PULSE ANALOG MODULATION TECHNIQUES

In Analog pulse modulation also known as continuous wave modulation, the sine wave is used as a carrier wave. In pulse communication system a train of pulse is used as a carrier wave. On superimposing the sinusoidal signal, one of the parameters of the pulse can be varied with respect to signal. Hence we have three types of pulse modulation.

- Pulse Amplitude Modulation (PAM)
- Pulse Width or Pulse Duration modulation (PWM)
- Pulse Position modulation.(PPM)

Pulse amplitude modulation is defined as the process of varying the amplitude of the pulse in proportion to the instantaneous variation of message signal.

Pulse width modulation is defined as a process of varying the width of the pulse in proportion to the instantaneous variation of message.

Pulse position modulation (PPM) is defined as the process of varying the position of the pulse with respect to the instantaneous variation of the message signal.

Difference between Pulse modulation and Analog modulation

Pulse modulation	Analog modulation
The modulated signal is in the form of pulses.	The modulated signal is in the form of continuous signals.
It is used sampling technique.	It is not used sampling technique.
It has required large bandwidth.	It has required less bandwidth.
Pulse modulation has both analog and digital nature.	It has only analog modulation.
In pulse modulation, the train of pulses is used as a carrier.	High frequency sine wave is used as carrier.
The input signal is either analog or digital.	Input signal is analog signal only.
The example of pulse modulation is PAM, PPM, PWM, DPCM, ADM etc.	The example of continuous wave modulation is AM , FM and PM
It is used in satellite communication.	It is used in radio and TV broadcasting.

Pulse Amplitude Modulation (PAM)

Pulse amplitude modulation is defined as the process of varying the amplitude of the pulse in proportion to the instantaneous variations of message signal.

Let the message signal be given by

$$v_m = V_m \sin \omega_m t$$

If x(t) is a periodic signal with period T_0, then it should satisfy the definition x(t) = x (t + T_0) . The pulse train is a periodic signal with some fundamental period T_0• Then the information present in each period of the pulse train is given by

$$P = Vp \qquad 0 \le t \le \Delta$$

$$=0 \qquad \Delta \le t \le T0$$

Where Δ is the width of the pulse and the leading edge of the pulse is assumed to be coinciding with the starting of the interval in each period.

The pulse amplitude modulated wave in the time domain is obtained by multiplying the message with the starting of the interval in each period and is given by

$$P0 = p \text{ x } v_m$$

Substituting *p* in the above equation we get

$$P0 = V_p V_m \sin \omega_m t \qquad 0 \le t \le \Delta$$

$$=0 \qquad \Delta \le t \le T0$$

The following figure shows the message, pulse train and PAM signal.

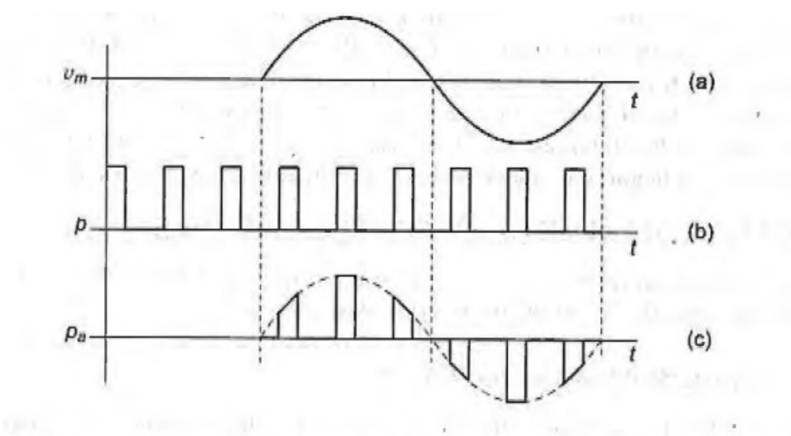
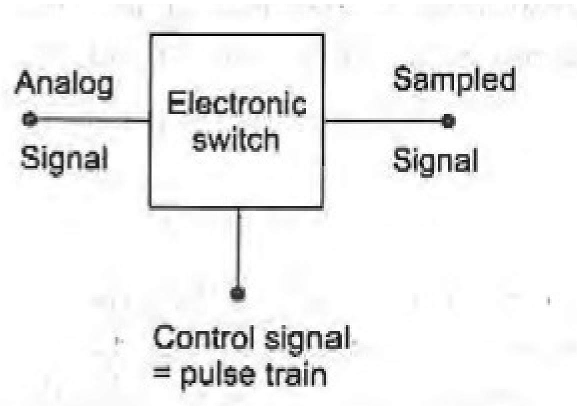


