Rizvi

Ch-4

Digital Modulation

*) **Advantages of Digital transmission** (2013

(I) Noise immunity

(II) Better suited to processing & multiplexing

(III) More noise resistant than their analog counterparts

(IV) Simpler to measure and evaluate

(V) Better suited to evaluate error performance.

*) **Disadvantages of Digital transmission**

(I) Requires more bandwidth than simply transmitting the original analog signal.

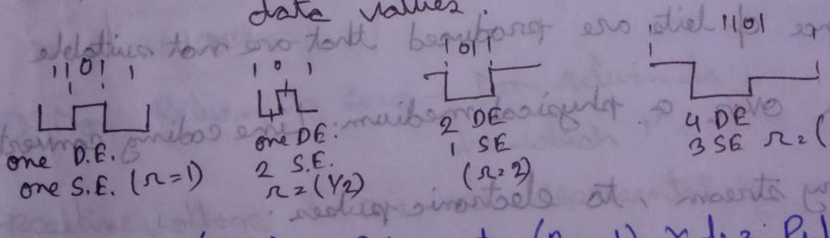
(II) Necessity of additional encoding and decoding circuitry.

(III) Requires precise time synchronization between transmit and receiver clock.

(IV) Incompatible with older analog transmission facilities.

*) **Signal level** – number of different voltage levels allowed in a signal.

Data level – " " voltage levels that acutually represent data values.



*) Bit rate (BPS) = Pulse rate (Baud) $\times$ b = Pulse rate $\times \log_2 L$

b – number of bits per pulse; L – number of different signal elements.

* Encoding considerations

(i) Signal spectrum

(a) Lack of DC component – DC component in signal are not desirable, because

– Can't pass through certain devices

– Leave extra energy on the line.

(ii) Clocking / Synchronization

(iii) Error detection

(iv) Noise immunity (v) Cost & Complexity

* Line code

The output of an ADC can be transmitted over baseband channel. The digital info. must first be converted into a physical signal. This signal is called line code.

* Why we need line coding

Line coding is the process of converting digital data to digital signal.

From Shannon capacity theorem, we can determine a channel capacity. But it doesn't give any info. about how to do transmission. Line coding is used for transmission purpose. In PCM, a sequence of bits are produced that are not suitable for transmission over a physical medium. Line coding converts the bits or binary stream, to electronic pulses.

That's why line coding is necessary.

* Some desirable properties of Line code (2014, 10,

① Self synchronization — There is enough timing info. built into the code so that bit synchronizers can be designed to extract the clock signal.

(II) Low Probability of errors — Receivers can be designed that will recover binary data with low probability of bit error.

(III) Spectrum is suitable for channel.

(IV) Transmission BW — should be as small as possible.

(V) Error detection capabilities — either by adding channel encoder and decoder or by incorporating into the line code.

(VI) Transparency of data protocol and line code to receive every possible sequence of data.

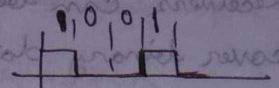
* Define Return-to-zero, non-return to zero, manchester code. (2011,

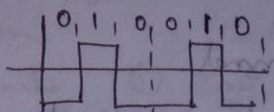
① Return to zero — Return to zero describes a line code in which the signal drops between each pulse. This takes place even if a number of consecutive 0's or 1's occur.

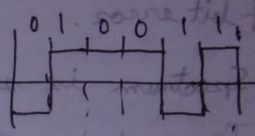
(II) Non-Return to Zero — Non return to zero is a binary code in which 1's are represented by positive voltage & 0's are represented by negative/zero voltage. There is no neutral condition. These pulse have more energy than RZ.

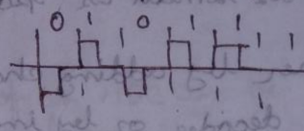
(iii) Manchester Code — Here 0 is expressed by low to high transition and 1 is represented by high to low ".
The significant transition of 0 or 1 occurs in the mid point of a period.

** For drawing

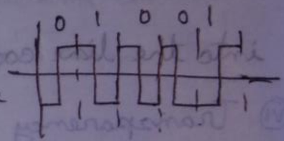
Unipolar — 

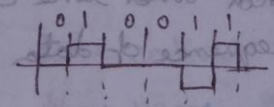
NRZ-L — 

NRZ-I 

RZ — 

Manchester — 

Differential manchester (invert). 

Bipolar AMI — 

** Comparison between NRZ & Manchester Code

	Bandwidth	Timing	DC value
Unipolar NRZ	Low	No info.	High DC component
Bipolar NRZ	Lower	"	No DC component
Differential NRZ	Lower	"	Little / No DC
Manchester	High	Good Clock recovery	No DC
Diff. Manchester	Moderate	"	No DC

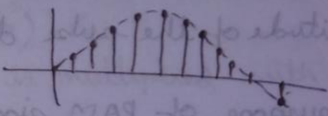
Describe PCM (2014, 13, 12, 11, 10)

PCM or Pulse code modulation is essentially analog to digital conversion of a signal where the info. contained is represented by digital word in a serial bit stream.

To do it, following steps are taken

① Sampling

In sampling analog signal is sampled every T_s, where T_s is the sample interval. Sampling can be done through ideal, natural or flat top sampling. The most common sampling method is simple and hold.



