Analog Communication Systems
EC-413-F

Lecture no 3

Topics to be covered

Sampling Theory

Sampling and Pulse Modulation

- **The Sampling Theorem**
- **PAM -- Natural and Flat-Top Sampling**
  ***Time-Division Multiplexing* (*TDM*)**
  ***Intersymbol Interference* (*ISI*)**
- **Pulse Width and Pulse Position Modulation**
  Demodulation
- **Digital Modulation**
  ***Pulse Code Modulation* (*PCM*)**
  ***Delta Modulation* (*DM*)**
- **Qualitative Comparisons Of Pulse and Digital Modulation Systems**

The Sampling Theorem

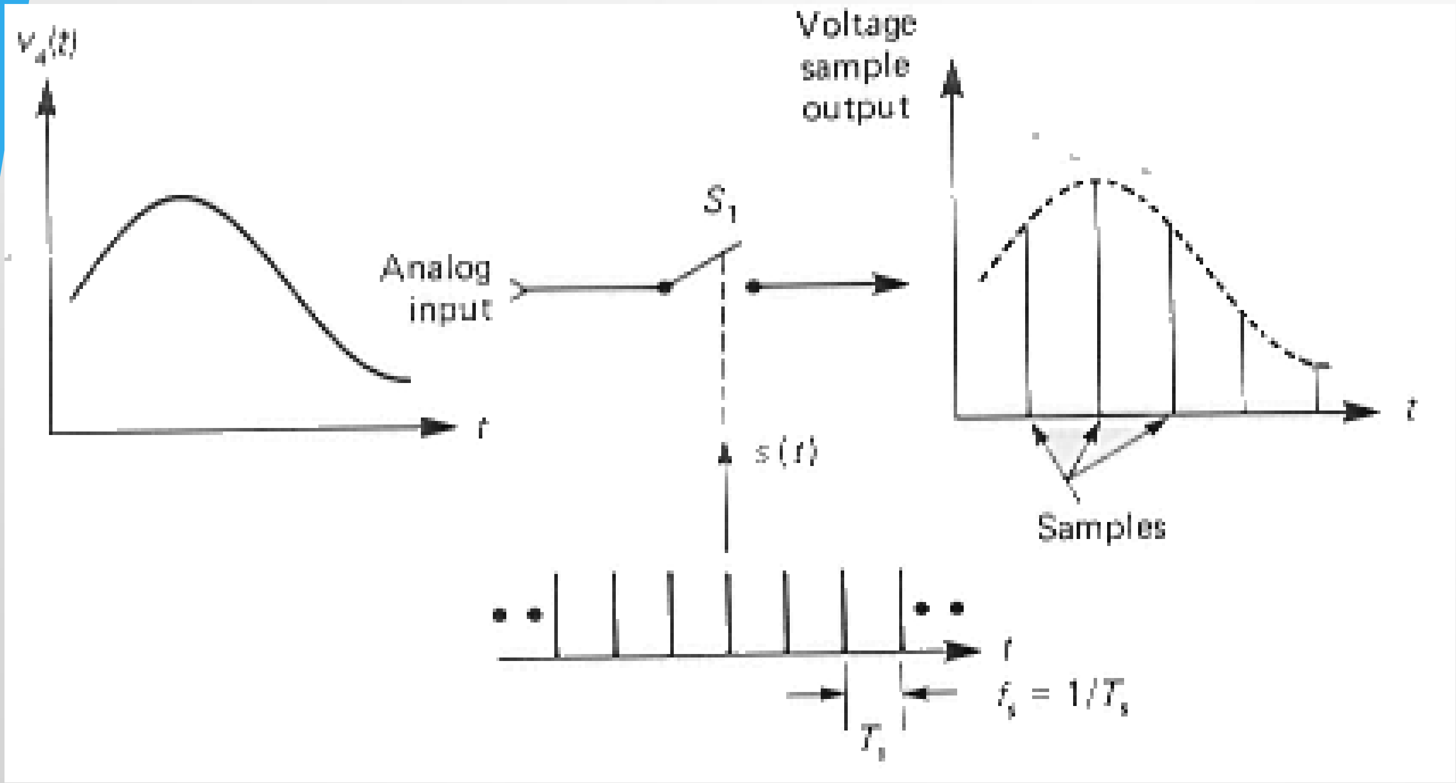


Figure 11-1. Impulse sampling of an analog voltage.

The Sampling Theorem

- **A sampler is a mixer with a train of very narrow pulses as the local oscillator input.**
- **If the analog input is sampled instantaneously at regular intervals at a rate that is at least twice the highest analog frequency**

$$f_s \geq 2f_a(\text{max})$$

- **then the samples contain all of the information of the original signal.**

The Sampling Theorem

- **The analog signal *v*(*t*) has a signal spectrum represented by the Fourier transform *V*(*f*),**
- **and the sampling signal**

$$s(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT_s)$$

consists of instantaneous impulses every nT_s sec, where $n = 0, \pm 1, \pm 2, \ldots$

$$S(f) = \frac{1}{T_s} \sum_{n=-\infty}^{\infty} \delta(f - nf_s)$$

- **The Fourier transform of *s*(*t*) is**

The Sampling Theorem

- The time-domain product performed by the sampler produces a sampled output spectrum given by

$$V_s(f) = \frac{1}{T_s} \sum_{n=-\infty}^{\infty} V(f - nf_s)$$

- where this spectrum consists of replicas of the analog signal spectrum $V(f)$, translated in frequency by each of the sampling frequency harmonics.

The Sampling Theorem

- The sampler is a wideband (harmonic) mixer producing upper and lower sidebands at each harmonic of the sampling frequency.
- Figure 11-2a illustrates the correct way to sample: if sampling is done at $f_s > 2f_A(\max)$ the upper and lower sidebands do not overlap each other,
- and the original information can be recovered by passing the signal through a low-pass filter (see Figure 11-2c and d).

The Sampling Theorem

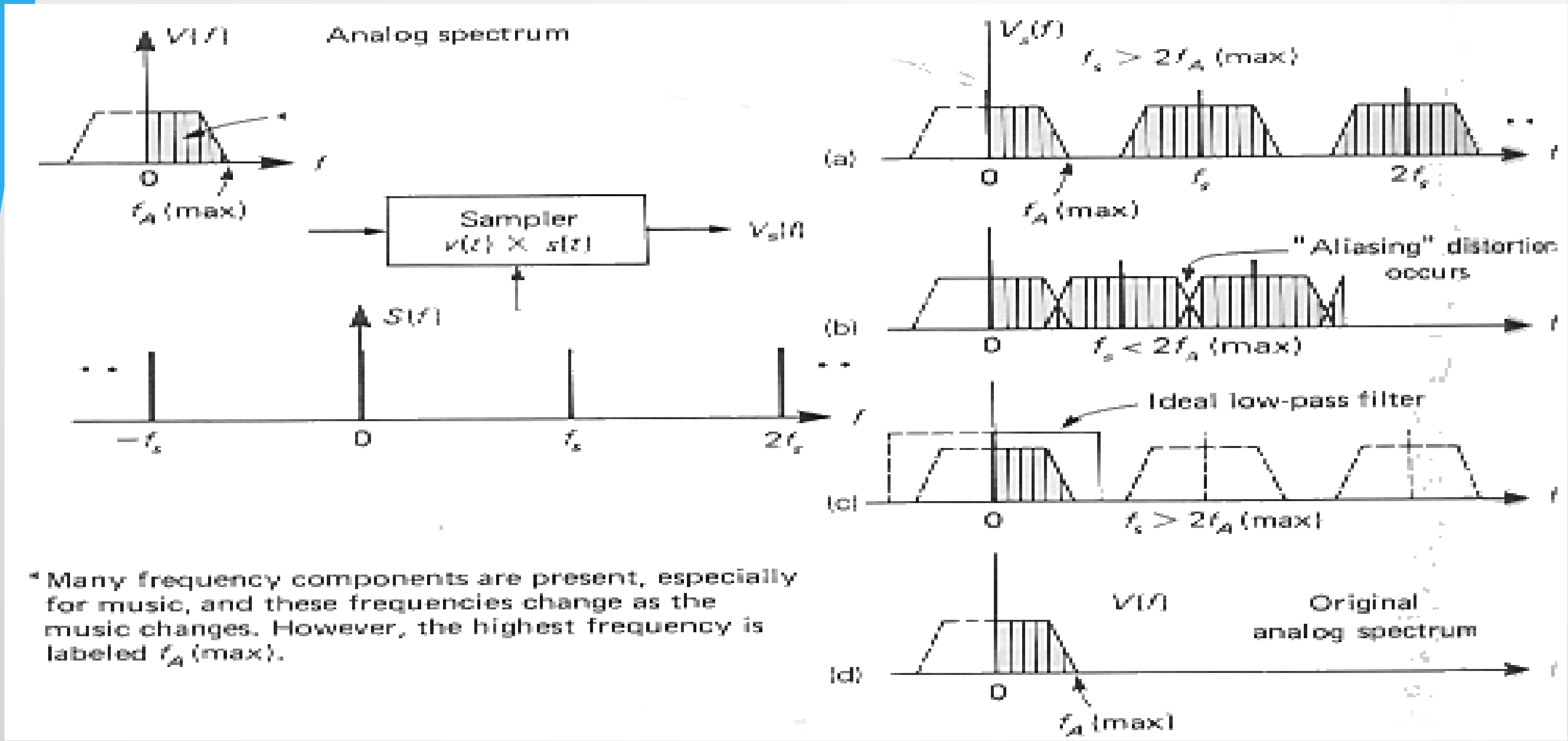
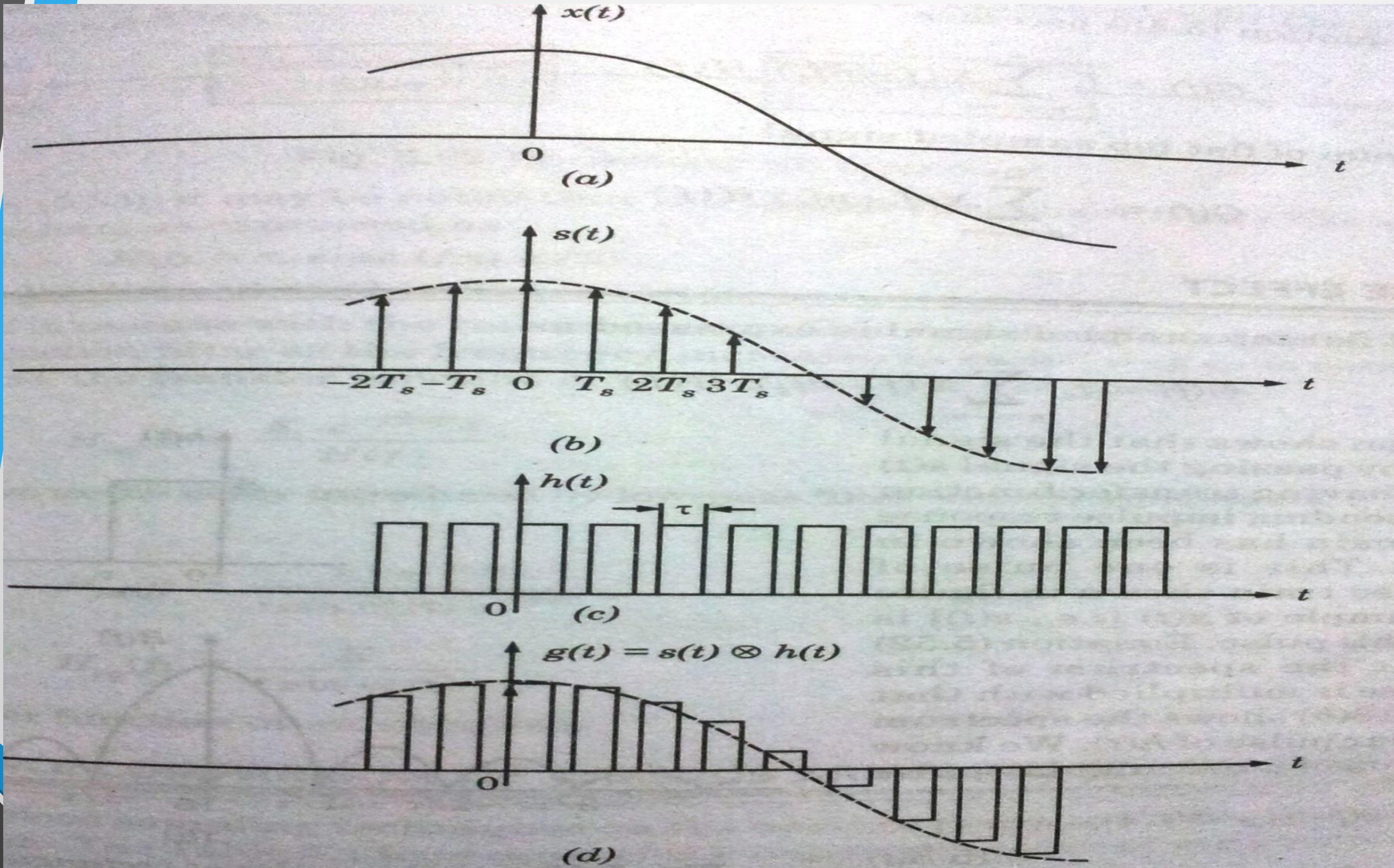