Fig. 5.1 Generation of PAM signal: (a) Message, (b) Pulse train, and (c) PAM.

The amplitude of the PAM signal is proportional to the instantaneous value of the message signal. The message signal is a low frequency signal. Multiplication of the two for generating the PAM signal results in the convolution of their spectra in the frequency domain. Thus PAM signal still retains the message spectrum in the low frequency range after modulation. PAM is not useful like AM for communication.

Sampling Process

Sampling is a signal processing operation that helps in sensing the continuous time signal values at discrete instants of time. The sampled sequence will have amplitudes equal to signal values at the sampling instants and undefined at all other times. This process can be conveniently performed using PAM described above.



The sampling process is as shown in Fig, The continuous time signal to be sampled, is applied to the input terminal. The pulse train is applied as the control signal of the switch. When the pulse occurs, the switch is in ON condition, that is, acts as short circuit between input and output terminals.

The output value will therefore be equal to input. During the other intervals of the pulse train, the switch is in OFF condition, that is, acts as open circuit. The output is therefore undefined. The output of the switch will be essentially a PAM signal. Any active device like diode, transistor or FET can be used as a switch

It is necessary to know how often the signal needs to be sampled, so that when needed an approximate version of the continuous time signal can be reconstructed. This is based on the sampling theorem which states that **"the sampling frequency (Fs) i.e., number of samples per second should be greater than or equal to twice the maximum frequency component (Fm) of the input signal".**

$$Fs \geq 2Fm$$

The minimum possible value of sampling frequency is known as Nyquist rate. Thus the sampling theorem will decide the periodicity associated with the pulse train. The second important aspect is the width of the pulse should not influence the amplitude of the sampled value. To minimize this effect, for all practical processing $\Delta \rightarrow 0$, so that the pulse train becomes on impulse train. The Fourier transform of an impulse train is also an impulse train in the frequency domain. Therefore convolution will not affect the shape of the sampled signal.

Problems:

1. A message signal made of multiple frequency components has a maximum frequency value of 4 kHz. Find out the minimum sampling frequency required according to the sampling theorem.

   Solution

   F s=4 kHz

$Fs \geq 2Fm = 2 X 4$ kHz = 8 kHz

2. A message signal has the following frequency components: a single tone sine wave of 500 Hz and sound of frequency components with lowest value of 750 Hz and highest value of 1800 Hz. What should be the minimum sampling frequency to sense the information present in this signal according to the sampling theorem?

   Solution

   F s=1800 Hz

   $Fs \geq 2Fm = 2 X1800$ Hz = 3600 kHz

Pulse Width Modulation

<u>Pulse width modulation (PWM) is defined as the process of varying the width of the pulse in proportion to the instantaneous variations of message. The amplitude and the position of the pulse remain constant.</u>

Let Δ be the width of the pulse in the unmodulated pulse train. ln PWM

$$\Delta \ \dot{\alpha} \ v_m$$

Mathematically, the width of pulse in PWM signal is given by

$$\Delta m = \Delta.(1 + v_m)$$

When there is no message, i.e., $v_m = 0$, then the width of the pulse will be equal to the original width Δ. For positive values of message, the width will be proportionately increases by $(1 + v_m)$ factor. For negative values of message, the width decreases by $(1 - v_m)$factor.