Ideal sampling

⑪ Quantization

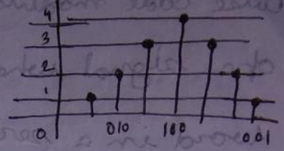
It is the second step. The result of sampling is a series of pulse with amplitude values between the maximum & minimum amplitude value. These values are quantized with following steps —

ⓐ We assume that the original signal has instantaneous amplitude between V_{max} & V_{min}.

ⓑ We divide range into L zones with each of height Δ.
where $\Delta = \frac{V_{max} - V_{min}}{L}$.

ⓒ We assign quantized value 0 to L-1.

ⓓ We approximate value of sample amplitude to the quantized values.



ⓘⓘⓘ Encoding

In encoding after each sample is quantized each sample can be converted to code word.

*) Define PAM? Why do we need PAM (2009, 07)

Pulse Amplitude modulation (PAM) is used to describe the conversion of the analog signal to a pulse type signal where the amplitude of the pulse denotes the analog info.

The purpose of PAM signalling is to provide another waveform that look like pulses, yet contains the information that was present in the analog information.

*) Describe two sampling ① Natural, ⓘⓘ Flat-top (2013, 09, 07)

① Natural Sampling (Grating)

It takes a slice of the waveform and top of slice preserves the shape of the waveform. If $W(t)$ is an analog waveform bandlimited to B hertz, the PAM signal that uses natural sampling is, $W_s(t) = W(t)\, s(t)$

where $s(t) = \sum_{k=-\infty}^{\infty} \Pi\left(\frac{t-kT_s}{\tau}\right)$

which is a rectangular wave switching waveform & $f_s = \frac{1}{T_s} \geq 2B$.

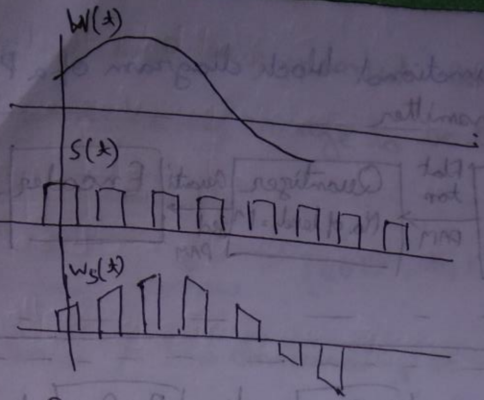


Fig: PAM signal with natural sampling.

The PAM waveform with naturally sampling is relatively easy to generate since it only requires the use of an analog switch that is readily available in CMOS hardware.

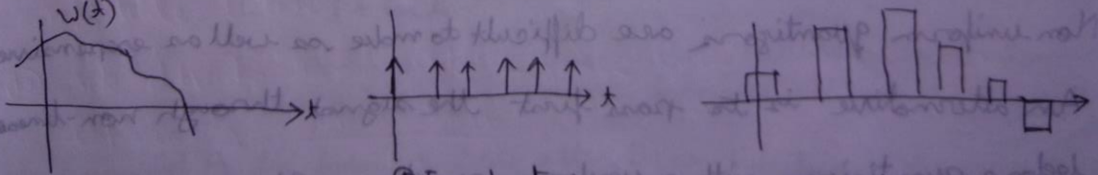
(ii) Instantaneous Sampling (Flat-top)

In flat top sampling, a signal is multiplied with impulse train. It uses sample and hold operation. We get flat top samples that do not preserve the shape of waveform.

If $w(t)$ is an analog waveform bandlimited to B hertz the flat top PAM signal is

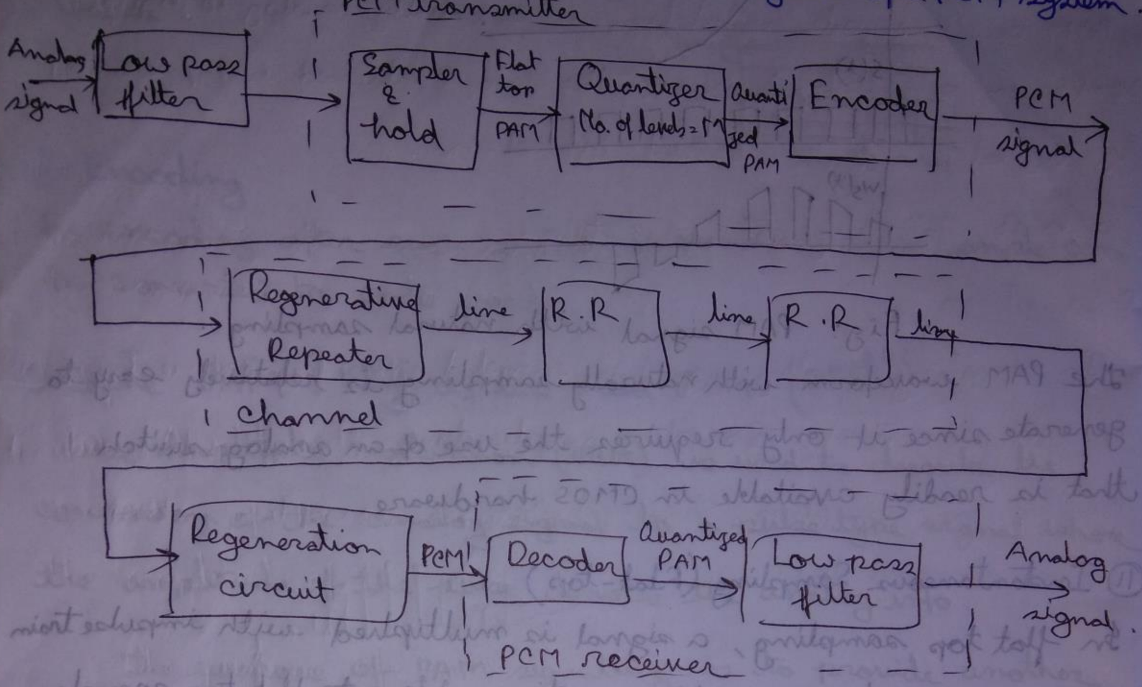
$$W_s(t) = \sum_{k=-\infty}^{\infty} w(kT_s)\, h(t - kT_s)$$

