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In

ELECTRONICS AND TELECOMMUNICATION ENGINEERING

DEPARTMENT OF ELECTRONICS AND TELECOMMUNICATION

ENGINEERING

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INDIA

EXPERIMENT NO :

Title   : Study of Sampling and Quantization

Class   : Final year B.Tech (Extc)

Batch   :

Roll No.   :

Date   :

EXPERIMENT NO.

AIM: To study the sampling and quantization of signal.

THEORY:

A signal is defined as some variable which changes subject to some other independent variable. We will assume here, that the independent variable is time, denoted by t and the dependent variable could be any physical measurement-variable which changes over time - think for example of a time varying electric voltage. We will denote the generic measurement variable with x or x(t) to make its time-dependence explicit. An elementary example of such a signal is a sinosoid. When we want to represent such a sinusoid in the digital domain, we have to do two things: sampling and quantization which are described in turn.

Sampling:

The first thing we have to do, is to obtain signal values from the continuous signal at regular time-intervals this process is known as sampling. The sampling interval is denoted as Ts and its reciprocal, the sampling frequency or sample-rate is denoted as fs, where fs = 1/Ts. The result of this process is just a sequence of numbers. We will use n as an index into this sequence and our discrete time signal is denoted as x[n] - it is customary in DSP literature to use parentheses for continuous variables such as the time t and brackets for discrete variables such as our sample index n. Having defined our sampling interval Ts, sampling just extracts the signals value at all integer multiples of Ts such that our discrete time sequence becomes:
$$x[n] = x(n \cdot Ts) \ (1)$$
Note that at this point (after sampling), our signal is not yet completely digital because the values x[n] can still take on any number from a continuous range - that's why we use the terms discrete - time signal here and not digital signal. Figure 1 illustrates the process of sampling a continuous sinusoid.
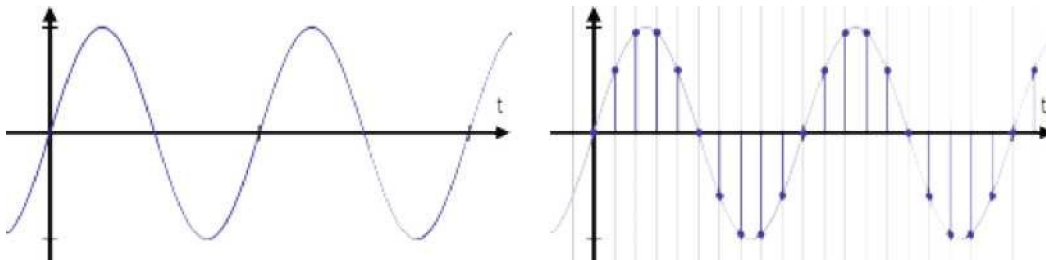

Fig.

The sapling theorem is significant in communication system because it provides the basis for transmitting analog signal by use of digital techniques. The sampling theorem can be defined in two ways:

1) A band limited signal having no frequency components higher than fm Hz is completel[y] described by its samples values at uniform intervals less than or equal to 1/2*fm sec apart.

2) A band limited signal having no frequency components higher than fm Hz may be completel[y] recovered from the knowledge of its samples taken at rate of at least 2fm samples per second.

Quantization:

After the sampling we have a sequence of numbers which can theoretically still take on an[y] value on a continuous range of values. Because this range in continuous, there are infinite[ly] many possible values for each number, in fact even uncountably infinitely many. In order to b[e] able to represent each number from such a continuous range, we would need an infinite numb[er] of digits - something we don't have. Instead, we must represent our numbers with a fini[te] number of digits, that is: after discretizing the time-variable, we now have to discretize th[e] amplitude-variable as well. This discretization of the amplitude values is called quantizatio[n]. Assume, our sequence takes on values in the range between -1... + 1. Now assume that we mu[st] represent each number from this range with just two decimal digits: one before and one after th[e] point. Our possible amplitude values are therefore: -1.0,-0.9, . . . ,-0.1, 0.0, 0.1, . . . , 0.9, 1.[0]. These are exactly 21 distinct levels for the amplitude and we will denote this number [of] quantization levels with Nq. Each level is a step of 0.1 higher than its predecessor and we w[ill] denote this quantization stepsize as q. Now we assign to each number from our continuous rang[e] that quantization level which is closest to our actual amplitude: the range -0.05... + 0.05 maps [to] quantization level 0.0, the range 0.05...0.15 maps to 0.1 and so on. That mapping can be viewe[d] as a piecewise constant function acting on our continuous amplitude variable x.