The following Figure shows the generation of PWM signal

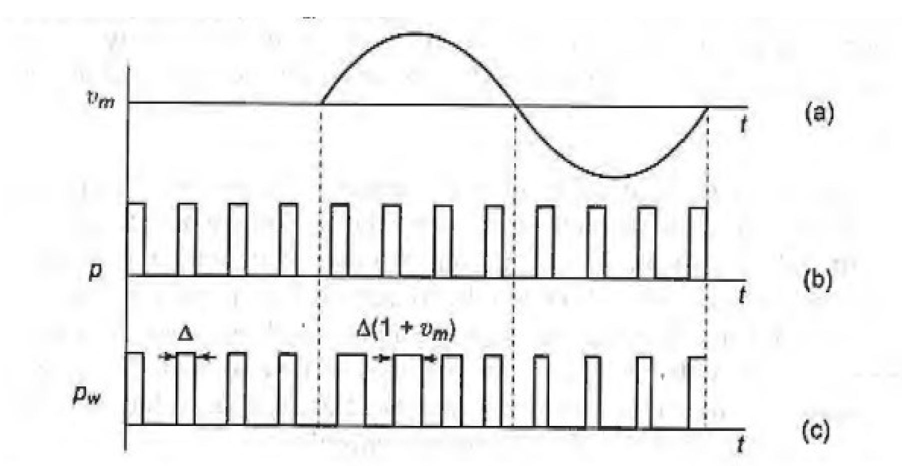


Fig. 5.3 *Generation of PWM signal. (a) Message, (b) pulse train and (c) PWM.*

There are three types of modulation

a) Trailing edge modulation
b) Leading edge modulation
c) Leading-trailing edge modulation.

Trailing edge modulation:

In this method, the position of the leading edge is fixed. But the position of the trailing edge is changed according to the required width of the pulse.

Leading edge modulation:

In this method, the position of the trailing edge is fixed. But the position of the leading edge is changed according to the required width of the pulse.

Speed Control of DC Motors using PWM

The speed of the dc motor depends on the average dc voltage applied across its terminals. Suppose V volts is the voltage for running the dc motor at its full speed, then 0 volt is the voltage for the rest condition of dc motor. Now, the speed of the dc motor can be varied from its rest to full speed value by varying the dc voltage. This can be conveniently performed with the help of PWM as illustrated in the following Fig.

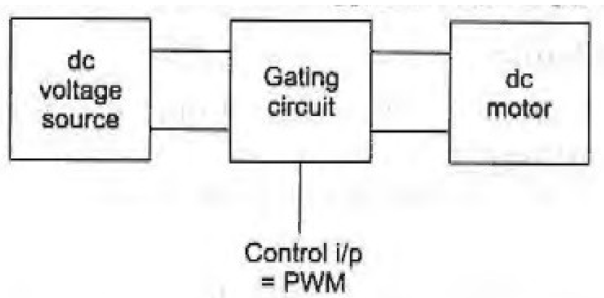


Fig. 5.4 *Speed control of dc motor using PWM.*

The constant dc voltage source is applied across the terminals of dc motor through a gating circuit controlled by the PWM signal. The gating circuit will essentially convert the constant dc source into a variable dc source. Suppose when there is no modulation, the width of the pulse will be the original value A and let this run the dc motor at some speed. Now when the width increases, the voltage value increases hence the speed increases. Similarly, if the width of the pulse decreases, the speed of the motor also decreases. Thus, PWM provides a convenient and efficient approach for the speed control of dc motors.

Pulse Position Modulation

Pulse position modulation (PPM) is defined as the process of varying the position of the pulse with respect to the instantaneous variations of the message signal.

