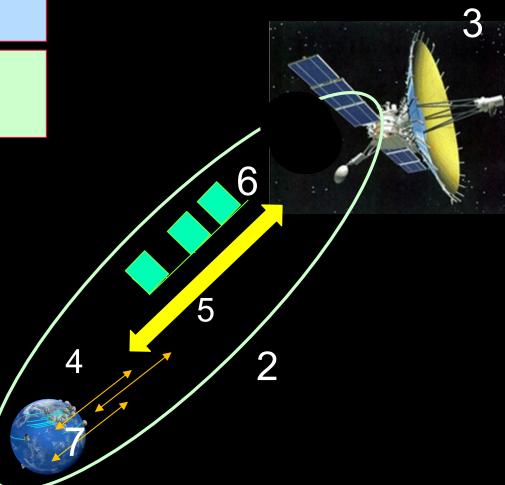
## PHYS 3250 Introduction to space communications

## Professor N Bartel Sketch of the 7 chapters

- 2 Orbital aspects
- 3 Spacecraft
- 4 Earth station
- 5 Communications link
- 6 Modulation and multiplexing techniques
- 7 Multiple access to a satellite





# 6. Modulation and multiplexing techniques

6.1 Introduction to analogue and digital modulation
6.2 Analogue modulation
6.3 Frequency division multiplexing
6.4 Digital baseband signal
6.5 Digital modulation
6.6 Time division multiplexing



#### 6.1 Introduction to analogue and digital modulation

First we want to compare the basics of the transmission with analogue and digital modulation

The quality of transmission with analogue modulation is S/N (signal to noise ratio).

signal power

S/N = -----

baseband noise power

S/N can be greater than C/N. For instance for FM (frequency modulation) TV transmission: [S/N] can be up to 35 dB larger than [C/N].

The quality of transmission with digital modulation is BER (bit error rate).

number of improperly detected bits during time t

BER = -----

total number of bits detected during time t

e.g. BER =  $10^{-5}$   $\rightarrow$  on average 1 bit in 100,000 is wrong.

S/N and BER depend on C/N and on the modulation technique.



#### Analogue modulation

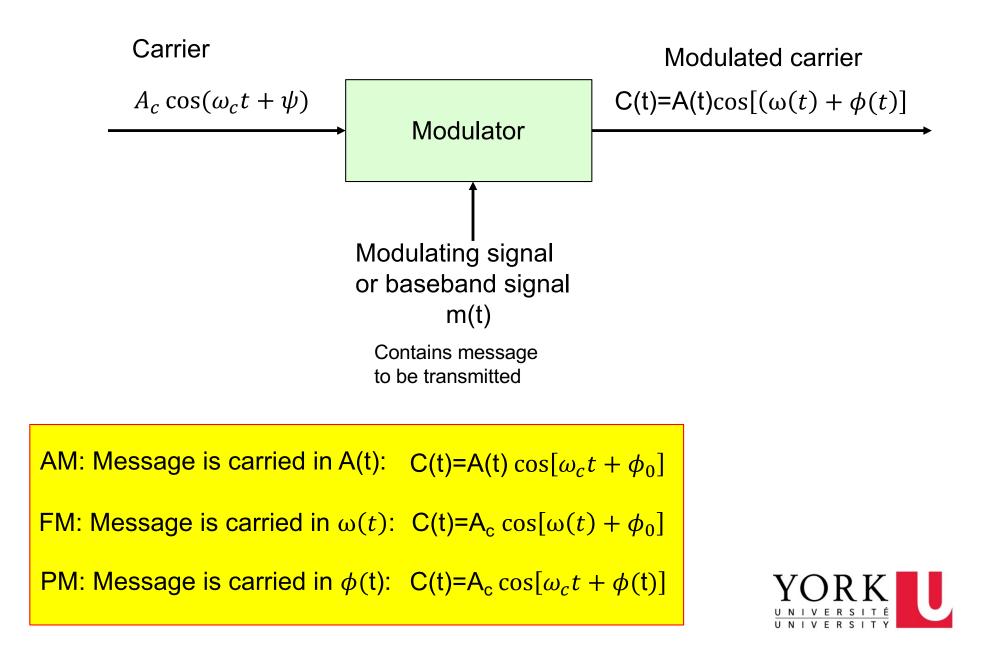
- Amplitude modulation (AM)
- Frequency modulation (FM)
- Phase modulation (PM)

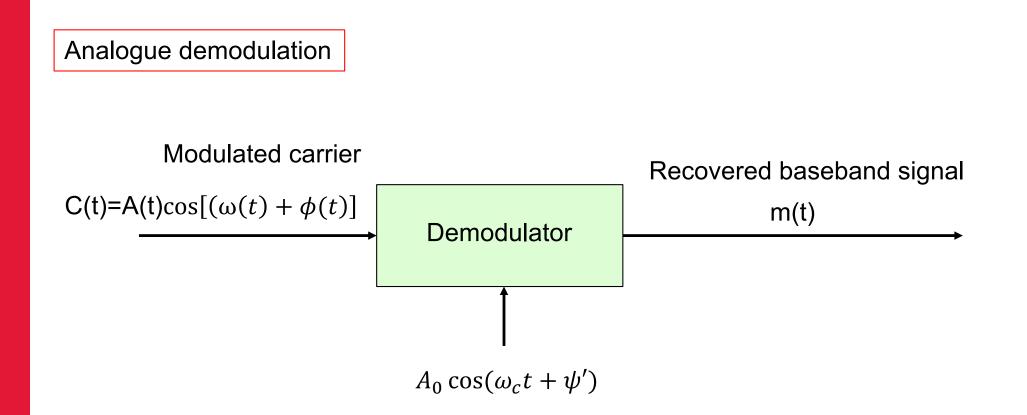
#### **Digital modulation**

- Amplitude shift keying (ASK)
- Frequency shift keying (FSK)
- Phase shift keying (PSK)
  - o Binary PSK (BPSK)
  - Quadrature PSK (QPSK)



#### 6.2. Analogue modulation





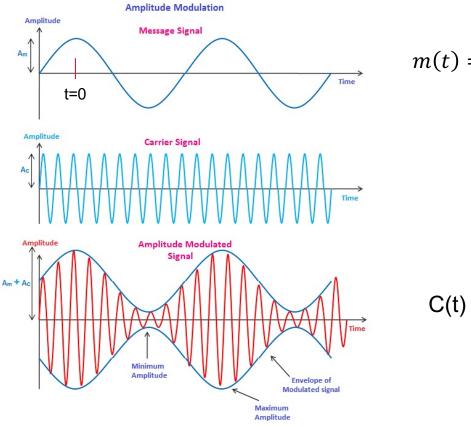
6.2.1 Amplitude modulation (AM)

$$\begin{split} \mathsf{C}(\mathsf{t}) &= \mathsf{A}(\mathsf{t}) \cos \left( \omega_{\mathrm{c}} \mathsf{t} + \phi_{0} \right) & \omega_{\mathrm{c}} \text{: carrier angular frequency} \\ &= \mathsf{A}_{\mathrm{c}} \left[ 1 + \frac{A_{m}}{A_{c}} \mathsf{m}(\mathsf{t}) \right] \cos \left( \omega_{\mathrm{c}} \mathsf{t} + \phi_{0} \right) \end{split}$$

 $\frac{A_m}{A_c}: 0 \le \frac{A_m}{A_c} \le 1 \quad : \text{ amplitude modulation index}$  $m(t): \ |m(t)| \le 1 \quad : \text{ baseband signal}$ 

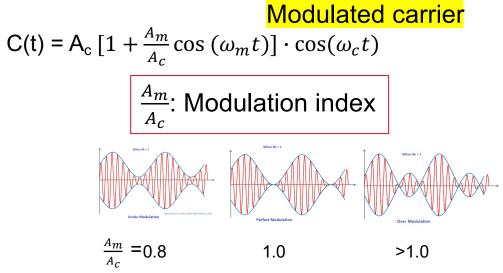


Graphical example for a special case:  $m(t) = \cos \omega_m t$ 



 $m(t) = A_m \cos(\omega_m t)$  Baseband signal

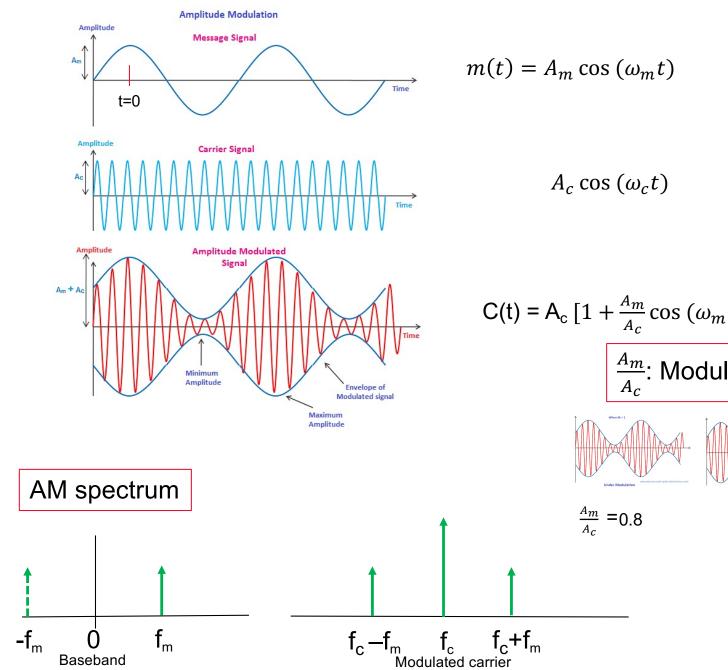
 $A_c \cos(\omega_c t)$  Carrier

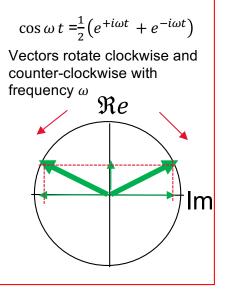


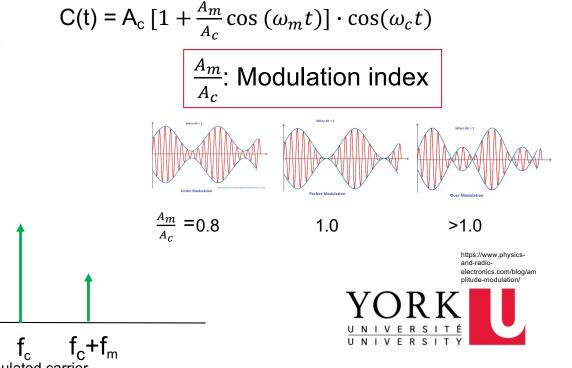
https://www.physics-and-radio-electronics.com/blog/amplitude-modulation/



