23-9-2019

introduction to communication

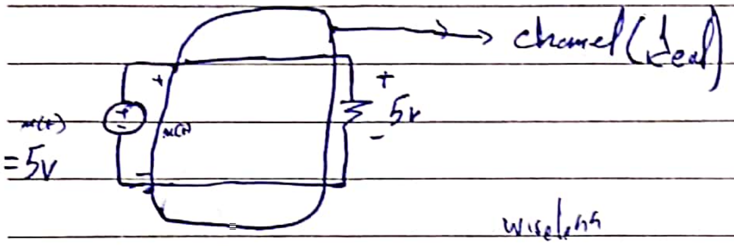
hw 1 : W. home work
hw 2 : Signals
hw 3 : Solve Problems

Purpose of Comm system
- carry info from A to B

Typical com system consists of 3 things:
- Source
- channel
- Destination

Source → (channel → how info travels) → Destination
↳ Generation of info ; ↳ makes use of info

baseband ← m(t)
ex Message signal

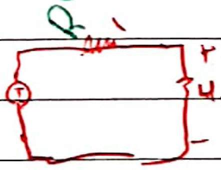


channels examples: copper wire, Co axil cable, optical fibre, wave guide → wired
wireless → most important

more cost, more mobility, more expensive

- channels are not ideal in life
  - they have problems, Limitations, Distortions, noise, attenuation

real life



→ m'(t) → not m(t)

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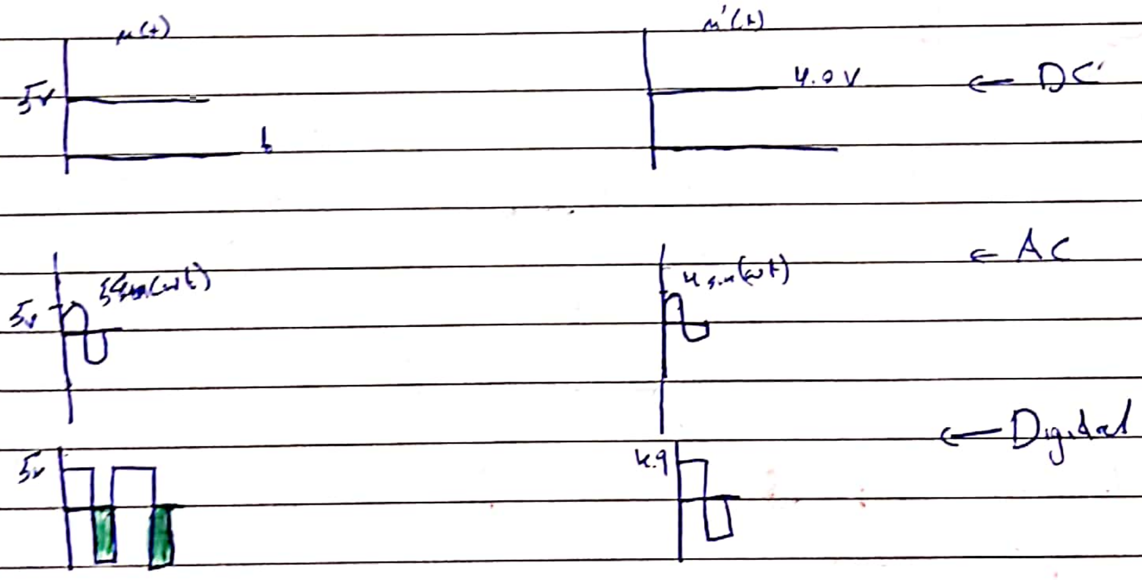
$m'(t) = m(t) \dfrac{R}{R+R'}$ → is not 0 → Voltage divider

* attenuation is Power loss

attenuation happend
→ Loss

Attenuation Facts

1. attenuation exists for all signal types
2. " " for all channel types
3. but attenuation level is different for different channel types
4. attenuation increases as the chanel length increases
   $L\uparrow \to R\uparrow \Rightarrow$ more Loss



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for sinusoidal

$f = \frac{1}{T}$

$\omega = 2\pi f$

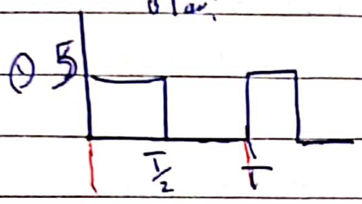
$DC = \text{average} = 0$

$P_{avg} = \text{Power average} = \frac{A^2}{2} = \frac{V_{rms}^2}{1} = \left(\frac{A}{\sqrt{2}}\right)^2$

here → signal's revision

P_{av}?

(1)

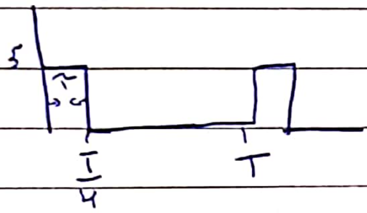


$P_{av} = \frac{A^2}{2} = 12.5$

→ more att↑
more loss↑

$P_{avg} = \frac{1}{T}\int_0^T x^2(t)\,dt$

→ less att↓
less loss↓

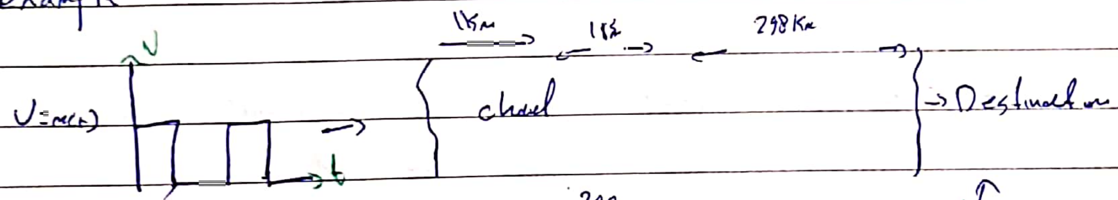


$P_{av} = \frac{A^2}{4} = 6.25$

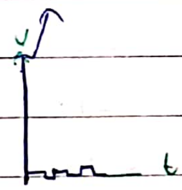
. adv of wireless

more mobility

example



after channel $\Rightarrow (0.9)^{300} \cdot m(t) = 1.874 \times 10^{-14} \times m(t) \approx 0$



signal attenuated

. because channel is not ideal

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Solution to attenuation

ⓐ use amplification → Devices that Does opposite of attenuation
↳ Power gain

$P_r = (0.9)^{300} m(t)$
$\neq 1$

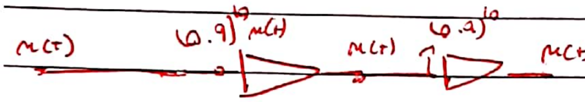
Pout = Pin × Gain

So $P_r = 1.8 \times 10^{-14} m(t)$

to get m(t) : divide on
↳ real life no amplifier can't do that
↳ Very expensive

• Doesn't work in Long Distance channels

↳ sol → split the channel and put multiple amplifiers



→ and so on

↳ this has Problems → no accessable Place to put amplifiers
• Like in the middle of ocean

ⓑ use less attenuation channels
Like optical Fibre
↳ drops the loss
↳ 0.9 → 0.99
↳ more expensive though

ⓒ Digital signals

(4)

Signals revision

1- Fourier series

the idea :- express any Periodic Signal $x(t)$ as the sum of an infinite number of (cosin, sine) Sinusoidal functions.

- There are three ways of doing that:
  1- complex form
  2- trigonometric form.
  3- compact form.