Let t_P indicate the timing instant of the leading or trailing edge of the pulse in each period of the pulse train. In PPM

$$t_P \propto v_m$$

Mathematically, the position of the leading or trailing edge of the pulse (in each period) in PPM signal is given by

$$t_P = f(v_m)$$

When there is no message, then the position of the leading or trailing edge of the pulse will be equal to the original position and hence $t_P = 0$. For positive values of message, the position will be proportionately shifted right by

$$t_P = f(v_m).$$

For negative values of message, the position will be proportionately shifted left by $-t_P = -f(v_m)$ factor.

The following Figure shows the generation of PPM signal.

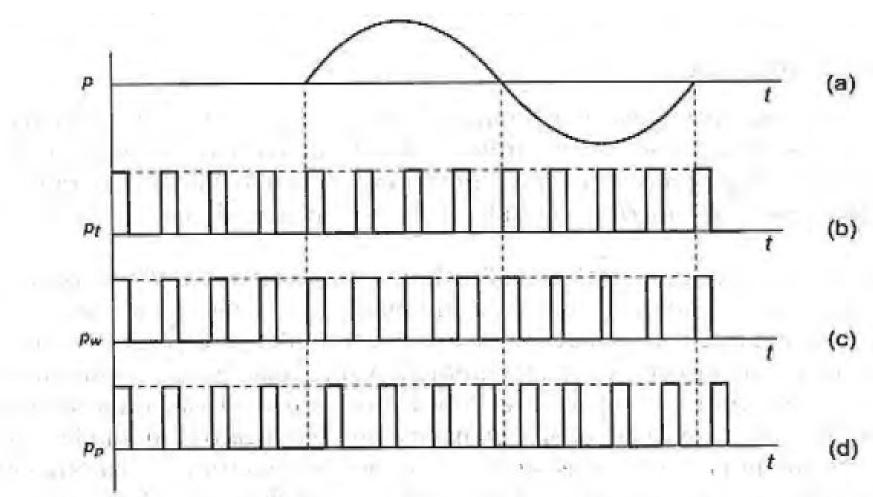


Fig. 5.5 *Generation of PPM. (a) Message, (b) pulse train, (c) PWM and (d) PPM.*

Let PWM is generated by varying the width of the trailing edge. Then this edge will be extracted to get the position of the pulse in each period. Once the position is extracted, the leading or trailing edge of the pulse is placed at this instant. The amplitude and width of the pulse remain constant as in the original pulse train. The resulting PPM will also have the spectrum in the baseband region itself:

Alternatively, if PWM is generated by varying the leading edge, then this edge is extracted to generate PPM and any edge can be used in case of modification of both edges. Even though, the PPM signal also contains the message information in the pulse train; it is seldom used due to its indirect way of storing message.

Comparison of PAM, PWM & PPM

Sl. N0	PAM	PWM	PPM
1	Amplitude of the pulse is proportional to amplitude of the modulating signal	Width of the pulse is proportional to amplitude of the modulating signal	The relative position of the pulse is proportional to amplitude of the modulating signal
2.	Band width of the transmission depends on the pulse width	Band width of the transmission depends on the rise time of the pulse.	Band width of the transmission depends on the rise time of the pulse.
3	Instantaneous power of the transmitter varies	Instantaneous power of the transmitter varies	Instantaneous power of the transmitter remains constant.
4	Noise interference is high	Noise interference is high	Noise interference is minimum
5	System is complex to implement	System is simple to implement	System is simple to implement
6	Similar to AM	Similar to FM	Similar to PM

DIGITAL TRANSMISSION

Digital Modulation is the technique of using digital signals such as binary or any other forms of discrete signals to modulate a carrier wave.

Advantages of Digital transmission:

1. Digital signals do not get corrupted by noise etc
2. Digital signals use less bandwidth. That is more information can be transmitted into the same space.
3. Digital transmission can be encrypted so that only the intended receiver can decode it
4. Digital signals can be checked for errors
5. Easier multiplexing of various forms of information like voice, data, video
6. Security – by using coding techniques to avoid jamming

Disadvantages

1. When analog signals are converted into digital signal, the conversion results in Quantization error
2. As more data is to required to be transmitted the power consumption high.
3. To transmit more data, the frequency of the signal is high which requires large bandwidth
4. As bandwidth is large, it is difficult transmit digital signals.

