Along with the sampled pulses, one synchronizing pulse is sent in each cycle. These data pulses along with the control information form a *frame*. Each of these frames contain a cycle of time slots and in each frame, one or more slots are dedicated to each data source. The maximum bandwidth (data rate) of a TDM system should be at least equal to the same data rate of the sources.
Synchronous TDM is called synchronous mainly because each time slot is preassigned to a fixed source. The time slots are transmitted irrespective of whether the sources have any data to send or not. Hence, for the sake of simplicity of implementation, channel capacity is wasted. Although fixed assignment is used TDM, devices can handle sources of different data rates. This is done by assigning fewer slots per cycle to the slower input devices than the faster devices.

Short Answers

1. What is meant by aliasing effect?

Aliasing effect takes place when sampling frequency is less than Nyquist rate.Under such condition, the spectrum of the sampled signal overlaps with itself. Hence higher frequencies take the form of lower frequencies. This interference of the frequency components is called as aliasing effect.

2. State Sampling theorem.

A bandlimited signal of finite energy, which has no frequency components higher than W Hz, may be completely recovered from the knowledge of its samples taken at the rate of 2W samples per second.

3. Mention the merits of DPCM.

1 .Bandwidth requirement of DPCM is less compared to PCM.

2. .Quantization error is reduced because of prediction filter

3.Numers of bits used to represent one sample value are also reduced compared to PCM.

4. What is the main difference in DPCM and DM?

DM encodes the input sample by one bit. It sends the information about + δ or - δ, ie step rise or fall. DPCM can have more than one bit of encoding the sample. It sends the information about difference between actual sample value and the predicted sample value.

5. How the message can be recovered from PAM?

The message can be recovered from PAM by passing the PAM signal through reconstruction filter integrates amplitude of PAM pulses. Amplitude reconstruction signal is done to remove amplitude discontinuities due to pulses.

6. What are the advantages of the Delta modulation?

1 Delta modulation transmits only one bit for one sample. Thus the signalling rate and transmission channel bandwidth is quite small for delta modulation.

2. The transmitter and receiver implementation is very much simple for delta modulation. There is no analog to digital converter involved in delta modulation.

7. How does Granular noise occurs?

It occurs due to large step size and very small amplitude variation in the input signal

8. What are the two limitations of delta modulation?

1 Slope of overload distortion.

2. Granular noise.

9. What is the advantage of delta modulation over PCM?

Delta modulation uses one bit to encode on sample. Hence bit rate of delta modulation is low compared to PCM.

10. What is meant by adaptive delta modulation?

In adaptive delta modulation, the step size is adjusted as per the slope of the input signal. Step size is made high if slope of the input signal is high. This avoids slope overload distortion.

11. What is meant by quantization?

While converting the signal value from analog to digital, quantization is performed.The .analog value is assigned to nearest digital value. This is called quantization.The quantized value is then converted into equivalent binary value. The quantization levels are fixed depending upon the number of bits. Quantization is performed in every Analog to Digital Conversion.

13. Define Pulse Amplitude modulation

The amplitude of a carrier pulse is altered in accordance to that of amplitude of message signal to make it accommodate the information signal.

14. Define Pulse code modulation

Pulse code modulation refers a form of source coding. It is a form of digital modulation techniques in which the code refers a binary word that represent digital data.With PCM, the pulses are of fixed length and fixed amplitude

15. Define sampling rate

The sampling rate fs must be atleast two times the highest frequency component of the original signal to be accurately represented fs>=2fm