where $h(t)$ denotes the sampling-pulse shape



(a) Baseband (b) Impulse train (c) Resulting PAM signal.

*) Draw and describe functional block diagram of a PCM system.



(Describe — picture describe

*) Why companding is necessary (2014, 12

Companding is compression of a signal at the transmitter using a non linear function. Compressed signal is quantized. At receiver an expansion function is applied.

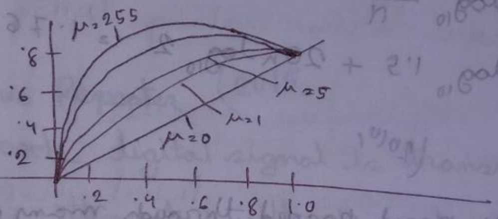
Non uniform quantizers are difficult to make as well as expensive. An alternative is to pass first the signal through non-linearity before quantizing with a uniform quantizer. It causes the signal amplitude to be compressed.

At receiver it will be expanded. Companding allows signals with a large dynamic range to be transmitted through smaller dynamic range capability.

*) Explain A-law, μ-law. (2015, 14, 13, 12

In USA & Japan μ-law companding is used. It can be defined that

$$V_{out} = \frac{V_{max} \cdot \ln\left(1+\mu\, V_{in}/V_{max}\right)}{\ln(1+\mu)} \quad \text{where } 0 \leq \mu \leq 255$$



A-law companding is used to approximate true logarithmic companding. In Europe, this type of companding used for transmission. A-law companding can be defined-

$$V_{out} = V_{max} \frac{A\, V_{in}/V_{max}}{1+\ln A} \quad \text{wher } 0 \leq \frac{V_{in}}{V_{max}} \leq \frac{1}{A}$$

$$= \frac{1+\ln\left(A\, V_{in}/V_{max}\right)}{1+\ln A}$$

*) What is 6 DB rule? Prove SQR = 1.76 + 6.02 n where SQR means signal to quantization ratio and n is the number of bits used. (2012, 09, 07)

Peak signal to noise ratio, $\left(\frac{S}{N}\right) = 10 \log_{10} \frac{S}{N} = 10 \log 3 + 20 n \log_{10} 2$

$= 4.77 + 6.02\, n$

It is called 6 DB rule.

Now, Average SQNR is

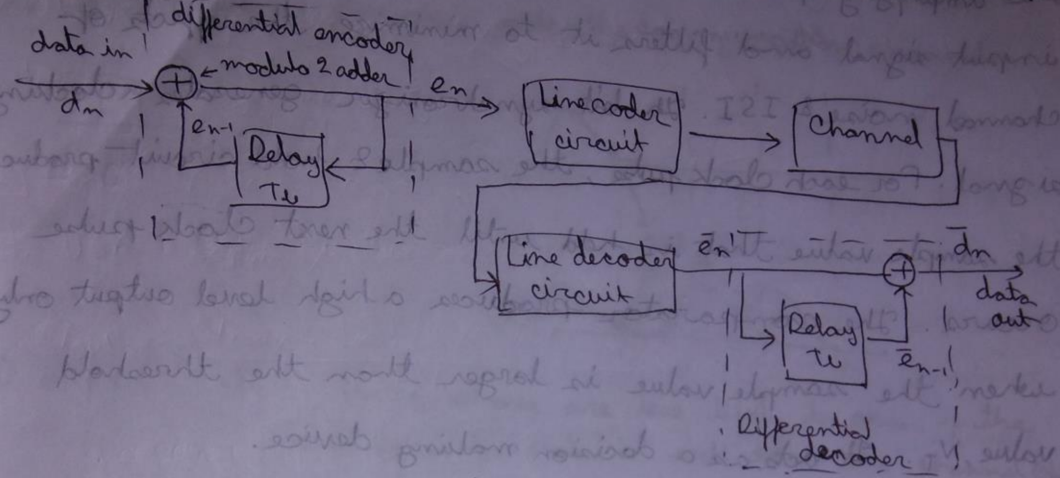
$\left(\frac{S}{N}\right) = 10 \log_{10} \frac{S}{N}$

$= 10 \log_{10} 1.5 + 20 n \log_{10} 2 = 1.76 + 6.02\, n$.