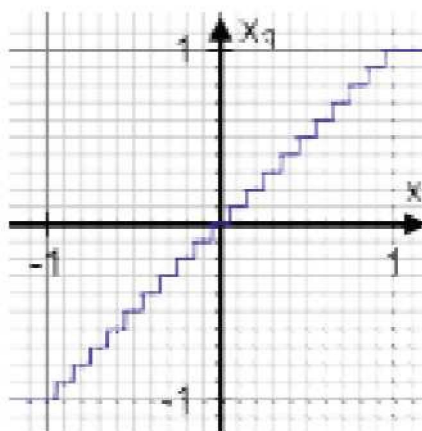Fig: Characteristic line of a quantizer

Conclusion: Hence we have studied the sampling and quantization.

EXPERIMENT NO :

Title   : Study of Delta Modulation.

Class   : Final year B. Tech (Extc)

Batch   : H

Roll No.   : 201203164

DOP   **:**

DOS   :

EXPERIMENT NO.

Aim: Study of Delta Modulation.

Theory:

Delta modulation is a system of digital modulation developed after pulse code modulation. In this system, at each sampling time, say the K th sampling time, the difference between the sampling value at sampling time K and the sampling value at the previous sampling time (K-1) is encoded into just a single bit, i.e. at each sampling time we ask simple question.

Has the signal amplitude increased or decreased since the last sample was taken?

If signal amplitude has increased, then modulator's output is at logic level 1.

If signal amplitude has decreased, then modulator's output is at logic level 0.

Thus, the output from the modulator is a series of zeros and ones to indicate rise and fall of the waveform since the previous value. One way in which delta modulator and demodulator is assembled is as shown in fig.1 and fig.2

The delta modulator works as follows:

The analog signal which is to be encoded into digital data is applied to +ve input of the voltage comparator which compares it with the signal applied to its –ve input from the integrator output(more about this signal in fourth coming paragraph).

The comparators output is logic '0'or '1' depending on whether the input signal at +ve terminal is lower or greater than the –ve terminal input signal.

The comparators output is then latched into a D flip-flop which is clocked by the transmitted clock. Thus the output of D flip-flop is latched '1' or '0' synchronous with the transmitter clock edge.

This binary data stream is transmitted to receive and is also fed to the unipolar to bipolar converter. This block converts logic '0' to voltage level of +4v and logic '1' to voltage level -4v.

The bipolar output is applied to the integrator whose output is as follows:

a. Rising linear ramp signal when -4v is applied to it. (corresponding to binary 1)
b. Falling linear ramp signal when +4v is applied to it. (corresponding to binary 0)

The integrator output is then connected to the–ve terminal of voltage comparator, thus completing the modulator circuit.

Let us understand the working of modulator circuit with the analog input waveform as in graph:

Suppose at some time instant t=0, the integrator input voltage is lower than the analog input. This causes the voltage comparator voltage to go high i.e. logic'1', this data is latched in the D flip-flop at the rising edge of transmitter clock. The latched '1' output of D flip flop is translated to -4v by the unipolar to bipolar converter block. The integrator then ramps up to catch analog signal.

At the next clock cycle t=1, the integrator output becomes more than the analog input, so '0' is latched into D flip-flop. The integrator now ramps downwards as +4v signal from unipolar to bipolar converter appears at its input. Thus the ramp signal again tries to catch the fallen analog signal.

As we can observe, after several clock cycles the integrator output is approximation of the analog input which tries to catch up the analog input at each sample time. The data stram from D flip-flop is the delta modulators output.

