and such systems might find application for in-car stereos for example. As will be discussed in the next section, it is possible to reduce the master clock speed below this figure by resorting to a hybrid modulation technique.

It was stated earlier that the distortion levels can be reduced by increasing the pulse repetition frequency. This is achieved by introducing a sample-rate increase device (an interpolator) prior to the modulation stage. It is now common for CD manufacturers to use such devices to give the 'oversampling' method of D/A conversion in order to reduce the in band effect of quantization noise, as was discussed in an earlier paper [11]. Thus it is expected that amplifiers such as that under discussion would interface to CD player after the sample rate increase stage, by-passing the the on-board D/A converter.

It was also mentioned that class AD 2-sided modulation offers great promise as the method to adopt for real implementations of digital power amplifiers. Its main advantages are that it offers lower distortion than 1-sided modulation and that the output stage is easier to construct than for BD modulation (compare figures 8 & 9). The problem, highlighted in figure 6, is currently under investigation and so it is expected to be able to produce a working modulator which conforms to the predicted performance in the near future. This will initially be a demonstration of the principles, operating at a lower pulse repetition rate or lower precision than would be necessary for full high-fidelity performance and constructed from fast TTL devices.

## Linearization

Expressions (4) indicate that DPWM performance becomes more linear with increasing pulse repetition rate, but as the previous section shows, this compromises the realizability of the modulator. Pulse Width Modulation is very obviously an inherently non-linear process, so that it is never going to be possible to completely linearize it. However during the course of the investigations that have been made into PWM, a number of possible techniques have come to light.

The first, and to date the only one to be investigated, is the use of a hybrid Pulse Amplitude/Pulse Width modulation technique. This involves ensuring that the area under each pulse is proportional to the magnitude of the digital sample (as it is in conventional PAM and PWM) and to use the most significant bits to control the pulse amplitude and the rest to control the pulse duration. This has the added advantage that the modulator master clock frequency may be reduce by $2^e$, where $e$ represents the number of most significant bits used and corresponds to the exponent in floating point representations in computers. The least significant bits are similarly called the mantissa, $m$ - do not confuse this use of $m$ with the modulation index.

This technique was investigated by simulation for a variety of combinations of $e$ and $m$; 16 bit signals were used throughout and more bits were always assigned to the mantissa - typically 3 bit exponents were used with 13 bit mantissas. This method proved to most effective in linearizing the modulation, but it must be noted that implementation requires that power supply lines must be varied at a speed commensurate with the pulse repetition frequency.

A summary of the results obtained is shown in graph 6. This presents only the results for class BD 1-sided modulation as class AD cannot be adapted to this technique as discussed in [3]. It can be seen that for comparable p.r.f. the distortion is lower than with conventional DPWM.

The technique for linearizing conventional class AD 2-sided modulation has already been mentioned, but others actively under consideration include placing an inverse non-linearity prior to the modulator, its approximate shape having determined by testing a modulator in simulation with Gaussian white noise and comparing the probability density functions and investigating the feasibility of of using feedback around the modulator, using an A/D converter in feedback loop. This would be driven by the demodulated signal at the output of the low-pass filter. Another possibility is to predict the distortion using the inverse non-linearity mentioned above and adding in a distortion correction term to the analogue representation of the signal either before or after the low-pass filter in much the same way as the current dumping technique used in some analogue amplifiers.

## Applications and Advantages

The most obvious potential application for such devices is in consumer and professional power amplification. Also, as previously mentioned, these devices may well be used in in-car hi-fi systems, using less than 16 bits of the available signal. It has been shown in [3] that with more than 8 or 9 bits, the distortion is independent of the number of bits used: only the noise floor is changed. This seems a particularly suitable application as the background noise level in a car is very high. Also, it was mentioned earlier that DPWM might be adopted in active loudspeakers to be interfaced directly to digital sources of audio.

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In general, DPWM techniques may be used anywhere power signals are required to be derived from digital representations. In this respect, it offers significant advantages over conventional D/A converter techniques since longer wordlengths may easily be accommodated as long as reductions in the conversion rate can be accommodated and the linearity problem can be overcome.

## Conclusions

A technique has been described which looks suitable for adoption as a technique for digital to power-analogue conversion. Various investigations have been discussed and these have highlighted which of the forms of DPWM are suitable for further investigation: it seems particularly likely that 2-sided AD modulation will offer suitably low distortion, and that 2-sided BD modulation will offer even better performance once the problems associated with implementing a power switching stage have addressed. Also showing significant promise is the hybrid modulation scheme combined with class BD modulation.