Pulse Code Modulation (PCM)

PCM is the digitally encoded modulation technique that is commonly used for digital transmission. In PCM, the analog signal is sampled and then converted to a serial n-bit binary code for transmission. Each code has the same number of bits. A pulse or lack of a pulse within a prescribed time slot represents either a logic 1 or a logic 0 condition. It requires the same time for transmission.

PCM is shown in Figure.

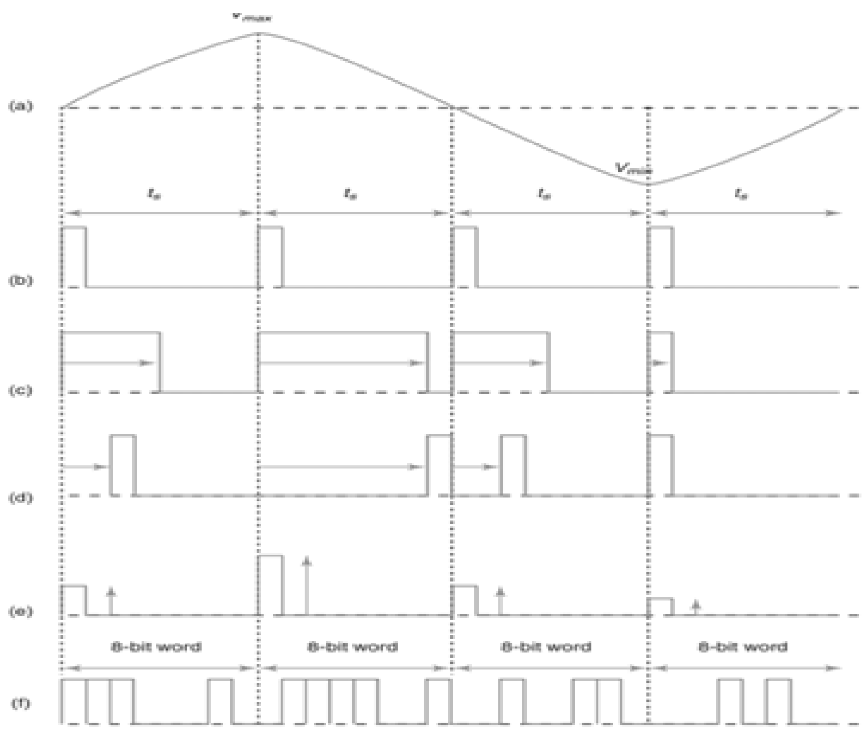


FIGURE 1 Pulse modulation: (a) analog signal; (b) sample pulse; (c) PWM; (d) PPM; (e) PAM; (f) PCM

Simplified block diagram of a single-channel, simplex PCM transmission system is as shown below.

The PCM transmission system has two sections – Transmitter and Receiver.

Transmitter:

The band pass filter limits the frequency of the analog input signal to the standard voice-band frequency range of 300 Hz to 3000 Hz. The sample-and-hold circuit periodically samples the analog input signal and converts those samples to a multilevel PAM signal. The analog-to-digital converter (ADC) converts the PAM samples to parallel PCM codes. These codes are converted to serial binary data in the parallel-to-serial converter and then outputted onto the transmission line as serial digital pulses. The transmission line has number of repeaters which are placed at prescribed distances to regenerate the digital pulses.

Receiver:

In the receiver, the serial-to-parallel converter converts serial pulses received from the transmission line to parallel PCM codes. The digital-to-analog converter (DAC) converts the parallel PCM codes to multilevel PAM signals. The hold circuit is basically a low pass filter that converts the PAM signals back to its original analog form.

PCM SAMPLING

The function of a sampling circuit in a PCM transmitter is to periodically sample the continuously changing analog input voltage and convert those samples to a series of constant amplitude pulses which can be easily converted into binary PCM code. For the ADC to accurately convert a voltage to a binary code, the voltage must be relatively constant so that the ADC can complete the conversion before the voltage level changes. If not, the ADC would be continually attempting to follow the changes and may never stabilize on any PCM code.