Figure 11-2. Sample spectra and their outputs. (a) $f_s > 2f_A$(max) Nyquist criteria met. (b) $f_s < 2f_A$(max) Frequency foldover of "aliasing" distortion occurs. (c) $f_s > 2f_A$(max) and recovery of original information with low-pass filter. (d) The original analog signal spectrum following recovery as in (c).

The Sampling Theorem

- However, if the sampling rate is less than the Nyquist rate, $f_s < 2f_A(\text{max})$ the sidebands overlap, as shown in Figure 11-2b.
- The result is *frequency-folding* or aliasing distortion, which makes it impossible to recover the original signal without distortion.

Wave form for flat top sampling



5.17 (a) Baseband signal $x(t)$ (b) Instantaneously sample signal $s(t)$ (c) Constant pulse width function $h(t)$ (d) Flat top sampled signal $g(t)$ obtained through convolution of $h(t)$ and $s(t)$

Pulse Amplitude Modulation – Natural and Flat-Top Sampling

- The circuit of Figure 11-3 is used to illustrate pulse amplitude modulation (PAM). The FET is the switch used as a sampling gate.
- When the FET is on, the analog voltage is shorted to ground; when off, the FET is essentially open, so that the analog signal sample appears at the output.
- Op-amp 1 is a non-inverting amplifier that isolates the analog input channel from the switching function.

Pulse Amplitude Modulation – Natural and Flat-Top Sampling

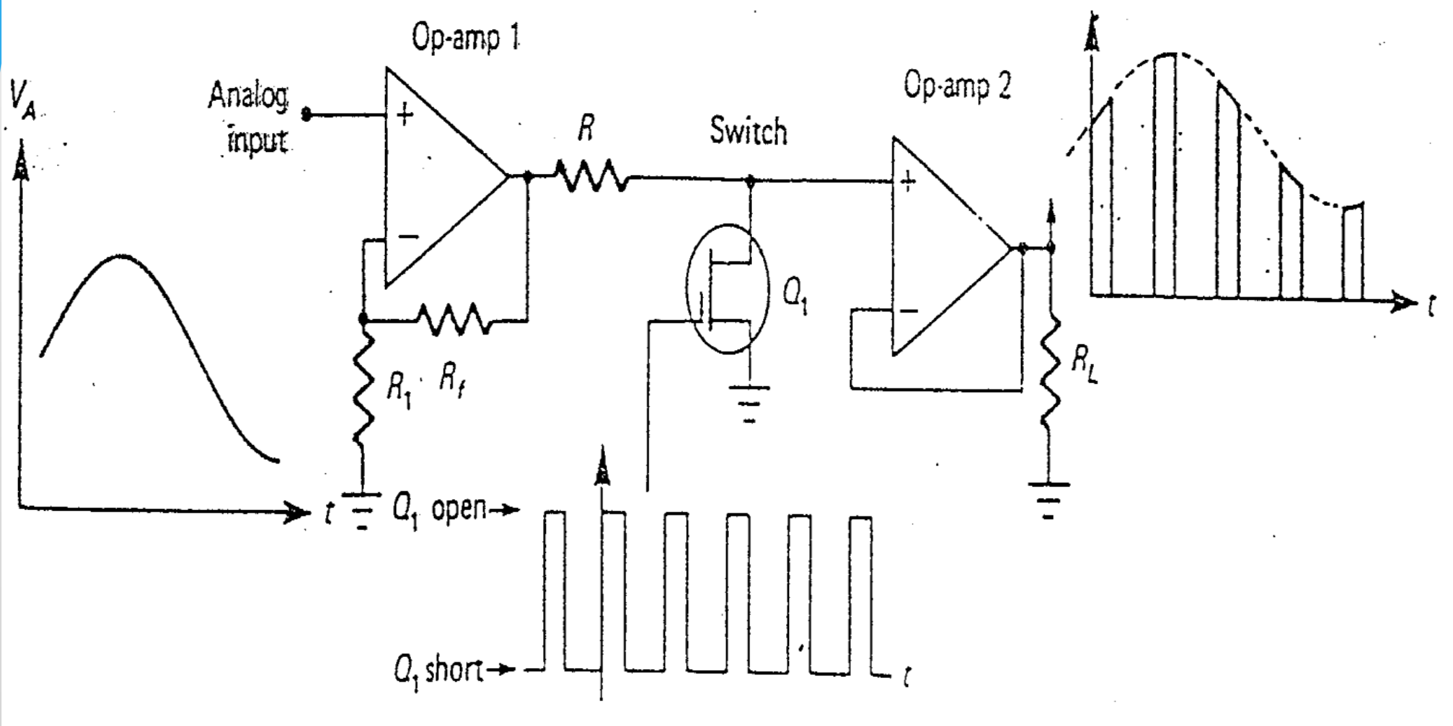


Figure 11-3. Pulse amplitude modulator, natural sampling.

Pulse Amplitude Modulation – Natural and Flat-Top Sampling

- Op-amp 2 is a high input-impedance voltage follower capable of driving low-impedance loads (high “fanout”).
- The resistor R is used to limit the output current of op-amp 1 when the FET is “on” and provides a voltage division with r_d of the FET. (r_d, the drain-to-source resistance, is low but not zero)

Pulse Amplitude Modulation – Natural and Flat-Top Sampling

- **The most common technique for sampling voice in PCM systems is to a sample-and-hold circuit.**

- **As seen in Figure 11-4, the instantaneous amplitude of the analog (voice) signal is held as a constant charge on a capacitor for the duration of the sampling period T_s.**

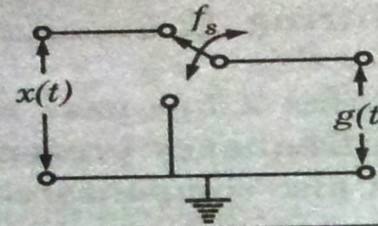
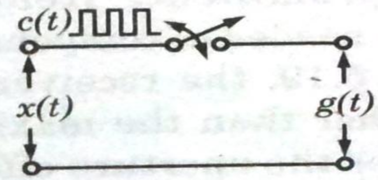
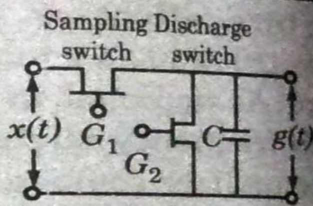
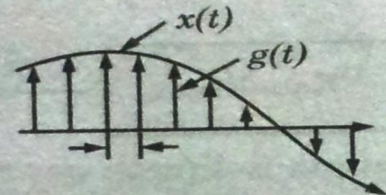
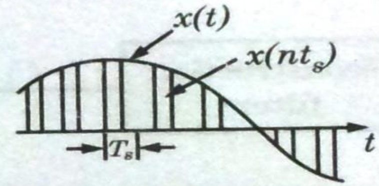
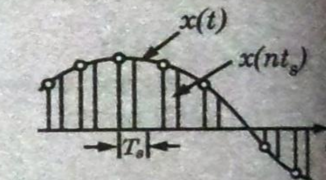
- **This technique is useful for holding the sample constant while other processing is taking place, but it alters the frequency spectrum and introduces an error, called aperture error, resulting in an inability to recover exactly the original analog signal.**

