Lecture 1: Digital Baseband Transmission and Match Filter

Digital Baseband Transmission

- A method used in communication sysems to transmit digital data without modulating it onto a carrier frequency.
- Instead of modulating the data onto a carrier wave, as in analog transmission, digital baseband transmission sends the data directly over the transmission medium.
- This can be achieved in various techniques,

line coding - converts the digital signal into a format suitable for transmission over a communication channel.

Pulse shaping - minimize intersymbol interference.

- One of the advantages of digital baseband transmission is its simplicity and efficiency in transmitting digital data directly without the need for analog modulation and demodulation processes.

Base band Pulse transmission

Generally, there are two types of digital data transmission techniques,

1. Base Band pulse transmission
2. Pass Band Pulse transmission

Baseband pulse Transmission - The digital data is transmitted over the channel directly. There is no carrier or any modulation. This transmission is suitable for transmission over short distance communication ~~transmission~~

Passband pulse Transmission - The digital data can be transmitted ~~are~~ over the channel by using modulation techniques. This one can be used for the longer distance communication.

In baseband transmission, the data bits are directly converted into signals. Generally a higher voltage level represents the bit 1, while a lower voltage level represents bit 0.

• Baseband transmission is the simplest form for the commun... of information. Discrete information is communicated with specific symbols selected from a finite set of symbols.

• In baseband transmission, symbols are simply communicated... a pulse with a discrete voltage level and, for binary transmiss... only two voltages are used.

• Transmission of digital data (bit stream) over a noisy baseband channel typically suffers ~~from~~ two channel limitations.

Inter symbol interference (ISI)

Background noise (e.g. AWGN)

• Baseband pulse transmission is a method used in data communic-ation where digital signals are transmitted quickly directly without modulation onto a transmission medium such as a wire or fiber optic cable. In this technique, the baseband signal typically consists of binary data represented as pulses.

• Here is a breakdown of how it works:

1. Binary Representation: Data to be transmitted is represented in binary form, where each bit is either a 0 or a 1.

2. Pulse Generation: Each bit in the binary data is represented by a pulse. For example, a high voltage pulse could represent a binary 1, and a low voltage pulse could represent a binary 0.

3. Transmission: These pulses are then directly transmitted over the communication medium. This could be a copper wire, fiber optic cable, or even wireless transmission.

4. Reception: At the receiving end, the pulses are detected and interprete-d back into binary form. This involves converting the received pulse signals back into digital data.

Format Analog Signals:

To transform an analog waveform into a form that is compatible with a digital communication, the following steps are taken:

Sampling

Quantization and encoding

Baseband transmission

Slide 6 - this is how Baseband transmission is carried out in

Binary pulse format

Waveform representation of binary digits

- Using PCM, analog waveforms are transformed into binary digits
- Binary digits will be represented with electrical ~~pulse~~ pulse in order to transmit through a baseband channel.

Pulse code Modulation and demodulation

- Pulse code Modulation is a method that is used to convert an analog signal ~~to~~ into a digital signal, so that modified analog signal can be transmitted through the digital communication network.
- PCM is in binary form, so there will be only two possible states high and low (0 and 1).

(We can also get back our analog signal by demodulation)

- The pulse code modulation process is done in three steps
  - Sampling
  - Quantization and
  - Coding

PCM Standard

There are two standards of PCM namely

- The European Standard
- The American Standard

• They differ slightly in the detail of their working the principles are the same.

European - PCM = 30 channels

North American - PCM = 24 channels

Japanese - PCM = 24 channels

Sampling

Analog signal is sampled every T_s sec

T_s is referred to as the sampling interval (time)

$f_s = 1/T_s$ is called the sampling rate or sampling frequency

There are 3 sampling methods:

Ideal - an ~~impulse~~ impulse at each sampling instant

Natural - a pulse of short with with varying amplitude

Flattop - sample and hold, like natural but with single amplitude value

Quantizing

• The process of measuring the numerical values of the sample in a suitable scale.