The implementation problems have been covered and although these push audio systems close to the performance limits of available technologies, this is not an insurmountable problem.

Whilst conventional amplification systems offer a performance which has yet to demonstrably matched by the techniques discussed in this paper, it is worth bearing in mind that it has taken some 60 years or more to reach this level of performance, with untold numbers of workers contributing to the body of knowledge that has accumulated. In contrast, very little has been done in the field of digital amplification, at least so far as the author is aware.

## A Personal Request

Much of the above reported work has been executed in isolation, and so the author would be very pleased to hear from others working in this or similar fields who have useful comments or criticisms to make. Equally, suggestions for collaboration will be favourably regarded.

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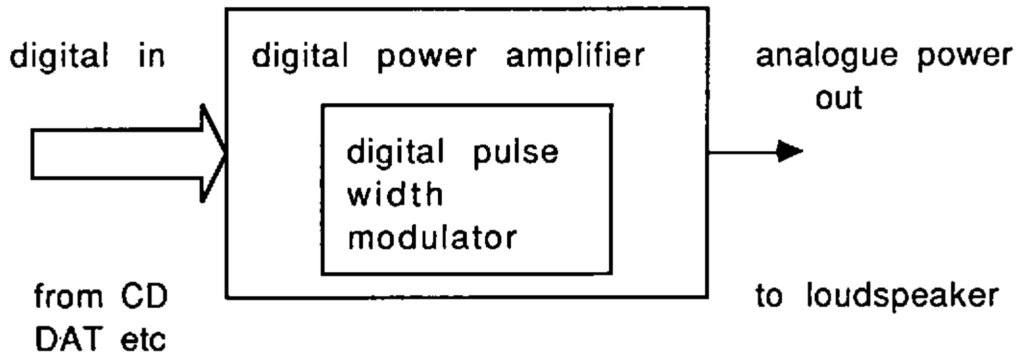



