PULSE-ANALOGUE MODULATION

In pulse analogue modulation (PAM) a parameter of a pulse train is varied in sympathy with the signal. Since the modulated signal is discrete in time we have to be able to recover it from its discrete values. Pulse modulation requires the sampling theorem also known at ***Kotelnikov's theorem or Shannon's theorem;*** which states ***that a signal with no frequency components greater than f_m Hz is uniquely specified by its instantaneous values uniformly separated by time intervals of T where $f_s = 1/T \geq 2f_m$***

NOTE

1. T is related to f_m - which is the maximum frequency component in the signal. This is expected since the faster are the variations of the signal with time, the closer the samples should be to follow the signal.

2. Provided that $f_s \geq 2f_m$ then no information is lost by sampling and m(t) may ideally be precisely reconstructed from the samples.

In PAM there are three possible parameters that can be changed: (1) amplitude resulting in PAM (Pulse Amplitude Modulation), duration giving PWM (Pulse Width Modulation) and position giving PPM (Pulse Position Modulation). Since PWM and PPM relate to the time axis they are usually referred to as Pulse Time Modulation. See Figure 1 for the different types of PAM.

1. In PAM: the modulated pulses have a fixed width, but the amplitude varies in proportion to the value of the sample of the signal at that instant in time. As shown in Figure 1.

2. In PWM, the pulse duration (width) varies in proportion to the value of the sample of the signal while the amplitude of the pulse remains unchanged.

3. In PPM, the position of the pulse is delayed from some instant in proportion to the value of the sample of the signal while again the amplitude of the pulses and their duration remain unchanged.

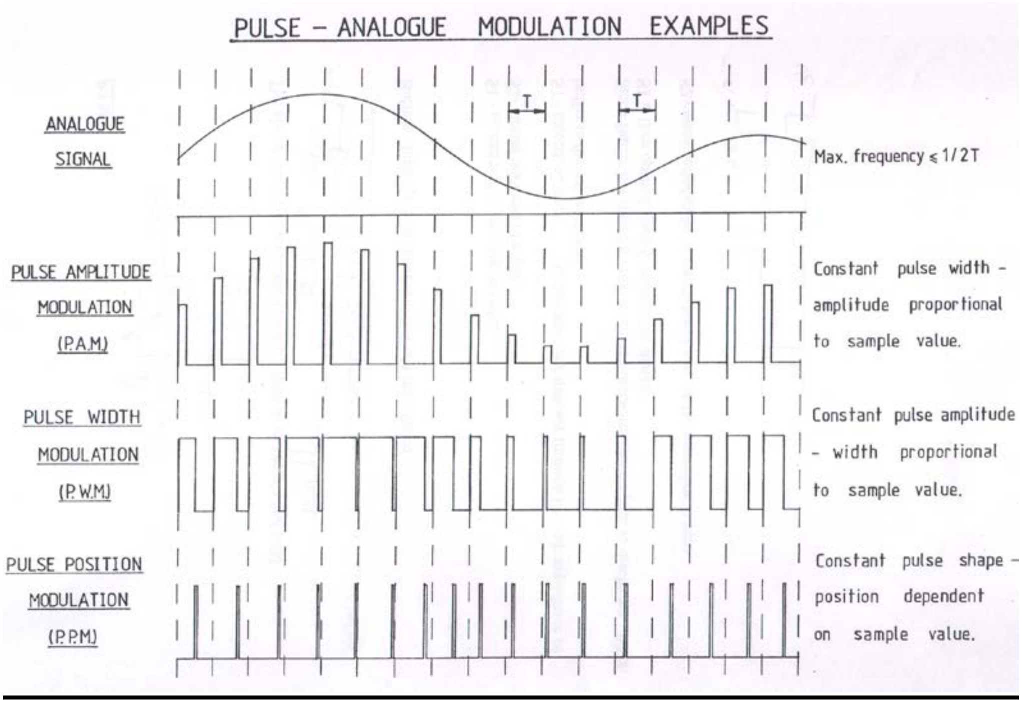


Figure 1 Illustration of Pulse Analogue Modulation

PAM generation and demodulation

Generation of PAM can be achieved simply by using a gate that opens to let the signal through when the pulse is on and closes it when the pulse is off as shown in Figure 2.