Pulse Amplitude Modulation – Natural and Flat-Top Sampling

- **The amount of error depends on how much the analog changes during the holding time, called aperture time.**

- **To recover signal reconstruction filter is used with an equalizer ckt.**

- **To estimate the maximum voltage error possible, determine the maximum slope of the analog signal and multiply it by the aperture time ΔT in Figure 11-4.**

S.No.	Parameter of comparison	instantaneous sampling		
1.	Sampling principle	It uses multiplication	It uses chopping principle	It uses sample and hold circuit
5.	Generation circuit			
3.	Waveforms involved			
4.	Feasibility	This is not a practically possible method	This method is used practically	This method is also used practically
5.	Sampling rate	Sampling rate tends to infinity	Sampling rate satisfies Nyquist criteria	Sampling rate satisfies Nyquist criteria
6.	Noise interference	Noise interference is maximum	Noise interference is minimum	Noise interference is maximum
7.	Time domain representation	$g(t) = \sum_{n=-\infty}^{\infty} x(nT_s)\delta(t - nT_s)$	$g(t) = \frac{\tau A}{T_s} \sum_{n=-\infty}^{\infty} x(t)\sin c(nf_s\tau) e^{j2\pi nf_s t}$	$g(t) = \sum_{n=-\infty}^{\infty} x(nT_s)h(t - nT_s)$
8.	Frequency domain representation	$G(f) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s)$	$G(f) = \frac{\tau A}{T_s} \sum_{n=-\infty}^{\infty} \sin c(nf_s\tau)X(f - nf_s)$	$G(f) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s)H(f)$

...signal is

Pulse Amplitude Modulation – Natural and Flat-Top Sampling

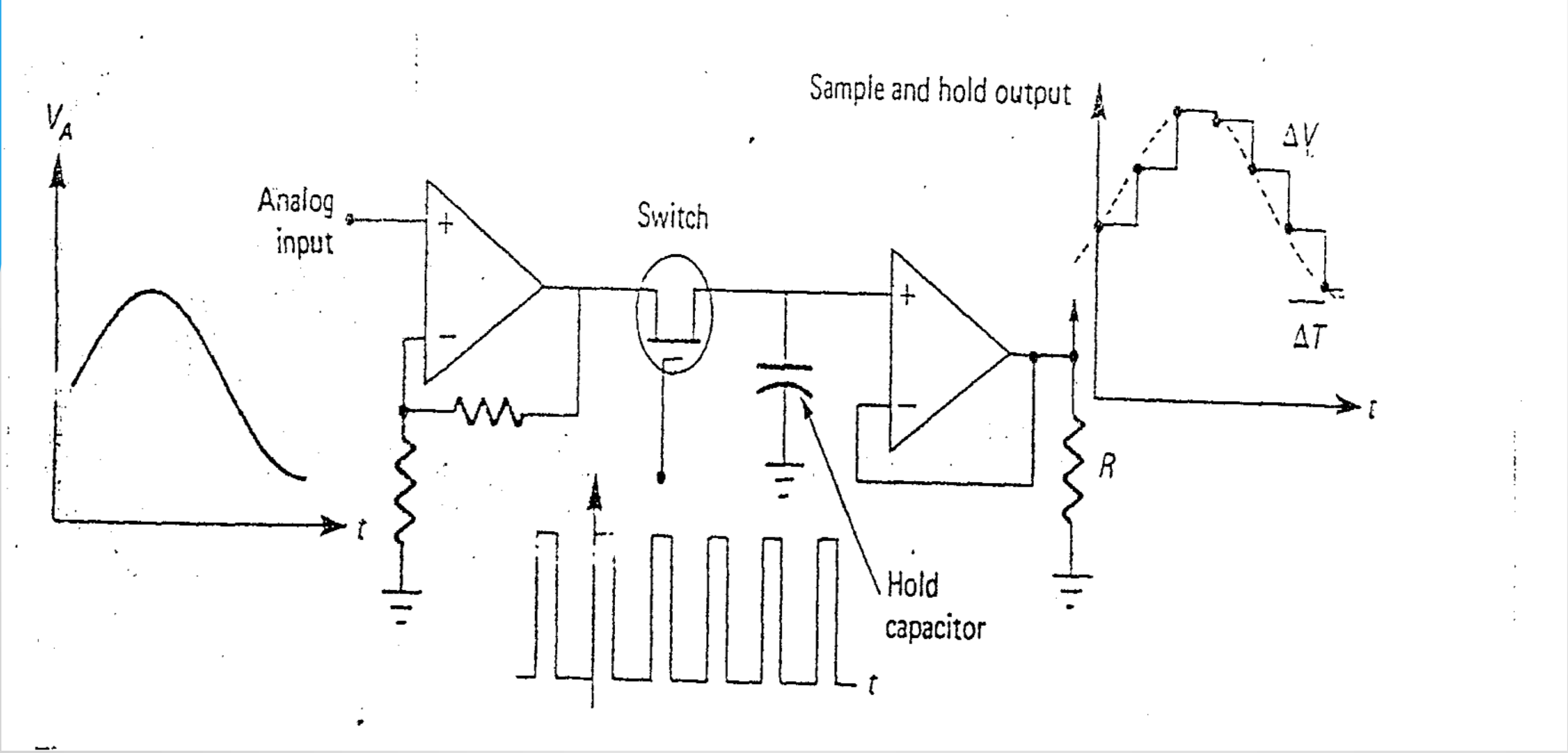


Figure 11-4. Sample-and-hold circuit and flat-top sampling.

Time-Division Multiplexing

- In the three-channel multiplexed PAM system of Figure 11-6, each channel is filtered and sampled once per revolution (cycle) of the commutator.
- Notice that the commutator is performing both the sampling and the multiplexing.
- The commutator must operate at a rate that satisfies the sampling theorem for each channel.
- Consequently, the channel of highest cutoff frequency determines the commutation rate for the system of Figure 11-6.

Time-Division Multiplexing

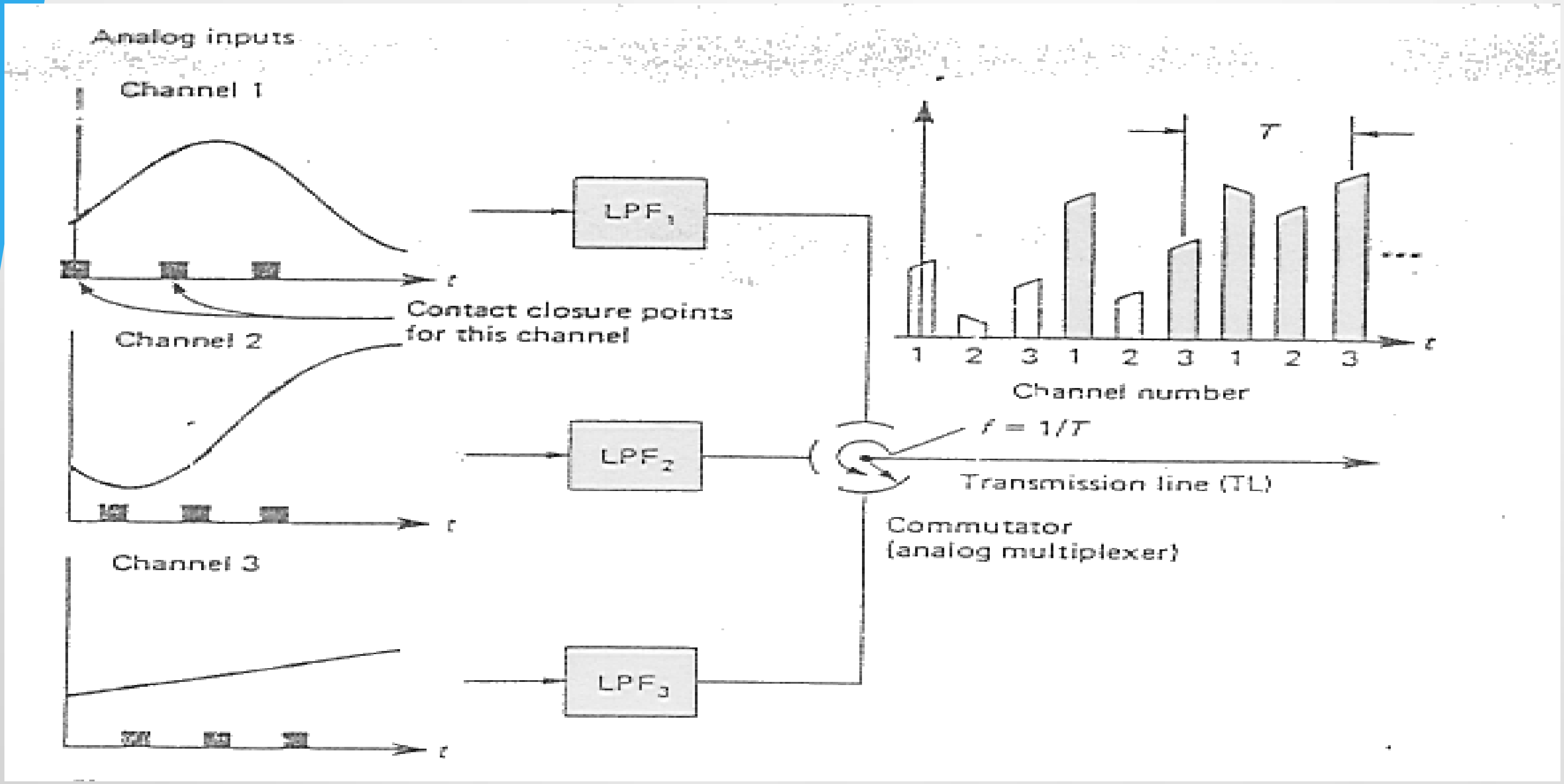


Figure 11-6. Time-division multiplex of three information sources.