*) Differential coding (2010)

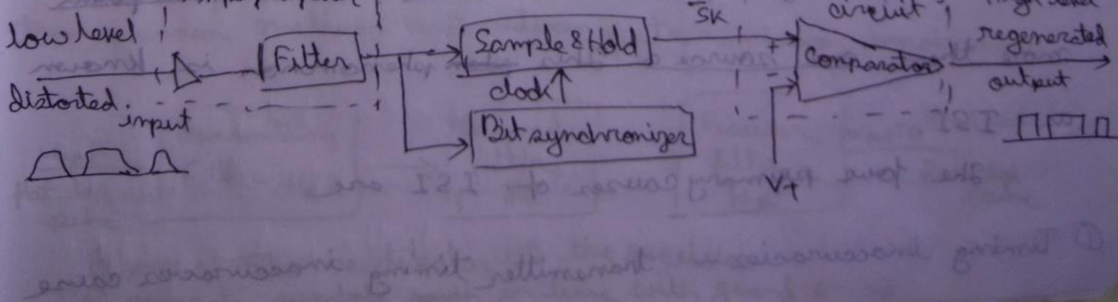
When serial data are ~~pased~~ passed through many circuits along a communication channel, the waveform is often unintentionally inverted. To ameliorate this problem, differential coding is often employed. The encoded differential data are generated by $e_n = d_n \oplus e_{n-1}$ where $\oplus$ is a modulo 2 adder/ exclusive OR operation. The received data are decoded by $d_n = \bar{e}_n \oplus \bar{e}_{n-1}$

Each digit in the encoded sequence is obtained by comparing the present input bit with the past encoded bit. A binary 1 is encoded if the present input bit and the past encoded bit are opposite state and binary 0, if ~~[illegible]~~ state is same.



* Regenerative Repeater (2015,

When a line code digital signal is transmitted over a hardwire channel, it is attenuated, filtered and corrupted by noise. For long lines data can't be recovered at the receiving end, unless repeaters are placed in cascade along the line to amplify and 'clean-up' the signal periodically. Nonlinear processing can be used to regenerate a 'noise-free' digital signal, this type of processing is called regenerative repeater



The amplifying filter increases the amplitude of the low level input signal and filters it to minimize the effects of channel noise & ISI. The bit synchronizer generates a clocking signal. For each clock pulse, the sample & hold circuit produces the sample value that is held until the next clock pulse occurs. The comparator produces a high level output only when the sample value is larger than the threshold value, V_T. It acts as a decision making device.

*) Why do we uses regenerative repeater

Because it

- regenerates a noise free digital signal
- Amplifies and cleans the signal periodically.

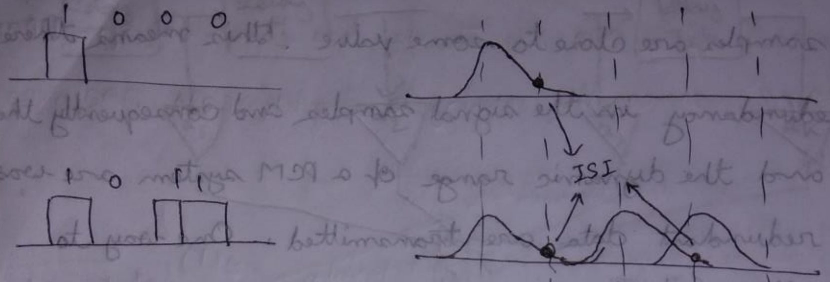
*) Intersymbol Interference (ISI) (2014, 13, 12

Rectangular pulses will not remain rectangular ~~if~~ in less than an infinite BW. The narrower the BW, the more rounded the pulse. If the phase distortion is excessive, the pulse will tilt and consequently, affect the next pulse. ~~When pulses from more than one source a~~ This ~~phen~~ phenomenon is known as ISI.

The four primary causes of ISI are

① Timing inaccuracies - transmitter timing inaccuracies cause

ISI if the rate of transmission does not conform to the ringing frequency designed into the communication channel.

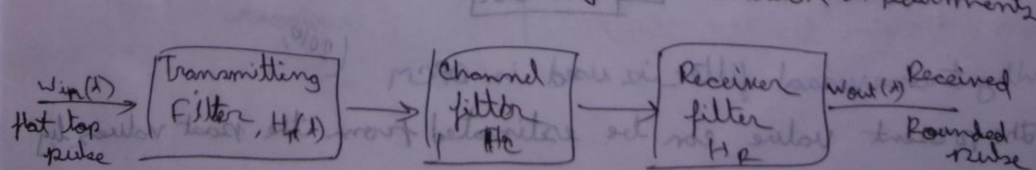


(II) Insufficient BW – Timing errors are less likely to occur if the transmission rate is well below the channel BW.

(III) Amplitude distortion – When the frequency characteristics of a channel depart from normal value, pulse distortion results. Pulse distortion occurs when the peaks of pulses are reduced, causing improper ringing frequencies.

(IV) Phase distortion – Phase distortion occurs when frequency components undergo different amount of time delay while propagating through the transmission medium.

In order to mitigate ISI (I) 'equalizer' filter is used. It equalize the distortion for all frequencies, creating a uniform transmission medium and reducing transmission impairments.



(II) use a sequence detector at the receiver.

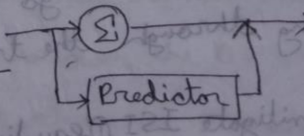
(III) seperate symbols ~~with~~ in time with guard period.