1- complex exponential Fourier series

$$x(t) = \sum_{n=-\infty}^{\infty} \alpha_n \, e^{(jn\omega_0 t)} \qquad \omega_0 = \frac{2\pi}{T_0},\ 2\pi f_0$$

where $\alpha_n = \frac{1}{T}\int_{t_1}^{t_1+T} x(t)\, e^{-(j\omega_0 n t)}\, dt.$

$n = \pm 1, \pm 2, \pm 3 \ldots$

* note: for real valued $x(T) \rightarrow$ $\alpha_n = \alpha_n^*$ conjugate

2- Trigonometric fourier Series

$$x(t) = \frac{a_0}{2} + \sum_{n=1}^{\infty} \left(a_n \cos(n\omega_0 t) + b_n \sin(n\omega_0 t)\right)$$

$\omega_0 = \frac{2\pi}{T_0},\ 2\pi f_0$

where • $a_n = 2\,\mathrm{Real}\{\alpha_n\} = \frac{2}{T}\int_{t_1}^{t_1+T} x(t)\cos(n\omega_0 t)\, dt,$

$n = 0, 1, 2, 3 \ldots$

• $b_n = -2\,\mathrm{Imaginary}\{\alpha_n\} = \frac{2}{T}\int x(t)\sin(n\omega_0 t)\, dt$

$n = 0, 1, 2, 3, \ldots$

* notice $\alpha_n = \frac{1}{2}(a_n - j b_n),$

$n = 0, 1, 2, 3 \ldots$

Signals revision

3. Compact Fourier Series:

$$x(t) = \frac{C_0}{2} + \sum_{n=1}^{\infty} C_n \cos(n\omega_0 t - \theta_n)$$

$\omega_0 = \frac{2\pi}{T_0} , 2\pi f_0$

- where: $C_n = \sqrt{a_n^2 + b_n^2} = 2|a_n|$

$n = 0, 1, 2, 3 \ldots$

$\theta_n = \tan^{-1}\left(\frac{b_n}{a_n}\right) = \tan^{-1}\left(\frac{-Im\{a_n\}}{Re\{a_n\}}\right) = -\angle a_n$

$n = 0, 1, 2, 3 \ldots$

- Notice that: $a_n = C_n \cos(\theta_n) = 2Re\{a_n\}$

$n = 0, 1, 2, 3 \ldots$

$b_n = C_n \sin(\theta_n) = -2Im\{a_n\}$

$n = 0, 1, 2, 3 \ldots$

Cards] I - memorize and practice them on different signals.

notes: - the complex Fourier series coefficients a_n's represent the Fourier spectrum of that signal $x(t)$.

- since these coefficients are complex numbers, they generate two spectra:
  1- the magnitude spectrum of $x(t)$
  2- the phase spectrum of $x(t)$

- the magnitude spectrum can also be drawn using the compact coefficients C_n since $C_n = 2|a_n|$

- the C_n spectrum is called the one sided magnitude spectrum (because $n > 0$)

Signals revision

- while the a_n spectrum is called the two-sided magnitude spectrum (because $-\infty < n < \infty$)

2- Fourier Transform

- the idea: it's a mathematical tool that converts a signal $x(t)$ from the time domain into the frequency domain as $X(\omega)$.

  - the inverse of F.T does the same in opposite manner

- the F.T of a general signal $x(t)$, wether periodic or aperiodic is given by:

$$X(\omega) = \mathcal{F}\{x(t)\} = \int_{-\infty}^{\infty} x(t)\, e^{-j\omega t}\, dt$$

- and the inverse fourier transform is given by

$$x(t) = \mathcal{F}^{-1}\left[X(\omega)\right] = \frac{1}{2\pi}\int_{-\infty}^{\infty} X(\omega)\, e^{j\omega t}\, d\omega$$

notes

- the fourier transform $X(\omega)$ represents the fourier spectrum density of the signal $x(t)$
- notice that the fourier spectrum density consists of a group of impulses $\delta(\omega)$'s while aperiodic signals is a smooth curve
- this is due to the fact that periodic signals are actually the sum of an infinite number of sinusoidals

Signal's revision

3 Energy and Power spectral densities.

3.1 Energy Spectral Density (ESD)

- the idea: The ESD is a function that describes the relative amount of energy of a given signal relative to frequency

- the total area under the ESD is the total energy in the signal $x(t)$ defined as E_x

$$ESD = \Psi_x(\omega) = |X(\omega)|^2$$

3.2 Parseval's theorem:

it gives a way to calculate the total energy $x(t)$

$$E_x = \int_{-\infty}^{\infty} |x(t)|^2 \, dt \overset{or}{=} \frac{1}{2\pi} \int_{-\infty}^{\infty} |X(\omega)|^2 \, d\omega$$

3.3 Power spectral Density (PSD)

- the idea: the PSD is a function that describes the relative amount of Power of a given signal relative to frequency

- the total Area under the PSD is the total Power in the signal $x(t)$ defined as P_x

- notice that the PSD of Periodic signals consists of a group of impulses $\delta(t)$'s while the PSD of aperiodic signals is a smooth curve

Signals revision

- this due to the fact that Periodic functions are actually the sum of an infinite number of sinusoidal functions

- $PSD = S_x(w) = \lim_{T \to \infty} \frac{1}{T} |X_T(w)|^2$

- most of the signals we consider in communications theory exist for a long time, they are Power signals, some of them Periodic and some of them are aperiodic

- Power signals have a PSD not an ESD (their ESD is infinite)

3.4 Average Power (P_x)

- the P_x of a signal can be calculated using two different methods :

$$P_x = \lim_{T \to \infty} \frac{1}{T} \int_{-\frac{T}{2}}^{\frac{T}{2}} |x(t)|^2 dt = \frac{1}{2\pi} \int_{-\infty}^{\infty} S_x(\omega)\, d\omega$$

Signals revision

4. Average and RmS

. consider $x(T)$ is a continuous message signal and a discrete version of it x_n with a sampling period Δt and a sampling frequency $f_s = \frac{1}{\Delta t}$:

4.1 average of a message signal:

$$\text{Signal average} \leftarrow \bar{X} = \lim_{T \to \infty} \frac{1}{T} \int_{-T/2}^{T/2} x(t)\, dt$$

. and for the sampled version of $x(t)$, we have N samples Δt apart, the average is :

$$\text{Sampled signal average} \leftarrow \bar{X}_n = \frac{1}{N.\Delta t} \sum_n X_n . \Delta t$$

$$= \frac{1}{N} \sum_n X_n \qquad \text{where } \Delta t = \frac{1}{f_s}$$

4.2 RMS of a message signal:

$$\text{Signal rms} \leftarrow X_{rms} = \sqrt{\left(\lim_{T \to \infty} \frac{1}{T} \int_{-T/2}^{T/2} x(t)^2\, dt \right)}$$

. and for the sampled Discrete Version

$$\text{Sampled signal rms} \leftarrow X_{n\,rms} = \sqrt{\left(\frac{1}{N.\Delta t} \sum_n x_n^2 . \Delta t \right)}$$

$$= \sqrt{\left(\frac{1}{N} \sum_n X_n^2 \right)} \qquad \text{where } \Delta t = \frac{1}{f_s}$$