Time-Division Multiplexing

- As an example, suppose the maximum signal frequency for the three input channels are

  $f_{A1}(\text{max}) = 4 \text{ kHz}$, $f_{A2}(\text{max}) = 20 \text{ kHz}$,

  and $f_{A3}(\text{max}) = 4 \text{ kHz}$.

- For the TDM system of Figure 11-6,

  the multiplexing must proceed at

  $f \geq 2f_A(\text{max}) = 40 \text{ kHz}$

  to satisfy the worst-case condition.

Time-Division Multiplexing

- We can calculate the transmission line pulse rate as follows:

  The commutator completes one cycle, called a frame, every 1/40 kHz = 25 μs.

- Each time around, the commutator picks up a pulse from each of the three channels. Hence, there are

  3 pulses/frame x 40k frames/s = 120k pulses/s.

Time-Division Multiplexing

- The 4 kHz channel is being sampled at five times the rate required by the sampling theorem. But if we slow down the commutator, the 20-kHz channel will be inadequately sampled.
- One the thought might be to multiplex at 8 k-frames/sec and sample the 20-kHz channel 5 times per frame.
- If you sketch this, as is done in Figure 11-7,

  you discover that there are

  7 pulses/frame x 8k frames/s = 56k pulses/s, which looks good.

Time-Division Multiplexing

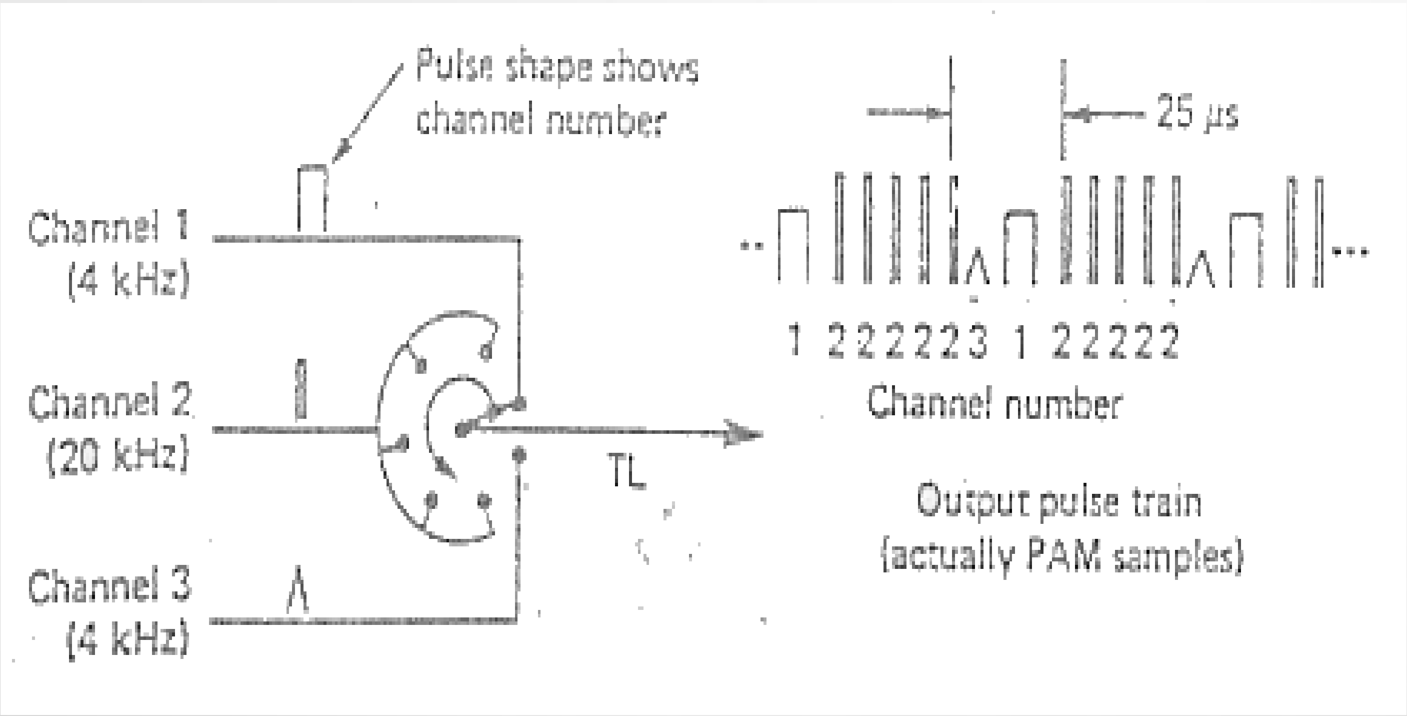


Figure 11-7. Possible TDM solution.

Time-Division Multiplexing

- The two missing samples stolen from the 20-kHz channel results in inadequate sampling and periodic aliasing distortion.
- For no errors, the commutation rate must be 17.14 kHz, producing 120k samples/s on the transmission line.

- A better scheme is shown in Figure 11-8 with insertion of channel 1 and 3 between two samples of channel 2.
- With 12.5 μs/pulse and 7 pulses/frame, the multiplexing rate can be

    (2 pulses/25μs)/(7 pulses/frame) = 11.428k frames/s

and

    (11.428k frames/s) x (7 pulses/frame) = 80k pulses/s

with no errors.

Time-Division Multiplexing

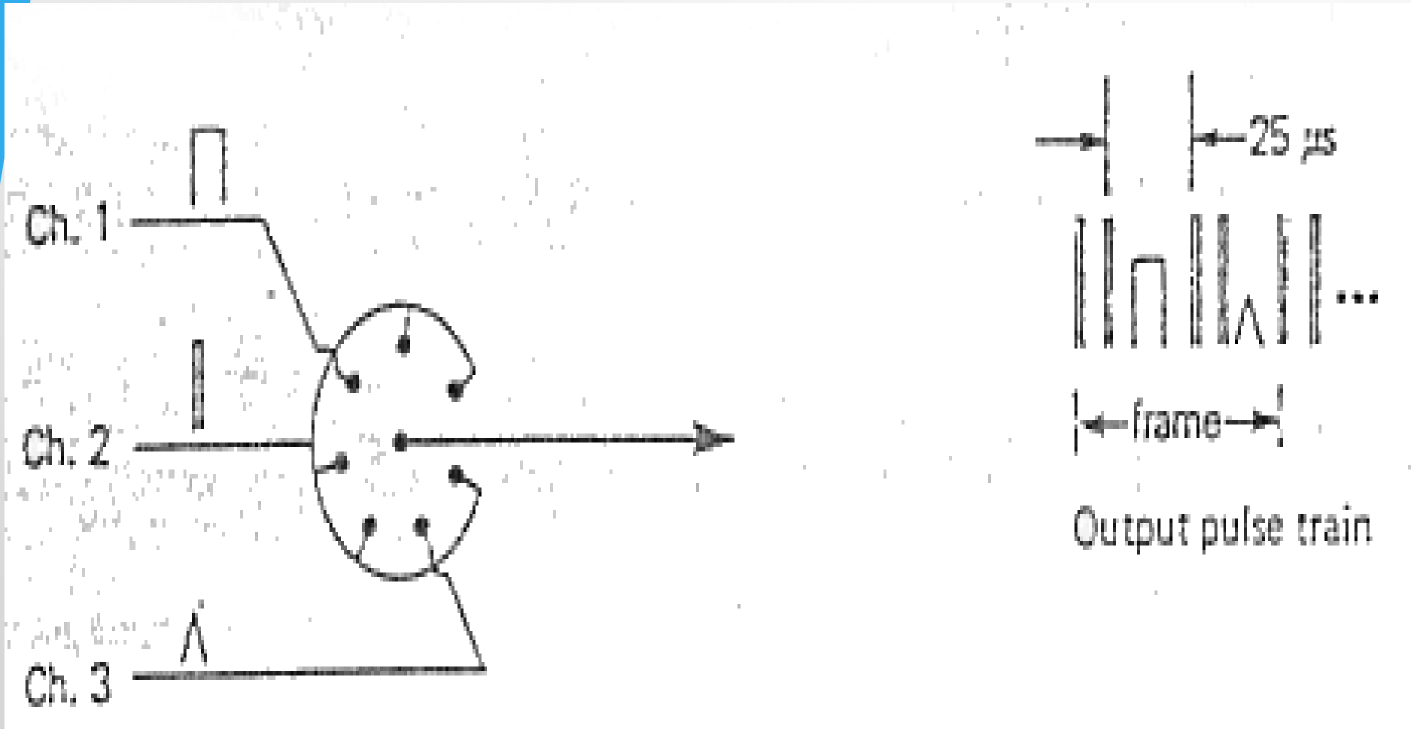
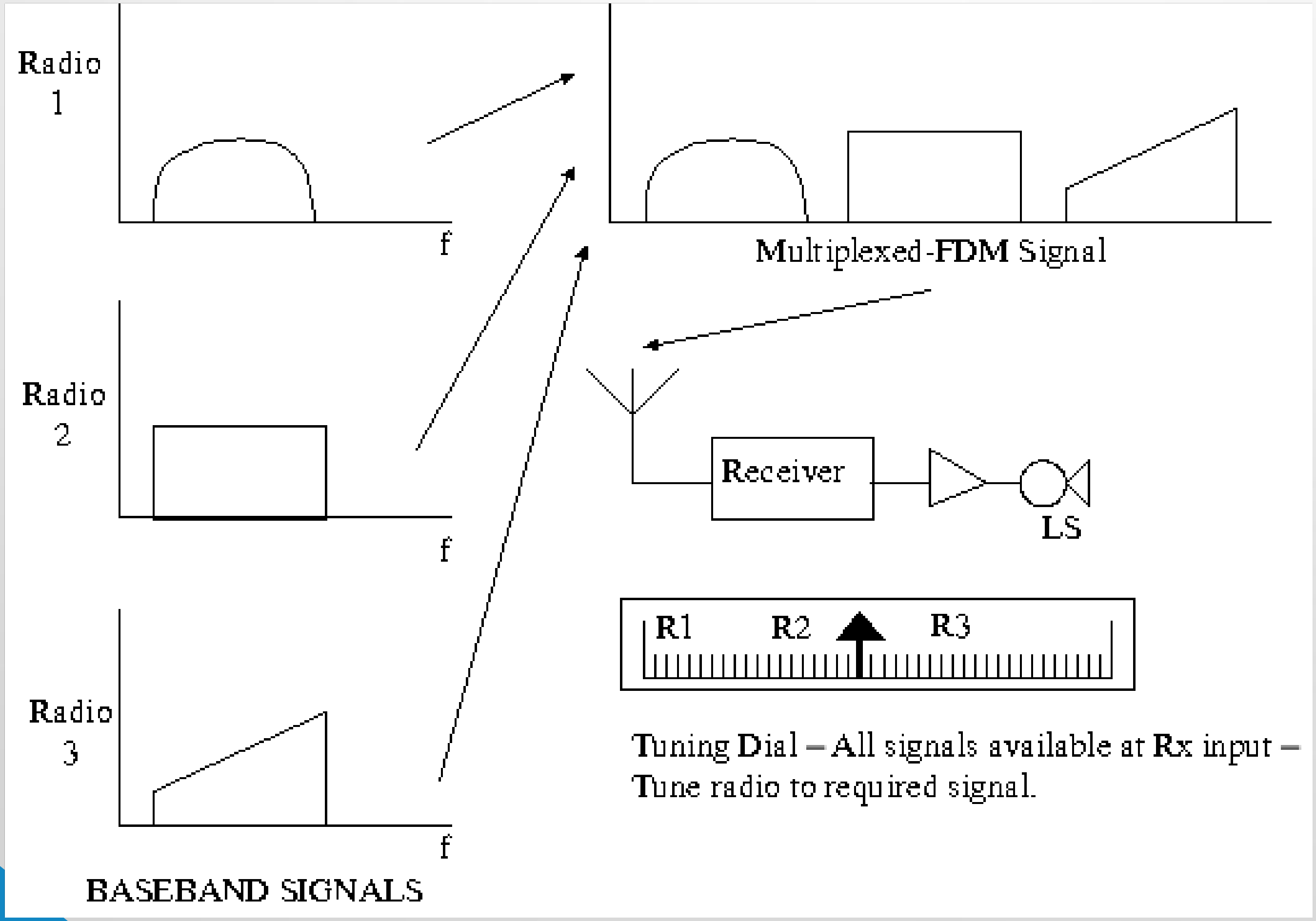