Conclusion:

Hence we have studied Delta Modulation and Demodulator.

EXPERIMENT NO:

Title   : Study of Adaptive Delta Modulation.

Class   : Final year B. Tech (Extc)

Batch   : H

Roll No.   : 201203164

DOP   **:**

DOS   :

EXPERIMENT NO.

Aim: Study of Adaptive Delta Modulation.

Theory:

As it has been seen, delta modulation system is unable chase the rapidly changing information of the analog signal which gives rise to distortion & hence poor quality reception. This is known as slope overloading phenomenon. The problem can be overcome by increasing the integrator gain(i.e. step size).But using high step size integrator would lead to a high quantization noise.

Quantization Noise -

It is defined as error introduced between the original signal, & the quantized signal due to the fixed step-size in which the signal is incremented .As the error is random in nature & hence unpredictable, it can be treated as noise. High quantization noise may play on small amplitude signal. The solution to this problem is to increase the integrator gain for fast changing input & to use normal gain for small amplitude signal.

The basic idea is to increase the integrator gain when slope overload occurs. If still it is unable to catch up with the signal, the integrator gain is double again. The integrator on board has four available gains standards, standard X2, standard X4 & standard X8, The integrator this adapts itself to the gain where its lowest value can just overcome the slope overloading effect.

Functionally, the adaptive delta modulator/demodulator is shown in fig.

As it can be observed, the adaptive delta modulation is similar to the delta modulator except for few blocks namely the counter & the control circuit.

The input to control circuit is the latched data from D flip flop. The counter is reset whenever 'high' appears at the output of the control circuit.

The input to the control circuit is the latched data from the D flip-flop. The counter is reset whenever 'high' appears at the output of the control circuit. Both the counter & the control circuit are clocked by the same TX clock. The input to the integrator from the counter 2-bit

control word which control the gain of the integrator. When the output of counter is '00' the gain is lowest where as it is highest for counter output '11'.

The control circuit works as follows-

The control circuit compares the preset data bit from D-flip-1=flop with the previous two data bits. Its output to the counter is high when the bit data is identical, the control circuits output goes low, thus letting the counter advance with every clock cycle. The adaptive delta modulator is same as delta demodulator except for two blocks namely, the control circuit & the counter. They function in the same as in modulator part, except for the fact that they are clocked by the receiver clock.

Consider the adaptive delta modulation in operation. In normal case, when slope overloading is not occurring, the integrator output always hunt above & below the analog input even after it has caught up with it. The output from the D flip-flop is a constantly changing '1' to '0' at each TX clock edge. Even when the analog is changing at a slightly higher rate, the integrator ramp output is able to catch it in two clock cycles, thus, the output of D-flip-1=flop is never a three or more consecutive '0' or '1's.

The changing input to the control circuit ensures that its output to the counter is high & hence the counter is rested at every clock cycle. Thus the control word from the counter is always '00'forcing the integrator gain at its lowest value, thereby reducing quantization noise. Here the adaptive delta modulator is behaving just as a delta modulator.

Suppose now a fat changing analog signal appears at the input of the modulator such that the slope overloading occurs. The integrator output no longer follows the analog signal but it spends its time trying to catch up the analog signal. As a result of continuous ramping in one direction, the D flip-flop output is either '0' or '1' for three or more consecutive time.

Conclusion:

Hence, we have studied Adaptive Delta Modulation and Demodulator.

Write all types manchster biplor unipolar

EXPERIMENT NO:

Title   : To study different Line coding techniques

Class   : Final year B. Tech (Extc)

Batch   : G

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DOS   :

EXPERIMENT NO.

Aim: To study different coding techniques.

Theory:-

Introduction:-

Unipolar scheme(return to zero):-

Traditionally a unipolar scheme was designed as a zero voltage define signal zero. It is called RZ because the signal does not return to zero at the middle of the bit fig. shows unipolar RZ scheme.

Compare this polar quarter parts these scheme is very costly. The normalized power (power needed to send one bit per unit line resistance) double that for polar NRZ. For this reason these scheme is not used in data communication time axis.