Signals revision

4.3 Average Power of a message signal:

- as in 3.4, the Average Power (P_x) of $x(t)$ is given by:

$$P_x = \lim_{T \to \infty} \frac{1}{T} \int_{-T/2}^{T/2} |x(t)|^2 \, dt$$

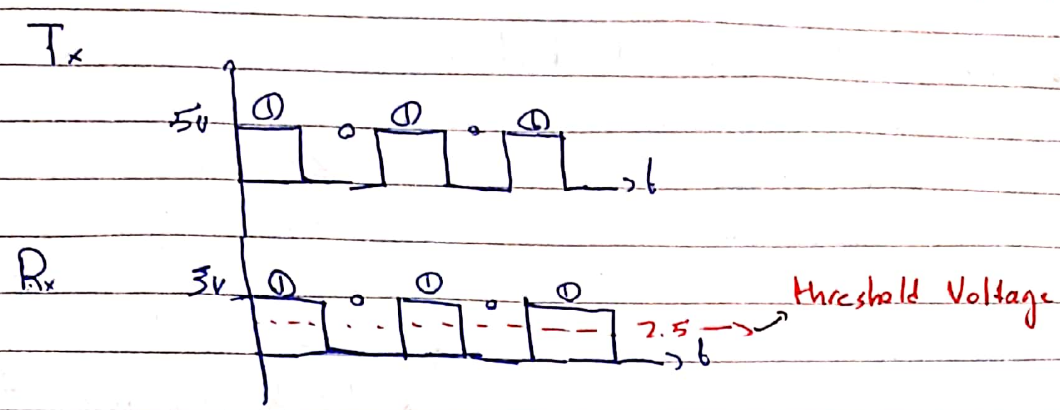
- and the average Power of the sampled version

$$P_{xn} = \frac{1}{N} \sum_n x_n^2$$

- Notice that the (RMS value)^2 is the P_x, this is because we assume a normalized load impedance of 1Ω, hence P_x of $x(t)$ is

$$P_x = X_{rms}^2$$

$$P_{xn} = X_{n\,rms}^2$$

③ 

- the reciever Doesn't care about signal str, if its higher than its threshold.

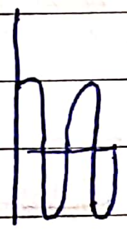
- if its lower than the threshold it gives 0,0,0 for ever.

Problem ②

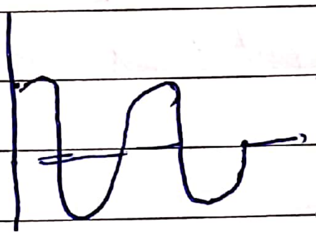
Linear Distortion :- channel attenuation changes to respect to Frequency.

- usually higher f is more attenuation.

- hence the channel acts as a Low Pass filter



$A\cos(f_2 2\pi t) = m(t) \rightarrow 0.8\,A\cos(2\pi f_2 t)$



$A\cos(f_1 2\pi t) \rightarrow 0.9\,A\cos(2\pi f_1 t)$

$f_2 > f_1$

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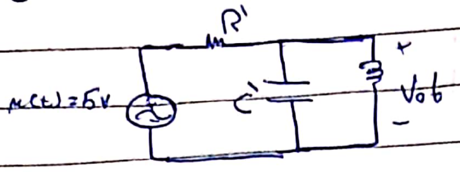
• Cause of Linear Distortions



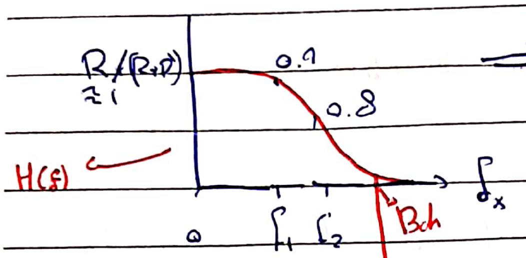
$\bar{Z}_R = R$

$\bar{Z}_C = \frac{1}{jwC'}$

$\bar{Z}_L = jwL'$

Very low freq ≈ DC $C \to Z_C \uparrow \to$ open circuit → Leads to low attenuation level

// high freq, $C \to Z_C \downarrow \to$ short circuit → Leads to higher attenuation levels



⟹ $H(f) = V_o(f)/V_i(f)$

→ transfer function, frequency Response

behavior of a LPF

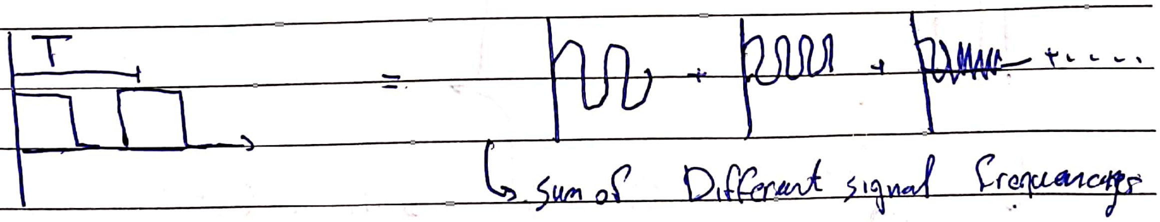
B channel

→ cut off frequency

→ Bandwidth of the channel

→ its like a body Gaurd.

→ signals that need to be sent are usually high f phase



→ sum of Different signal frequencies

$f = \frac{1}{T}$

→ fundamental frequency

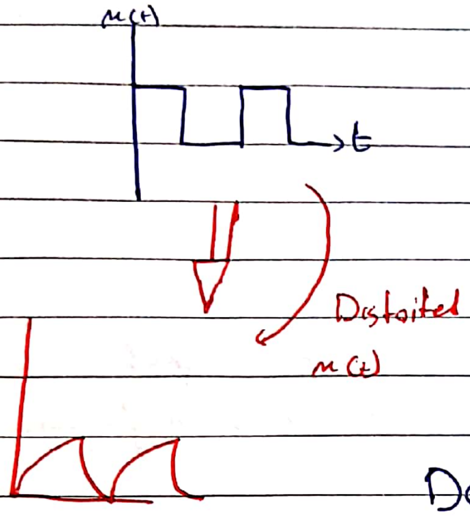
$x(t) = \sum c_n \cos(2\pi f_n t)$

→ Multiple frequencies

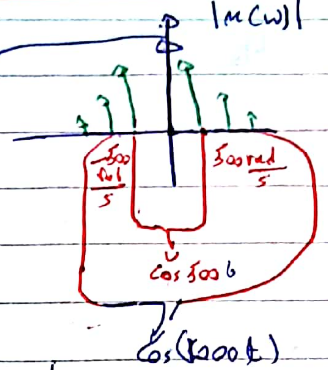
25 - 9 - 2019 ...mmunication I

Sinusoidals need 3 things
1- Amplitude } we can get them by mag spectrum density
2- frequency
3- phase → phase spectrum density

Fourier series spectrum
- Mag
- phase → Phase spec density



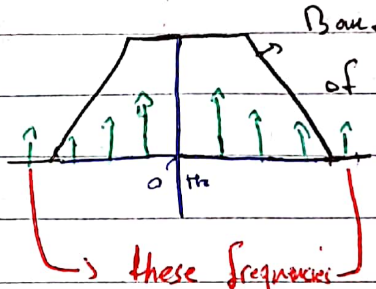
⟺



DC Value
$\int = 0$

→ Amplitude of these cos is the Area of those Arrows

fibre is Band Pass filter
copper is a LPF
→ depends on physical shape



→ these frequencies won't Pass in the LPF because of a C in wires

$B_{m(t)}$ → Band width of the message

Band width of the message
→ it's where we can neglect all frequencies because they don't matter

→ sol ⇒ Band width of the channel > Band width of the message

(7)

25-9-2019

- channel Bandwidth $B_{channel}$:
  - a property of the channel
  - get it from data sheet of the channel
  - the frequency where it starts to attenuate heavily.

