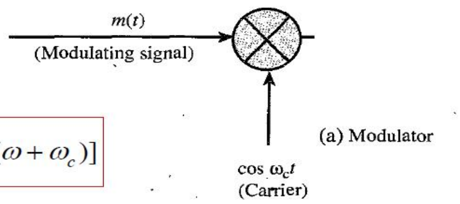
Amplitude modulation: Double Side Band (DSB)

Amplitude modulation is characterized by the fact that the **amplitude** A of the **carrier** $A\cos(\omega_c t+\theta_c)$ is varied **in proportion** to the **baseband** signal $m(t)$, the **modulating signal**.

The frequency ω_c and the phase θ_c are **constant**.
We can assume $\theta_c = 0$ without a loss of generality.

If the **carrier amplitude** A is made **directly proportional** to the modulating signal $m(t)$, the modulated signal is $m(t)\cos\omega_c t$.

As was seen earlier this type of **modulation simply shifts** the **spectrum** of **m(t)** to the **carrier frequency**. Thus, if then

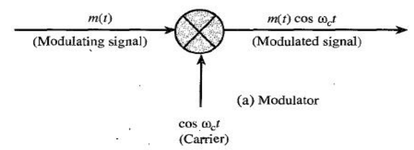


$$m(t) \Leftrightarrow M(\omega)$$

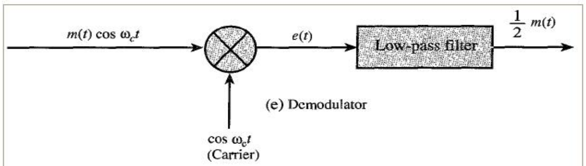
$$\boxed{m(t)\cos\omega_c t \Leftrightarrow \frac{1}{2}[M(\omega-\omega_c)+M(\omega+\omega_c)]}$$

Demodulation/ Detection

Thus, **demodulation**, which is almost identical to modulation, consists of **multiplication** of the **incoming modulated signal** $m(t)\cos\omega_c t$ by a **carrier** $\cos\omega_c t$ followed by a **low pass filter**.



Modulation



Demodulation

GENERATION OF ANGLE-MODULATED SIGNALS

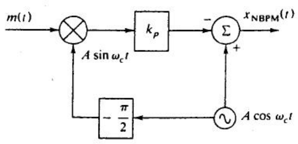
Narrowband Angle-Modulated Signals:

Neglecting all higher-power term of $\phi(t)$ the **angle modulated signal** becomes

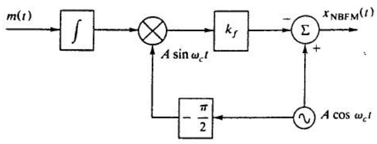
$$x_c(t) \approx A\cos\omega_c t - A\phi(t)\sin\omega_c t$$

The **narrowband (NB) angle-modulated signal** becomes

$$x_{\mathrm{NBPM}}(t) \approx A\cos\omega_c t - Ak_p m(t)\sin\omega_c t$$



(a) NBPM



(b) NBFM

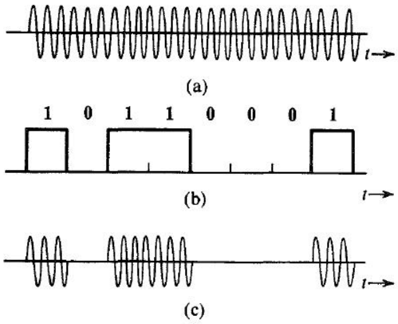
$$x_{\mathrm{NBFM}}(t) \approx A\cos\omega_c t - A\left[k_f\int_{-\infty}^{t} m(\lambda)\,d\lambda\right]\sin\omega_c t$$

DIGITAL CARRIER SYSTEMS: ASK

An unmodulated cartier $\cos\omega_c t$

The **on-off baseband signal m(t)** (the **modulating signal**)