[*] Differential Pulse Code Modulation (DPCM) (2009,

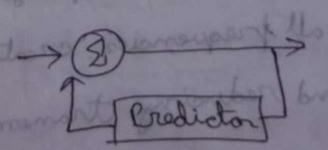
When audio or video signals are sampled it is usually found that adjacent samples are close to some value. This means there is a lot of redundancy in the signal samples and consequently that the BW and the dynamic range of a PCM system are wasted when the redundant data are transmitted. One way to minimize this is to transmit PCM signals corresponding to the difference in adjacent sample values. This is DPCM. At the receiver the present sample value is regenerated by using the past value plus the update differential value.

(Block diagram — same as PCM system,
difference all PAM → DPCM হবে
Sampler & hold বদলে Regeneration circuit এর পরে new ভাবে add হবে

Sampler এর পরে —

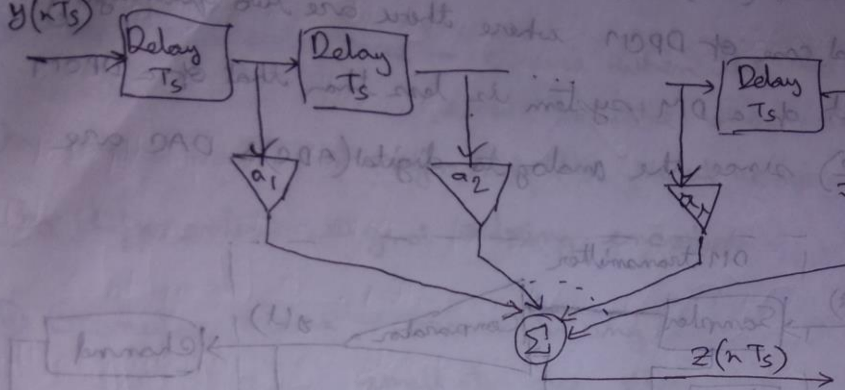


Regeneration —



[*] Why transversal filter is used in DPCM (2010,

The present value can be estimated from the past values by using a prediction filter. The filter may be realized by using tapped delay line to form a transversal filter.

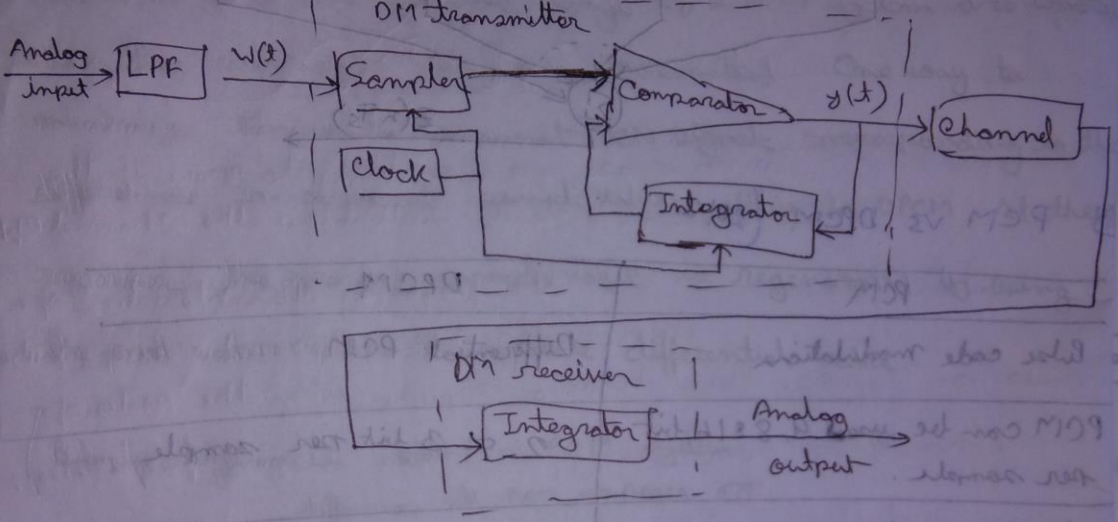


PCM vs DPCM (2012

PCM	DPCM
Pulse code modulation	Differential PCM
PCM can be used 4, 8 & 16 bit per sample.	2 or 3 bit per sample
It has required highest BW.	required small BW.
Quantization error depends on number of levels.	Slope over load distortion & quantization error is present.
number of level depends on number of bit.	has fixed number of levels.
No feedback in transmitter / receiver	The feedback is present.

Delta Modulation (DM) (2014, 13

DM is a special case of DPCM where there are two quantizing levels. The cost of a DM system is less than that of a DPCM system (M > 2) since the analog to digital (ADC) & DAC are not needed.



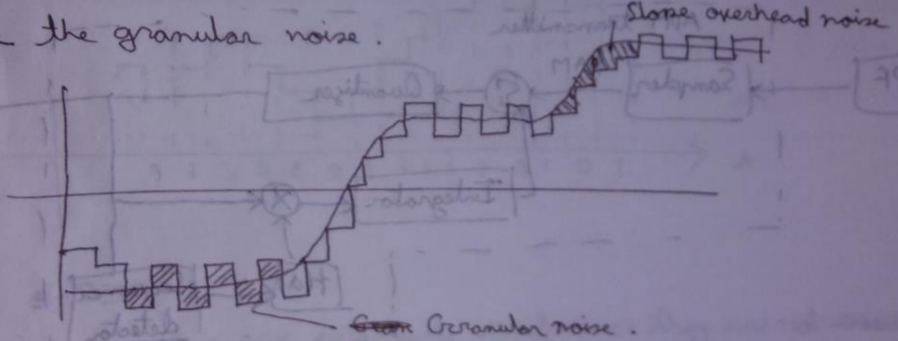
In the DM circuit, the operations of subtractor & two level quantizer are implemented by a comparator so that the output is $\pm V_c$. In this DM signal is a polar signal.