$B \rightarrow Hz$
$2\pi B \rightarrow rad/sec$

- Signal Bandwidth $B_{m(t)} = B$
  - a property of the signal
  - you get it by F.T of the signal
  - the f above where $m(t)$ has negligible harmonics

Rule $B_{m(t)} \lll B_{channel}$

- numbers

1 km of copper channel has $B_{ch} \approx 1-2$ MHz
" " coaxial cable " " " $\approx$ 1-2 GHz
" " optical fibre " " " $\approx$ so many THz

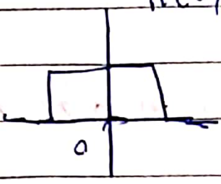
For copper wires we need amplifier every 5-10 km
" optical fibre " " " " 50-100

Sol ⓐ $\rightarrow$ send at smaller data rate $\rightarrow$ frequency compression
or better channel $\rightarrow$ more expensive
Less data speed
Worse connection

30 - 9 - 2019

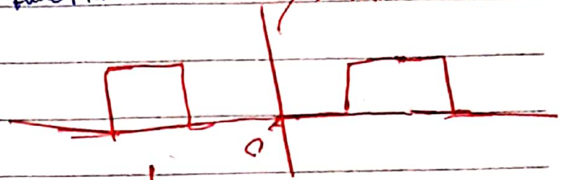
L3 -

$H(w)$ → Frequency response function



→ ideal LPF

$h(w)$ → fiber wires



↳ BPF (ideal

Sol 2- use equilizer

: its a device used at the reciever (destination)

• it performs the attenuation gain of the channel

• Very expensive → getting cheaper

↳ multiple amplifiers

↳ amplifies attenuated Signals (frequencies)

↳ comm 2 more info

Sol 3- Pre distortion at the transmitter. (amplifying before sending to neglect effect of channel)

a at the Tx

↳ opposite job of channel

↳ expensive

↳ amplifies before message sent.

* channel depends on channel type ①

// Length ②

Attenuation level $= \frac{P_o}{P_{in}} < 1$ = actual Math $\left| 10 \log_{10} \left(\frac{P_o}{P_{in}} \right) \right|$ dB

↳ always negative

Five Apple

30-9-2019

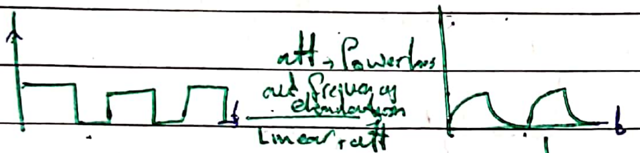
Cat 5 → Copper wire → Popular in ethernet

So att level = $\left| 10 \log_{10} \left(\frac{P_o}{P_{in}} \right) \right|$ → for data sheets

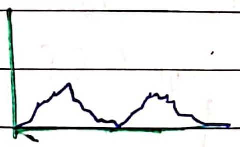
– So in exam paf = in front of att levels → if its attenuation

Problem 3 non-Linear distortions dependent

- attenuation changes with amplitude or Phase
- ~~at~~ usually higher amplitude gives higher attenuation



↳ Distorted signal



↳ non-Linear distortion

– reasons – non Linear Properties of elements such as diodes and Transistors

- equalizers and predistortion devices are expensive because they are complex

Problem 4 noise → undesired signals
↳ random by external and internal.

m(t) → [cable] → m(t) + n(t)

- Linear ↳ noised
- non Linear
- attenuation

(10)

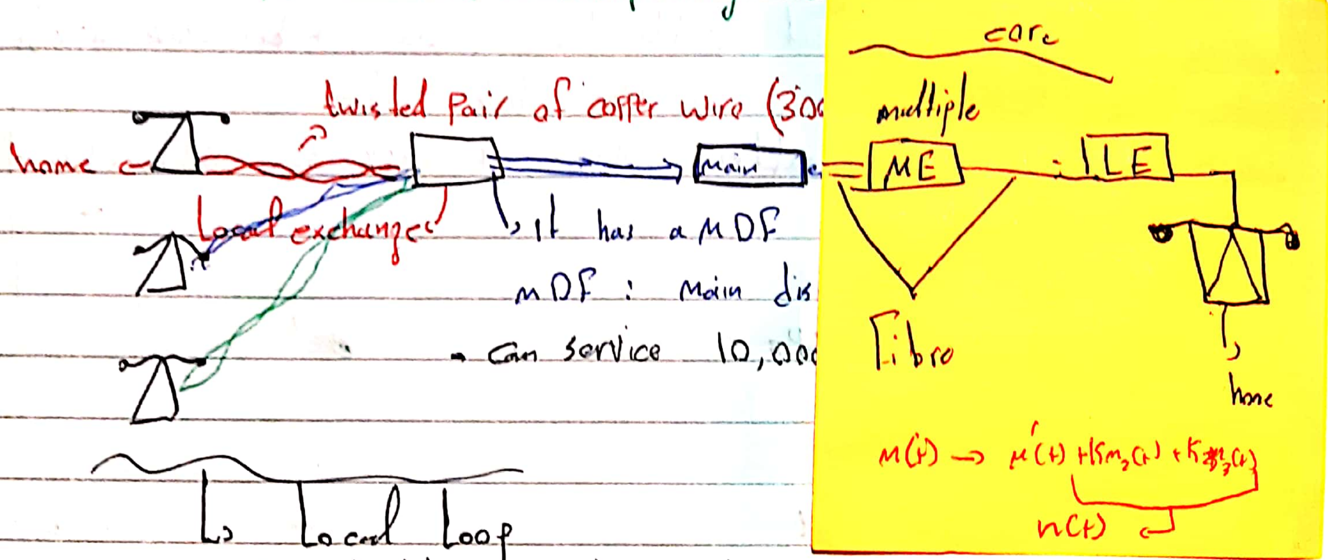
- deterministic → $\sin(100t - 50)$
  ↳ Known value at a time
- random → can't know
  ↳ Probability dependent (noise)

$n(t)$ → Some time it gives, sometime it takes
↳ 0.5V ↳ 0.1V

Sources: interference from signals transmitted on nearby channel (cross talk) → electromagnetic interference
↳ such as that to external electromagnetic waves
2. solar radiation
3. noise from different channels

example: Land line telephony systems
↳ PSTN
POTS

PSTN: Public Switched telephony network
POTS: Plain old telephoney service



↳ Local Loop
(Last mile) → most expensive part in telephony system

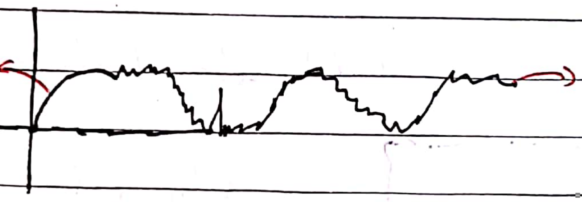
2/10/2019

L4. 2- internal sources.
↳ (thermal noise)

thermal noise:
↳ ① random motion of electrons in conductors, ② random diffusion and recombination of charged carriers in electronic devices

↳ because wires are not 0 K°
. it will have random motion

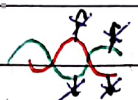
Smooth signal.



→ random motion of electrons

- external can be eliminated and reduced
- interal can be reduced but not be eliminated.

sol for ext



Like cat 5 1- shelding or twisting → good, reasonably price

STP 2- a different cable design (coax, fibre, wave guide)

(very expensive) 3- Proper design of the whole system

4- using filters at the reciever side: BPF, LPF, notch filter

electric field are vectors we can eleminate them by 2 vectors against each other cat 5 (UTP)

wifi → 2.4 GHz or 5 GHz

blue tooth → 2.4 GHz

electric oven → 24 GH

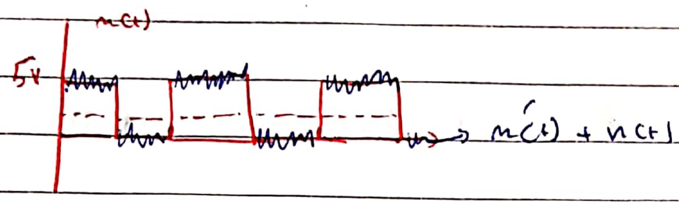
h.w → what is ISM band

2/10/2019

note on 4- Very cheap and very effective:
- Works for internal and external.

5- Digital transmission (Threshold detection, orthogonality, FEC, etc)
↳ getting cheaper.