• The finite number of amplitude intervals is called the "quantizing interval" ~~like~~

• Quantizing intervals are coded in binary form, and so the quantization intervals will be ~~provided~~ ~~by~~ powers of 2.

• In PCM, 8 bit code is used and so we have 256 intervals, for quantizing (128 levels in the positive direction and 128 levels in negative direction)

Lecture 2

Baseband Pulse Transmission

• The purpose of a communication system is to transmit information-bearing signals through a communication channel separating the transmitter from the receiver.

• Information bearing signals are also referred to as baseband signals. The term baseband is used to designate the band of frequencies representing the original signal as delivered by a source of information.

• The proper use of the communication channel requires a shift of the range of baseband frequencies into other frequency ranges suitable for transmission, and a corresponding shift back to the original frequency range after reception. For example, a radio system must operate with frequencies of 30kHz and upward, whereas the baseband signal usually contains frequencies in the audio frequency range, and so some form of frequency-band shifting must be used for the system to operate satisfactorily.

• A shift of the range of frequencies in a signal is accomplished by using modulation, which is defined as the process by which some characteristic of a carrier is varied in accordance with a modulating wave (signal). A common form of the carrier is a sinusoidal wave, in which case we speak of a continuous wave modulation process. The baseband signal is referred to as modulating wave, and the result of the modulation process is referred to as the modulated wave.

• Modulation is performed at the transmitting end of the communication system. At the receiving end of the system, we usually require the original baseband signal to be restored. This is accomplished by using a process known as demodulation, which is the reverse of the modulation process.

• In basic-signal processing terms, we thus find that the transmitter of an analog communication system consists of a modulator and the receiver consists of a demodulator as depicted in figure. In addition to the signal received from the transmitter, the receiver input channel noise. The degradation in receiver performance due to channel noise is determined by the type of modulation used.

Continuous - wave modulation

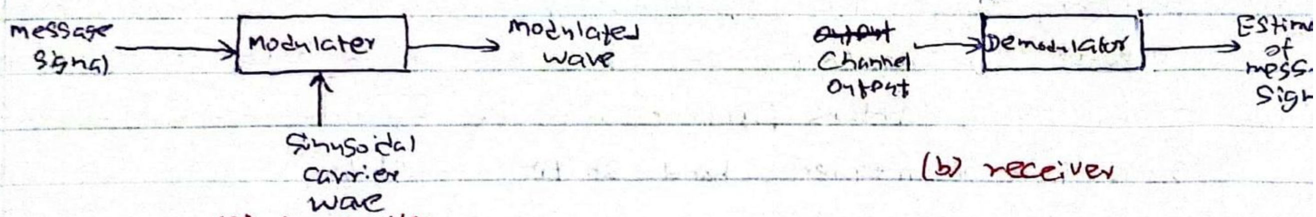
amplitude modulation

The amplitude of the Sinusoidal carrier wave is varied in accordance with the baseband signal

angle modulation

The angle of the Sinusoi[d]al carrier wave is varied in accordance with the baseband signal.

Components of a continuous - wave - modulation system



(a) transmitter

(b) receiver

• In continuous - wave (CW) modulation, some parameter of a sinusoidal carrier wave is varied continuously in accordance with the message signal. This is in direct contrast to pulse modulatio[n]

• In pulse modulation, some parameter of a pulse train is varied in accordance with the message signal. We may distinguish two families of pulse modulation: analog pulse modulation an[d] digital pulse modulation.

• In analog pulse modulation, a periodic pulse train is [u]sed as the carrier wave, and some characteristic feature of each pu[lse] (eg. amplitude, duration or position) is varied in a continuous manner in accordance with the corresponding sample value of t[he] message signal.

• Thus in analog pulse modulation, information is transmit[ted] basically in analog form, but the transmission takes place at discr[ete] times.

• In digital pulse modulation on the other hand, the mes[sage] signal is represented in a form that is discrete in both time and amplitude, thereby permitting its transmission in digital form as a sequence of coded pulses: this form of signal transmission has no CW counterpart.