1) NRZ (non return to zero):-

In polar NRZ encoding to levels of voltage amplitude are used. The two versions of NRZ i.e.1st version NRZL and NRZI as shown in fig. In 1st version i.e. NRZL (NRZ level) the level of voltage determines the value of the bit. In 2nd variation NRZI (NRZ invert) the change or lack of change in level of voltage determines the voltage of bit. If there is no change the bit is zero, if there is change bit is one. Although this line wondering to the problem for both variation. It is twice as server than NRZL. If there is long sequence of 0's and 1's the average signal power becomes skewed in NRZI. This problem occurs only for long sequence of 0's.

The synchronization problem that in(sender & receiver clocks are not synchronized) also serious in NRZL than in NRZI while along zero's can cause a problem in both scheme. A long sequence of it affects only NRZI.

Another problem with NRZL with occurs when there sudden change in polar scheme. NRZI does not have this problem. Both signal have an average rate of NRZ bandwidth.

2) RZ (return to zero)

The main problem with NRZ encoding when sender & receiver clock are not synchronized .The receiver does not known when one bit ended & next bit starting.

One solution is RZ scheme. In RZ, this uses three values i.e. positive negative & zero. In RZ signal not change in bit during the bits. In fig we see that signal goes to 0's in the middle of each bit. The main disadvantage of RZ encoding is that it requires 2 signal changes to encode a bit therefore occupies greater bandwidth.

The same problems we maintain a sudden change of polarity resulting in all zero's still exist here, but there is no dc component problem. Another problem is complexity. It has been replaced by better performing Manchester & differential Manchester scheme.

Conclusion:- Thus we have studied RZ and NRZ codes.

EXPERIMENT NO:

Title : Study of FDM

Class : Final year B. Tech (Extc)

Batch : G

Roll No. :

DOP :

DOS :

<div align="center">**EXPERIMENT NO.**</div>

Aim: Study of the Frequency Division Multiplexing/De-multiplexing with sinusoidal wave.

Equipment Required:

1. ST2211 trainer with power supply cord.
2. Oscilloscope with connected probe.
3. Patch cords.

Theory:

The deriving of two or more simultaneous, continuous channel from a transmission medium by assigning a separate portion of the available frequency spectrum to each of the individual channel is known as FDM.

It is possible by simultaneous transmission of multiple separate signals through a shared medium (such as a wire, optic, or light beam) by modulating, at the transmitter. These separate signals have separable frequency bands and by adding those results linearly either before transmission or within the medium. While thus combined, all the signals may be amplified, conducted, translated in frequency and routed toward a destination as a simple signal.

This technique is cost effective, which are the motivation for multiplexing. Apparatus at the receiver separates the multiplexed signals by means of frequency passing or rejecting filters, and demodulates the result individually, each in manner appropriate for the modulation scheme used for that band or group. Neither the transmitters nor the receivers need be close to each other; ordinary radio, television, and cable service are example of FDM. It was once the mainstay of the long distance telephone system. The more recent developed time division multiplexing in its several forms lends to the handling of digital data, but the low cost and high quality of available FDM equipment, especially which intended for television signals, makes it a reasonable choice for many purpose.

Frequency division multiplexing is the process of combining several information channels by shifting their signals to different frequency groups within the frequency spectrum so that they can all be transmitted over a common transmission channel.
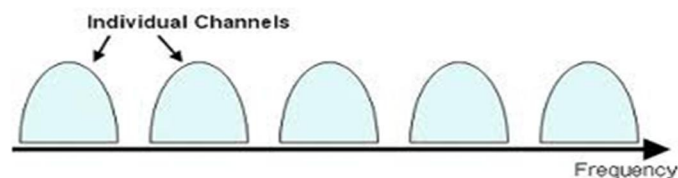
<div align="center">Frequency Division Multiplexing (FDM)</div>