When the **carrier amplitude** is varied in **proportion** to **m(t),** we have the **modulated carrier** $m(t)\cos\omega_c t$



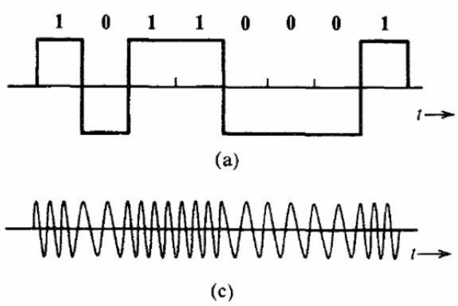
Note that the **modulated signal** is still an **on-off** signal.

$$x_c(t) = \begin{cases} A\cos\omega_c t & \text{symbol } 1 \\ 0 & \text{symbol } 0 \end{cases}$$

This **modulation scheme** of transmitting **binary data** is known as **on-off keying (OOK)** or **amplitude-shift keying (ASK).**

FSK

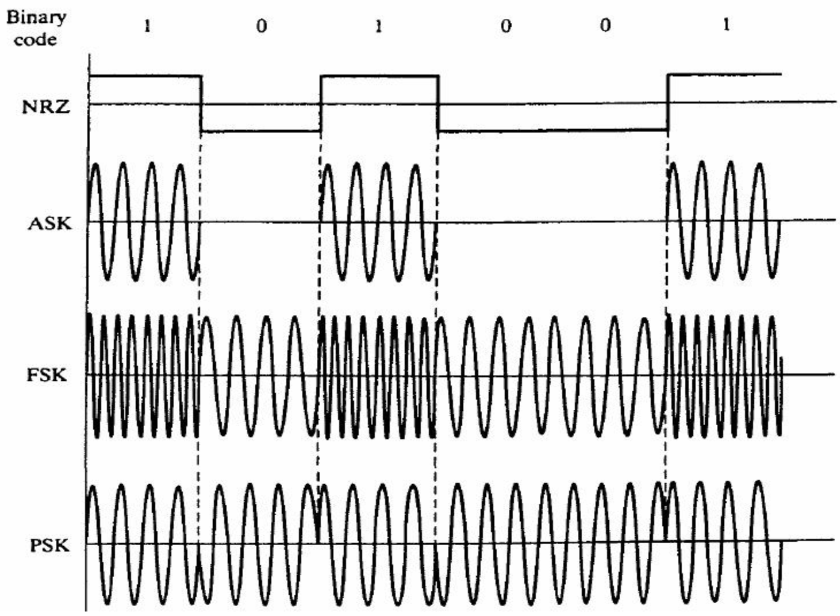
When the **data** are **transmitted** by **varying** the **frequency**, we have the case of **frequency shift keying (FSK).**



A **0** is transmitted by a **pulse** of **frequency** ω_{c0} and **1** is **transmitted** by a **pulse** of **frequency** ω_{c1}..

The information about the **transmitted data resides** in the **carrier frequency**.

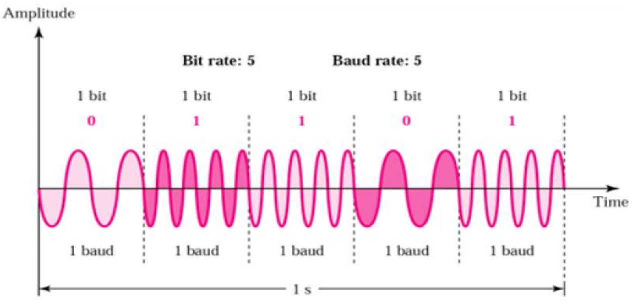
$$x_c(t) = \begin{cases} A\cos\omega_1 t & \text{symbol } 1 \\ A\cos\omega_2 t & \text{symbol } 0 \end{cases}$$



FSK

- Frequency of the carrier is varied to represent digital data (binary 0/1)
- Peak amplitude and phase remain constant.
- Avoid noise interference by looking at frequencies (change of a signal) and ignoring amplitudes.
- Limitations of FSK is the physical capabilities of the carrier.
- f_1 and f_2 equally offset by equal opposite amounts to the carrier freq.