There are two basic techniques to perform the sampling function:

1. natural sampling and
2. Flat-top sampling.

Natural Sampling:

Natural sampling is shown in Figure

Natural sampling is one in which the tops of the sampled pulses retain their natural shape during the sample interval. This makes it difficult for an ADC to convert the sampled pulses to a PCM code. With natural sampling, the frequency spectrum of the sampled output is different from that of an ideal sample. The amplitude of the frequency components produced from narrow, finite-width sampled pulses decreases for the higher harmonics. This alters the information frequency spectrum requiring the use of frequency equalizers (compensation filters) before recovery by a low-pass filter

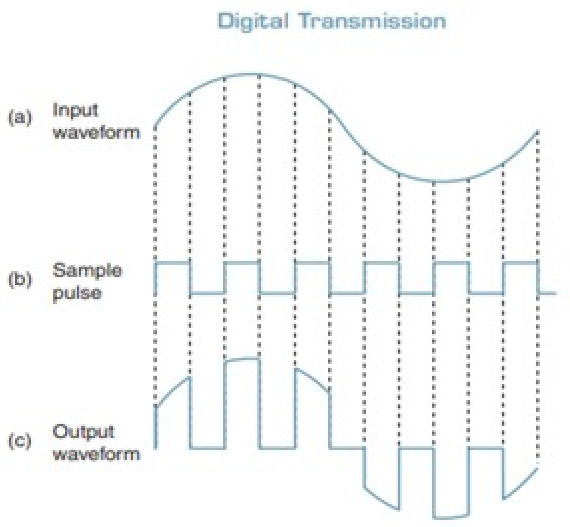


FIGURE 3 Natural sampling: (a) input analog signal; (b) sample pulse; (c) sampled output

Flat-top sampling

Flat-top sampling is shown in Figure

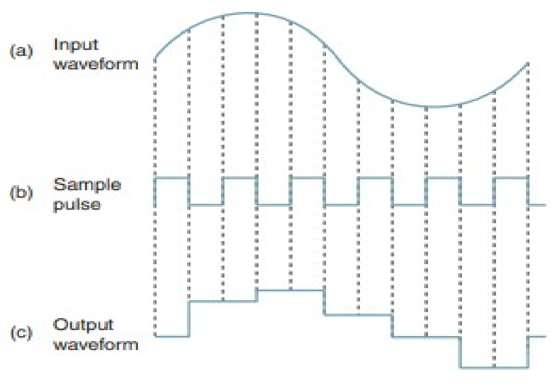


FIGURE 4 Flat-top sampling: (a) input analog signal; (b) sample pulse; (c) sampled output

The most common method used for sampling voice signals in PCM systems is flattop sampling, which is accomplished in a sample-and-hold circuit. The purpose of a sample and-hold circuit is to periodically sample the continuously changing analog input voltage and convert those samples to a series of constant-amplitude PAM voltage levels. With flat-top sampling, the input voltage is sampled with a narrow pulse and then held relatively constant until the next sample is taken.

The sampling process alters the frequency spectrum and introduces an error called **aperture error**. This prevents the recovery circuit in the PCM receiver from reproducing the original analog signal voltage. The magnitude of error depends on how much the analog signal voltage changes while the sample is being taken and the width (duration) of the sample pulse.

Sample and Hold Circuit

The schematic diagram of a sample-and-hold circuit is as shown below.