**Figure 1: Digital power amplification**

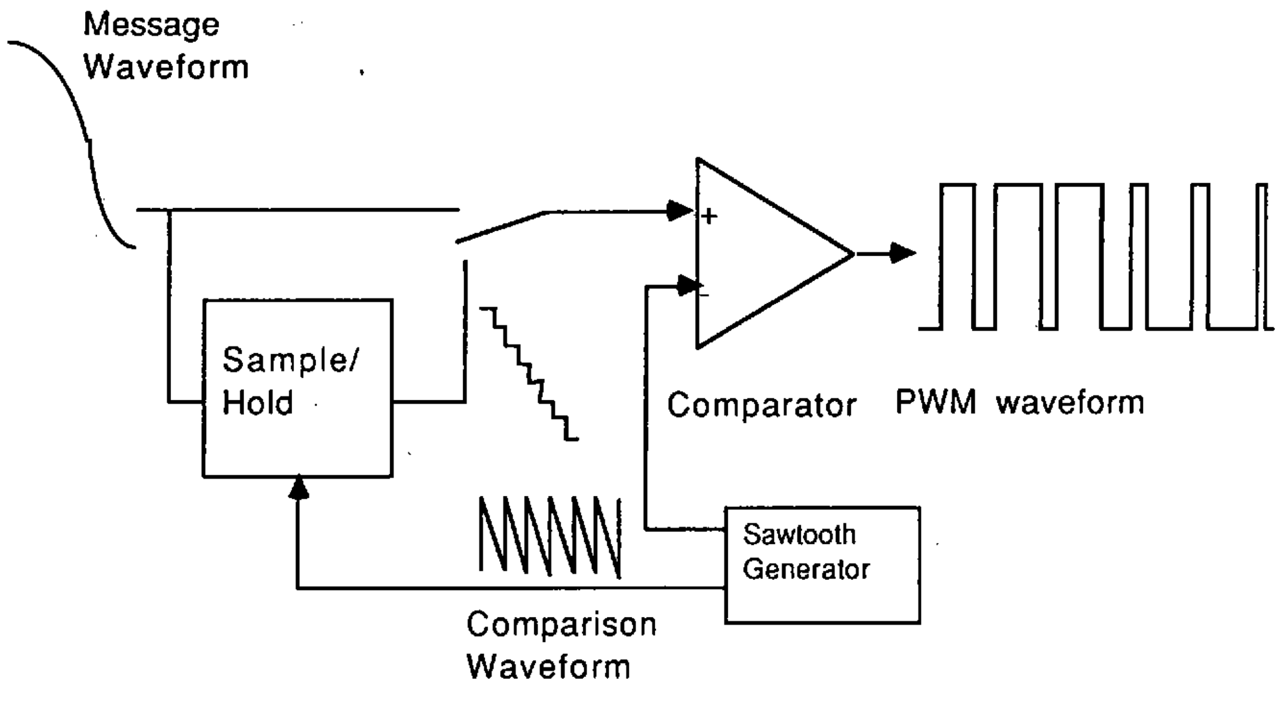



Figure 2: Operation of Pulse Width Modulator

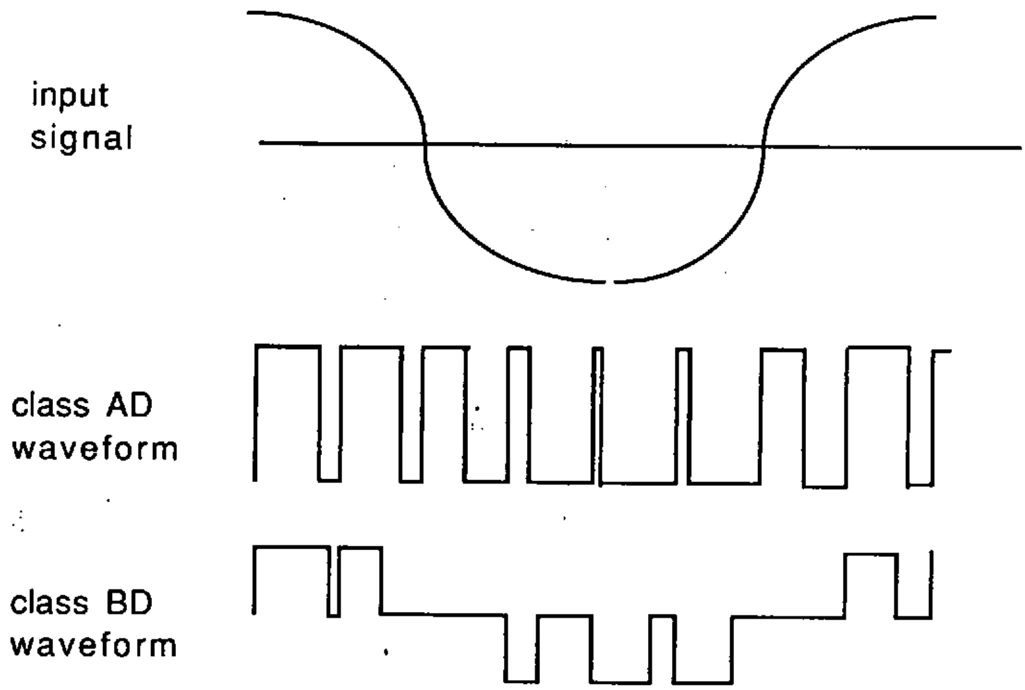