FEC: forward error correction



If noise too big to pass threshold voltage
↳ using other sol

Solution for internal Noise.

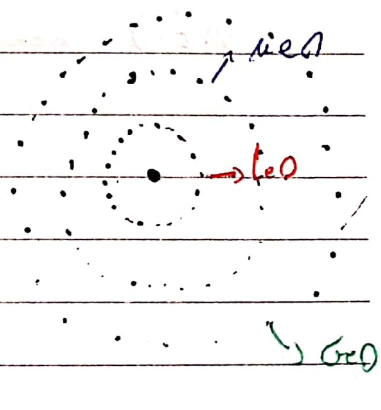
1- Cooling
↳ no thermal energy
↳ very expensive (not work for normal ~~app~~ ~~...~~) apps
↳ to push to 0.15°

example
↳ satellite system

2, 3 are the same as external → 4, 5

example

↳ 11 orbits



Leo: low-earth orbit (~~4~~ 1000 Km) (Iridium)
Meo: medium-earth orbit (10000 Km)
GEO: geo stationary earth orbit
geo synchronous earth orbit
35,768 Km ≃ 36000 Km (Inmarsat)

- orbits of satellites

2/10/2019

the orbits that satelite orbits

GEO satelite covers 1/3 of earth surface Area

LEO satelites are about 66 to cover the earth surface Area

only 3 satelites to cover the surface of the earth

needs 4-6 hrs to rotate full cycle around earth

needs 24 hours to rotate around earth

so it seems to be fixed on the sky

thats why we need 35,768 Km

- because it gives 24 hour to rotate
- easier to track
- controling it to stay in orbit with fuel injectors

MEO needs 4-6 satelites to cover the earth

2/10/2019

LEO satellite examples:
1- Iridium
2- Global star
3- Space X.
4- ISS (International space station)

MEO
↳ (GPS) global positioning system → US and earth
(GlONASS) → Russian } Don't want to depend on US
(Galleleo) → europe
(Baidu) → chineese

GEO
↳ TV broad casting (nile sat, hot bird, badr sat)
↳ Turnar sat
↳ thuraya

Geo suffers from 1- heavy attenuation (Long way to earth.
2- to much noise

in Dishes they use LNA, LNB
↳ big ones
↳ Low noise Amplifier
↳ Low noise Block

the 4 problem are important because the Co-exist in all channels (combined effects)

↳ Sol for att and noise
↳ use Regenerator → works on 2 problems.
(Repeater)

7/10/2019

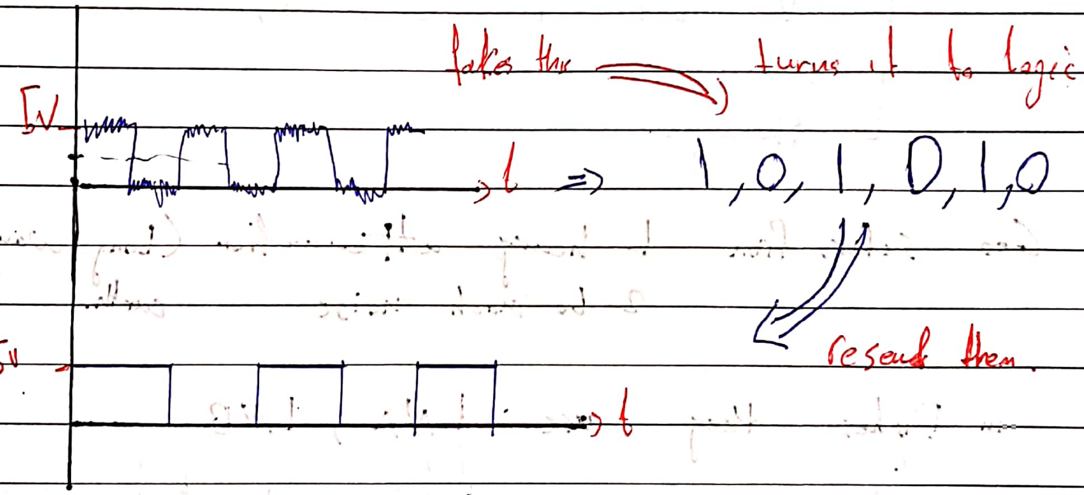
Regenerator $R = R_x + T_x$

reciever — Transmitter

it takes naisey signal with help of Voltage threshold
→ then resending a clear signal

1- reads the bits
2- Sends them as Voltage

Regenerators → 1- dont work on really naisey Signals
2- Very expensive



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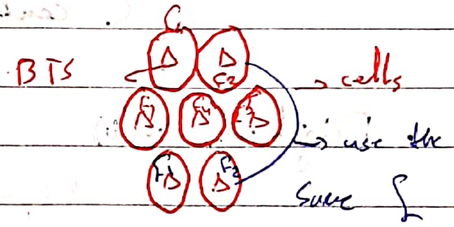
1.5 impairments that exist in some channels

5- Fading: Variable attenuation with time of day ① and reciever ② Location (wireless system)

6. Doppler Shift: shift in the frequency of the Tx signal shows up in a wireless channel and fast moving objects → exist in wireless.

7. Frequency re-use interference: Shows up in wireless systems (reduce cost) when we re-use the same frequency at multiple nearby Location to increase system capacity. same as WiFi home 2.4 friend 2.4

→ exist in wireless



8. chromatic Dispersion:
Specific to optical fibre channels
≈ Distortion in a way.

- shannon's Limit

$$C = B_{ch} \times \log_2 (1 + SNR)$$ → Linear Distortion

C: capacity of the channel bits/s
B_{ch}: channel bandwidth (units of Hz)
SNR: Signal to noise ratio (unitless)

SNR = (Signal Power / noise Power)
attenuation ↲ ↳ noise

7-10-2019

Comm systems are classified based on the Type of Signal Sent on the channel

	base band		Carrier
analog	① analog baseband Comm sys	analog modulation → ← demodulation	② analog carrier Comm sys
digital	③ digital baseband Comm sys	Digital modulation → ← demodulation	④ digital carrier Comm sys

(Digitization ↓ / ↑ De-digitization)

why 4? each one has different adv

- analog m(t) can assume any Value in a continuous range of Values at any Point in time t.
  (continuous of Value - continuous in time)

- Digital m(t) can assume only finite Values or shapes and uses threshold detection

- Base band m(t) has a frequency-domain Spectrum clustered around Zero frequency (the base)

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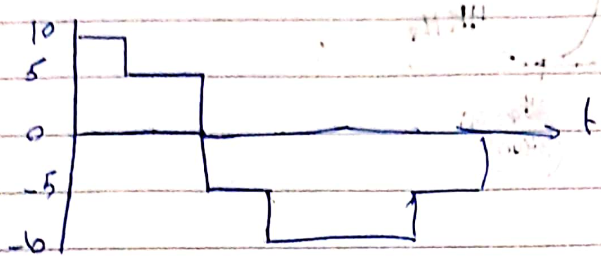
• Digital and Periodic