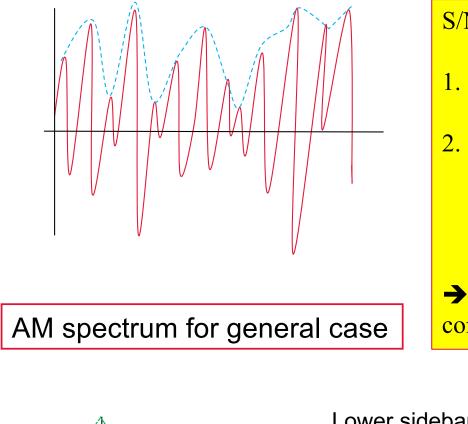
#### Graphical example for a special case: $m(t) = \cos \omega_m t$

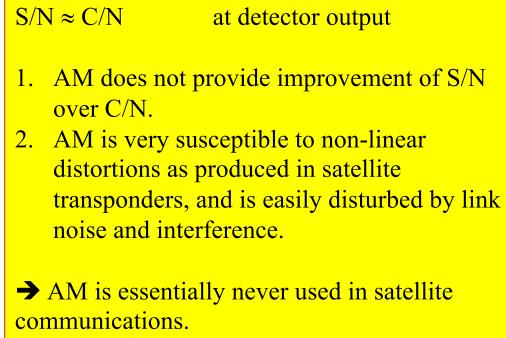


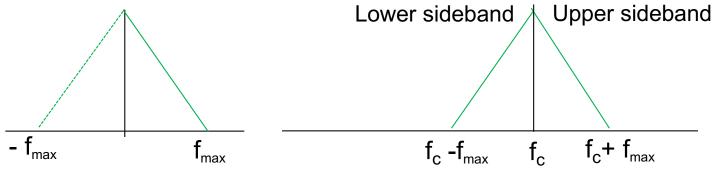




#### General case: baseband signal m(t) consists of cosine waves with a continuous range of frequencies.









Triangular baseband spectrum

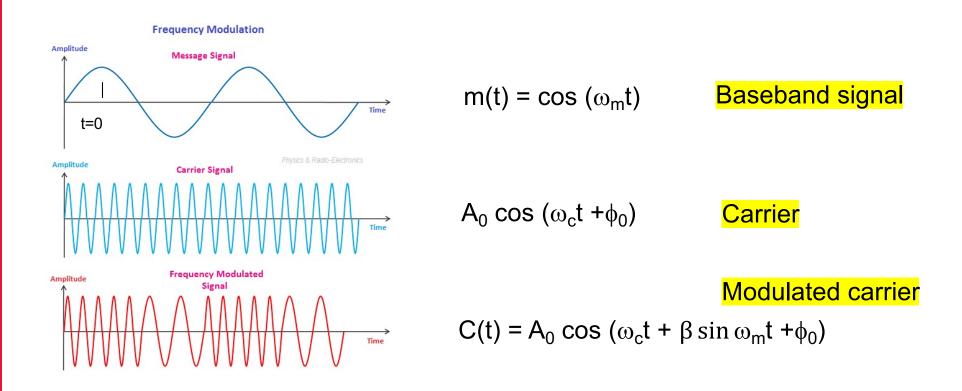
### 6.2.2 Frequency modulation (FM)

### Modulated carrier

$$\begin{split} C(t) &= A_0 \cos (\omega(t) t + \phi_0) & \omega_c: \text{ carrier angular frequency} \\ &= A_0 \cos \left( \int_0^t (\omega_c + k_\omega m(t)) dt + \phi_0 \right) & \omega_i(t) = \omega_c + k_\omega m(t): \text{ instantaneous frequency,} \\ &= A_0 \cos (\omega_c t + k_\omega \int_0^t m(t) dt + \phi_0) & k_\omega: \text{ frequency modulator=constant} \\ & \underbrace{\text{Example 6-1}}_{\text{Integration of frequency over time gives the phase}} & \text{If } \omega_i(t) = \text{constant, then } C(t) = A_0 \cos \left( \int_0^t \omega_i dt + \phi_0 \right) = A_0 \cos (\omega_i t + \phi_0) \\ & \text{Integration of frequency over time gives the phase} & \text{If } \omega_i dt + \phi_0 \\ & \text{Integration of frequency over time gives the phase} & \text{If } \omega_i dt + \phi_0 \\ & \text{Integration of frequency over time gives the phase} & \text{If } \omega_i dt + \phi_0 \\ & \text{Integration of frequency over time gives the phase} & \text{If } \omega_i dt + \phi_0 \\ & \text{Integration of frequency over time gives the phase} & \text{Integration of frequency over time gives the phase} & \text{Integration of frequency over time gives the phase} \\ \end{array}$$

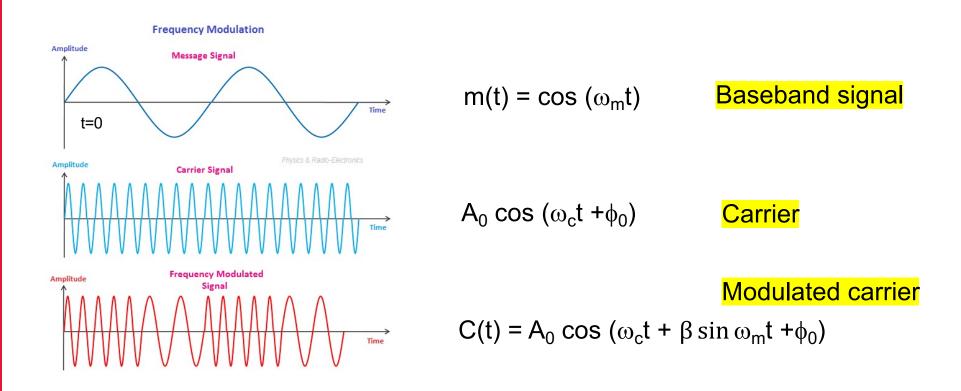
Graphical example for a special case:  $m(t) = \cos \omega_m t$ 

 $\omega_{i} = \omega_{c} + k_{\omega} \cos (\omega_{m} t)$   $C(t) = A_{0} \cos (\omega_{c} t + \beta \sin \omega_{m} t + \phi_{0})$   $\beta = \frac{k_{\omega}}{\omega_{m}} = \frac{\Delta \omega}{\omega_{m}} = \frac{\Delta f}{f_{m}} : \text{Modulation index}$   $\omega_{m} = 2\pi f_{m} : \text{Modulation frequency}$   $\Delta \omega = 2\pi \Delta f : \text{Max. frequency deviation, the frequency}$ of the modulated carrier varies between  $\omega = \omega_{c} - \Delta \omega \text{ and } \omega_{c} + \Delta \omega$ YORK



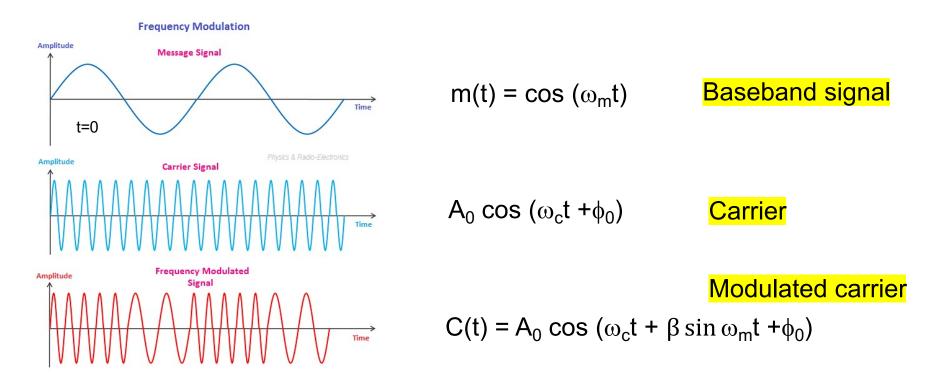
How does the spectrum look? What are the frequency components of cos  $[(\omega_c + \beta \sin \omega_m)t + \phi_0]$ ?



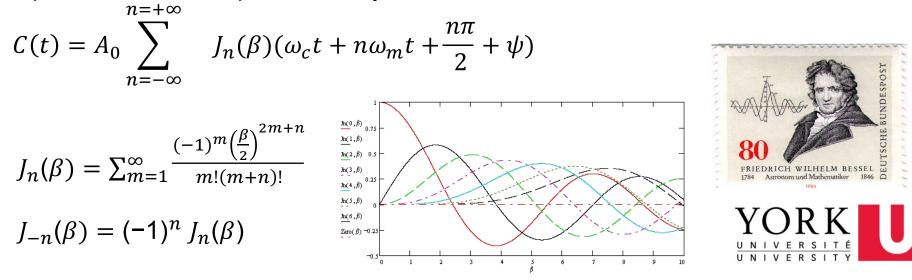


How does the spectrum look? What are the frequency components of cos  $(\omega_c t + \beta \sin \omega_m t + \phi_0)$ ?