At the receiver the DM signal may be converted back to an analog signal approximation. This is accomplished by using an integrator for the receiver.

⁂ Explain two different errors in delta modulation. (2014, 12, 07, 09

(i) Slope overhead noise — occurs when the step size δ is too small for the accumulator output to follow quick changes in the input waveform. It occurs due to differential signal being encoded.

(ii) Granular noise — occurs for any step size but is smaller for a small step size. Thus the smaller the δ, the lower the granular noise.



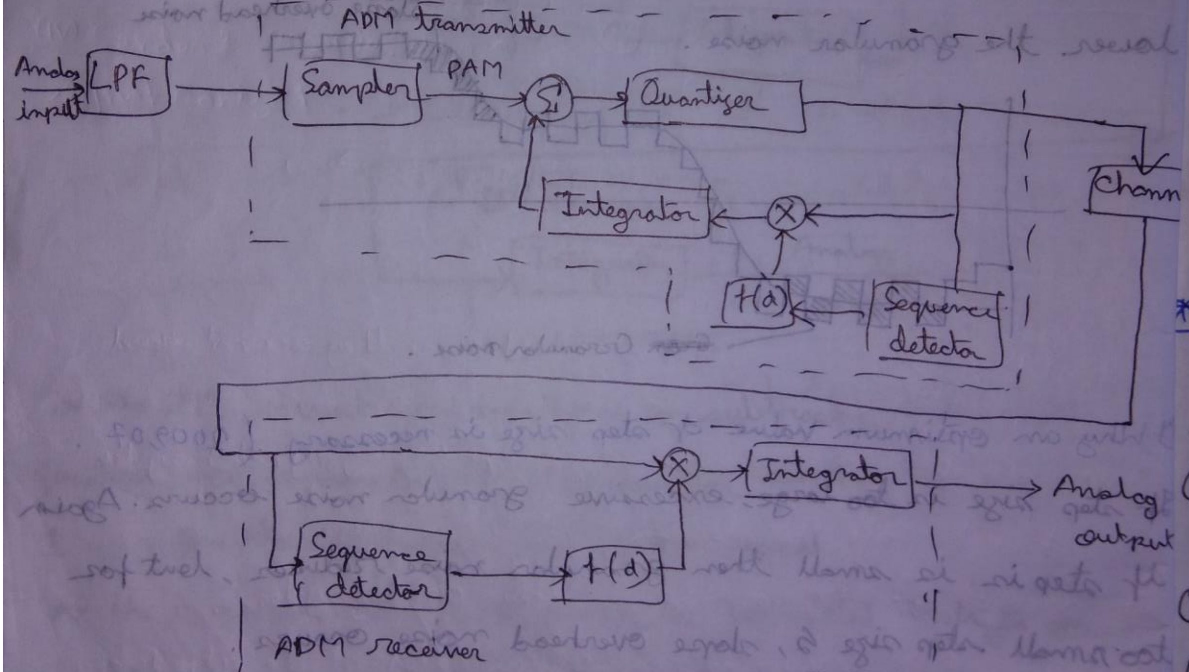
⁂ Why an optimum value of step size is necessary. (2009, 07.

If step size is too large, excessive granular noise occurs. Again if step is is small then granular noise reduces, but for too small step size δ, slope overhead noise occurs.

That's why, to reduce both the noises an optimum step size is choosen.

Adaptive Delta Modulation (ADM) (2014, 12, 10

To minimize the slope overhead noise while holding granular noise at a reasonable value, ADM is used. Here the step size is varied as a function of time as the input changes. The step size is kept small to minimize the granular noise until the slope overhead Noise begins to dominate. Then the step is increased to reduce S.O.N.



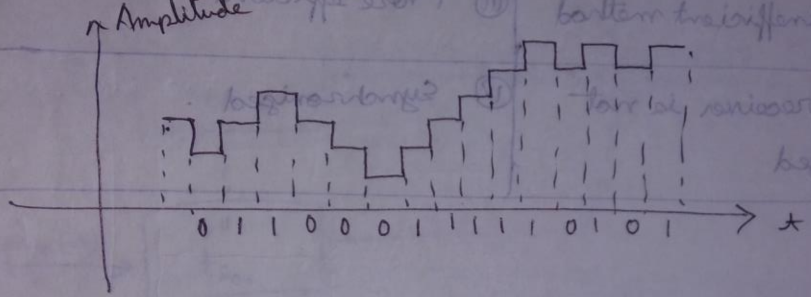
Advantages of ADM over conventional DM (2014, 12

① In ADM step size is variable and ~~const~~ step size varies. But in DM step size is fixed and cannot be varied.

(ii) In ADM quantization error is present but others are absent. While in DM slope overhead & granular noise is present.

(iii) In DM, SNR is poor while in ADM, SNR is better than DM.

* Sketch 011 0001111 0101 which is transmitted by a delta modulator and analog waveform (2011, 10, 09, 07



* Adv. of PCM

(i) Digital circuitry are extensively used since they are not ~~used~~ expensive

(ii) PCM signals derived from any type of analog signal may be merged with data signals.

(iii) Noise of of PCM is superior than analog system.