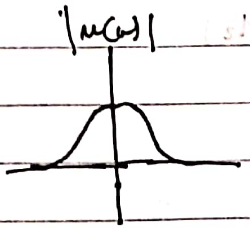


Digital & aperiodic

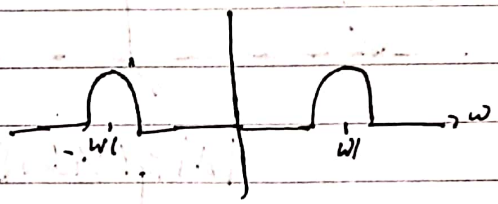
base band Vs Carrier

frequencies are clustered around the base

frequencies are away from the base



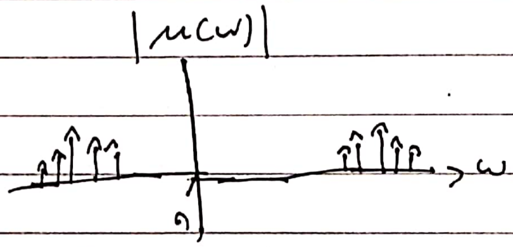
base band



carrier

• Periodic signals give pulses in frequency domain

So F.T can give:
1- carrier or base band
2- Periodic or aperiodic



Periodic and carrier

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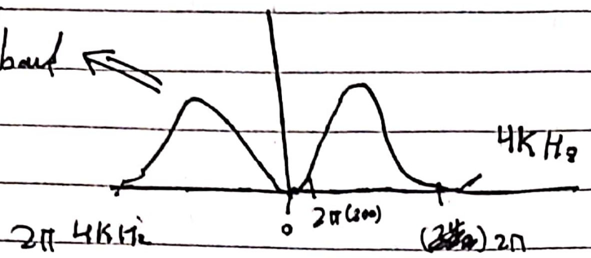
|M(Voice)|

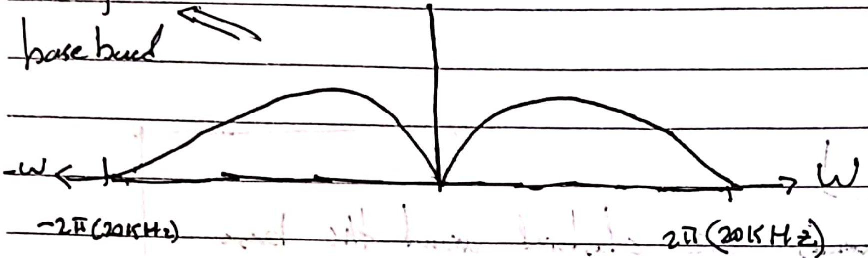
analog
Baseband

$B_{Voice} = 4000\ Hz$
4K Hz



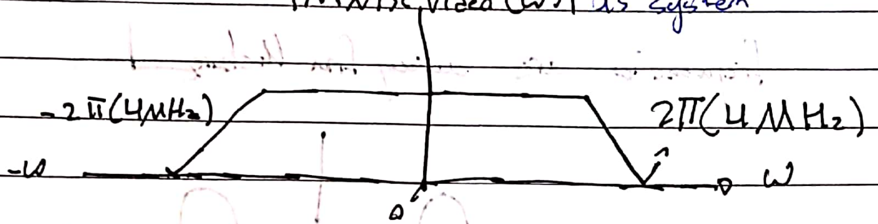
|Music M|

analog
base band

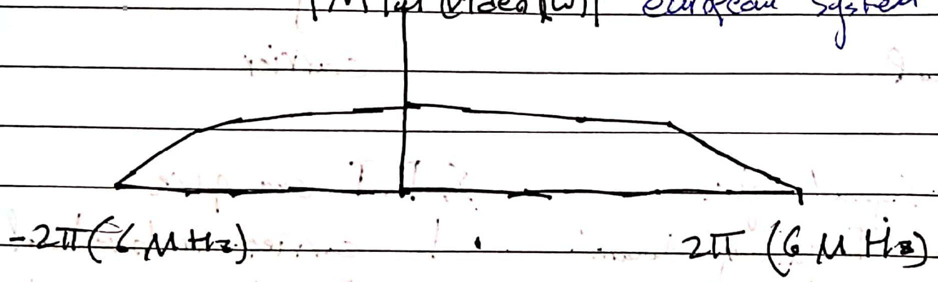


$|M_{NTSC}\ Video(\omega)|$ us system

(Base band)



$|M_{Pal}\ Video(\omega)|$ european system



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Analog base band

- an analog base band sends the analog signal $m(t)$ as it is (without modulation)

- advantages
  - simple
  - inexpensive
- used in low distance applications

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26-10 extra lecture

modulation → analog and digital

modulation: a process to turn from baseband to carrier

- The signal $m(t)$ is combined with high frequency called the carrier

$m(t) \longrightarrow \boxed{\text{mixer}} \longrightarrow \phi(t)$

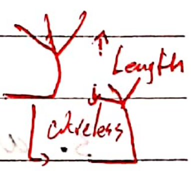
$c(t) = \cos(\omega t)$

1- Periodic
2- high Frequency

hence Frequencies are shifted

adv of modulation.

1- allows the use of antenna length.



$L_{ant} = \frac{\lambda}{2}$ λ is wave length

$\lambda \rightarrow c/f = \frac{3\times10^{8} m/s}{f}$

$\lambda_{voice} = \frac{3\times10^{8}}{3\times10^{3}} = 1\times10^{5} m = 100 km$ → not Practical.

Wireless band carrying V, I

it carries $\vec{E}, \vec{B}$

9 - 10 - 2019

2.4 Ghz ≈ 3GHz

$\lambda_{wifi} = \frac{c}{f} = \frac{3\times10^8}{3\times10^9} = 0.1$

$L = \frac{0.1}{2} = 5cm$ → Very Practical

2. allows multiplexing (FDM) as well as CDMA and OFDMA in digital system

multiplexing : Sending multiple signals on the same channel simultaneously.

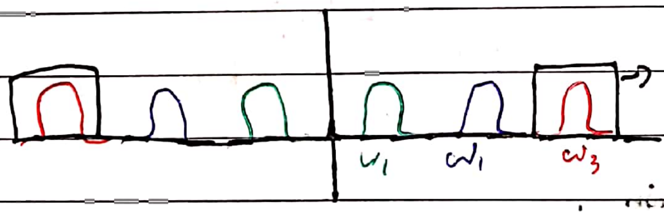
FDM : frequency division multiplexing → isolating frequencies

CDMA : code division multiple access

OFDMA : orthogonal frequency division multiple access.

↳ enabled by modulation

TDM : time division multiplexing



→ using tuneable Filter we get what we want

3. allows exchanging SNR for band width

ex of modulation systems
analog carrier
1- AM , FM (Amplitude mod), (Freq mod)
2- TV (Pal, NTSC)

digital carrier
1- digital radio broadcasting (DAB)
2 " tv " (DVB-S, DVB-T, ATSC)
digit Video broadcast
3- WiMax metropolitan
4- WiFi
5- cellular telephony (2nd, 3rd and 4th gens)
6- (blue tooth, Zigbee, NFC)
7- old dial-up modem
8- Adsl modem
copper wire ?
to multiplexing

modem: modulation demodulation

ADSL: asymmetric digital subscriber Line

• Digitization to convert from base band analog to baseband digital

ACD
1- Sampling
2 - Quatization
3- Mapping
4- encoding (coding)
5 - Pulse shaping

→ steps of converting from analog base band to digital base band. in respect to thier number

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- digital base band advantages.

1- immunity to noise and attenuation.
2-
3 } handout
4

- examples of digital base band systems.