**Figure 3 Pulse Width Modulation Schemes**

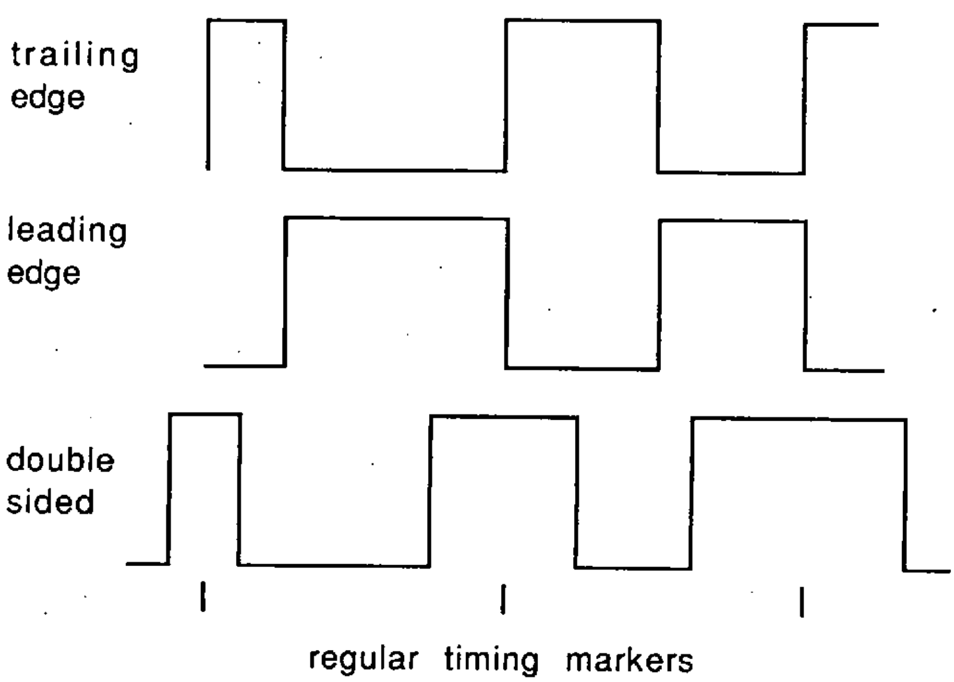



**Figure 4: Class AD Modulation Types**

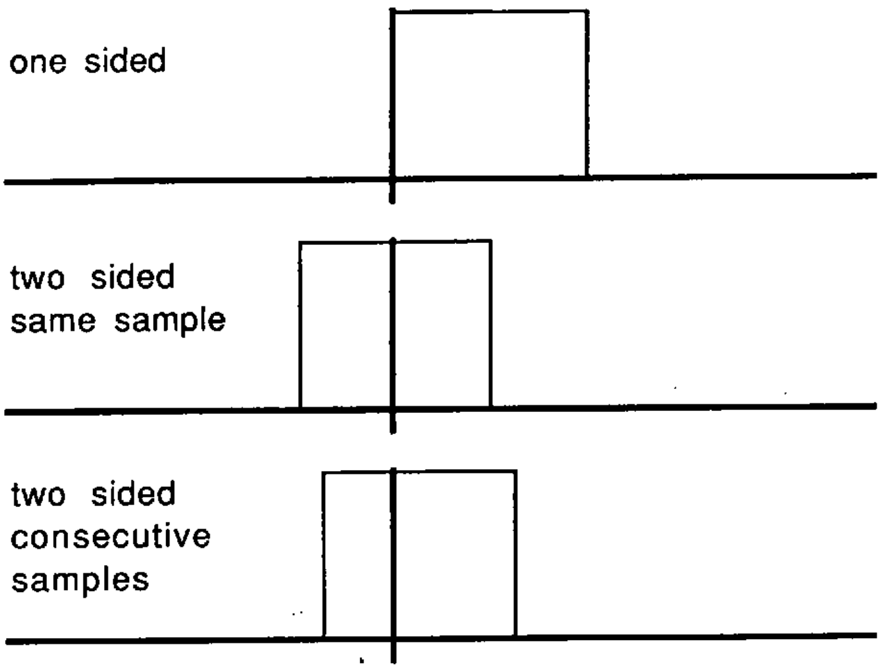



**Figure 5: Pulse types for AD**