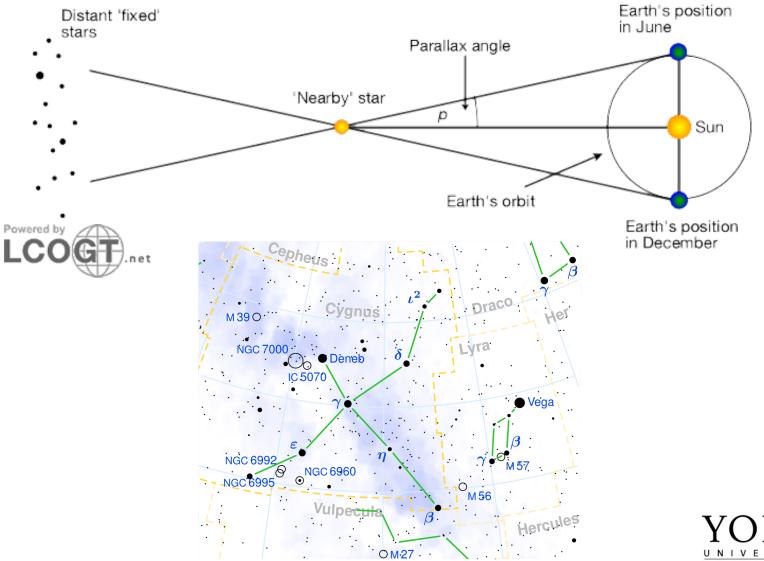




Spectrum can be represented by Bessel functions

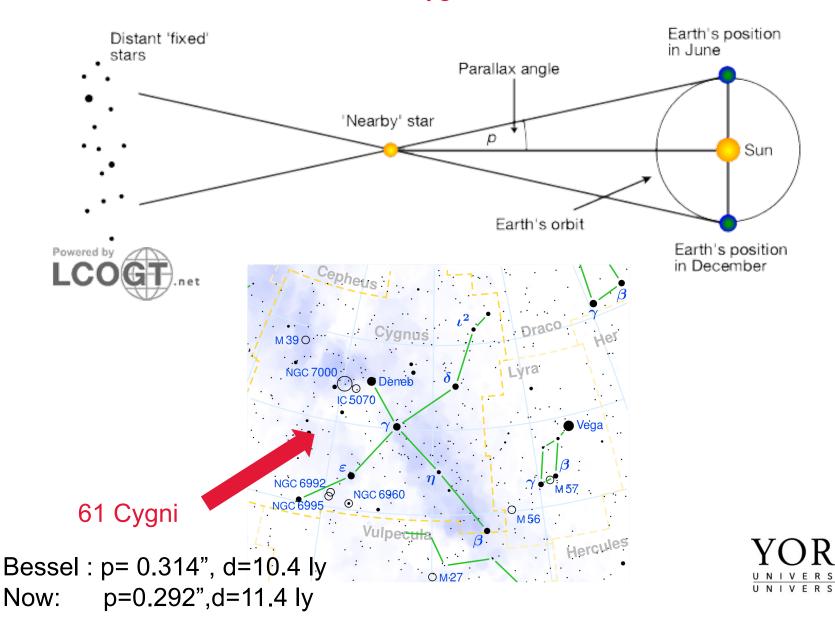


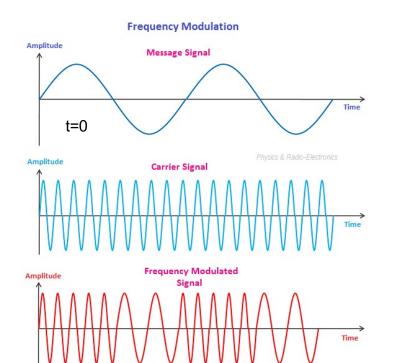
#### Bessel determined positions and proper motions of 50,000 stars. He was the first to determine the parallax (distance) of a star 61 Cygni

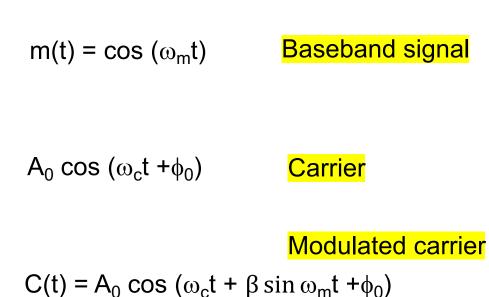


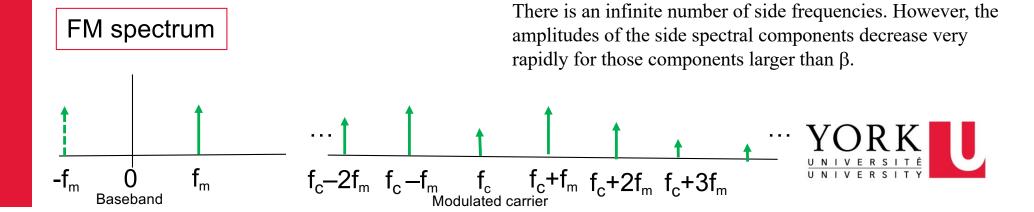


#### Bessel determined positions and proper motions of 50,000 stars. He was the first to determine the parallax (distance) of a star 61 Cygni

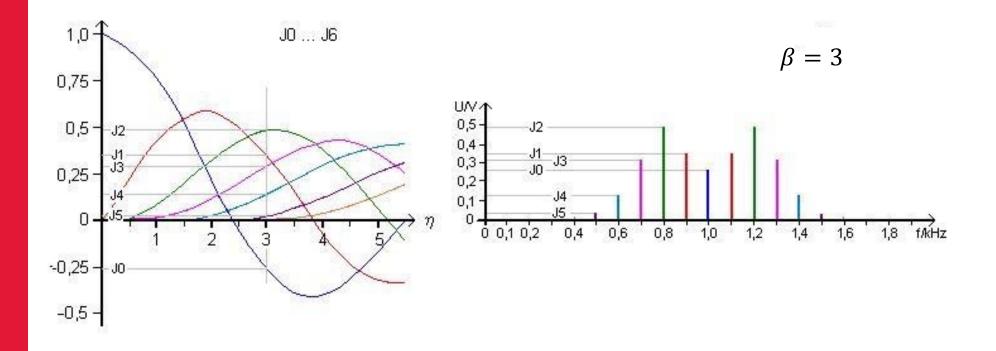






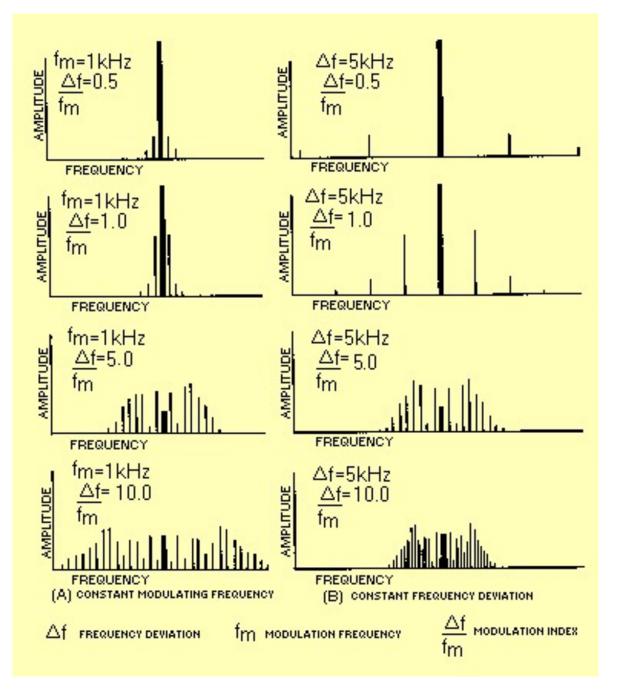


### How to determine the spectrum from the Bessel functions





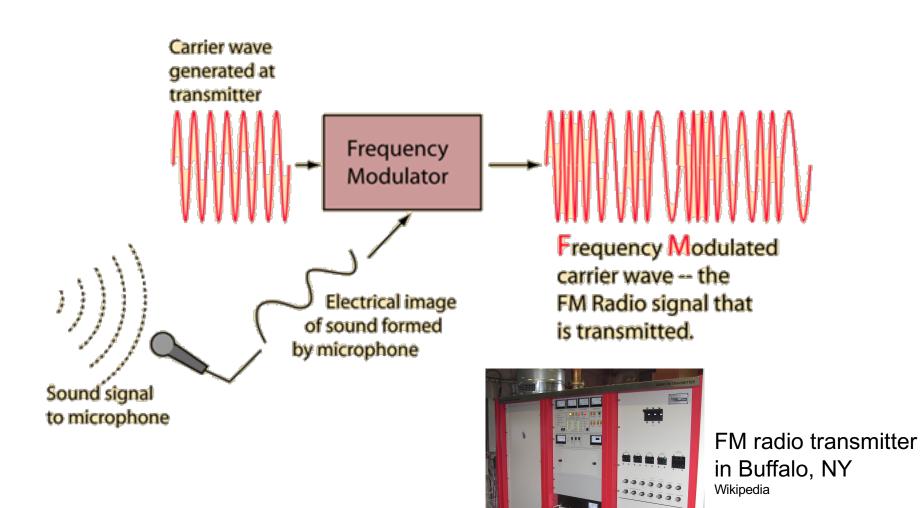
#### FM spectrum as a function of the modulation index





tpub.com

#### Sketch of FM with baseband signal





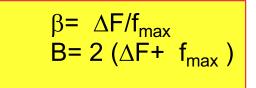
We do not need an infinitely wide filter bandwidth. Most of the information of a FM signal is in fact passed through a filter with a finite bandwidth, B, that depends on the modulation index  $\beta$ .

The bandwidth, B, is essentially the range in frequencies from  $f_c - (\beta + 1) f_m$  to  $f_c + (\beta + 1) f_m$ .

Carson's rule:	
$\begin{array}{l} B=2 \ (\beta+1) \ f_m \\ B=2 \ (\Delta f+f_m \ ) \end{array}$	

If m is not just a cosine wave but a baseband signal with a continuous range of frequencies, then we plot the spectrum as a generic spectrum as shown below,

then:



f<sub>max</sub>

 $\beta = \Delta F/f_{max}$ : modulation index for highest frequency in baseband spectrum. It is also sometimes called the deviation ratio, D.

Amplitude

 $\Delta F$  is similar to  $\Delta f$  that is it is the frequency range over which the carrier frequency is modulated.



# Example 6-1

A 100 MHz carrier is frequency modulated by a 1 kHz tone which produces frequency deviations of up to 75 kHz.

$$\beta = \Delta f/f_m = 75/1 = 75$$
  
B= 2 (\Delta f+ f\_m)  
= 2 (75 + 1)  
= 152 kHz

## Example 6-2

A video signal of bandwidth 4.2 MHz is used to frequency- modulate a carrier with  $\Delta F = 10.75$  MHz.

$$D = \beta = \Delta F/f_{max}$$
  
= 10.75/4.2 = 2.56  
B= 2 ( $\Delta F$ + f<sub>max</sub>)  
= 2 (10.75 +4.2)  
= 29.9 MHz



#### Signal-to-noise ratio

For FM, the S/N is not equal to C/N.

There are three factors that cause an improvement of S/N over C/N.