(iv) A clean PCM waveform can be regenerated at the output of each repeater from a ~~noi~~ noisy PCM input.

* Disadv. of PCM

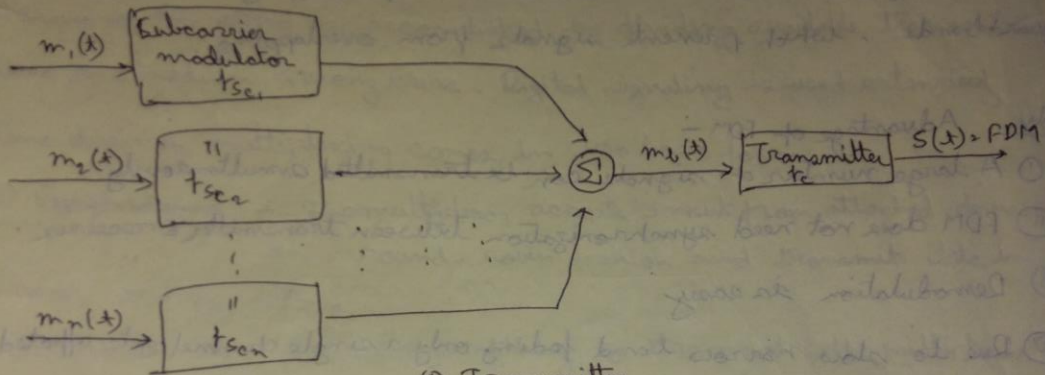
Main disadvantage is a much wider BW than that of the corresponding analog signal.

* Synchronous VS Asynchronous Transmission. (2012.

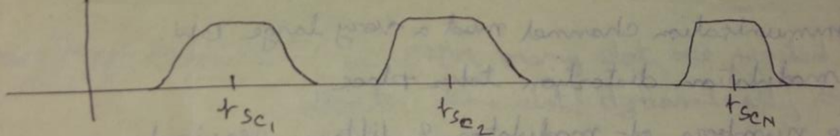
A.Synch.	Synch.
① Each character needs a start & stop bit.	① No need
⑪ There can be idle time	⑪ No idle time while sending data
⑪ Slow & inefficient method	⑪ More efficient
⑫ Sender & receiver is not synchronized	⑫ Synchronized.

* Frequency Division Multiplexing (FDM)

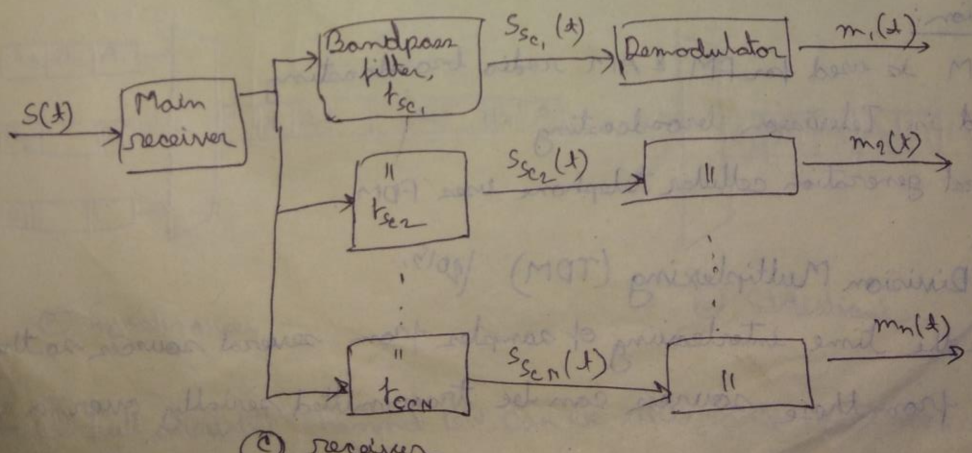
FDM is a networking technique in which multiple data signals of different frequency are combined for simultaneous transmission via a shared communication medium. Analog signaling is used to transmit the signal



(a) Transmitter



(b) Spectrum of composite baseband signal.



(c) receiver

The composite signal spectrum must consist of modulated signals that do not have overlapping spectra, otherwise crosstalk will occur between message signals at the receiver output.

Signals are combined into composite signal. Carrier frequencies are separated by sufficient bandwidth. These BW's are the channels through which various signals travel. Channels can be seperated by unused BW - called guardbands, which prevents signals from overlapping.

Adv. Advantage of FDM –

(I) A large number of signals can be transmitted simultaneously.

(II) FDM does not need synchronization between transmitter & receiver

(III) Demodulation is easy

(IV) Due to slow narrow band fading only a single channel gets affected.

Disadv.

(I) The communication channel must a very large BW.

(II) Intermodulation distortion takes place

(III) Large numbers of modulators & filters required

(IV) Suffers from the problem of crosstalk

Application:

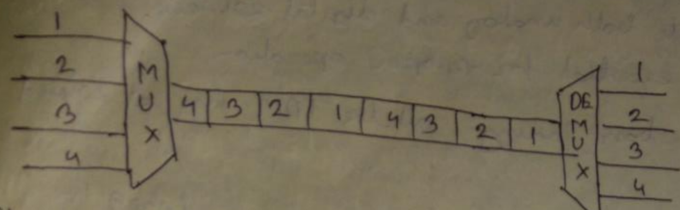
(I) FDM is used for FM & AM radio broadcasting

(II) Used in Television broadcasting

(III) First generation cellular telephone uses FDM.