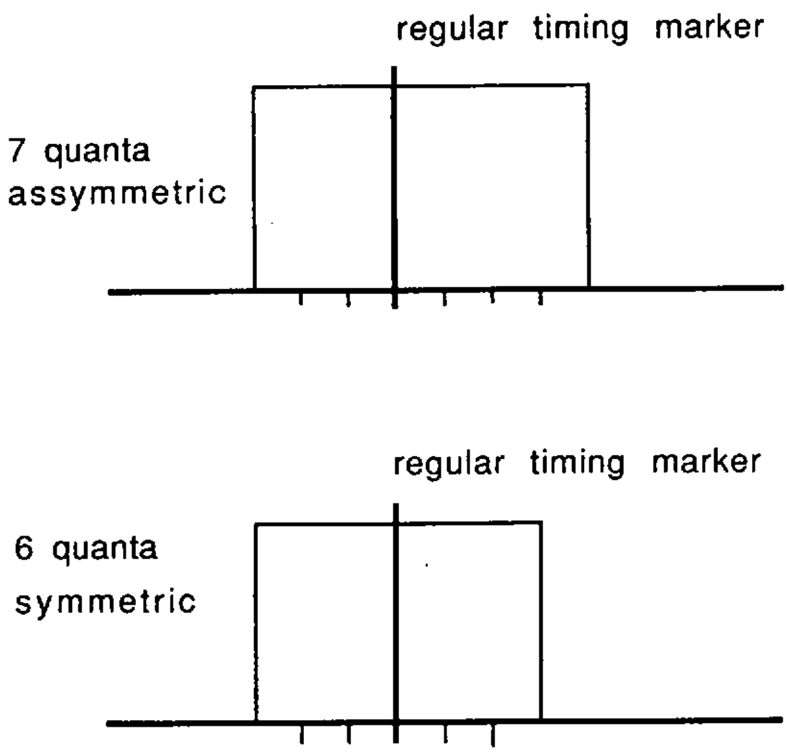



**Figure 6: Assymmetry with 2-sided modulation**

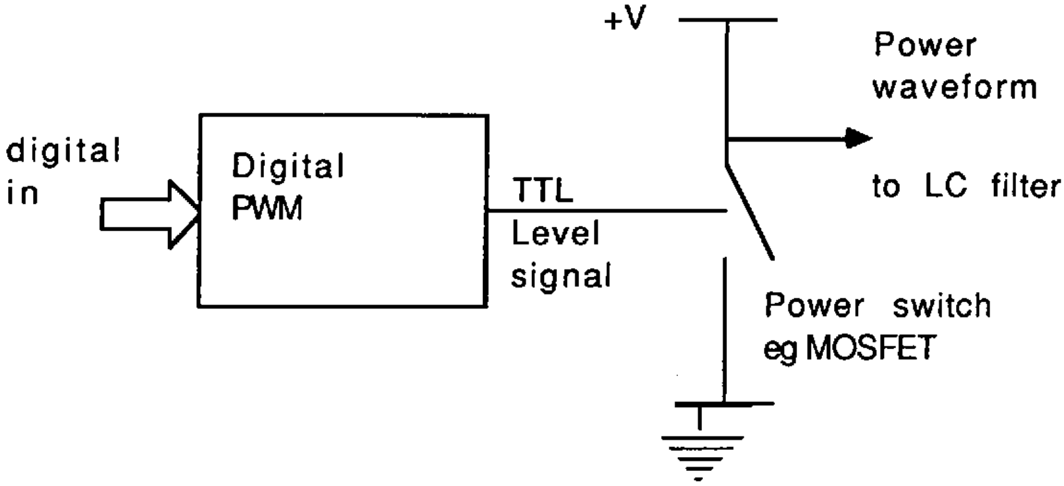



Figure 7: Simple Amplifier Schematic