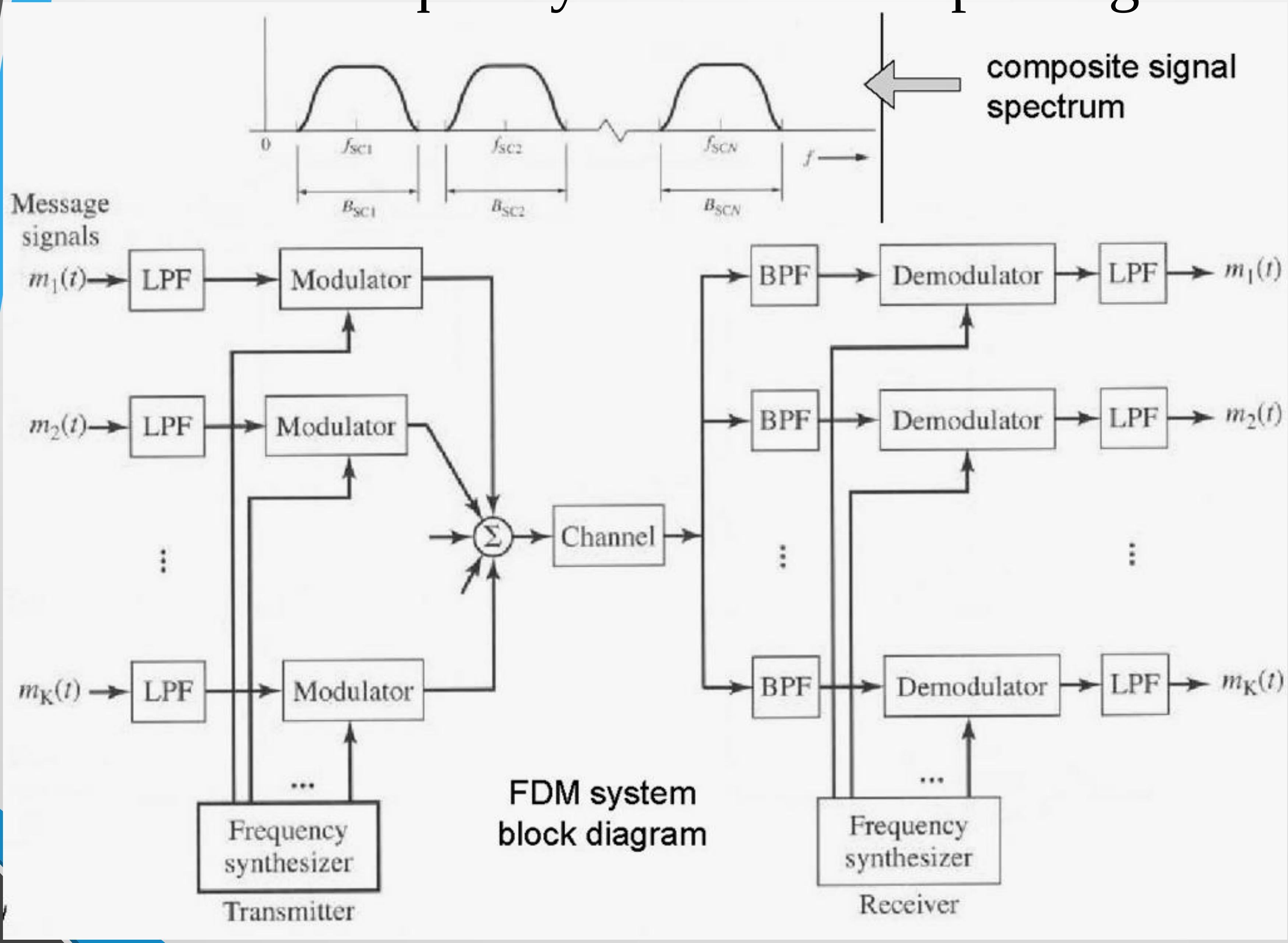
Figure 11-8. TDM solution for minimum transmission line pulse rate.

Frequency Division Multiplexing FDM

This allows several 'messages' to be translated from baseband, where they are all in the same frequency band, to adjacent but non overlapping parts of the spectrum. An example of FDM is broadcast radio (long wave LW, medium wave MW, *etc.*)



Frequency Divison Multiplexing



Pulse Width and Pulse Position Modulation

- In pulse width modulation (PWM), the width of each pulse is made directly proportional to the amplitude of the information signal.
- In pulse position modulation, constant-width pulses are used, and the position or time of occurrence of each pulse from some reference time is made directly proportional to the amplitude of the information signal.
- PWM and PPM are compared and contrasted to PAM in Figure 11-11.

Pulse Width and Pulse Position Modulation

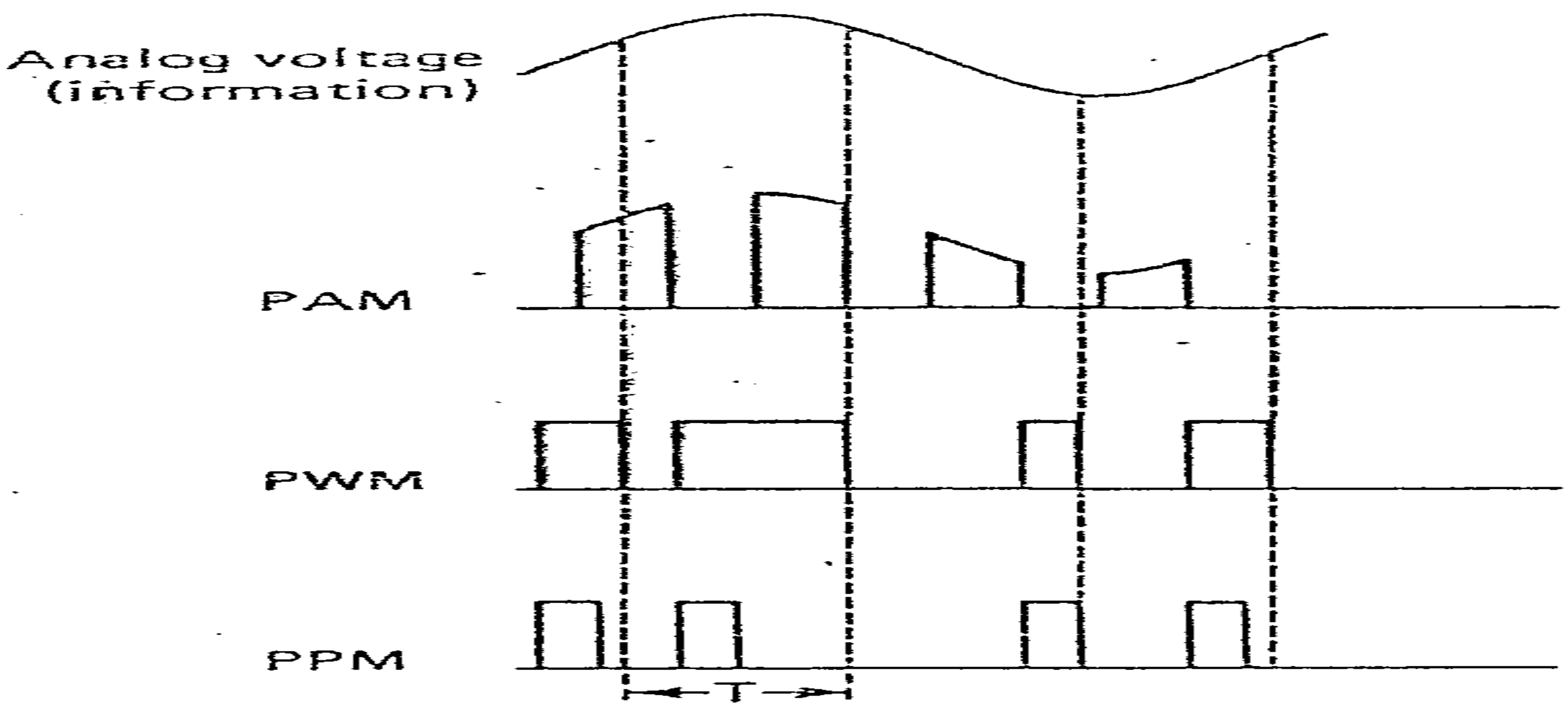
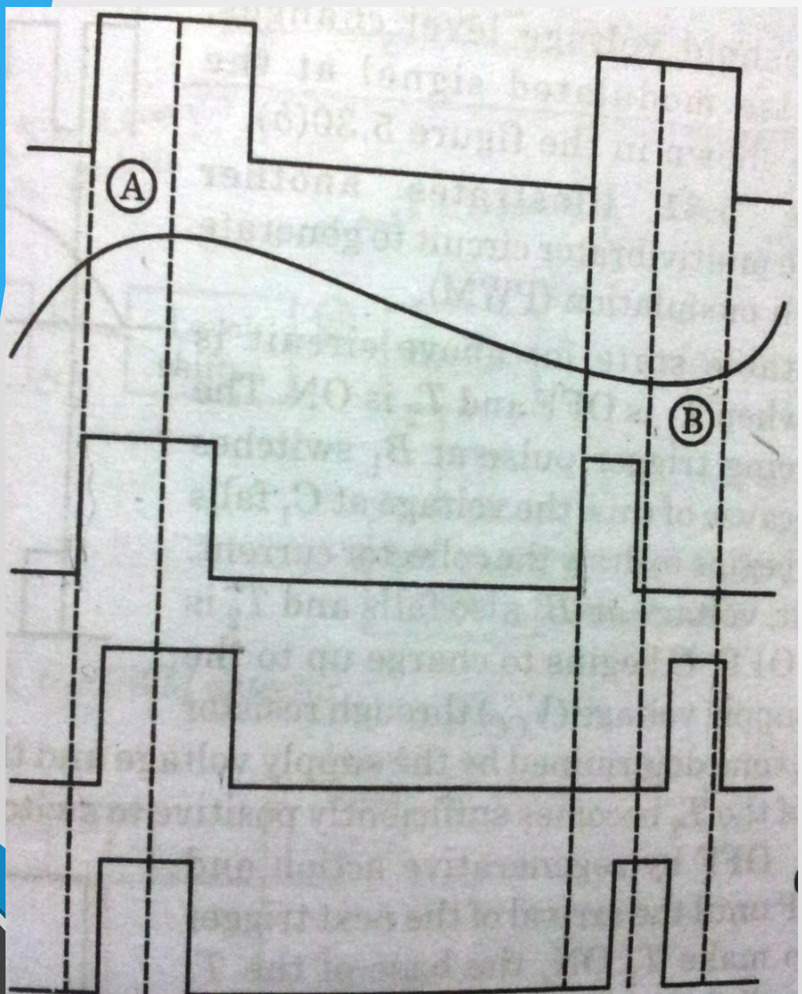