• The uses of coded pulses for the transmission of analog-information-bearing signals represents a basic ingredient in the application of digital communications.

What is modulation?

• Modulation refers to the process of modifying a carrier signal, typically a high-frequency waveform, to carry information or data.

• In modulation, the information signal (also called the baseband signal) is superimposed or impressed onto the carrier signal, altering its properties such as amplitude, frequency, or phase. The resulting modulated signal, known as the modulated carrier, carries the information to be transmitted.

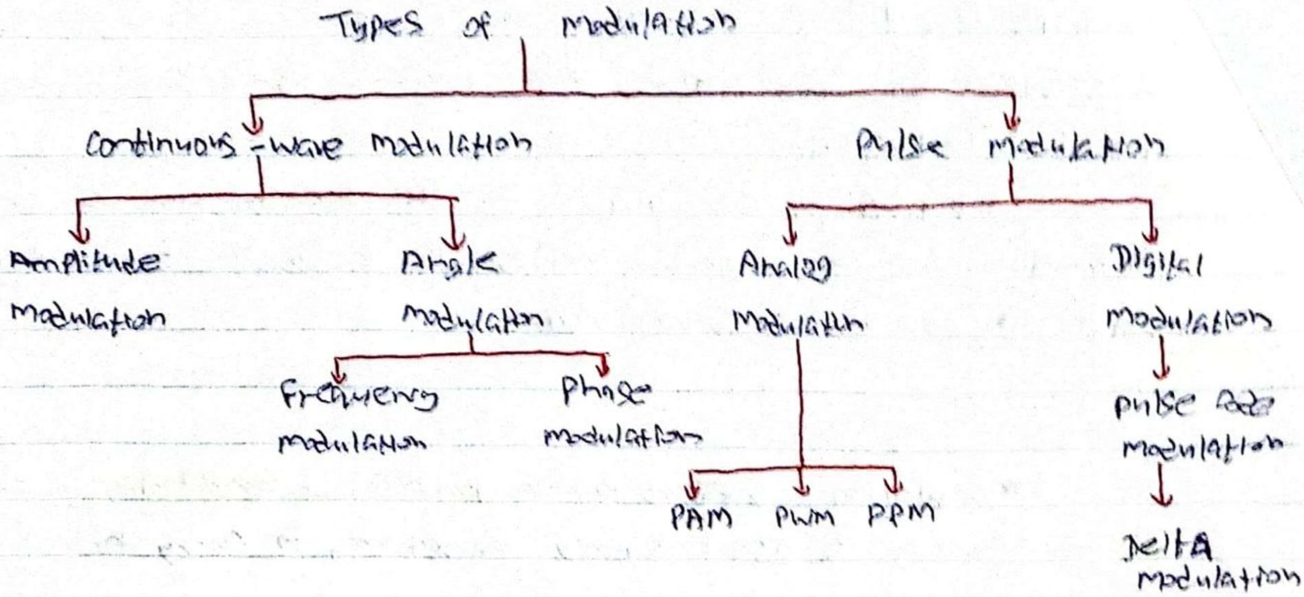
why we need modulation?

• Transmission Efficiency: Modulation enables the efficient use of the available bandwidth of communication channels. By shifting the frequency spectrum of the baseband signal to higher frequencies, multiple signals can be ~~transite~~ transmitted simultaneously without interference.

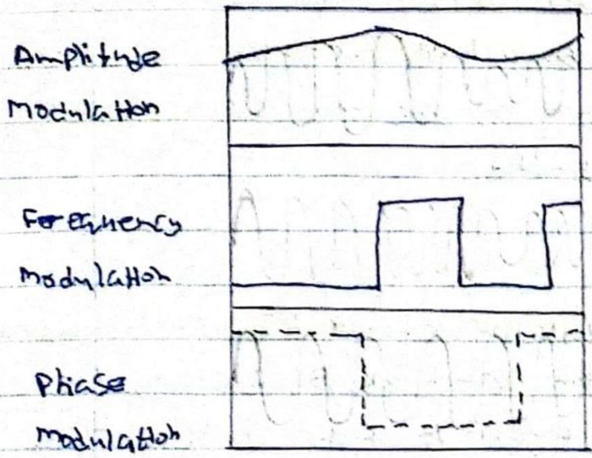
• Signal Integrity: Modulating the carrier signal helps in preserving the integrity of the information being transmitted. By utilizing specific modulation schemes, the modulated signal can be robust against noise, interference, and channel impairments, ensuring reliable transmission and reception of the information.