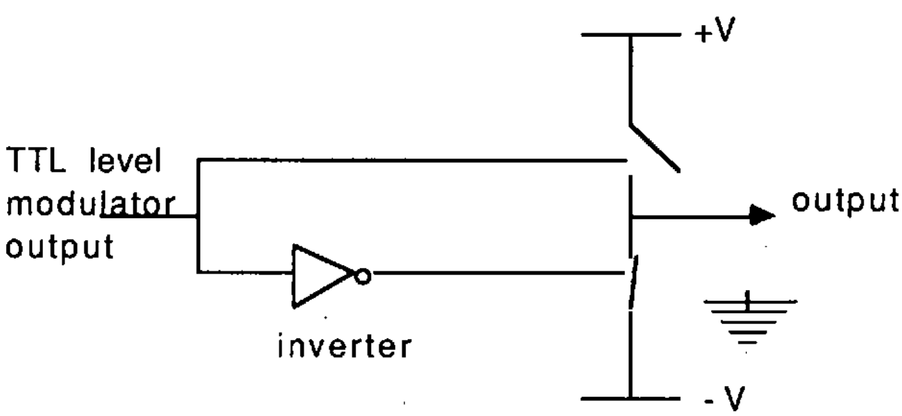



Figure 8: Symmetric AD output stage

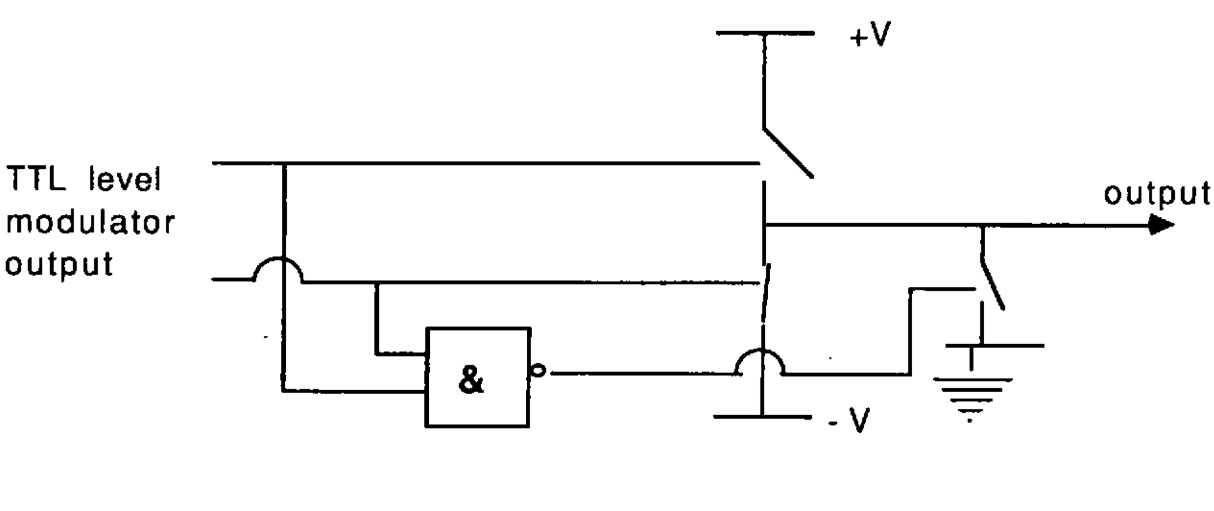
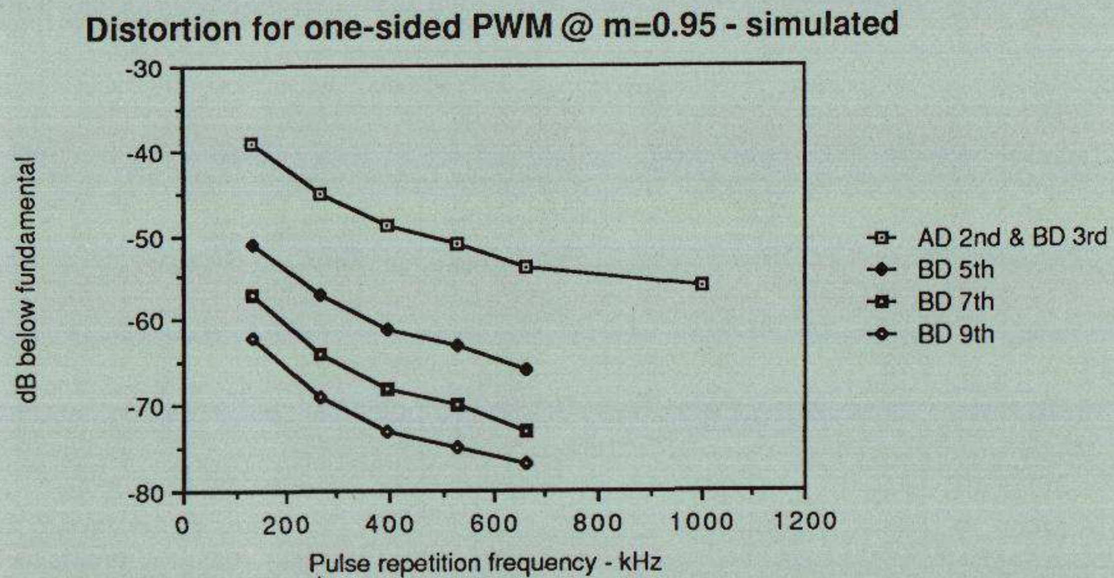
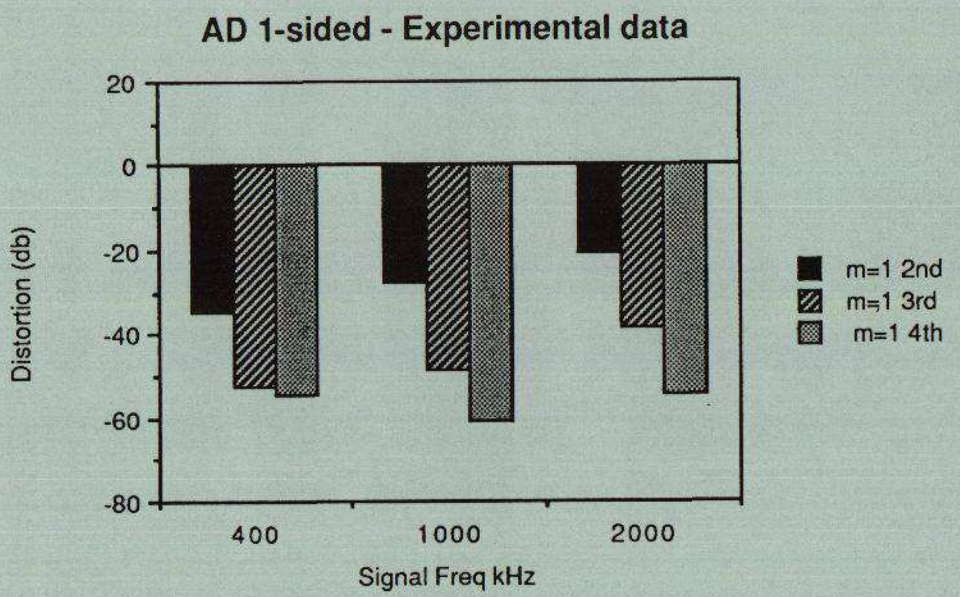