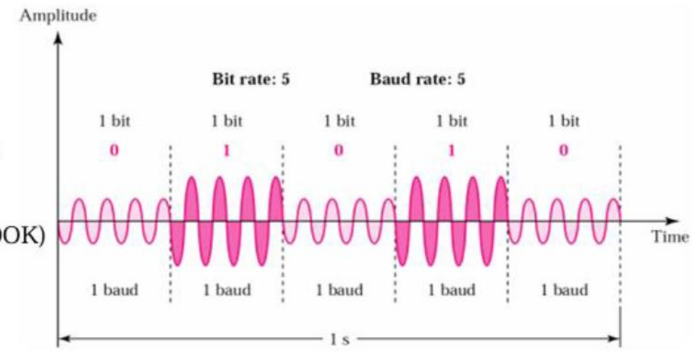
$$s(t) = \begin{cases} A\cos(2\Pi f_1 t) & binary1 \\ A\cos(2\Pi f_2 t) & binary0 \end{cases}$$



Amplitude Shift Keying (ASK)

$$s(t) = \begin{cases} A\cos(2\Pi f_c t) & binary1 \\ 0 & binary0 \end{cases}$$

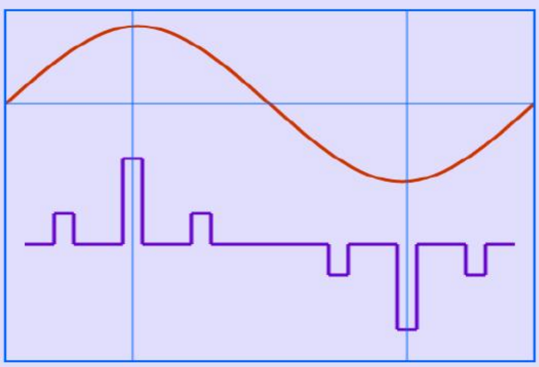
On/Off keying (OOK)

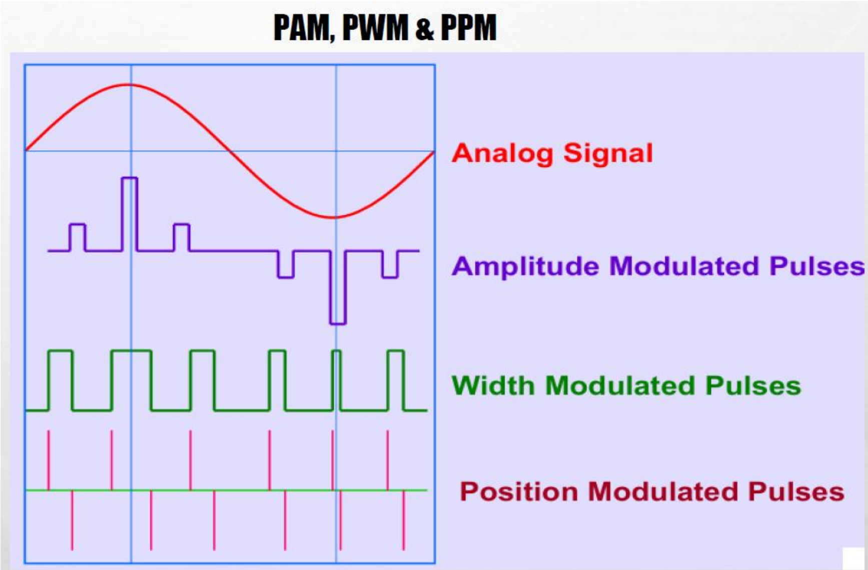


- The strength of the carrier signal is varied to represent binary 1 and 0.
- Frequency and phase remains the same.
- Highly susceptible to noise interference.