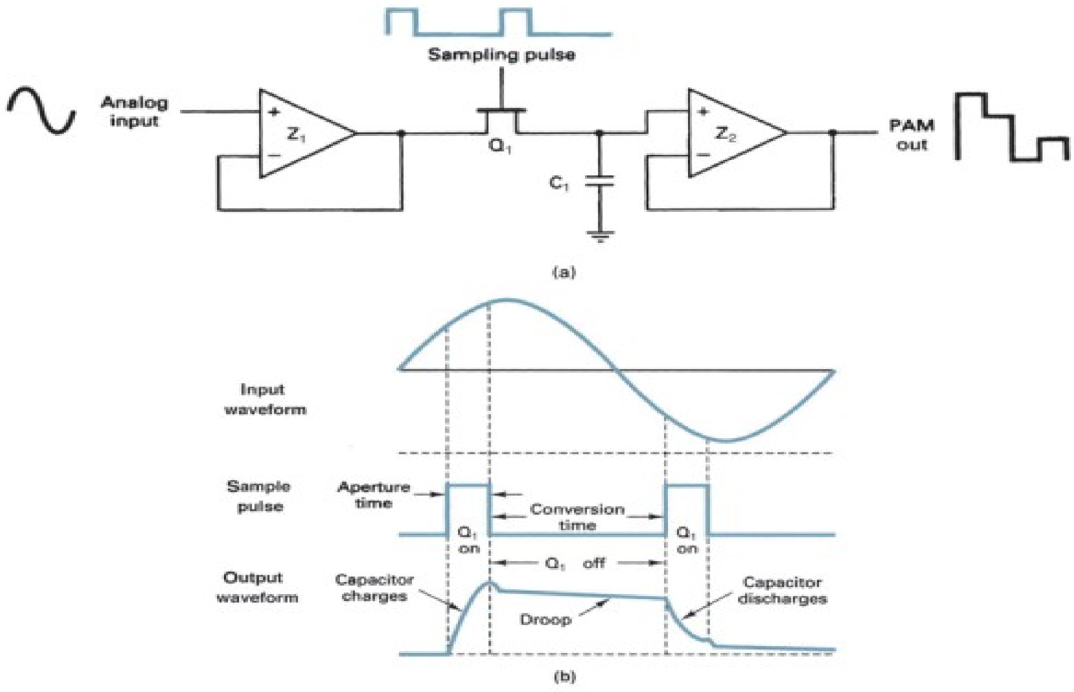


FIGURE 5 (a) Sample-and-hold circuit; (b) input and output waveforms

Working:

The FET acts as a simple analog switch. When Q1 is on, it provides a low-impedance path and the capacitor C1gets charged by the analog sample voltage. **The time that Q1 is on is called the aperture or acquisition time.** C1 is the hold circuit.

When Q1 is off, C1 does not have a complete path to discharge through and, therefore, stores the sampled voltage. The storage time of the capacitor is called the A/D conversion time because it is during this time the ADC converts the sample voltage to a PCM code. The acquisition time should be very short to ensure that a minimum change occurs in the analog signal while it is being deposited across C1. If the input to the ADC is changing while it is performing the conversion, aperture distortion results. Thus, by having a short aperture time and keeping the input to the ADC relatively constant, the sample-and-hold circuit can reduce aperture distortion.

To allow capacitor to charge or discharge rapidly during the short acquisition time, the RC charging time constant of the capacitor is kept very short. Hence the output impedance of voltage follower Z1 and the on resistance of Q1 be as small as possible. To limit the rapid drop in the capacitor voltage immediately following each sample pulse it is important that the input impedance of Z2 and the leakage resistance of C1 be as high as possible.

The wave forms of input analog signal, the sampling pulse, and the waveform developed across C1 is as shown above

Example

For the sample-and-hold circuit shown in Figure 5a, determine the largest-value capacitor that can be used. Use an output impedance for Z1 of 10 Ω, an on resistance for Q1 of 10 Ω, an acquisition time of 10 μs, a maximum peak-to-peak input voltage of 10 V, a maximum output current from Z1 of 10 mA, and an accuracy of 1%.