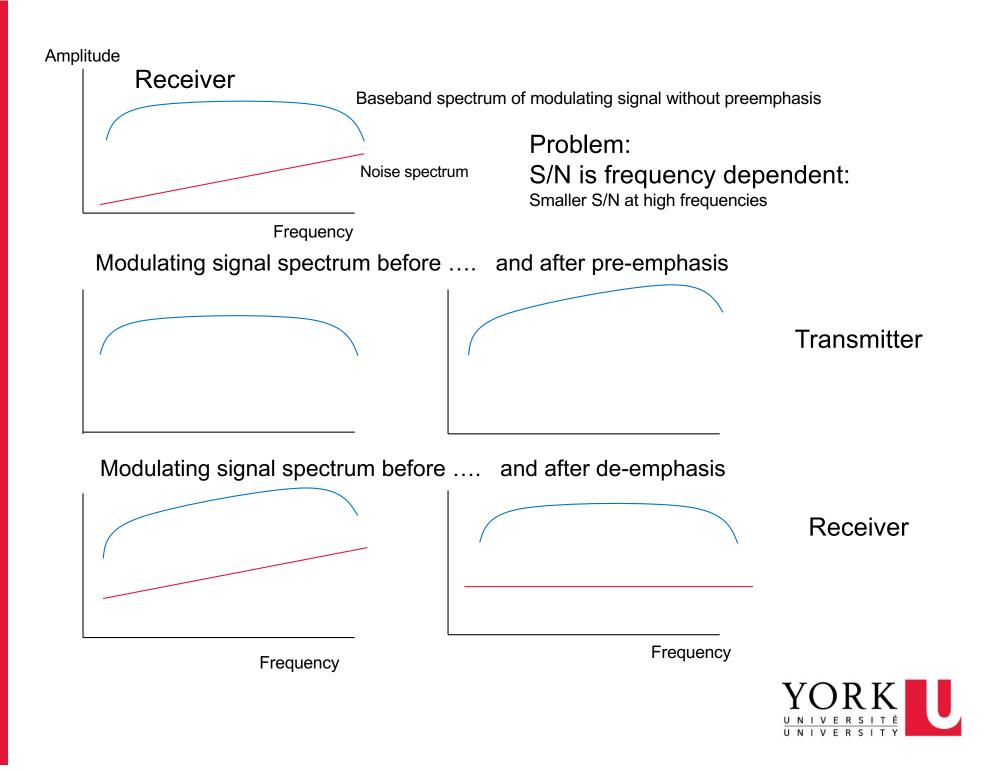
1. The processing gain  $K_R$  of the detector as a function of  $\beta$ .  $S/N = K_R \bullet C/N$  $K_R = 3 \ (\beta + 1) \ \beta^2$  (This is valid only if  $[C/N] \ge 10 \ dB$ )

# Example 6-3

For a video signal with  $\beta$  =2.56  $\rightarrow$  K<sub>R</sub> = 70 or 18.5 dB

2. *Pre-emphasis and de-emphasis*: If the noise at the output of an FM demodulator where conversion back to baseband takes place, increases with increasing frequency, the signal spectrum can be adjusted to the noise spectrum (pre-emphasis) at the transmitter and then re-adjusted at the receiver (de-emphasis). The result is a S/N independent of frequency.





Pre-emphasis and de-emphasis can increase the S/N by an emphasis improvement factor, P with

 $[P] \le 4 \ dB$  for telephony  $[P] \le 13 \ dB$  for TV

3. *Noise weighting:* By modifying the noise spectrum and adjusting it to the response of the output device and/or the response to the human ear, S/N can be further improved by [W]:

 $[W] \le 2.5 \ dB$  for telephony  $[W] \le 12 \ dB$  for TV

The S/N ratio is then given by:

 $S/N = C/N \cdot K_R \cdot P \cdot W$ 

 $[S/N] = [C/N] + [3 (\beta + 1)\beta^2] + [P] + [W]$ 



## Example 6-4

Example for S/N of a typical FDM system:

$$\begin{array}{ll} [C/N] &= 25 \text{ dB} \\ [P] &= 4 \text{ dB} \\ [W] &= 2.5 \text{ dB} \\ \Delta F &= 281 \text{ kHz} \\ f_{\text{max}} &= 108 \text{ kHz} \end{array} \qquad \begin{array}{l} \beta = \Delta F / f_{\text{max}} = 2.60 \\ 3 (\beta + 1) \beta^2 = 73.0 \end{array}$$

$$[S/N] = [C/N] + [3 (\beta + 1) \beta^{2}] + [P] + [W]$$
  
= 25 + 18.63 + 4 + 2.5  
= 50.13 dB  
S/N = 103,000

#### Peripheral information only

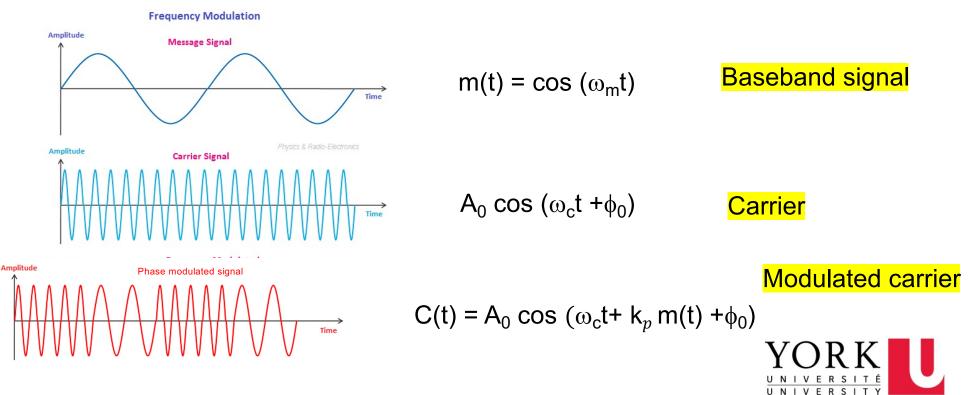
For worst channel (WC) the S/N is computed this way:  $\Delta F_{rms} = 35 \text{ kHz.} \quad (rms \text{ of frequency deviations})$   $b = 3.1 \text{ kHz.} \quad (bandwidth \text{ of individual channel})$   $B_{IF} = 778 \text{ kHz.} \quad (intermediate frequency bandwidth from Carson's rule, B_{IF} = B= 2 (\Delta F+ f_{max}))$   $[S/N]_{WC} = [C/N] + [(B_{IF}/b) (\Delta F_{rms} / f_{max})^2] + [P] + [W]$  = 25 + 14.21 + 4 + 2.5 = 45.71 dB



#### 6.2.3 Phase modulation (PM)

#### Modulated carrier

Graphical example for a special case:  $m(t) = \cos \omega_m t$ 



The spectrum is very similar to that of FM

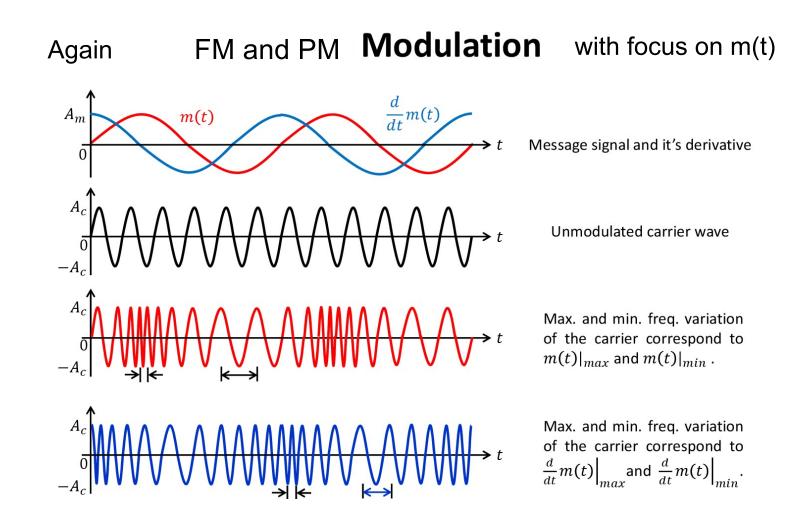
### FM and PM comparison

	Frequency modulation	Phase modulation
Phase	$\theta(t) = \omega_{c}t + k_{\omega} \int_{0}^{t} m(\tau)d\tau$	$\theta(t) = \omega_c t + k_p m(t)$
Frequency	$\omega_{i}(t) = \omega_{c} + k_{\omega} m(t)$	$\omega_{i}(t) = \omega_{c} + k_{p} \frac{d}{dt} m(t)$

 $\omega_{i}(t) = \frac{d}{dt} \theta(t)$  : Instantaneous frequency is the derivative of phase

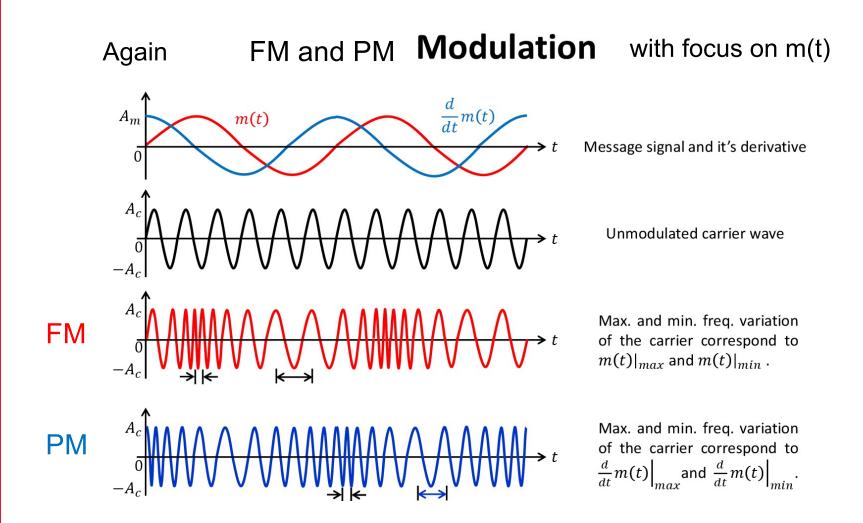
Frequency modulation:m(t) drives the frequency variationPhase modulation:m(t) drives the phase variation







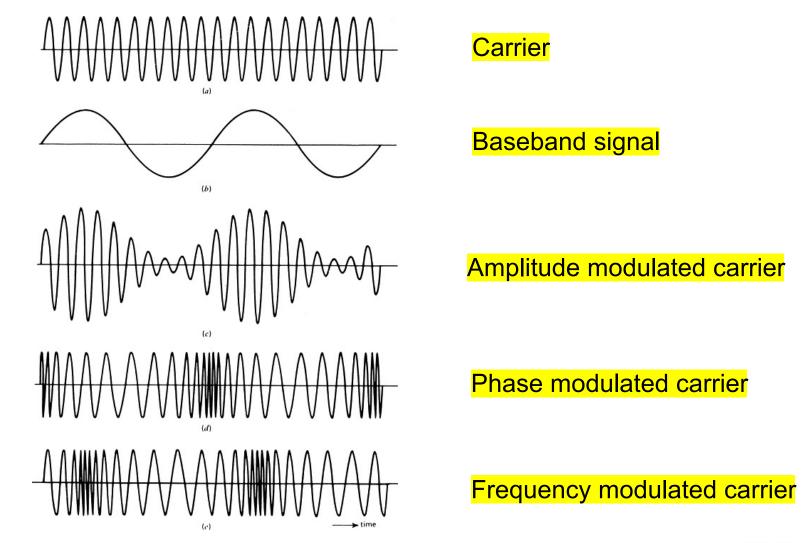
What is FM and what is PM?