* Time Division Multiplexing (TDM) | 2015,

TDM is the time interleaving of samples from several sources so that the info. from these sources can be transmitted serially over a single communication channel. TDM assigns time slots to each channel repeatedly.

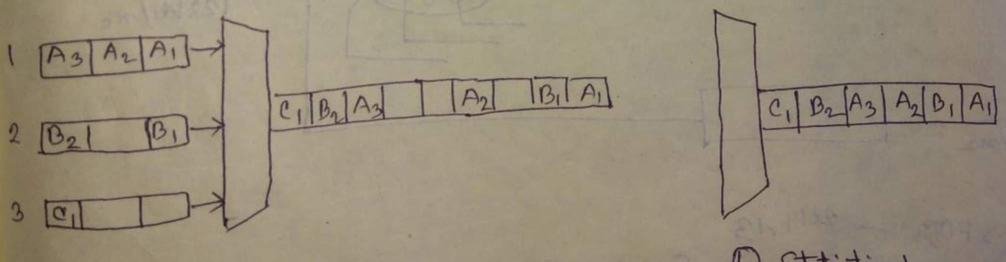


Sharing of the signal is accomplished by dividing available transmission time on a medium among users. Digital signaling is used extensively.
Time division multiplexing comes in two basic forms —

① Synchronous — The multiplexor accepts input from attached devices in a round-robin fashion and transmit data in never ending pattern

Problem — If a device has nothing to transmit, the multiplexor must still insert a piece of data from that device into the stream.

Ⓘ Stastical/Asynchronous — In STDM many slots are wasted, so this allocates time slots dynamically based on demand. A statistical multiplexor transmits only from active workstations



ⓑ Statistical

ⓐ Synchronous

Adv. ① Full available channel BW can be utilized for each channel.
Ⓘ Intermodulation distortion is absent.

(III) TDM circuitry is not very complex (IV) Problem of crosstalk is not severe
(V) TDM can easily accomodate both analog and digital sources.

Disadv. (I) Synchronization is essential for proper operation.

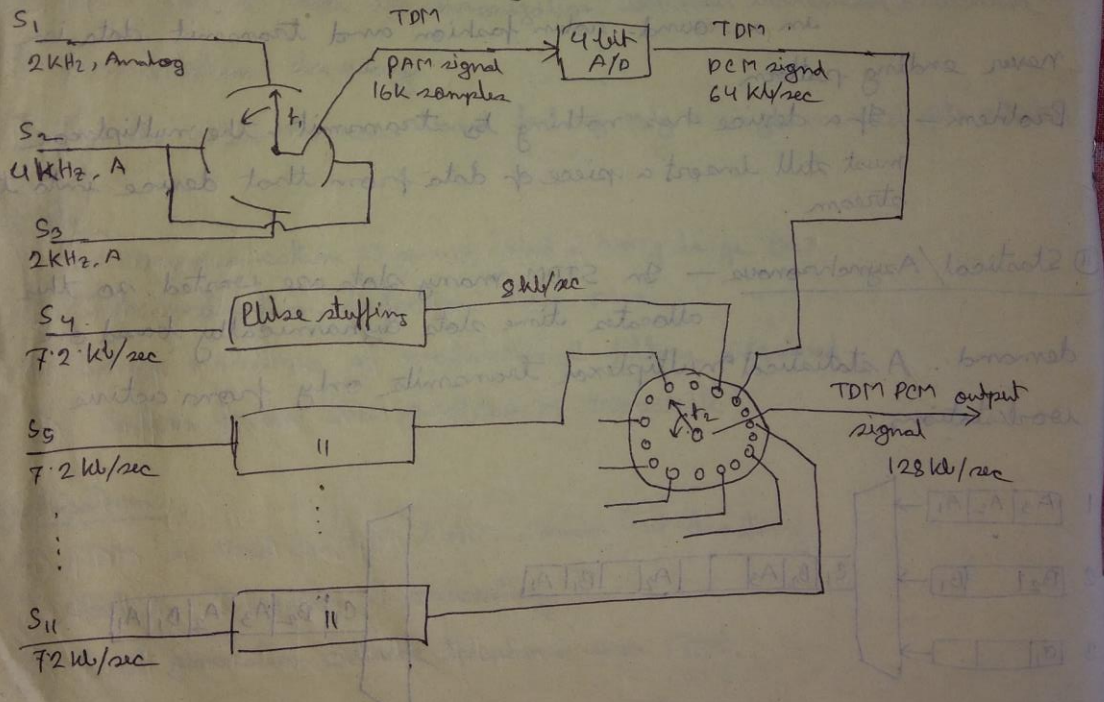
(II) Due to slow narrowband fading, all the TDM may get wiped out.

* Design a TDM multiplexer that will accomodate 11 sources. (2009,

S1 — Analog, 2KHz , S2 — Analog, 4KHz, S_3 — Analog 2KHz

S_{4-11} Digital, 7200 bits/sec. Analog will converted to 4 bit PCM

To satisfy Nyquist rate, S1, 2, 3 need to be sampled at 4, 8 & 4 KHz.
this can be accomplished by rotating 4KHz twice on each revolution.



(TDM vs FDM — 2014, 13
(Adv. & Disadv. দেখে নিতে হবে).