Procedure:-

1) Set the modulating frequency of channel 1 with the help of potentiometer to 2 kHz and channel 2 to 4 kHz.
2) Observe the carrier frequency 100 kHz and 200 kHz on the oscilloscope.
3) Connect the channel 1 output to left input of modulator channel 1.
4) Repeat step 3 for channel 2 also.
5) Connect carrier generator output (100 kHz and 200 kHz) to CH 1and CH 2 respectively.
6) Observe the modulator output on oscilloscope.
7) Connect the modulator output of CH 1 and CH 2 to adder circuit.
8) Connect the adder output to demodulator inputs in both the section.
9) Connect the respective carrier frequency to demodulator second input.
10) Connect the output of demodulator of CH 1 and CH 2 to LPF 1 and LPF 2.
11) Observe the output of low pass filter on the scope and compare it with modulating signal.

Conclusion:

During the process of frequency division multiplexing and de-multiplexing of signals both the signals are as same as applied input.

EXPERIMENT NO:

Title   : Study of Four channel analog TDM.

Class   : Final year B. Tech (Extc)

Batch   : G

Roll No. :

DOP   **:**

DOS   :

EXPERIMENT NO.

Aim: To Study the four channel analog time division multiplexing.

Apparatus: Four channel analog TDM channel, CRO, connecting wires, etc.

Theory:- Time Division Multiplexing(TDM)

Multiplexing is the process of combining signals from different information sources so that they can be transmitted over a common channel. Multiplexing is the advantage in cases where it is impracticable and uneconomical to provide separate links for different information sources. The price that has to be paid to acquire this advantage is in the form of increased system complexity and bandwidth.

It is a technique of transmitting more than one information on the same channel as can be noticed from fig.1 the samples consist of short pulses followed by another pulses after a long time interval. This no activity time interval can be used to include samples from the other channels as well. This means that several information can be transmitted over a single channel by sending samples from different information sources, as different moment in time. This technique is known as time division multiplexing (TDM).

Analog TDM:-

In this technique of multiplexing analog signals appear at the input of multiplexer and samples of signals are taken at instant of time and transmitted on the same channel by interweaving them.

In analog communication system like AM, FM the instantaneous value of the information signal is used to hang certain parameter of the carrier signal. The time division signal multiplexing signal can be simulated by two rotating switches, one at transmitter and other at receiver. The wipers rotate and establish electrical contact with one channel at a time.

Each signal is sampled over one sampling interval and transmitted one after the other along a common channel. Thus part of message is 1st followed by pair message 2, message 3 and then again message 1, so on. It can be anticipated from above process that the receiver switch has to follow two constraints.

1. It must rotate at the same as receiver switch.
2. It must start at the same time as transmitting switch and it must establish electrical contact with same channel no. as that of transmitter.

Procedure:-

1. Turn all potentiometers of analog signal generator unit to clockwise position.
2. Turn all balance adj. potentiometer of balanced modulator block to clockwise position.
3. Make connections.
4. Turn on the power of the trainer. It is indicated by lighting of switch.
5. Observe TDM output on oscilloscope by connecting TDM output terminal to oscilloscope.
6. Trigger signal to view clean multiplexed signal. This TDM output signal carries components of all four channels multiplexed in time domain.
7. Observe control signals tp10 and tp11.
8. Now vary the potentiometer of analog signal generator unit and balanced modulator block. Observe the effect of varying each potentiometer has on the TDM waveform at tp12. This point clears the concept of TDM.

Conclusion:-

Hence we have studied four channel analog time division multiplexing.

EXPERIMENT NO :

Title   : Study of Pulse Amplitude Modulation with

 Sampled, Hold, Flat-top.

Class   : Final year B. Tech (Extc)

Batch   : G

Roll No.   :

DOP   **:**

DOS   :

<center>**EXPERIMENT NO:**</center>

Aim: To study pulse amplitude modulation and demodulation with sample, hold, flat-top.

Apparatus: Kit

Theory:

Modulation is the process of changing parameters of carrier signal with respect to modulating signal. In pulse modulation system, the carrier is no longer a continuous signal but consist of pulse train same parameter of which is varied in accordance with instantaneous value of modulating signal.

There are two types of pulse modulation system.