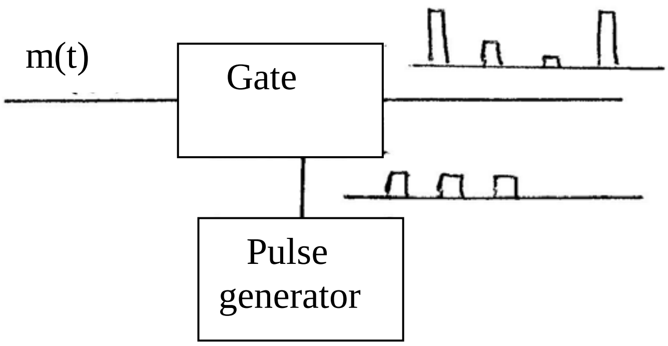
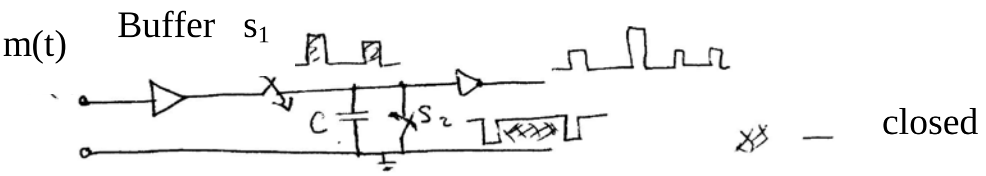


Figure 2. Block diagram for the generation of PAM

PAM can also be achieved by a sample and hold circuit as illustrated below.



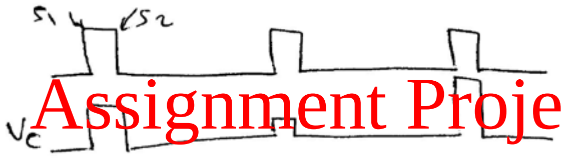
Buffers - high i/p impedance and low **o/p** impedance

S1 - closed with beginning of pulse
S2- closed with end of pulse

S1 - closed, s2 open   **C** charges very **quickly** through low o/p impedance of buffer to the value of m(t)

Since Buffer has high i/p impedance, the capacitor has no path to discharge. When **S1** is then opened, then **C** retains the charge.

S2 closes periodically to return the charge on the capacitor to zero.



Demodulation: Since the spectrum of a sampled signal contains the spectrum of the original signal then recovery of the original signal can be achieved simply by using a low pass filter. The filter can be also preceded by a sample and hold circuit to elongate the width of the pulses.

PWM and PPM generation and demodulation

Figure 3 displays a block diagram for the generation of PWM and PPM signals. In order to change the amplitude variations of the signal into time variations we use a saw-tooth signal which has a linear voltage versus time. The sawtooth signal is added to the waveform sampled using a sample and hold circuit (S/H). Using a fixed threshold the resulting width of the pulses is proportional to the amplitude of the signal as seen in the figure. For this circuit to work the signal must have positive voltage values only. Using the saw-tooth signal effectively changes the PAM signal at the output of the S/H into PWM.

A PPM signal is then generated from the PWM signal simply by using a negative edge triggered pulse generator to generate pulses of equal width but the actual location of the pulses on the time axis is related to the amplitude of the original signal.