• Compatibility: Modulation allows for compatibility between different communication systems and devices. By using standardized modulation techniques, communication systems can interoperate, enabling seamless communication between different technologies and platforms

Types of Modulation

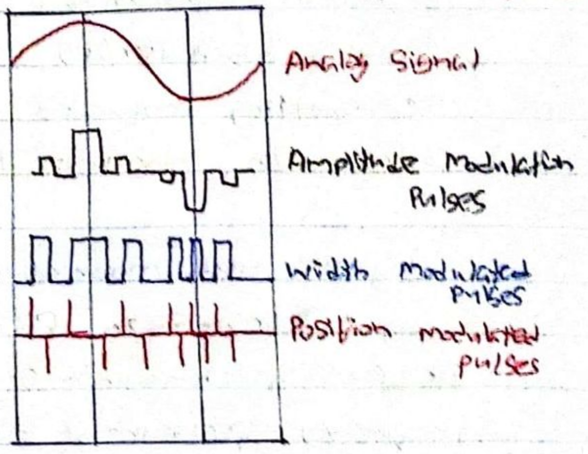


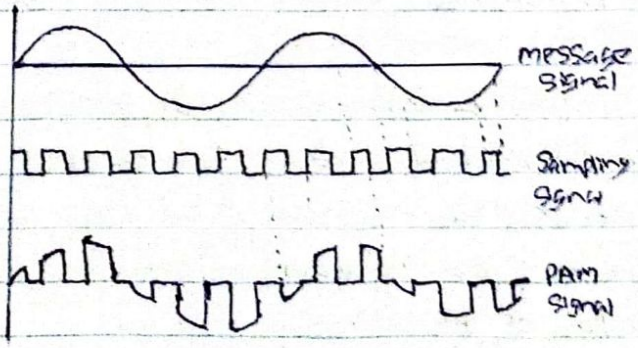
CW modulation



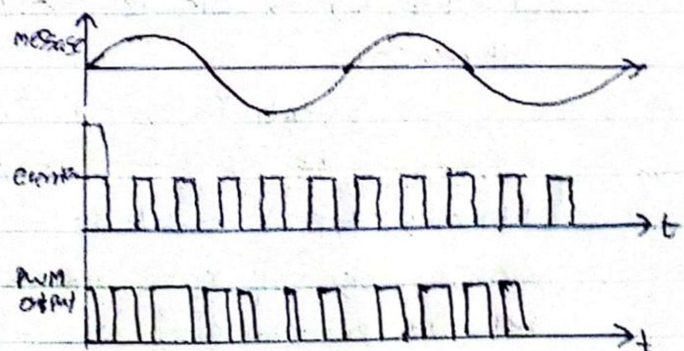
Analog pulse modulation

PAM, PWM and PPM at a glance



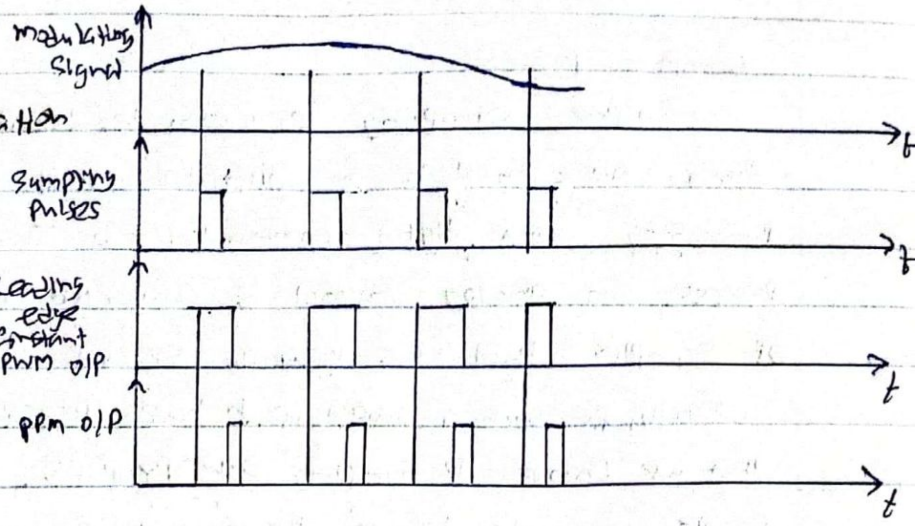


Pulse Amplitude Modulation
PAM



Pulse width modulation (PWM)

Pulse position modulation (PPM)



Types of Modulation

Pulse code Modulation

• Pulse code Modulation (PCM) is a digital modulation technique used for encoding analog audio or voice signals into a digital format for transmission or storage. It is widely used in telecommunication systems, audio recording, and playback devices.

The PCM process involves the following steps:

- Sampling
- Quantization
- Encoding
- Transmission or Storage

During Playback or reception, the PCM process is reversed

- Decoding
- Digital-to-Analog conversion
- Filtering and Reconstruction

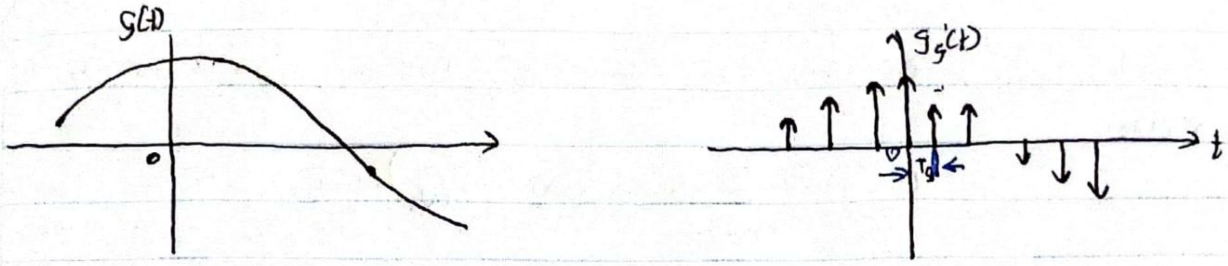
Sampling Process

- The sampling process is usually described in the time domain. As such it is an operation that is basic to digital signal processing and digital communications. Through use of the sampling process, an analog signal is converted into a corresponding sequence of samples that are usually spaced uniformly in time.