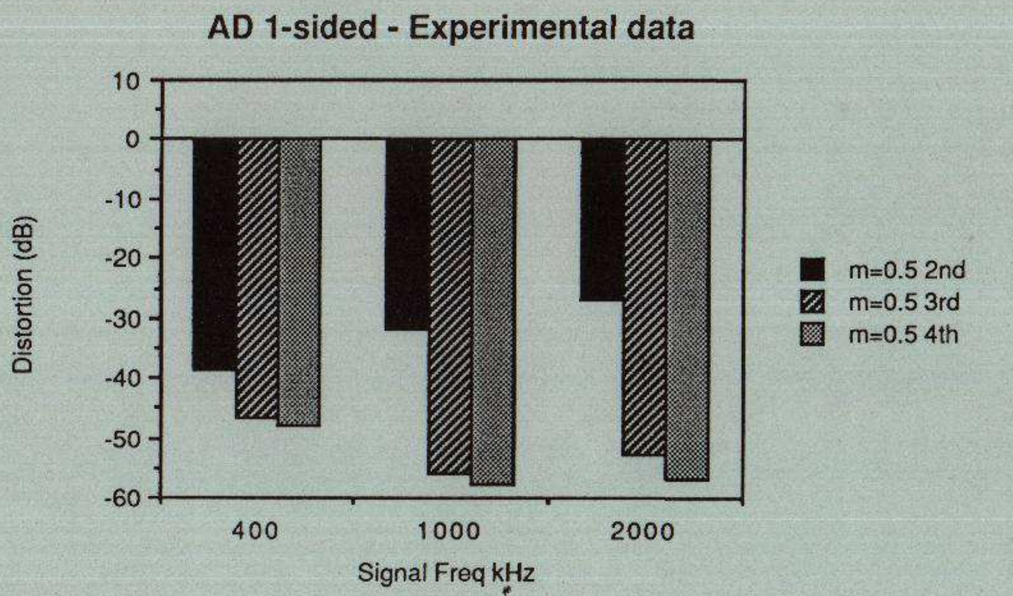

Figure 9: Class BD output stage



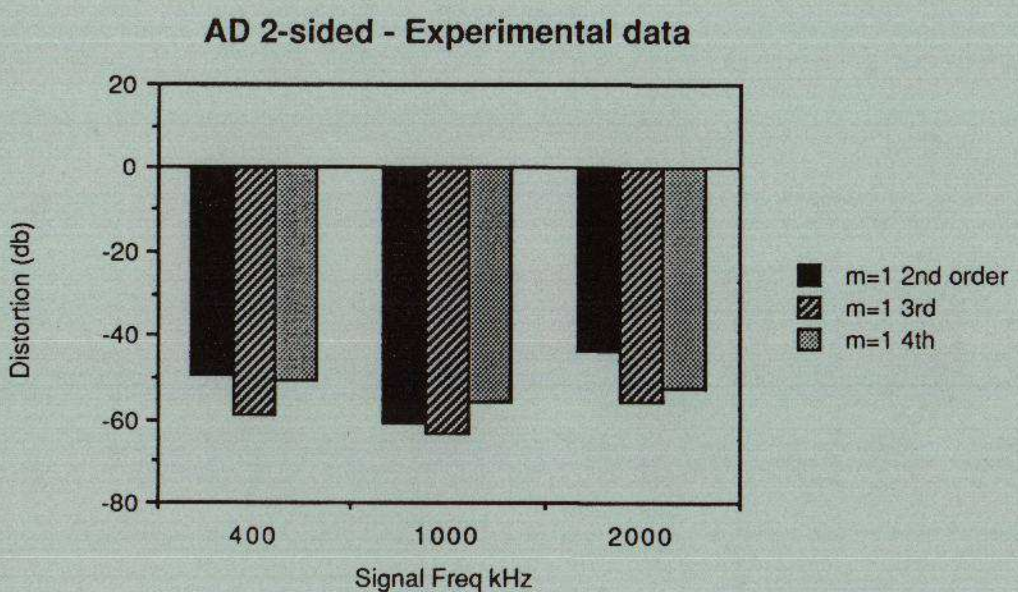

Graph 1