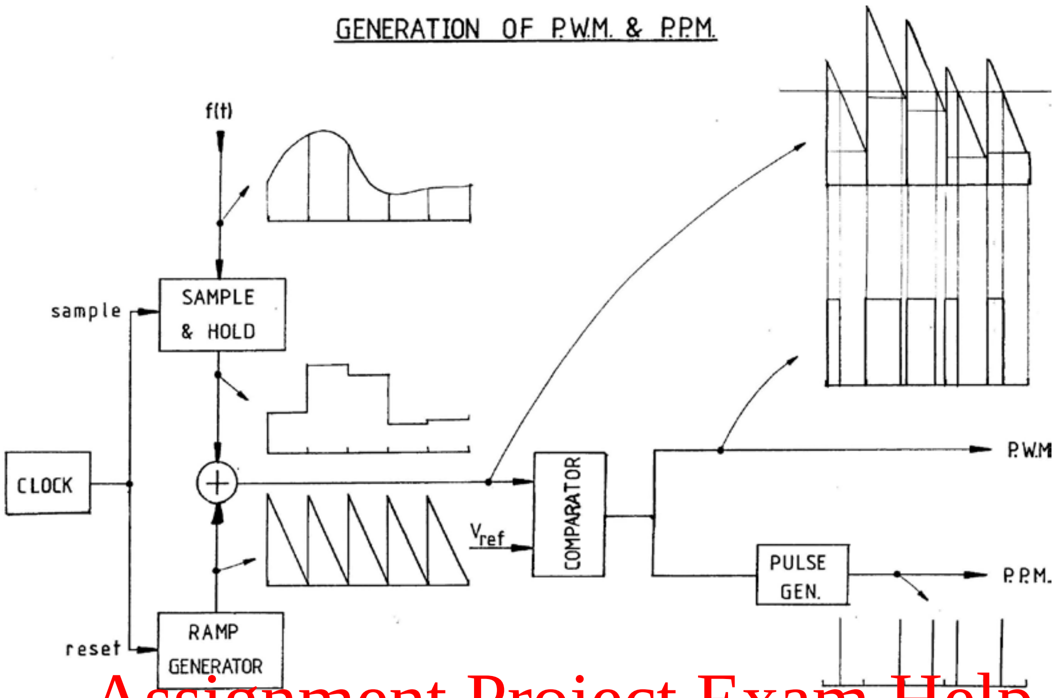


Figure 3 Block diagram of circuit used to generate PWM and PPM signals

Demodulation

To demodulate PWM signals we can also use a LPF since its spectrum contains a component at baseband [**proof** is **too** involved].

Alternatively we can exploit the position of timing edges to change PWM into PAM, which is similar to FM demodulation. A block diagram of a demodulating circuit is shown in Figure 4. In the figure there are two mono-stables one which is triggered by the positive edge of the PWM signal and the second one is a negative edge triggered mono-stable which is trigged by the negative edge of the PWM signal. Using an integrator which starts with the positive edge and stops with the negative edge, and sampling it following the negative edge, the voltage on the sample and hold circuit is then proportional to the duration of the pulse. This effectively changes a PWM signal into a PAM signal. In the case of PPM – the information regarding the start of the pulse is **lost, and** only information about the stop is available i.e. to detect PPM we need a generator or a circuit (CT) to derive the start pulses as shown in Figure 5 where the positive triggered mono-stable is replaced by a local clock generator synchronised to the transmitter.

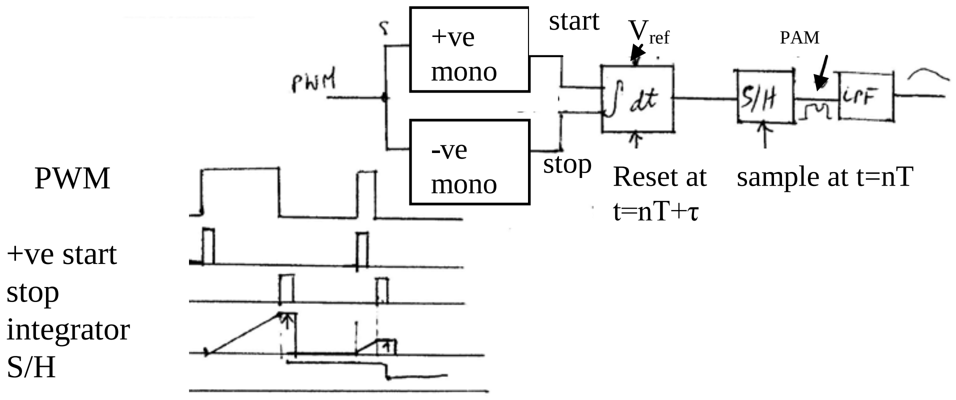


Figure 4 Block diagram of PWM demodulating circuit.

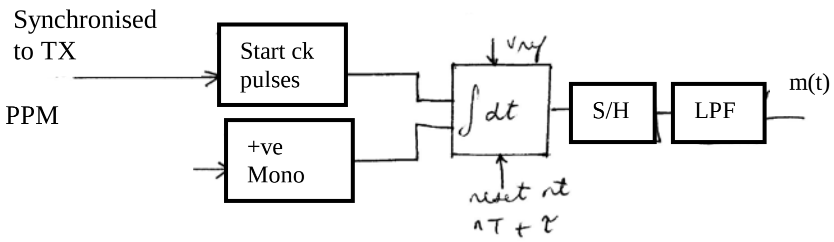


Figure 5 Block diagram for demodulating a PPM signal

General Comments

1. PAM is the most straightforward technique and is closely related to ideal sampling. Its performance is comparable to AM in CW modulation.