  Clearly, for such a procedure to have practical utility, it is necessary that we choose the sampling rate properly, so that the sequence of samples uniquely defines the original analog signal. This is the essence of the sampling theorem, which is derived in what follows.

- Consider an arbitrary signal $g(t)$ of finite energy, which is specified for all time. A signal segment of the signal $g(t)$ is shown in figure. Suppose that we sample the signal $g(t)$ instantaneously and at a uniform rate, once every T_s seconds. Consequently, we obtain an infinite sequence of samples spaced T_s seconds apart and denoted by $\{g(nT_s)\}$, where n takes on all possible integer values. We refer to T_s as the sampling period, and to its reciprocal $f_s = 1/T_s$ as the sampling rate. This ideal form of sampling is called instantaneous sampling.
  Let $g_\delta(t)$ denote the signal obtained by individually weighting the elements of a periodic sequence of delta functions spaced T_s seconds apart by the sequence of numbers $\{g(nT_s)\}$

$$g_\delta(t) = \sum_{n=-\infty}^{\infty} g(nT_s)\delta(t-nT_s) \quad -①$$

- We refer to $g_\delta(t)$ as the ideal sampled signal. The term $\delta(t-nT_s)$ represents a delta function positioned at time $t=nT_s$. From the definition of the delta function, we recall that such an idealized function has unit area; we may therefore view the multiplying factor $g(nT_s)$ in Equation (above) as a "mass" assigned to the delta function $\delta(t-nT_s)$. A delta function weighted in this manner is closely approximated by a rectangular pulse of duration Δt and amplitude $g(nT_s)/\Delta t$; the smaller we make Δt the better will be the approximation.