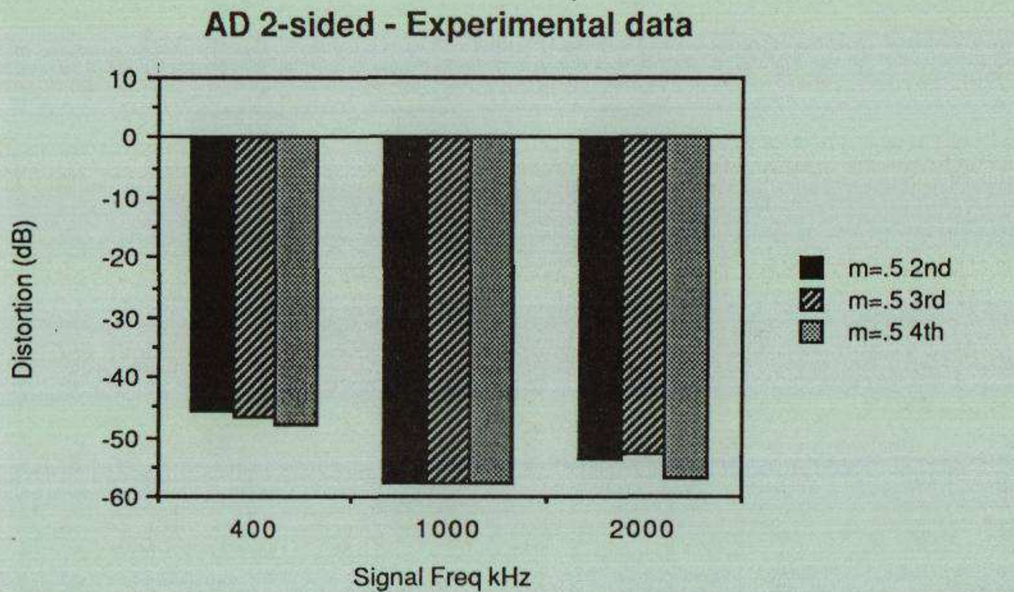

Graph 2



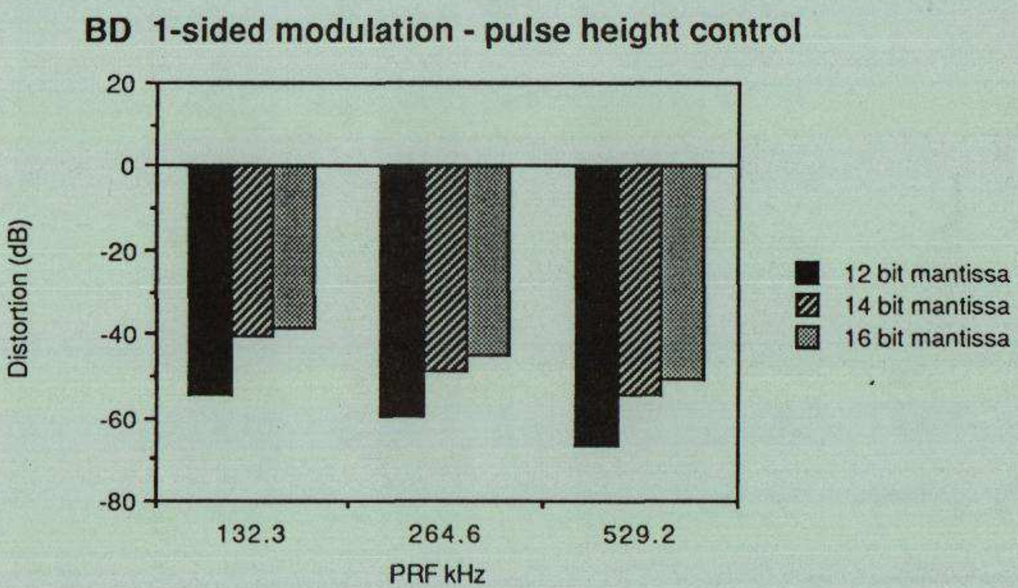

Graph 3




Graph 4




Graph 5




Graph 6