What is FM and what is PM?

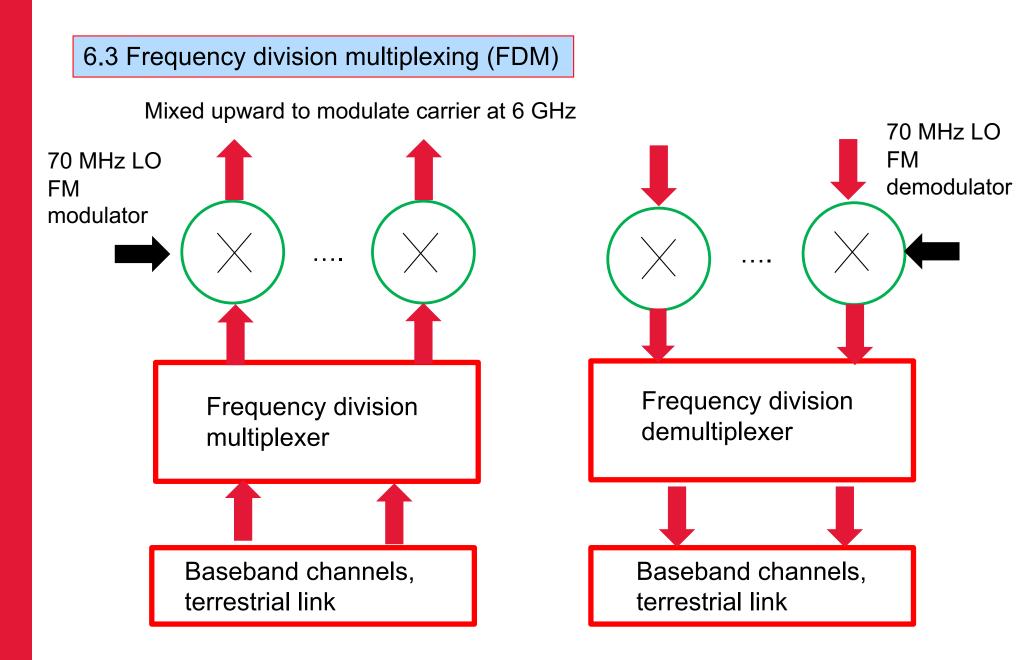
#### Graphical differences between AM, PM and FM carriers



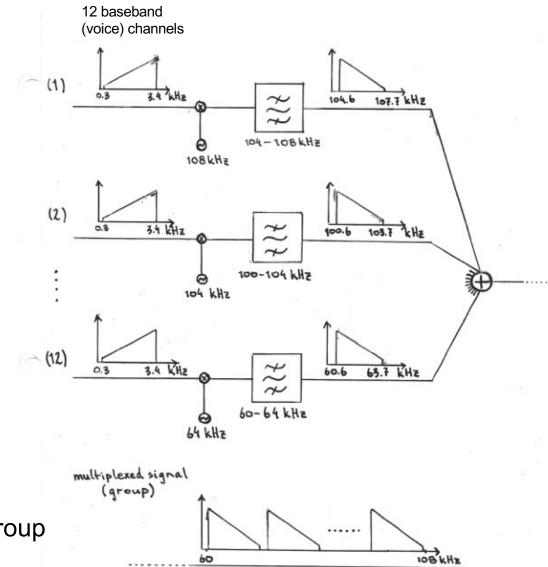


	FM	AM
1	S/N >C/N	S/N~C/N
2	<ul><li>S/N can be improved through:</li><li>1. processing gain</li><li>2. Pre-emphasis and de-emphasis</li><li>3. Noise weighting</li></ul>	No improvements possible
3	Bandwidth is in theory $\infty$ , in practise depends on modulation index and is given through Carson's rule	Bandwidth is given by 2 x $f_c$ - $f_m$ and is independent of modulation index
4	FM transmitters and receivers are relatively complex	AM transmitters and receivers are relatively simple
5	FM is power efficient	AM is not that power efficient







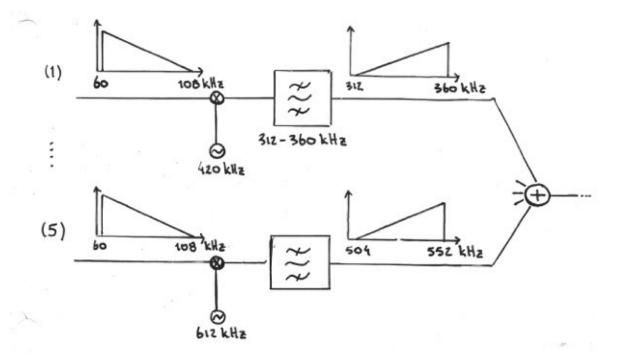


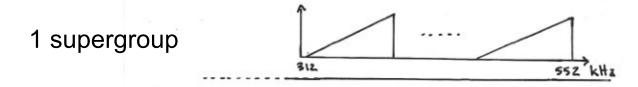
Spectra are separated by 4kHz to allow for guardband for filtering purposes so that distortions from neighbouring channels are limited.



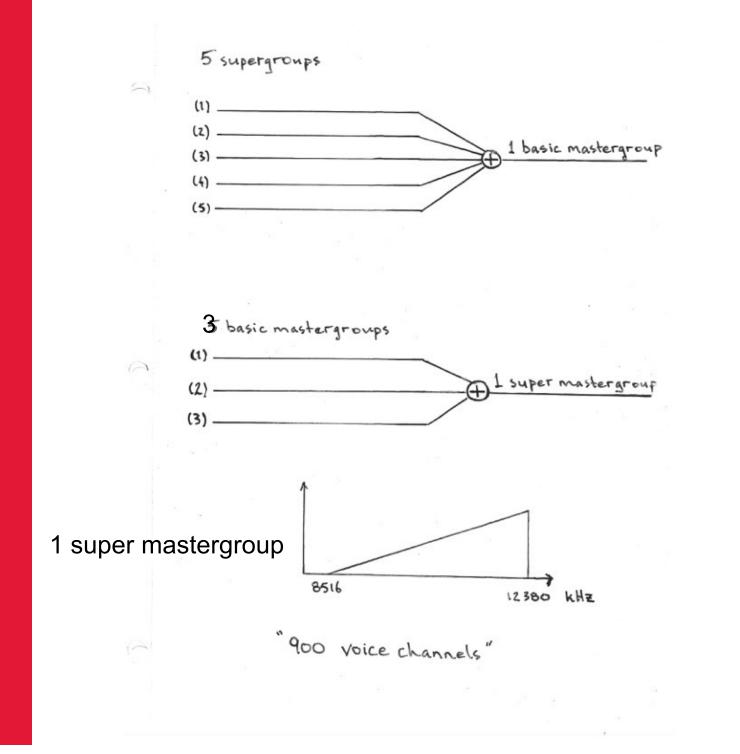
1 group

### 5 groups











### Hirarchical structure of FDM according to CCITT

(CCITT Comité Consultatif Internationale de Télégraphique et Téléphonique (International Telegraph and Telephone Consultative Committee) is an agency of the International Telecommunications Union (ITU). The ITU is an agency of the UN. The CCITT is an agency coordinating telephone and data communications systems on a worldwide basis and dealing with regulatory matters and with technical standards.)

12 (voice) cannels  $\rightarrow$  1 group

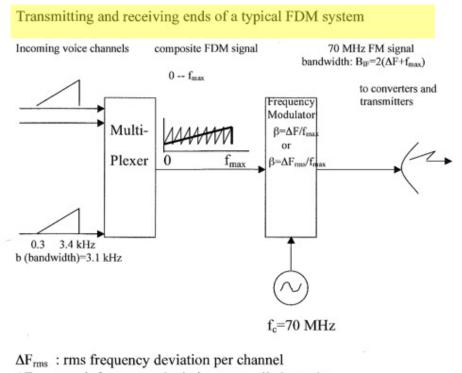
\_\_\_\_\_

- 5 groups
- 3 mastergroups
- → 1 supergroup 5 supergroups  $\rightarrow$  1 basic mastergroup
  - → 1 super mastergroup

The super mastergroup is the highest baseband unit.

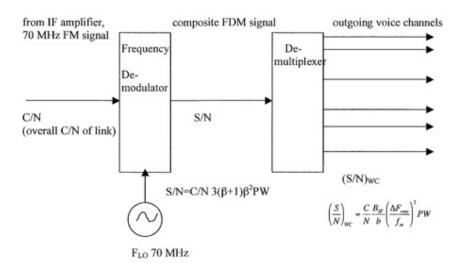
900 voice channels are frequency division multiplexed. However, lots of other schemes are in use.





ΔF : peak frequency deviation w.r.t. all channels

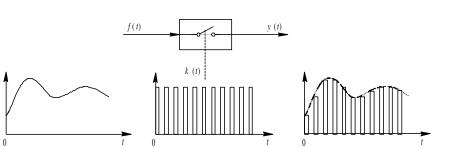
fm : center frequency of individual channel





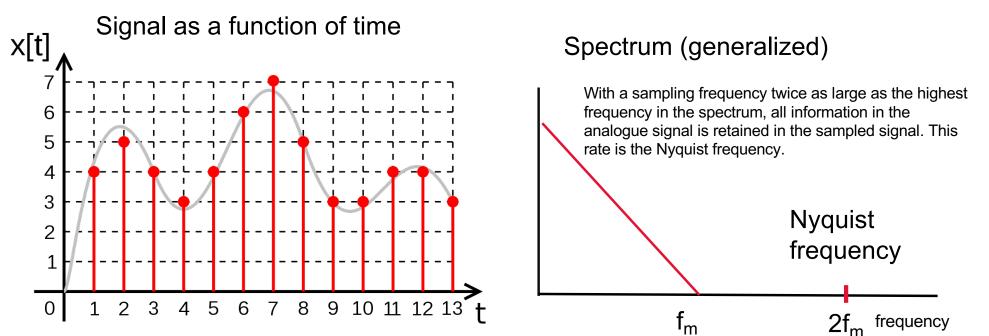
## 6.4 Digital baseband signal

A/D conversion: analogue signal converted into a digital signal



Sampler

A digital signal represents data as a sequence of discrete values.