Figure 11-11. Analog/pulse modulation signals.

PWM wave forms



- Three variation of pulses-
- Leading edge of the pulse held constant & change in pulse width with signal measures w.r.t leading edge.
- Tail edge is held constant and w.r.t to it, PWM varied and measured.
- Centre of the pulse is held constant & pulse width changes on either side of the pulse

Pulse Width Modulation

- Figure 11-12 shows a PWM modulator. This circuit is simply a high-gain comparator that is switched on and off by the sawtooth waveform derived from a very stable-frequency oscillator.
- Notice that the output will go to $+V_{cc}$ the instant the analog signal exceeds the sawtooth voltage.
- The output will go to $-V_{cc}$ the instant the analog signal is less than the sawtooth voltage. With this circuit the average value of both inputs should be nearly the same.
- This is easily achieved with equal value resistors to ground. Also the $+V$ and $-V$ values should not exceed V_{cc}.

Pulse Width Modulator

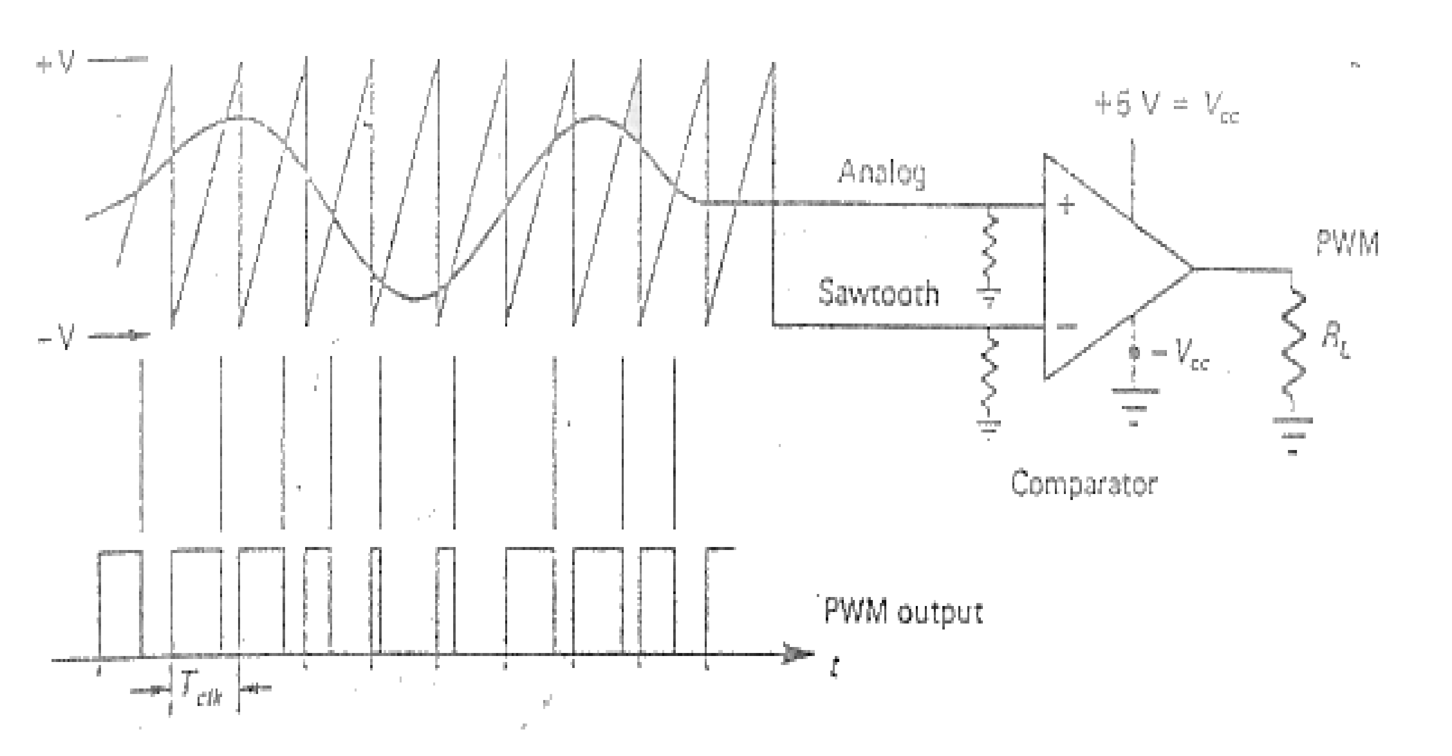
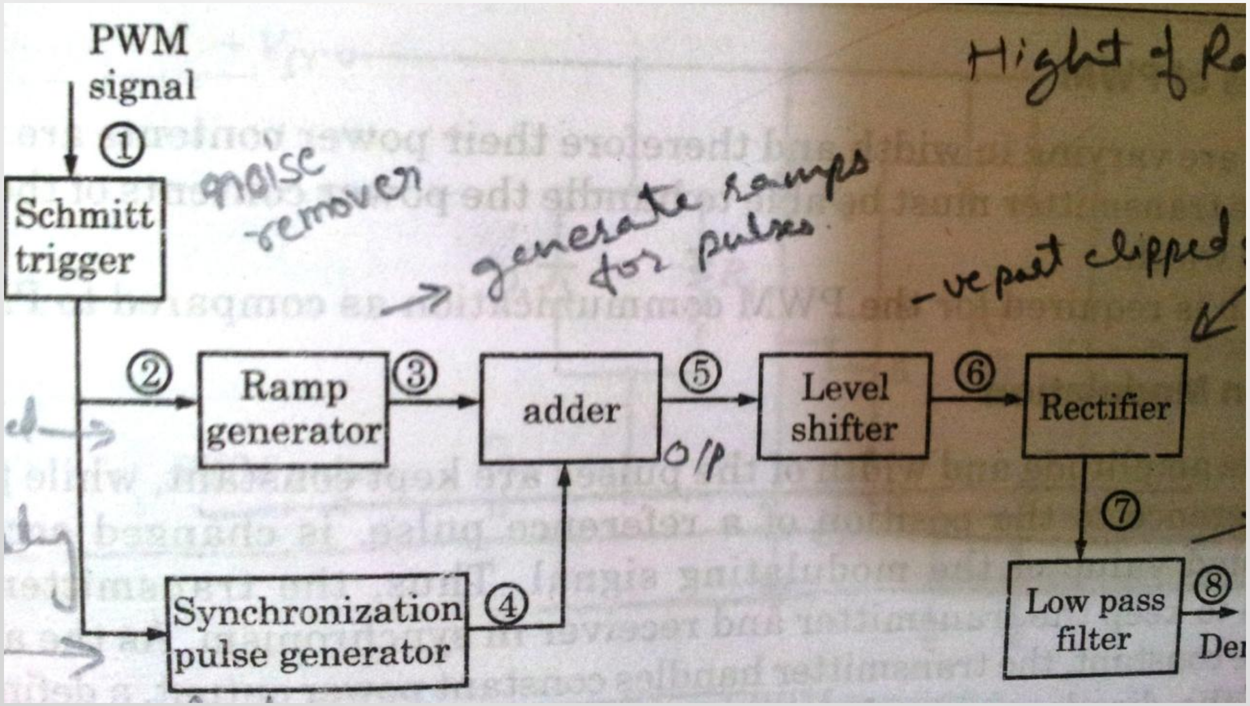


Figure 11-12. Pulse width modulator.