(a) (b)

The sampling process (a) Analog signal. (b) Instantaneously sampled version of the analog signal.

$$g_\delta(t) \rightleftharpoons f_s \sum_{m=-\infty}^{\infty} G(f - mf_s) \quad -(2)$$

- where $G(f)$ is the Fourier transform of the original signal $g(t)$, and f_s is the sampling rate.
- Above equation states that the process of uniformly sampling a continuous-time signal of finite energy results in a periodic spectrum with a period equal to the sampling rate.
- Another ~~Sam~~ useful expression for the Fourier transform of the ideal sampled signal $g_\delta(t)$ may be obtained by taking the Fourier transform of both sides of equation (1) and noting that the Fourier transform of the delta function $\delta(t - nT_s)$ is equal to $\exp(-j2\pi f T_s)$. Let $G_\delta(f)$ denote the Fourier transform of $g_\delta(t)$. We may therefore write

$$G_\delta(f) = \sum_{n=-\infty}^{\infty} g(nT_s)\exp(-j2\pi n f T_s) \quad -(3)$$

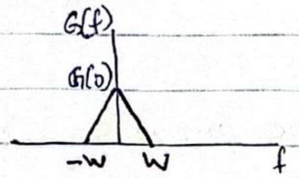
- The relation is called the discrete-time Fourier transform. It may be viewed as a complex Fourier series representation of the periodic frequency function $G_\delta(f)$, with the sequence of samples $[g(nT_s)]$ defining the coefficients of the expansion.
- The relations, as derived here, apply to any continuous-time signal $g(t)$ of finite energy and infinite duration. Suppose, however, that the signal $g(t)$ is strictly band-limited, with no frequency components higher than W Hertz. That is the Fourier transform $G(f)$ of the signal $g(t)$ has the property that $G(f)$ is zero for $|f| \geq W$, as illustrated in below figure 2, the shape of the spectrum shown in this figure is intended for the purpose of illustration only. Suppose also that we choose the sampling period $T_s = 1/2W$. Then the corresponding spectrum $G_\delta(f)$ of the sampled

signal $g_\delta(t)$ is as shown in below figure 2. Putting $T_s = 1/2W$ in Eqn 4 yields

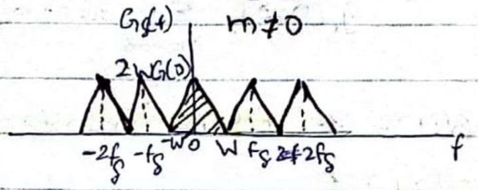
$$G_\delta(f) = \sum_{n=-\infty}^{\infty} g\left(\frac{n}{2W}\right) \exp\left(-\frac{j\pi n f}{W}\right) \quad -④$$

From Equation ②, we readily see that the Fourier transform of $g_\delta(t)$ may also be expressed as

$$G_\delta(f) = f_s G(f) + f_s \sum_{\substack{m=-\infty \\ m \neq 0}}^{\infty} G(f - mf_s) \quad -⑤$$



①



②

① Spectrum of a strictly band-limited signal

② Spectrum of the sampled version of $g(t)$ for a sampling period $T_s = 1/2W$.