2. PAM is less tolerant to additive noise than PWM or PPM. This follows since the information in PAM is related to the amplitude of the pulses whereas in PTM, the amplitude is constant. So as shown in Figure 6 if the amplitude of the signal is changed by additive noise; this cannot be eliminated from a PAM signal. Whereas for PTM we can use a slicer to eliminate the noise since the noise cannot move the edges unless the noise is of the same magnitude as the signal at which point it rapidly engulfs the signal threshold. That is if the edges of the pulses are vertical then additive noise does not have any time to exert any perturbations on the signal hence, additive noise has no effect because noise introduces only vertical perturbations. However in real systems, vertical edges can not be transmitted since channels tend to be bandlimited, hence PTM is also affected by noise.

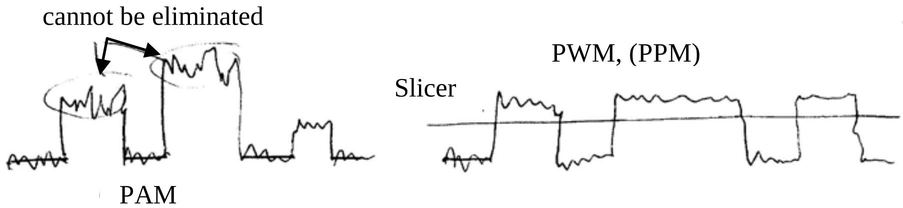


Figure 6 Illustration of effective of additive noise on PAM (left) and PWM, and PPM (right)

3. In PWM only the edges convey information. Hence, energy is wasted in long pulses, that is why PPM is a more useful technique since the width of the transmitted pulses is the same; i. e. no wastage of power. However, in PPM we need synchronisation to the transmitted pulses because the self-clocking of PWM is lost.

PPM utilises the transmitted power more efficiently than PWM since it avoids wastage of power in long pulses and yet retains information in the edges. i.e. **PPM** is the best pulse modulation scheme in terms of power and **SNR**

4. In obtaining the SNR advantage of pulse time modulation we had to pay in complexity and <u>time</u> since both PWM and PPM require more space on the time axis than PAM to be sure that adjacent samples do not overlap. If pulses are free to shift around or to get wider as in PPM and PWM, one cannot simply insert other pulses in the spaces i. e. multiplex and be confident that no interaction will occur.

5. From the sampling theorem, the spectrum of a sampled signal is infinite since it repeats at multiples of the sampling frequency. So what is the point of using **pulse** analogue modulation since we can not multiplex in frequency? Although we can not multiplex in frequency we can use a second form of multiplexing- known as Time Division Multiplexing TDM as shown in Figure 7 since to separate signals we need to separate them either in time or in frequency.

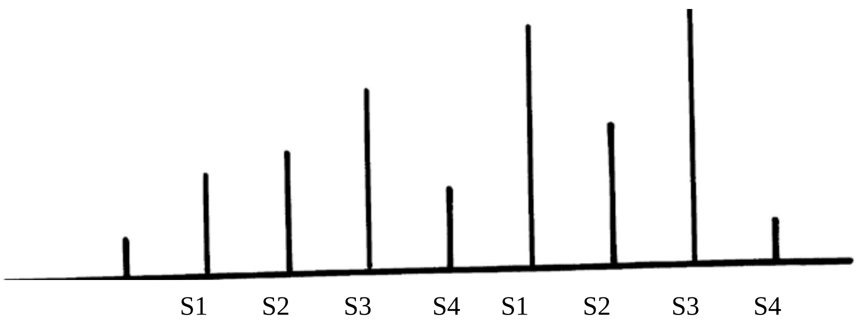


Figure 7. Illustration of time division multiplexing of four signals

Figure 8 shows a block diagram for generating a time division multiplexed signal using a commutator.