# Example 6-3

Decimal# 0,1,2,3,4,5,6,7,8,9,10 Binary# 000,001,010,011,100,101,110,111,1000,1001 1010



The *digital* information is transmitted as a waveform. The most fundamental such waveform is a string of 1's and 0's

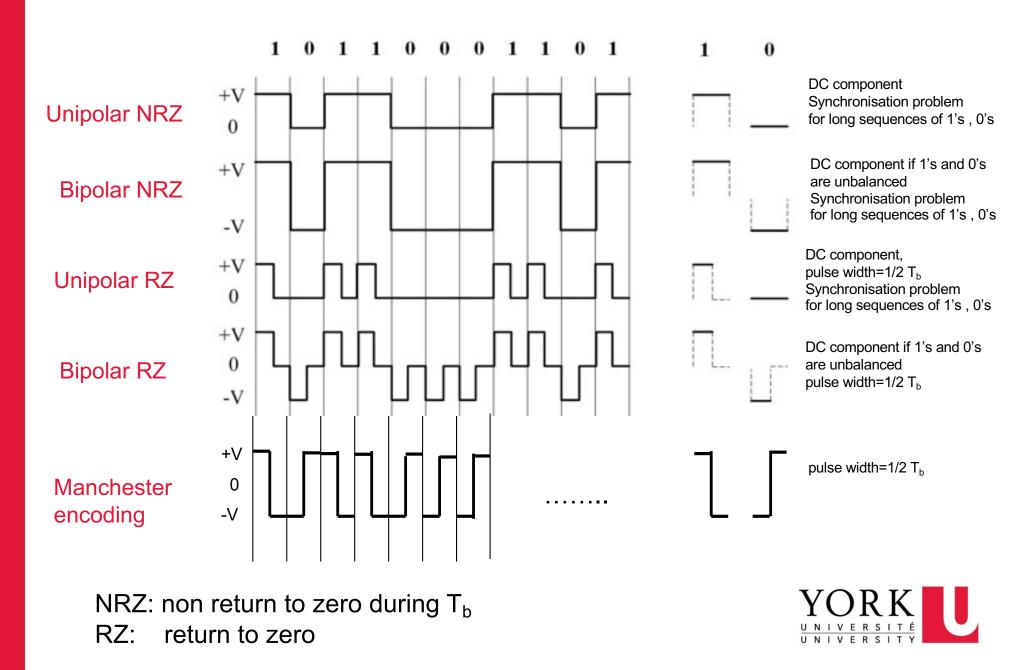
A unit of information is a *bit*. The duration of a bit is the *bit period*,  $T_b$  given in seconds or s. The inverse of the bit period is the bit rate,  $R_b$  given in bits per second of b/s.

$$R_b = \frac{1}{T_b}$$

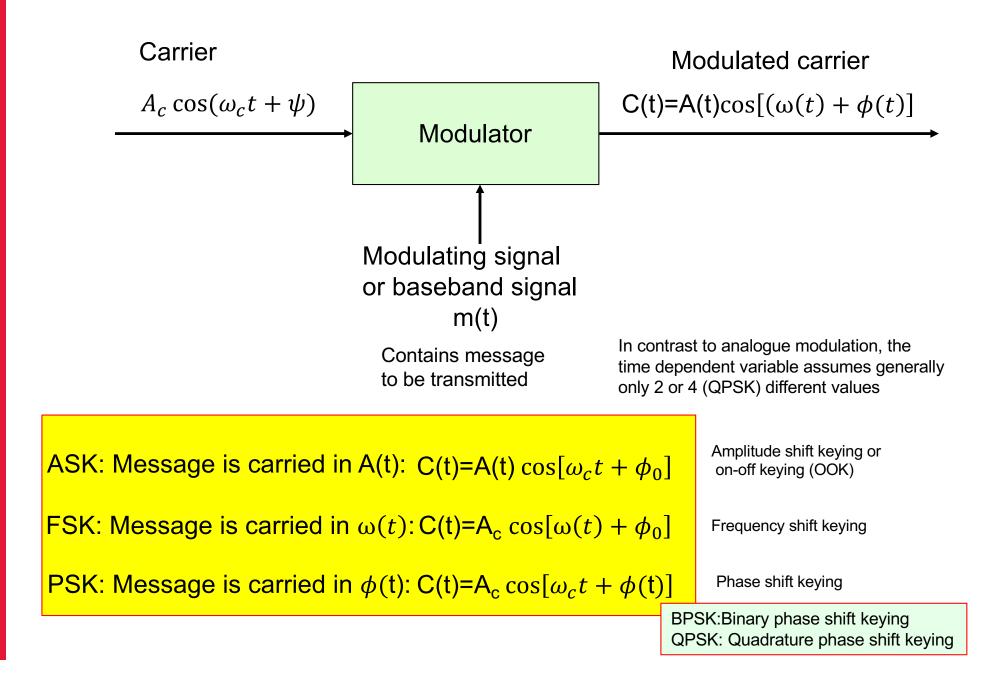


## Five different forms of coding

#### Disadvantages



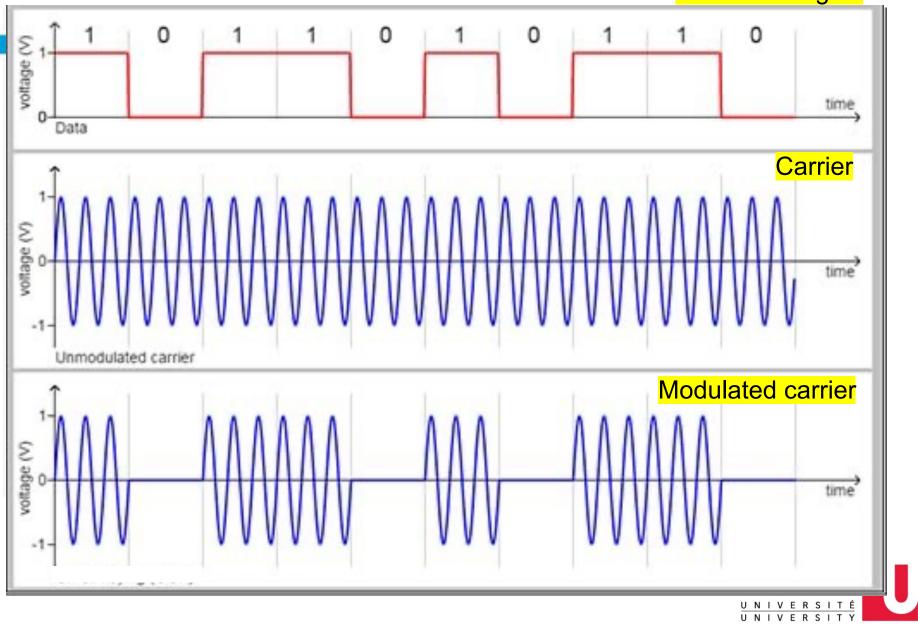
#### 6.5 Digital modulation

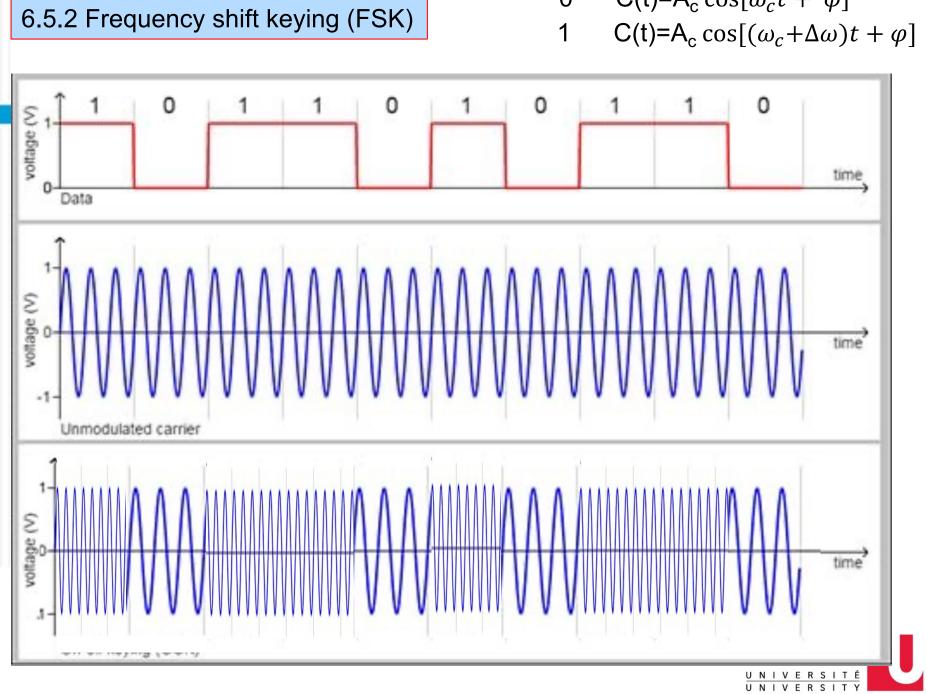


## 6.5.1 Amplitude shift keying (AM) or on-off keying (OOK)

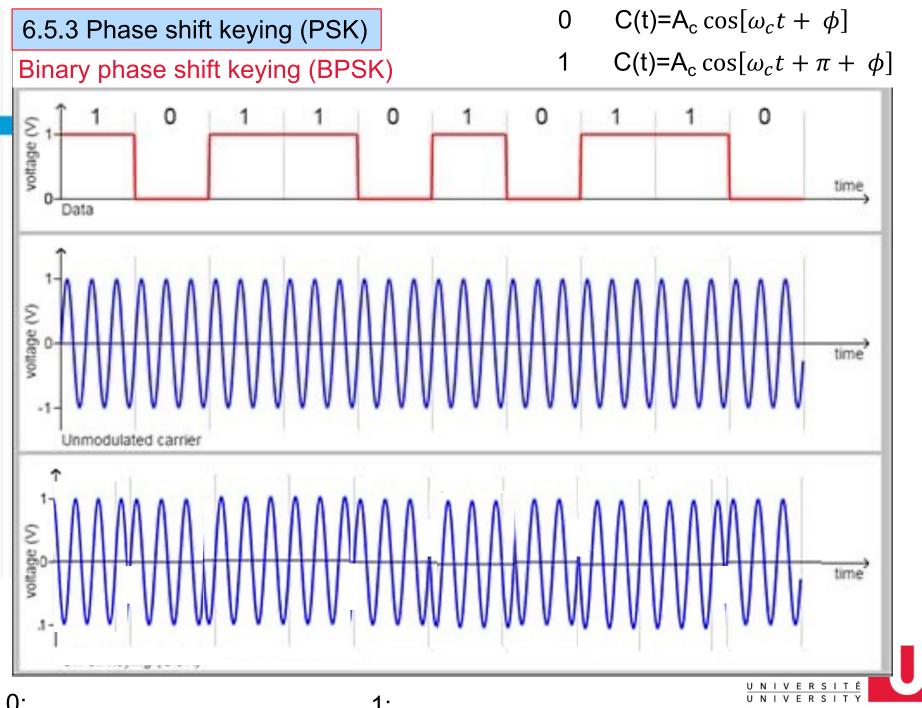
0 C(t)= 0 1 C(t)= $A_c \cos[\omega_c t + \phi]$ 