1) Pulse amplitude modulation (PAM).
2) Pulse time modulation (PTM).
      a) Pulse position modulation (PPM).
      b) Pulse width modulation (PWM).

A) Pulse amplitude modulation:
The amplitude of carrier pulse train is varied in accordance with the modulating signal.

Sampling Theorem:
It is significant in communication system. Because it provides the basis for transmitting analog signal is by use of digital tech. a band limiting signal having no frequency component higher than MHz may be completely recovered from knowledge of its sample taken at rate at least 2fm samples per second.
From fig. carrier pulse train fc(t) base band signal f(t) i.e. spectral range occupied by basic signal is called baseband signal. The frequency of carrier pulse train is decided by sampling theorem, fm is modulated amplitude signal which depends upon the value of f(t) during the time of pulse. The PAM signal fm(t) is known as discrete on time axis and continuous on amplitude axis. Modulated pulse will also be of positive as well as negative polarities. As the transmission of such bipolar pulse is inconvenient, a clamping circuit is used so that we always have a baseband signal with only the positive polarity.

Two methods of getting pulse amplitude modulation of waveform :

1) **Natural Sampling:-**
   Amplitude of carrier, pulse train is adjusted to 1, the duration of signal of pulse is T and they are separated by Ts. The PAM signal is obtained by f(t) and fc(t) in multiplier.
   The top of pulse amplitude modulated pulse are not flat but they follow the natural waveform of modulating signal f(t). During respective pulse intervals.

2) **Flat Top Sampling:-**
   The electronic circuitry needed to perform natural sampling completed because pulse top shape is to be maintained. These complications are reduced by flat top sampling shown in graph. Pulses have constant amplitude within the pulse interval and by making value of pulse amplitude constant within pulse interval some distortion is introduced

   PAM Modulator Circuit:-.
   This circuit is simple emitter follower in absence of clock signal that is output follower input. The modulating signal applied as an input signal. Another input to the base of the transistor is the clock signal. The frequency of clock is made equal to desire carrier pulse train frequency. The amplitude of clock signal is so chosen that the high level is over low level. Some negative voltage which is sufficient to bring transistor in cutoff region. When clock signal is low, transistor is cutoff ,output is zero and we get desired pulse amplitude modulating signal.

   Procedure:-
   1) Start the kit.
   2) Connect the wires as per the diagram.
   3) Observe the output.

Conclusion:-

Hence, we have studied pulse amplitude modulation with natural sampling and flat top sampling

EXPERIMENT NO:

Title   : Study of PWM using different Sampling

                Frequency.

Class   : Final year B. Tech (Extc)

Batch   : G

Roll No. :

DOP   :

DOS   :

Aim: Study of PWM using different Sampling Frequency.

Apparatus: MATLAB 2009 software.

Theory:

Modulation is process of changing parameters of carrier signal with respect to modulating signal. In pulse modulation system, the carrier is no longer a continuous signal but consists of pulse train same parameter of which is varied in accordance with instantaneous value of modulating signal.

There are two types of pulse modulation system'

1) Pulse amplitude modulation (PAM).
2) Pulse time modulation (PTM).
   a) Pulse position modulation (PPM).
   b) Pulse width modulation (PWM).

Pulse width modulation:

In Pulse width modulation of pulse amplitude modulation is also often PDM(pulse duration modulation) and less often, PLM (pulse length modulation). In system ,as shown in Figure a, we have fixed amplitude and starting time of each pulse, but the width of each pulse is made proportional to the amplitude of the signal at that instant.

PROGRAM:

```
%% MATLAB CODE FOR PWM
t=0:0.001:10;
f1=2*sin(0.5*pi*t);
f2=4*sawtooth(2*pi*t,1);
n=length(t);
for i=1:n
        if f1(i)>f2(i)
        z(i)=1;
    else
        z(i)=0;
        end
end
subplot(221);
plot(t,f1,'b');
hold on;
plot(t,f2,'r');
subplot(222);
plot(t,z,'g');
axis([0 10 -1.5 1.5])
```