Pulse Width Modulation

- A 710-type IC comparator can be used for positive-only output pulses that are also TTL compatible. PWM can also be produced by modulation of various voltage-controllable multivibrators.
- One example is the popular 555 timer IC. Other (pulse output) VCOs, like the 566 and that of the 565 phase-locked loop IC, will produce PWM.
- This points out the similarity of PWM to continuous analog FM. Indeed, PWM has the advantages of FM---constant amplitude and good noise immunity---and also its disadvantage---large bandwidth.

Demodulation of PWM signal



Demodulation

- Since the width of each pulse in the PWM signal shown in Figure 11-13 is directly proportional to the amplitude of the modulating voltage.
- The signal can be differentiated as shown in Figure 11-13 (to PPM in part a), then the positive pulses are used to start a ramp, and the negative clock pulses stop and reset the ramp.
- This produces frequency-to-amplitude conversion (or equivalently, pulse width-to-amplitude conversion).
- The variable-amplitude ramp pulses are then time-averaged (integrated) to recover the analog signal.

Pulse Position Modulation

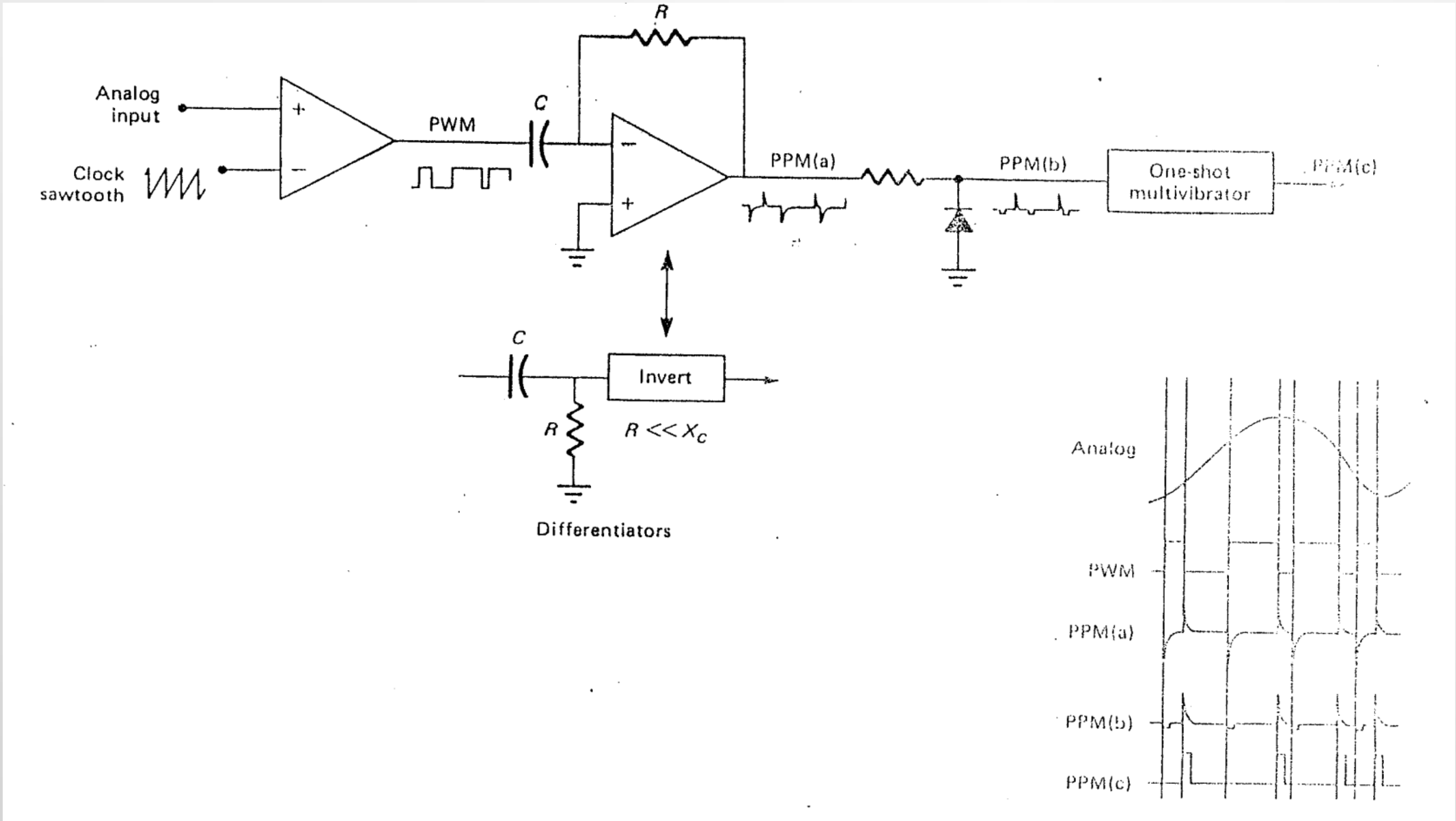
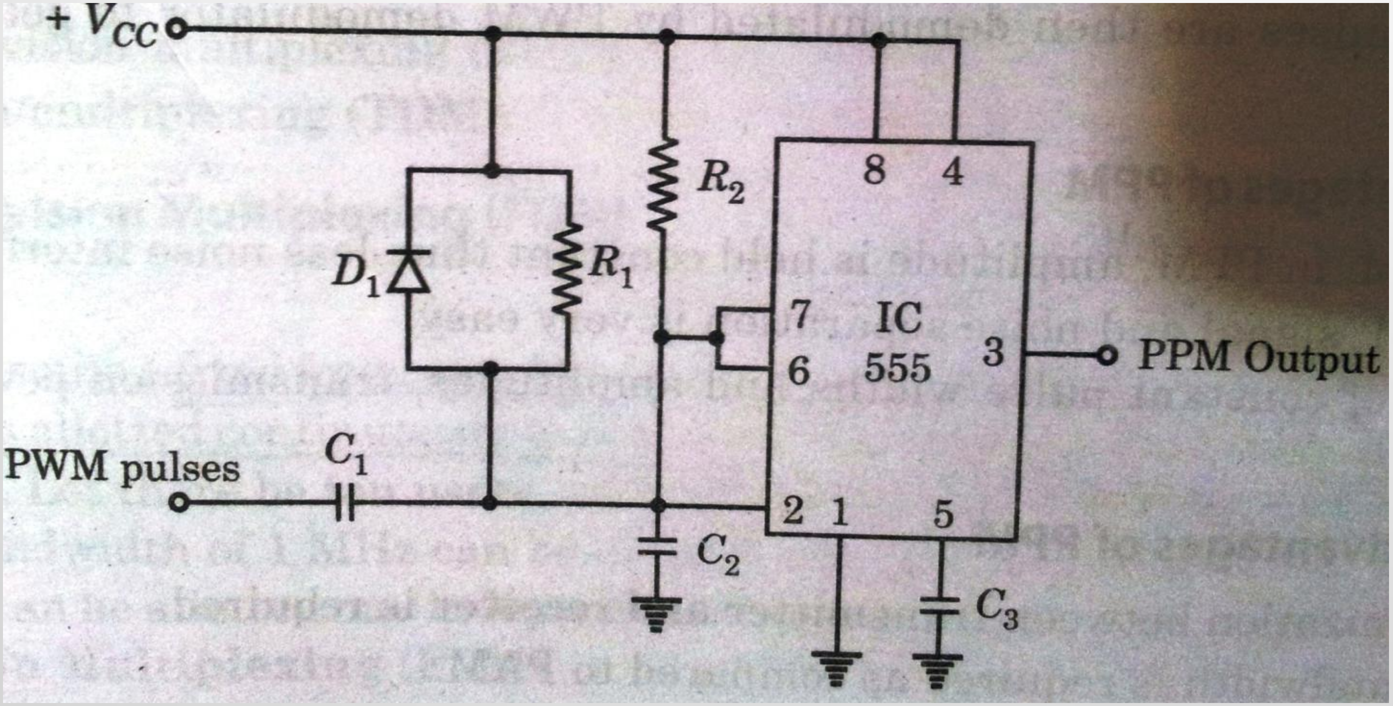


Figure 11-13. Pulse position modulator.

Ckt. diagram of PPM generator



Working Of PPM Generator -

- Contain differentiator & mono-stable vibrator
- In to Diff. is PWM waveform & it generate + & - ve spikes corresponding to leading & trailing edge.
- D1 bypasses the + spikes.
- -ve spike is used to trigger mono-stable vibrator
- Then it generate the pulses of same width and amplitude with reference to trigger to give pulse position modulated waveform.

Demodulation

- As illustrated in Figure 11-14, a narrow clock pulse sets an RS flip-flop output high, and the next PPM pulses resets the output to zero.
- The resulting signal, PWM, has an average voltage proportional to the time difference between the PPM pulses and the reference clock pulses.
- Time-averaging (integration) of the output produces the analog variations.
- PPM has the same disadvantage as continuous analog phase modulation: a coherent clock reference signal is necessary for demodulation.
- The reference pulses can be transmitted along with the PPM signal.

Demodulation

- This is achieved by full-wave rectifying the PPM pulses of Figure 11-13a, which has the effect of reversing the polarity of the negative (clock-rate) pulses.
- Then an edge-triggered flipflop (J-K or D-type) can be used to accomplish the same function as the RS flip-flop of Figure 11-14, using the clock input.
- The penalty is: more pulses/second will require greater bandwidth, and the pulse width limit the pulse deviations for a given pulse period.