Hence, under the following two conditions:

1. $G(f) = 0$ for $|f| \geq W$
2. $f_s = 2W$

We find from Equation 5 that

$$G(f) = \frac{1}{2W} G_\delta(f), \quad -W < f < W \quad -⑥$$

Substituting Equation 4 into 6, we may also write

$$G(f) = \frac{1}{2W} \sum_{n=-\infty}^{\infty} g\left(\frac{n}{2W}\right) \exp\left(-\frac{j\pi n f}{W}\right), \quad -W < f < W \quad -⑦$$

$$g(t) = \int_{-\infty}^{\infty} G(f) \exp(j2\pi f t)\, df$$

$$= \int_{-W}^{W} \frac{1}{2W} \sum_{n=-\infty}^{\infty} g\left(\frac{n}{2W}\right) \exp\left(-\frac{j\pi n f}{W}\right) \exp(j2\pi f t)\, df$$

Interchanging the order of summation and integration:

$$g(t) = \sum_{n=-\infty}^{\infty} g\left(\frac{n}{2W}\right) \frac{1}{2W} \int_{-W}^{W} \exp\left[j2\pi f\left(t - \frac{n}{2W}\right)\right] df \quad -⑧$$

The integral term in Equation is readily evaluated, yielding the final result

$$g(t) = \sum_{n=-\infty}^{\infty} g\left(\frac{n}{2W}\right) \frac{\sin(2\pi Wt - n\pi)}{(2\pi Wt - n\pi)}$$

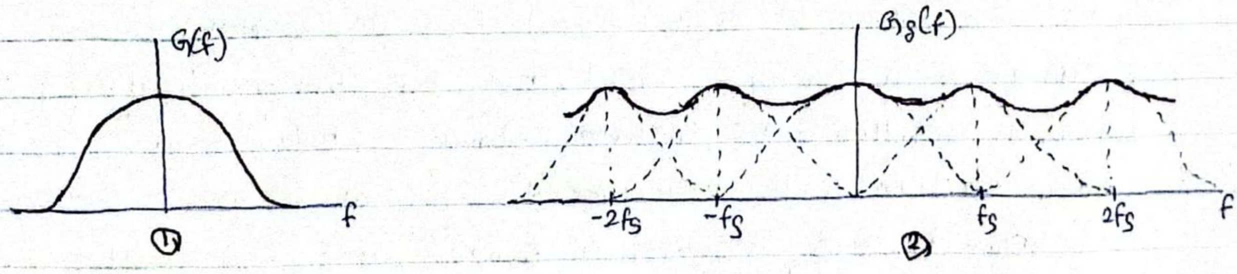
$$= \sum_{n=-\infty}^{\infty} g\left(\frac{n}{2W}\right) \operatorname{sinc}(2Wt - n), \quad -\infty < t < \infty \quad - (9)$$

(9) Provides an interpolation formula for reconstructing the original signal $g(t)$ from the sequence of sample values $[g(n/2W)]$, with the sinc function $\operatorname{sinc}(2Wt)$ playing the role of an interpolation function. Each sample is multiplied by a delayed version of the interpolation function, and all the resulting waveforms are added to obtain $g(t)$.

We may now state the sampling ~~function~~ theorem for strictly band-limited signals of finite energy in two equivalent parts, which apply to the transmitter and receiver of a pulse-modulation system, respectively:

1. A band-limited signal of finite energy, which has no frequency components higher than W Hertz, is completely described by specifying the values of the signal at instants of time separated by $1/2W$ seconds.

2. A band-limited signal of finite energy, which has no frequency components higher than W Hertz, may be completely ~~recorded~~ recovered from a knowledge of its samples taken at the rate of 2W samples per second.



(1) Spectrum of a signal

(2) Spectrum of an undersampled version of the signal exhibiting the ~~aliasing phenome~~ aliasing phenomenon

The sampling rate of 2W samples per second, for a signal bandwidth of W Hertz, is called the Nyquist rate; its reciprocal 1/2W (measured in seconds) is called the Nyquist interval.