**Baseband signal** 



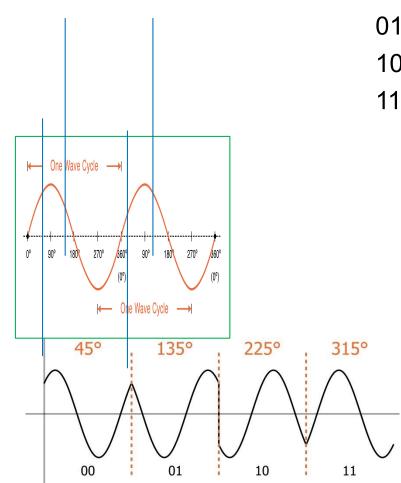


 $C(t)=A_c \cos[\omega_c t + \phi]$ 0



## Quadrature phase shift keying (QPSK)

Four phase shifts are used for 00, 01, 10 and 11



$$\begin{array}{ll} 00 & C(t) = A_c \cos[\omega_c t + 0 + \phi] \\ 01 & C(t) = A_c \cos\left[\omega_c t + \frac{\pi}{2} + \phi\right] \\ 0 & C(t) = A_c \cos[\omega_c t + \pi + \phi] \\ 1 & C(t) = A_c \cos\left[\omega_c t + \frac{3\pi}{2} + \phi\right] \end{array} \qquad \phi = + \frac{\pi}{4}$$

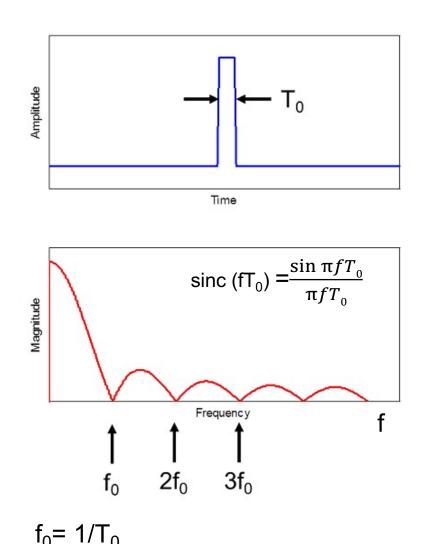
Advantage of QPSK (or 4-PSK):

- Higher data rate than with BPSK (doubled) while bandwidth remains about the same.
- 2) 4-PSK can be extended to n-PSK as far as equipment is able to distinguish between increasingly smaller phase differences.



## 6.5.4 Spectra

All digital modulation relies on the basic characteristic of a square pulse

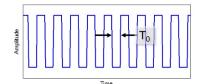


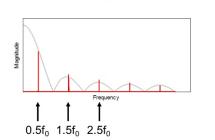
Single square pulse of width, T<sub>0</sub>

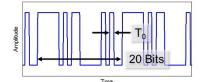
The Fourier transform of that pulse is a sinc function : sinc (x) =  $\frac{\sin x}{x}$ 

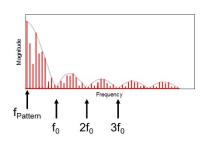
The magnitude of that function is given as the spectrum on the left. It is a continuum spectrum consisting of an infinite number of spectral components.

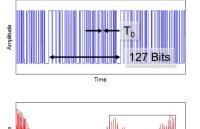


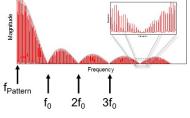












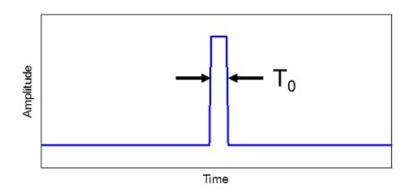
In infinitely long string of pulses with a period  $2T_0$ has a spectrum with components at f=  $n \cdot 1/2T_0$ The envelope of the frequency components is again a sinc function. Note, that in this example for n=2,4,6... The component amplitudes are zero

For a string of pulses with different durations but all equal to an integer number times  $T_0$ , the area under the envelope fills up. In this case the lowest frequency component is at f=1/20T<sub>0</sub>

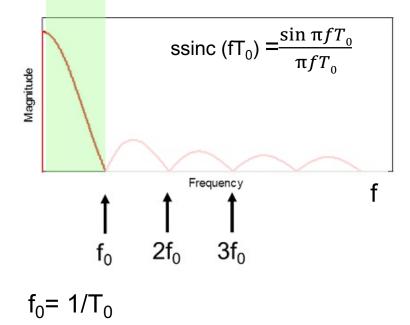
The longer the data pattern, the denser the components under the sinc function.



#### Back to the single square pulse



Rectangular lowpass filter



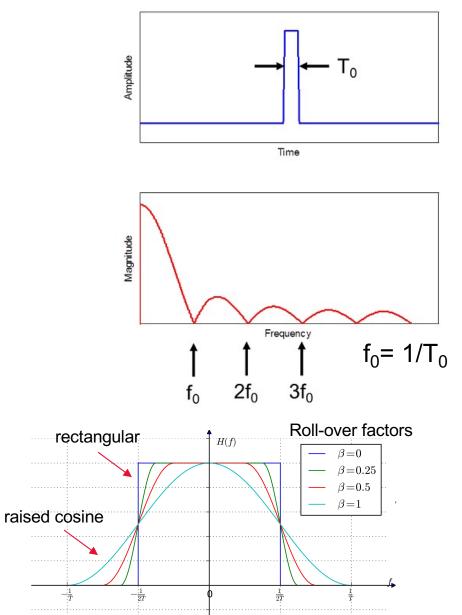
## Single square pulse of width, T<sub>0</sub>

The Fourier transform of that pulse is a sinc function : sinc (x) =  $\frac{\sin x}{x}$ 

The magnitude of that function is given as the spectrum on the left. It is a continuum spectrum consisting of an infinite number of spectral components. If we pass the pulse through a low-pass filter then the higher frequency would be absent in the out put spectrum. The output spectrum would only reproduce a smooth version of the square pulse with sidelobes.



## Raised cosine filter

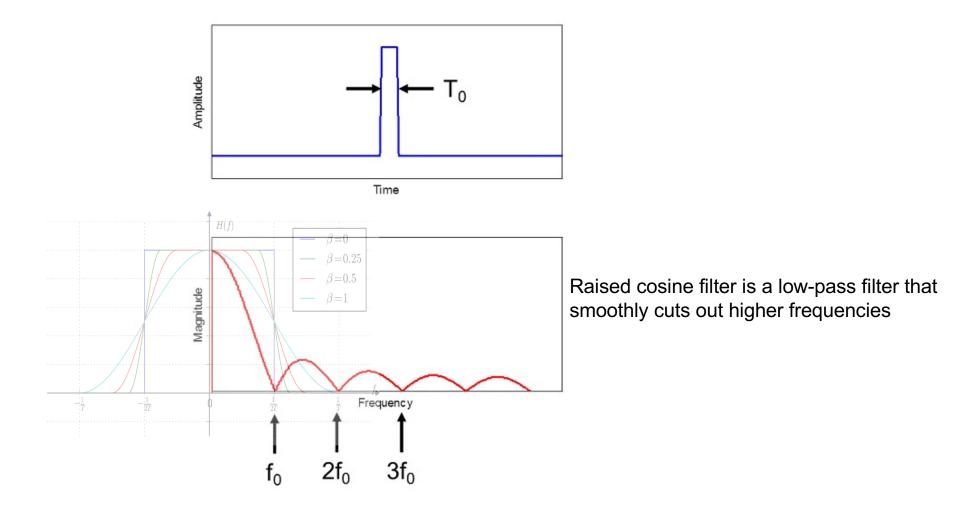


Instead of lowpass filters with a rectangular bandpass, in digital modulation raised cosine filters are frequently Used. Different roll-over factors characterize the raised cosine filters