Pulse Amplitude Modulation (PAM)

- Amplitude of regularly spaced pulses varies according to the instantaneous value of the analog signal.
- Series of pulses with varying amplitudes. Basis for digital schemes like PCM.





T1 Time Division Multiplexing

T1 line is a digital transmission system used in North America and Japan. Originally designed to carry 24 voice channels over twisted-pair copper lines

This allows multiple signals to share a single transmission medium

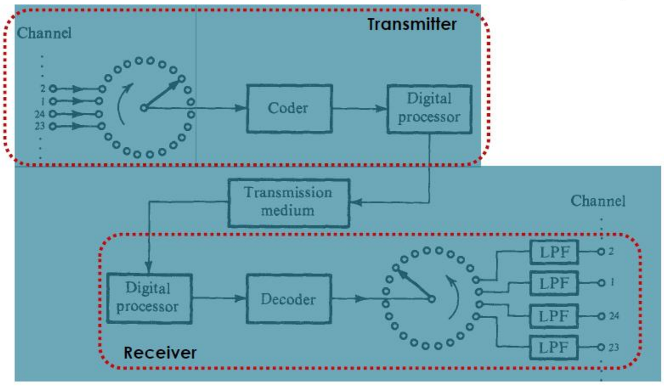
These 24 channels are then multiplexed using time division multiplexing. The T1 frame consists of 24 slots for the voice channels (24 x 8 bits = 192 bits) plus one framing bit, totaling 193 bits per frame.

Each voice channel is sampled at 8,000 times per second and converted into an 8-bit digital code using pulse code modulation (PCM). Total Data Rate = 193 bits x 8000 frames/sec = 1.544 Mbps

T1 is a synchronous system, meaning that each channel is allocated a fixed time slot, and the transmitting and receiving ends must be synchronized

Because the signal weakens over distance, T1 systems use regenerative repeaters every 6,000 feet to amplify and reshape the signa

T1 Time Division Multiplexing

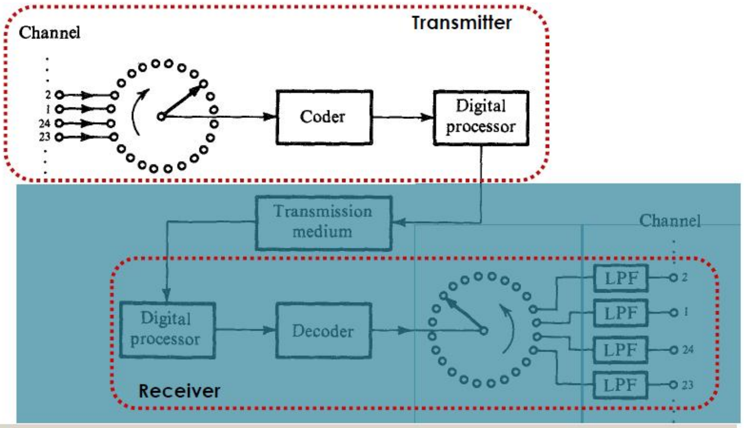


T1 Time Division Multiplexing has **24 channels**. All **24 channels** are **sampled in a sequence.**

The **sampler output** represents a **time-division-multiplexed PAM** (pulse amplitude modulated) signal.

The **multiplexed PAM signal** is now **applied to the input** of an **encoder** that **quantizes** each **sample** and **encodes** it into **eight binary pulses- a binary code word.**

T1 Time Division Multiplexing



The signal, now **converted** to **digital form**, is sent over the **transmission medium**.

Regenerative **repeaters** **spaced** approximately 6000 feet apart **detect** the **pulses** and **transmit new pulses**.

At the **receiver**, the **decoder** **converts** the **binary pulses** into **samples** (decoding).

The **samples** are then **demultiplexed** (i.e., **distributed** to each of the **24 channels**).

The **desired audio signal** is **reconstructed** by **passing** the **samples through a low-pass filter** in each channel.