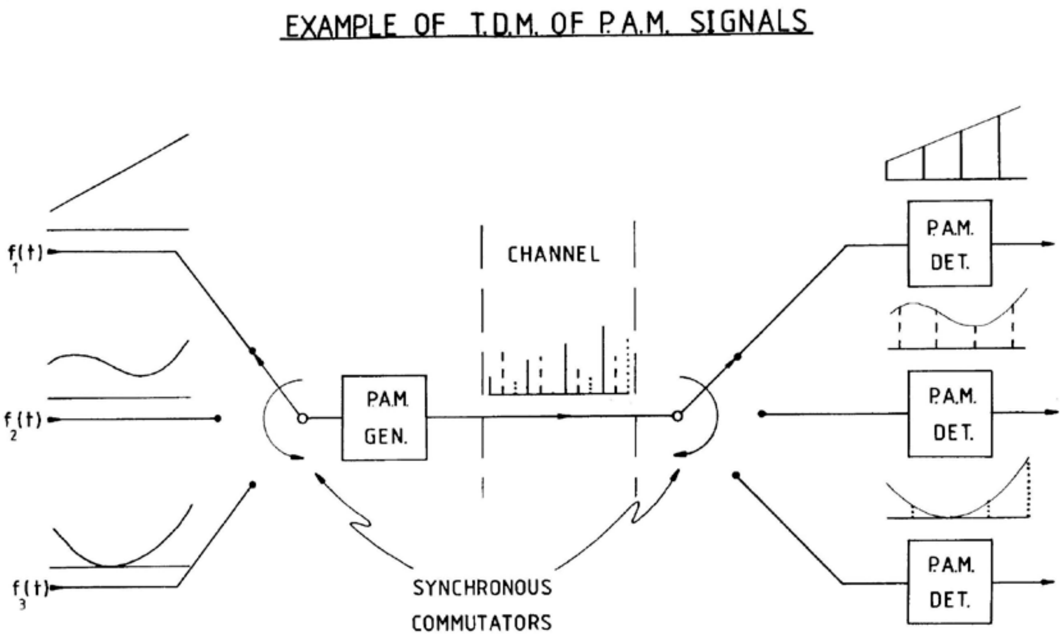


Figure 8. Generation of TDM signal

<u>NOTES on TDM</u>

1. It needs synchronous operation between the commutator and the de-commutator.

2. All input signals should have about the same bandwidth (BW) for efficient utilisation of **TDM** so that **they can** all be sampled at the same rate.

3. For a **TDM PAM** signal, **BW= n/2T Hz** for n channels. Recall from the sampling theorem that for a signal with f_{max}, we need **to sample at** $f_s \geq 2f_{max}$. For n channels we **need to** increase the sampling rate at the commutator by a factor equal to n

Hence, f^*_s =n/T or $f^*_{max} = f^*_s/2 = n/2T = n\, f_{max}$.

For n channels we can restrict the BW of transmission to $\geq$ n/2T Hz if we wish to further modulate the TDM pulses by continuous wave (CW) to transform the baseband signal to higher frequencies.

NOTE that limiting the BW leads to distortion of the edges of the transmitted pulses i. e. to transmit sharp edges we require an infinite BW. However, if only the amplitude is important, then we can limit the BW before transmission.

That is for PPM and PWM we require infinite BW. If the BW is limited then the identification of consecutive pulses is reduced. A limited bandwidth disperses the pulse (elongates the width of the pulses) as shown in Figure 9 resulting in their possible overlap.

Figure 9 Dispersion of pulse width due to limited bandwidth

$\Delta\tau$: timing accuracy
t_r : rising time

The timing accuracy is also affected by the presence of noise and in general it is related to the bandwidth B as follows:

$\Delta\tau \sim 1/2B$ for low SNR (Signal to Noise Ratio)

$\Delta\tau \sim 1/(B\,(SNR)^{1/2})$ for high SNR

When the **SNR=4 i.e.** 6 dB the timing accuracy $\Delta\tau$ using the relationship for high SNR is equal to using the relationship for low SNR. Thus this value is taken as the threshold point.

Comparison of **TDM** and **FDM**

a. BW efficiency - if all channels are of equal BW then TDM and Single Side Band, (SSB) Frequency Division multiplexed (FDM) have equal efficiency.

For FDM : BW = n x f_m Hz.

For **TDM: BW =n/2T=n*f*** f_m Hz where $1/2T = f_m$ in one baseband signal

b. effect of limiting BW:

FDM loses channels

TDM gets evenly spread cross talk (dispersion) i. e. spill over from one pulse over the next.

c. effect of non-linearities :

FDM: lower channels harmonics **give** rise to cross-talk in higher channels.
TDM: distortion occurs evenly across all channels i.e. less critical with TDM.

d. Typical applications - telephone systems

FDM: n =12, **BW ~ 4kHz =48** kHz group

TDM: 24 channels at 8 kHz x 4 kHz BW for each channel
BW= 96 kHz.