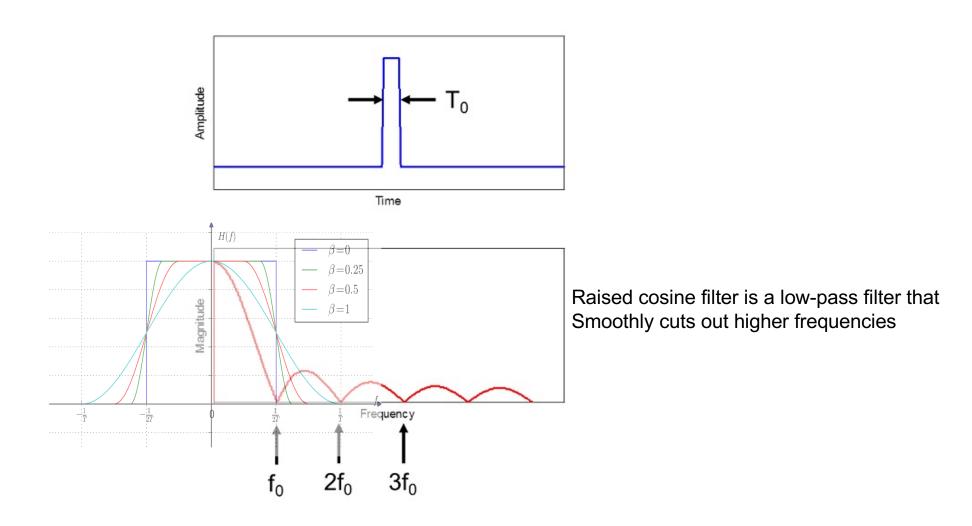


Frequency response of raised cosine filters with different roll-over factors

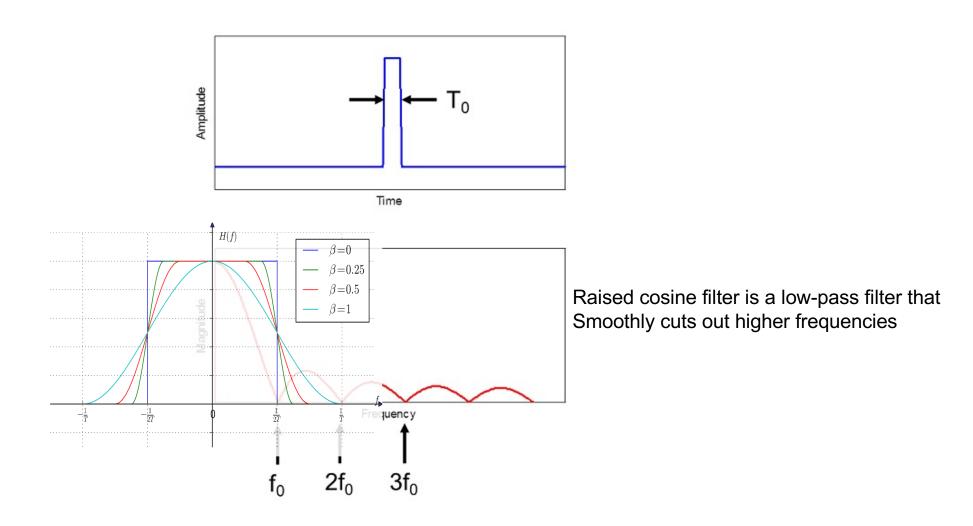
## Raised cosine filter



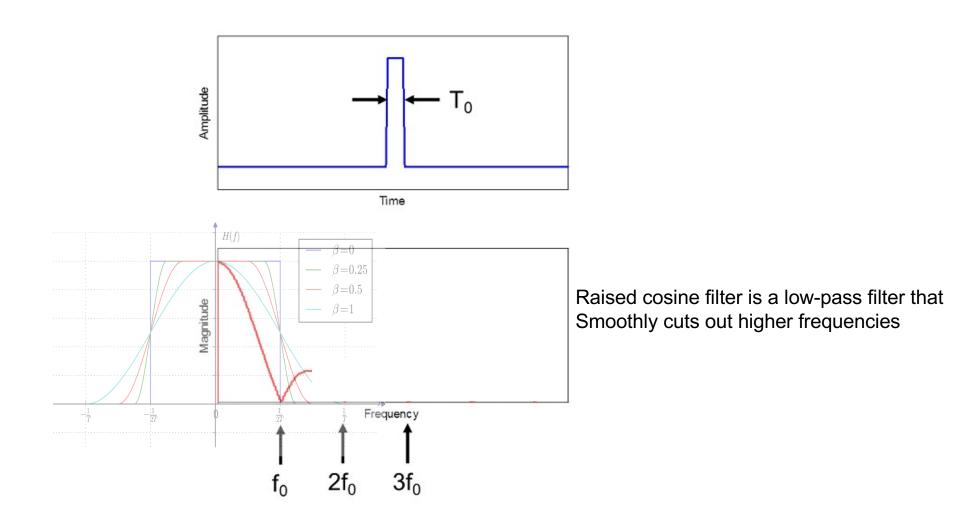






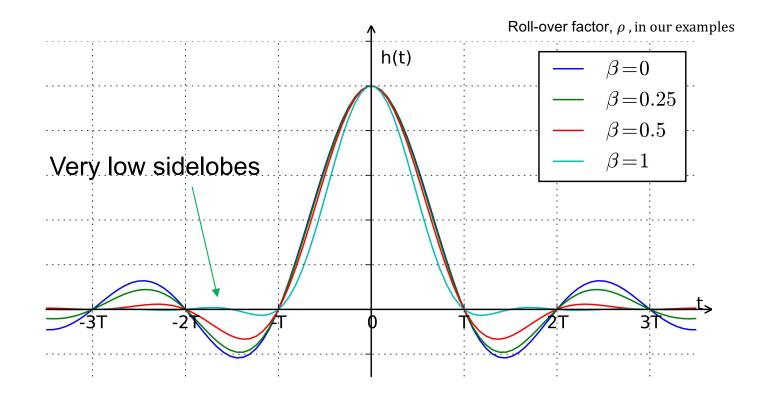




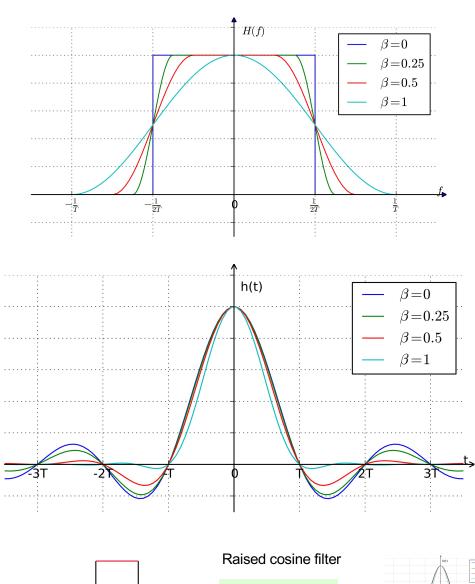




The output is a rounded pulse like a sinc function with lower sidelobes depending on the roll-over factor



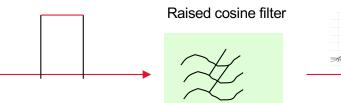


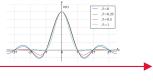


$$H(f) = egin{cases} 1, & |f| \leq rac{1-eta}{2T} \ rac{1}{2} \left[1+\cos\!\left(rac{\pi T}{eta}\left[|f|-rac{1-eta}{2T}
ight]
ight)
ight], & rac{1-eta}{2T} < |f| \leq rac{1+eta}{2T} \ 0, & ext{otherwise} \end{cases}$$

With a rectangular lowpass filter, the rectangular pulse would be converted to a pulse with a sinc function shape. It has ripples or sidelobes in the time Domain. These sidelobes could interfere with neighbouring pulses.

Sidelobes in time cause intersymbol interference (ISI). With appropriate roll-over factors, the sidelobes and thus ISI can be Minimized.







For BPSK:  $B = (1 + \rho) 1/(2T_b)$  $= \frac{1}{2} (1 + \rho) \bullet R_b$ 

 $\rho$ : roll over factor, defines raised cosine filter characteristics,  $0 \le \rho \le 1$ 

 $B_{IF} = 2B$  $= (1 + \rho) \bullet R_{b}$ 

Example 6-5

$$R_b = 100 \text{ kbps}, \rho = 1$$
$$\implies B_{IF} = 200 \text{ kHz}$$

For QPSK the information rate is doubled:

 $B = \frac{1}{2} (1 + \rho) \frac{1}{2T_b}$ =  $\frac{1}{4} (1 + \rho) \bullet R_b$ 

 $= \frac{1}{4} (1 + \rho) \bullet R_b \qquad \rho: \text{ roll over factor, defines raised cosine filter} \\ \text{ characteristics, } 0 \le \rho \le 1$ 

 $\mathbf{B}_{\mathrm{IF}} = \frac{1}{2} \left(1 + \rho\right) \bullet \mathbf{R}_{\mathrm{b}}$ 

Example 6-6

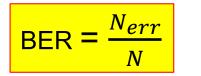
$$R_b = 100 \text{ kbps}, \rho = 1$$
  

$$\Rightarrow B_{IF} = 100 \text{ kHz}$$



## 6.5.4 Bit error rate (BER)

BER characterizes the quality of digital transmission

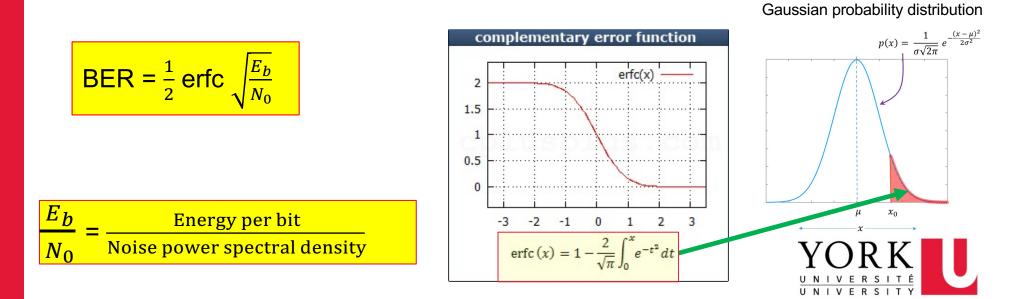


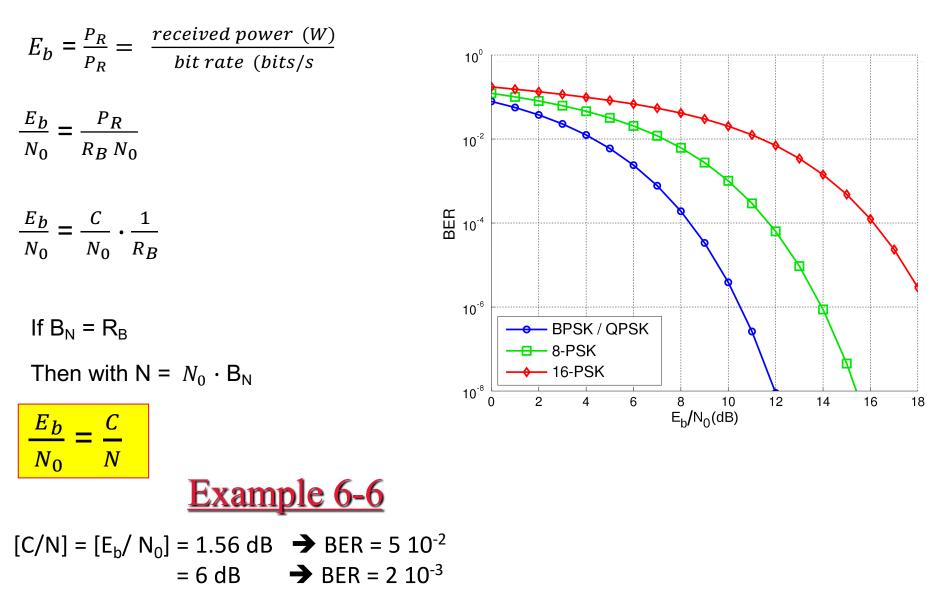
N<sub>err</sub>: Number of bits received in errorN: Total number of bits transmitted

# Example 6-5

100101110100Transmitted bit sequence100101100101Received bit sequence

$$N_{err} = 2$$
  
N = 12 BER = 0.167





Improvement of BER is possible through coding. By encoding extra bits (redundant bits) it is possible to detect and correct certain errors in the decoding process. However, transmission rate is affected.

For  $[E_b / N_0] \ge 5 dB$ , coding gain can be achieved. Example:  $[E_b / N_0] = 6 dB \rightarrow BER \le 10^{-7}$  can be achieved!



#### 6.6 Time division multiplexing (TDM)

#### Transmitting and receiving ends of a typical TDM system

The most common analogue multiplexing technique is <u>frequency division</u> <u>multiplexing (FDM).</u>

The most common digital multiplexing technique is <u>time division multiplexing</u> (TDM).