Solution

The expression for the current through a capacitor is

$$I = C\frac{dV}{dt}$$

Rearranging and solving for C

$$C = i\,\frac{dt}{dv}$$

where C = maximum capacitance (farads)
i = maximum output current from Z1, 10 mA
dv = maximum change in voltage across C1, which equals 10 V
dt = charge time, which equals the aperture time, 10 μs

Therefore,

$$\mathbf{C_{max}} = \frac{10\,mA\,x\,10\,\mu S}{10\,V} \ \mathbf{= 10\ nF}$$

The charge time constant for C when Q1 is on is

$$\tau = RC$$

where τ one charge time constant (seconds)

R output impedance of Z1 plus the on resistance of Q1 (ohms)

C capacitance value of C1 (farads)

Rearranging and solving for C gives us

$$C_{max} = \frac{\tau}{R}$$

The charge time of capacitor C1 is also dependent on the accuracy desired from the device

For an accuracy of 1%,

$$C = \frac{10\,\mu S}{4.6(20)} = 108\text{nF}$$

LINEAR VERSUS NONLINEAR PCM CODES

Linear versus Non-linear PCM codes

- For linear coding, accuracy of the higher amplitude analog signal is same as the lower order analog signal
- SQR for lower amplitude signal is less than the higher amplitude signal
- For voice transmission, low amplitude signals are more likely to occur than large amplitude signals.
- Thus non-linear encoding is the solution
- With non-linear encoding, the step size increases with the amplitude of the signal
- Non-linear encoding gives larger dynamic range
- SQR is sacrificed for higher amplitude signals to achieve more accuracy for the lower amplitude signals
- It is difficult to fabricate non-linear ADC

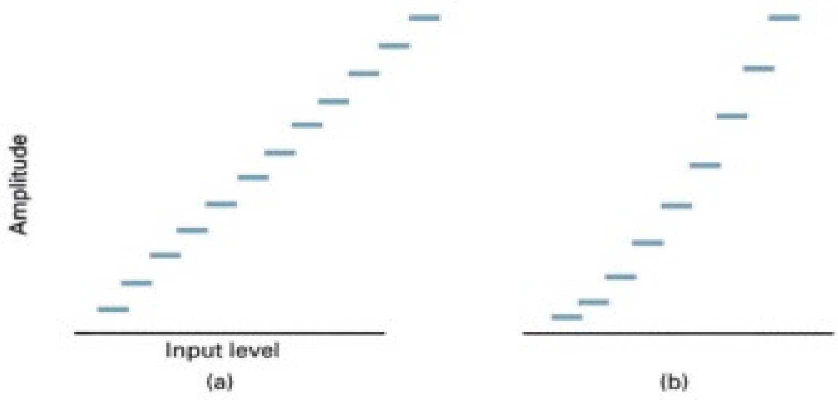


FIGURE 11 (a) Linear versus (b) nonlinear encoding

CODING METHODS

There are several coding methods used to quantize PAM signals into 2^n levels. These methods are classified as

- Level-at-a-Time Coding
- Digit-at-a-Time Coding
- Word-at-a-Time Coding

Level-at-a-Time Coding

This type of coding compares the PAM signal to a ramp waveform while a binary counter is being advanced at a uniform rate. When the ramp waveform equals or exceeds the PAM sample, the counter contains the PCM code.

- This type of coding requires a very fast clock if the number of bits in the PCM code is large.

- Also requires that 2^n sequential decisions be made for each PCM code generated.
- Limited to low-speed applications.

Digit-at-a-Time Coding

This type of coding determines each digit of the PCM code sequentially.

- It is analogous to a balance where known reference weights are used to determine an unknown weight.
- Digit-at-a-time coders provide a compromise between speed and complexity.
- One common kind of digit-at-a-time coder, called a feedback coder, uses a successive approximation register (SAR). With this type of coder, the entire PCM code word is determined simultaneously.

Word-at-a-Time Coding

- Word-at-a-time coders are flash encoders and are more complex; however, they are more suitable for high-speed applications.
- Logic circuits sense the highest threshold circuit sensed by the PAM input signal and produce the approximate PCM code. This method is again impractical for large values of n.