The derivation of the sampling theorem, as described herein, is based on the assumption that the signal g(t) is strictly band limited. In practice, however, an information-bearing signal is not strictly band limited, with the result that some degree of undersampling is encountered. Consequently, some aliasing is produced by the sampling process. Aliasing refers to the phenomenon of a high-frequency component in the spectrum of the signal seemingly taking on the identity of a lower frequency in the spectrum of its sampled version, as illustrated in above figures ③, ① and ...
The aliased spectrum, shown by the solid curve in Fig ① pertains to an "undersampled" version of the message signal represented by the spectrum of Figure ①

To combat the effects of aliasing in practice, we may use two corrective measures, as described here:

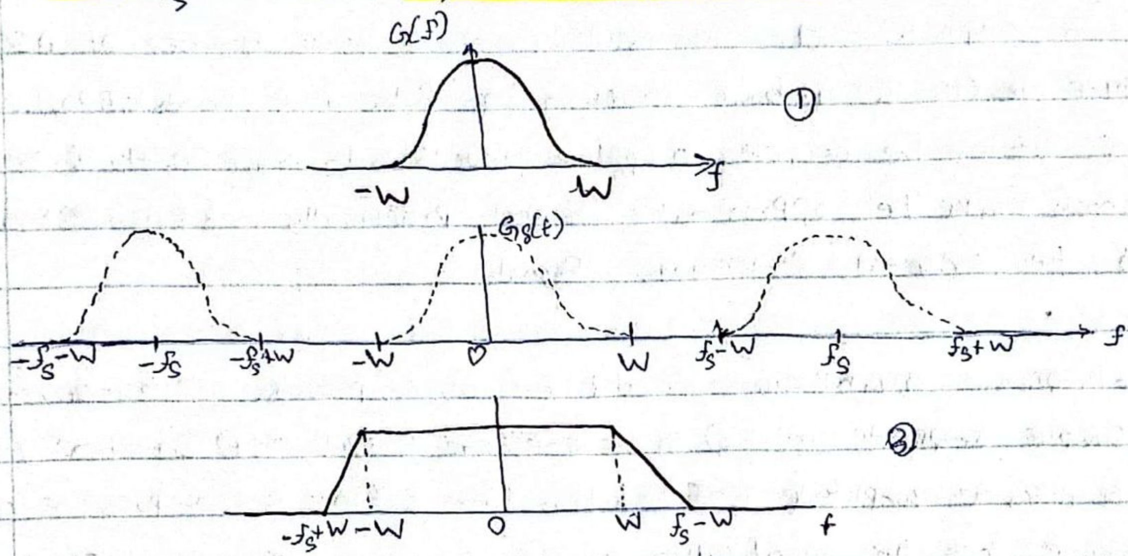
1. Prior to sampling: a low-pass anti-aliasing filter is used to attenuate those high-frequency components of the signal that are not essential to the information being conveyed by the signal.

2. The filtered signal is sampled at a rate slightly higher than the Nyquist rate.

The use of a sampling rate higher than the Nyquist rate also has the beneficial effect of easing the design of the reconstruction filter used to recover the original signal from its sampled version. Consider the example of a message signal that has been anti-alias (low-pass) filtered, resulting in the spectrum shown in below figure ①. The corresponding spectrum of the instantaneously sampled version of the signal is shown in below figure ②, assuming a sampling rate higher than the Nyquist rate. According to below figure ②, we readily see that the design of the reconstruction filter may be specified as follows (figure ③):

• The reconstruction filter is low-pass with a passband extending from $-W$ to W, which is itself determined by the anti-aliasing filter.

The filter has a transition band extending (for positive frequencies) from W to $f_s - W$ where f_s is the sampling rate.



① Anti-alias filtered spectrum of an information-bearing signal

② Spectrum of instantaneously sampled version of the signal, assuming the use of a sampling rate greater than the Nyquist rate.

③ magnitude response of reconstruction filter.

• The fact that the reconstruction filter has a well-defined transition band means that it is physically realizable.

quantization process

• A continuous signal, such as voice, has a continuous range of amplitudes and therefore its samples have a continuous amplitude range. In other words, within the ~~the~~ finite amplitude

Description of a memoryless quantizer

range of the signal, we find an infinite number of amplitude levels. It is not necessary in fact to transmit the exact amplitudes