Demodulation

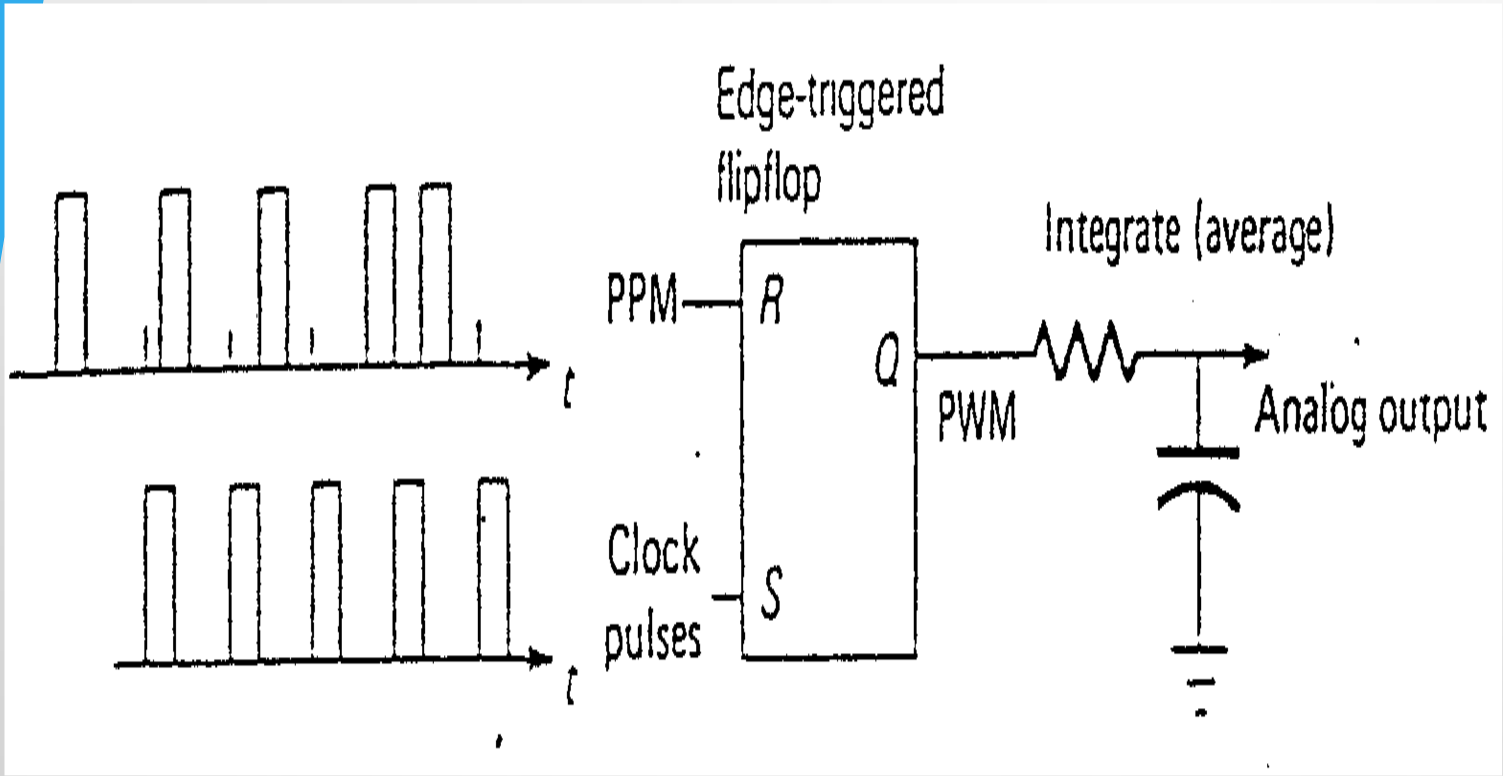
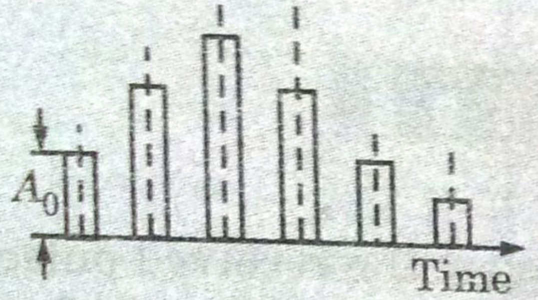
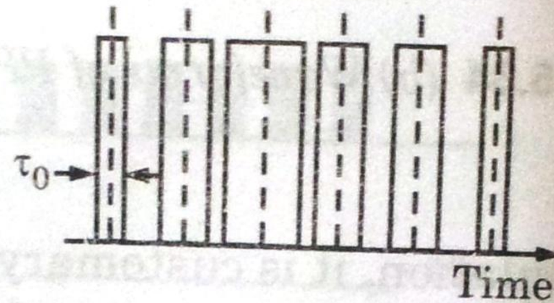
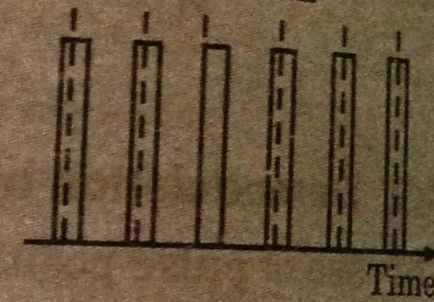


Figure 11-14. PPM demodulator.

Pulse Amplitude Modulation (PAM)	Pulse Width Modulation (PWM) or (PDM)	Pulse Position Modulation (PPM)
Waveform 	Waveform 	Waveform 
Amplitude of the pulse is proportional to amplitude of modulating signal.	Width of the pulse is proportional to amplitude of modulating signal.	The relative position of the pulse is proportional to the amplitude of modulating signal.
The bandwidth of the transmission channel depends on width of the pulse.	Bandwidth of transmission channel depends on rise time of the pulse.	Bandwidth of transmission channel depends on rising time of the pulse.
The instantaneous power of the transmitter varies.	The instantaneous power of the transmitter varies.	The instantaneous power of the transmitter remains constant.
Noise interference is high. System is complex	Noise, interference is minimum.	Noise, interference is minimum
Similar to amplitude modulation.	Simple to implement similar to frequency modulation.	Simple to implement similar to phase modulation.

Pulse Code Modulation (PCM)

- Pulse code modulation (PCM) is produced by analog-to-digital conversion process.
- As in the case of other pulse modulation techniques, the rate at which samples are taken and encoded must conform to the Nyquist sampling rate.
- The sampling rate must be greater than, or equal to, twice the highest frequency in the analog signal,

$$f_s \geq 2f_A(\max)$$

Pulse Code Modulation (PCM)

- A simple example to illustrate the pulse code modulation of an analog signal is shown in Figure 11-15.

- Here, an analog input sample becomes three binary digits (bits) in a sequence which represents the amplitude of the analog sample.

- At time t = 1, the analog signal is 3 V. This voltage is applied to the encoder for a time long enough that the three-bit digital "word", 011, is produced.

Pulse Code Modulation (PCM)

- The second sample at $t = 2$ has an amplitude of 6 V, which is encoded as 110.
- This particular example system is conveniently set up so that the analog value (decimal) is encoded with its binary equivalent.

Pulse Code Modulation (PCM)

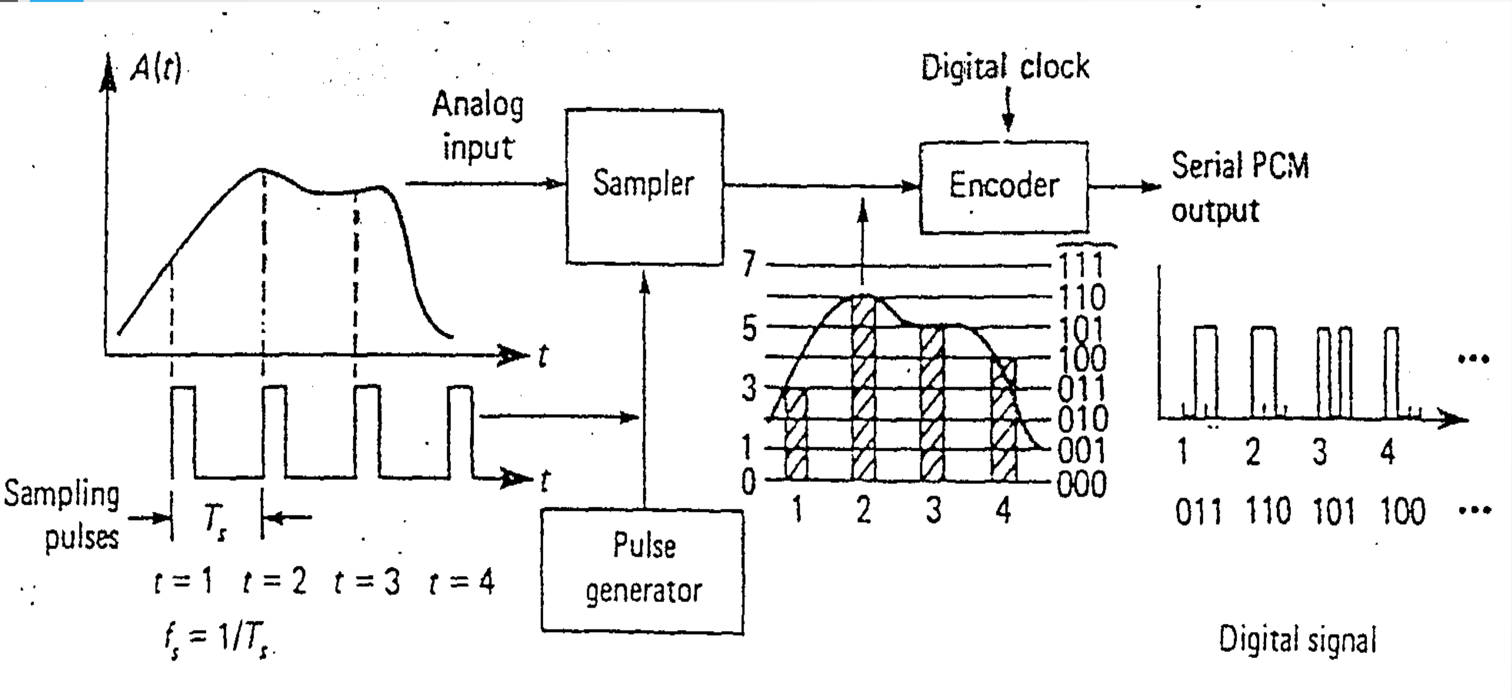


Figure 11-15. A 3-bit PCM system showing A/D conversion.