Imbedded system
- Serial (RS-232) and USB port connections
- ethernet (a popular Local Area network)

mouse

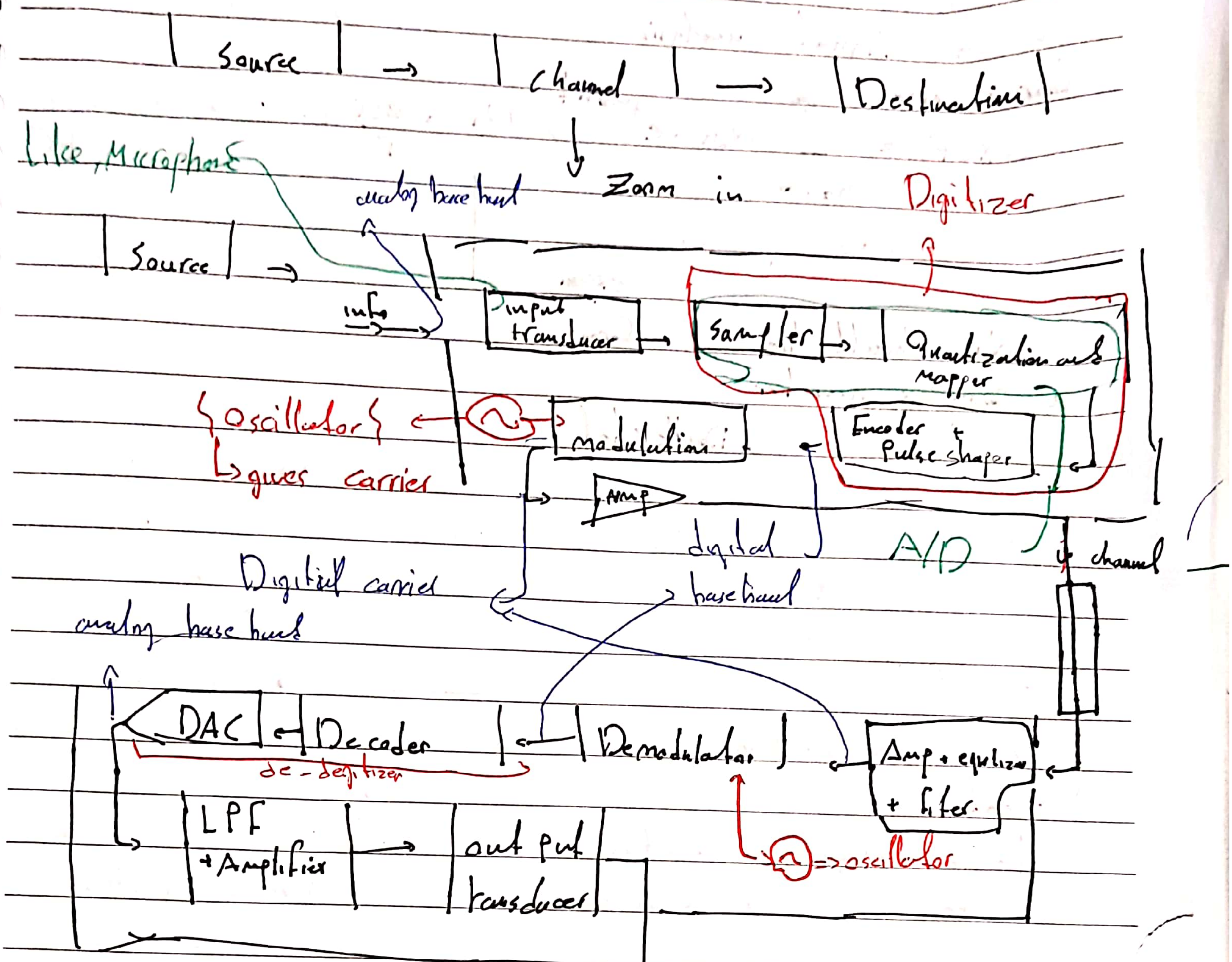
- telephony (between Local exchanges)

$\underbrace{T_1, T_2, ..}_{us}$ $\underbrace{E_1, E_2, ..}_{europ}$

→ Core parts in telephony systems

uses devices that are digital in nature

→ mouse, Keyboard



info $m(t) \approx m(t)$

. this example of digital carrier system

↳ because digital carrier moves in the channel. signal

. name of system depends on signal in channel.

14-10-2019

[-] Signal analysis

Fourier series

$$x(t) = \sum_{n=-\infty}^{\infty} a_n \cdot e^{j w_0 n t}, \quad w_0 = \frac{2\pi}{T}$$

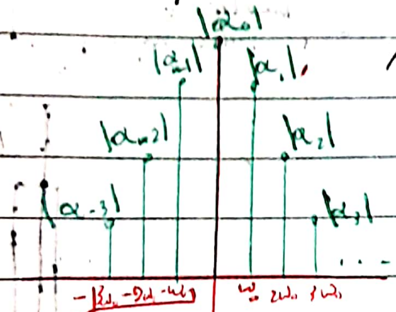
↳ periodic signal.

expon. form

a_n should be a complex number

- Research Project.
  - ↳ Near-field comm
  - ↳ 3 marks NFC
  - ↳ should cover
    - ↳ intro, overview, Design, standards, Applications, bluetooth comparison
  - ↳ in exam.

→ Double sided

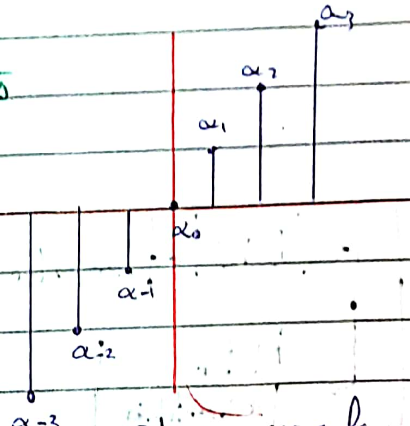


$a_0 = a_n \Rightarrow n=0$

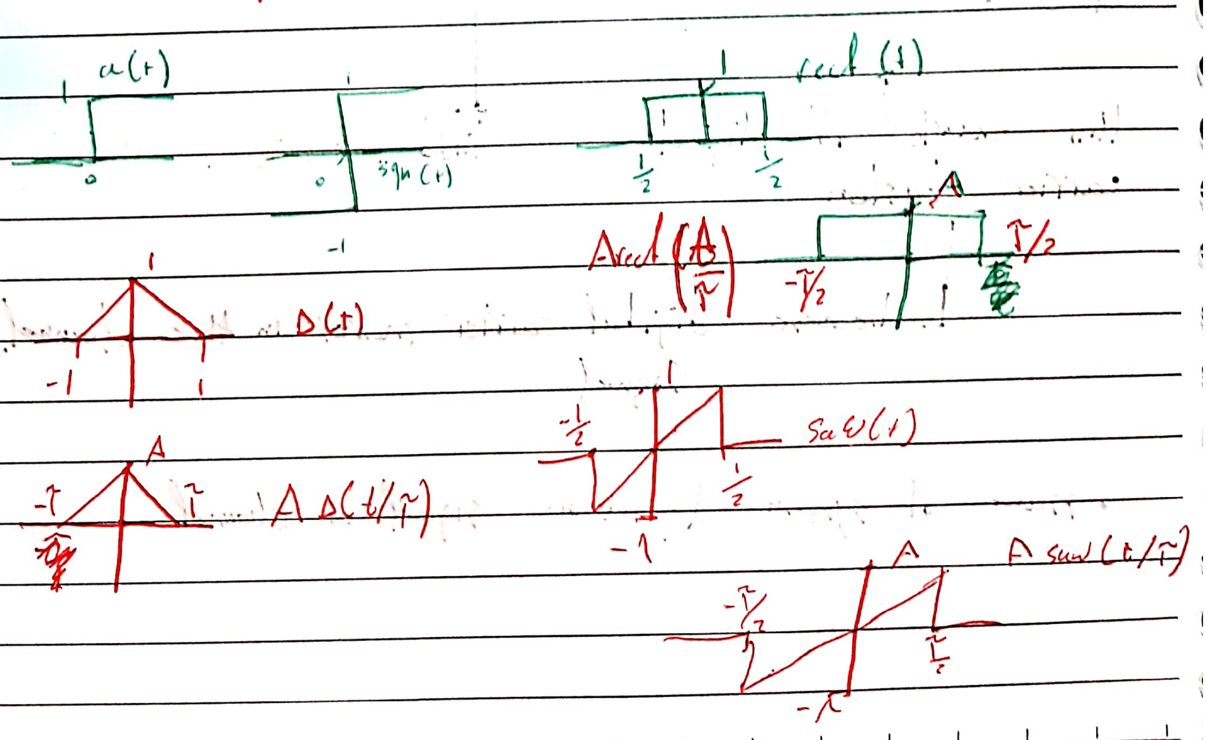
magnitude spectrum

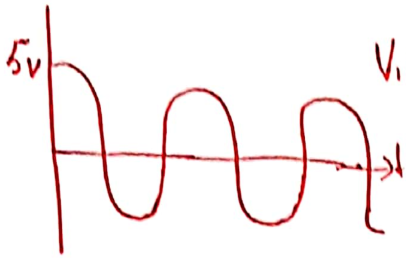
note:

↳ no frequency is negative
this represents a cosine



phase spectrum



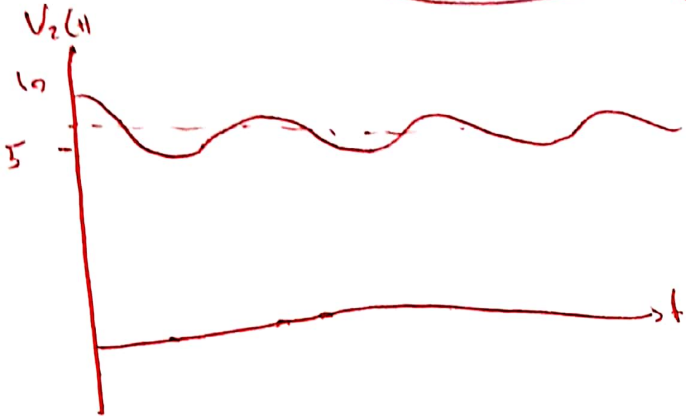


$V_1(t) = 5\cos(\omega_0 t)$

$\overline{x(t)} = 0 = DC$

$P_x = \overline{x^2(t)} = \frac{A^2}{2} = \frac{5^2}{2} = 12.5$

↳ (V_{rms}^2)



$V_2(t) = 7.5\cos(\omega t) + 7.$

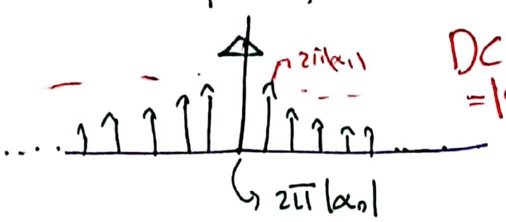
$\overline{x(t)} = 0 + 7.5 = 7.5$

$\overline{x^2(t)} = \frac{2.5^2}{2} + 7.5^2$

why like this ?

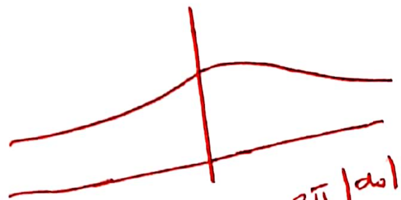
because 7.5 cosine and 7.5 are o

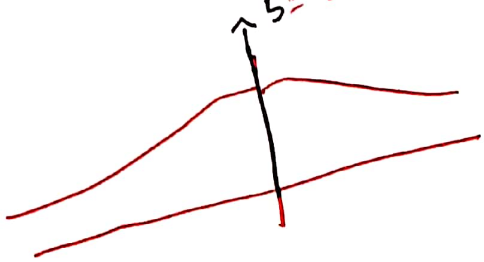
↳ we can use Principle on signals



DC in freq domain $= |\alpha_0|$

ex

1-  $\Rightarrow DC = 0$

2. 

$\Rightarrow DC = \frac{|5|}{2\pi}$

Dc Vs Average Power.

$$Dc = \overline{x(t)} = \lim_{T\to\infty} \frac{1}{T}\int_{-T/2}^{T/2} x(t)\, dt$$ ⟹ from time domain

$$Dc = \overline{x(t)} = a_0$$ ⟹ from frequency domain

↳ it shows you can get Dc value from α when it's α_0

* Sometimes time domain is harder than frequency domain

Average Power P_x

$$P_x = \overline{x^2(t)} = \lim_{T\to\infty} \frac{1}{T}\int_{-T/2}^{T/2} |x(t)|^2\, dt$$ ⟹ time domain

$$P_x = \overline{x^2(t)} = \frac{1}{2\pi}\int_{-\infty}^{\infty} S_x(\omega)\, d\omega$$ ⟹ frequency domain

↳ PSD: Power Spectral density

$$PSD = S_x(\omega) = \lim_{T\to\infty} \frac{1}{T}|X_T(\omega)|^2 = \mathcal{F}\{R_{xx}(\tau)\}$$

↳ auto correlation function

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PSD $= S_x(\omega)$

$= \lim_{T \to \infty} \frac{1}{T} |X_T(\omega)|^2$

or $= \mathcal{F}\{R_{xx}(\tau)\}$

$R_{xx}(\tau)$ auto correlation function.

hw 1- Find $R_{xx}(\tau)$ for $x(t) = A\cos(\omega_c t)$, then find PSD then find P_{avg}.

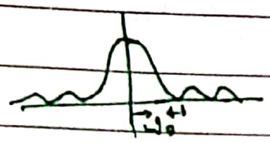
in power calculation, remember to consider pos and neg frequency.

$\int_{-\infty}^{\infty} = 2 \cdot \int_{0}^{\infty}$

ex 2- find $R_{xx}(\tau)$ for $x(t)$ periodic then PSD then P_{avg} from PSD

in band width calculations, only consider pos freq only!.

$x(t) = \sum_{-\infty}^{\infty} C_n \cos(\omega f_n)$ → orthogonal signals



$\sum$ PSD of $x(t)$ = $\sum$ PSD of cosines

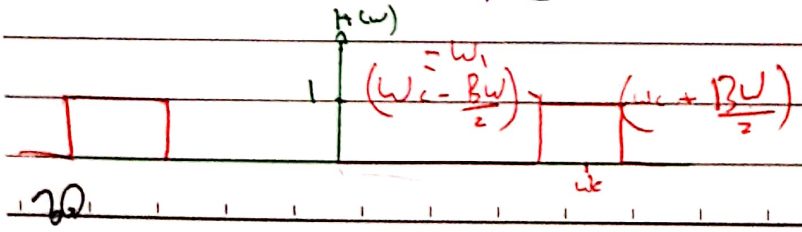
* Filters

f response of Filter $H(\omega) = \frac{V_o(\omega)}{V_{in}(\omega)} = \mathcal{F}\left\{\frac{V_o(t)}{V_{in}(t)}\right\}$

1st BPF : band pass filter.

$\omega_c = \frac{\omega_2 + \omega_1}{2}$, band width $= \omega_2 - \omega_1$ Don't forget 2π

ex if $BW = (80\,KHz)\,2\pi$, $\omega_c = (100\,MHz)\,2\pi$, gain = 1



$\omega_1 = (99.96\,MHz)\,2\pi$

$\omega_2 = (100.04\,MHz)\,2\pi$

how BPF modeled

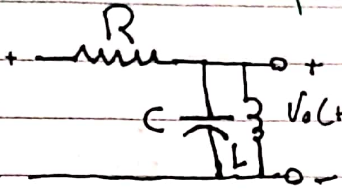
1- BPF

2- 

3- 

Gain can be Variable if it has other design

circuit of BPF :-



2nd order diff equation

$f_c = \frac{1}{2\pi\sqrt{LC}}$ Hz

B_{of} BPF $= \Delta f = \frac{R}{2\pi L}$, gain = 1 why?

because these are ideal Passive elements

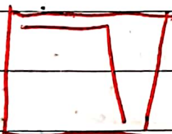
characteristics

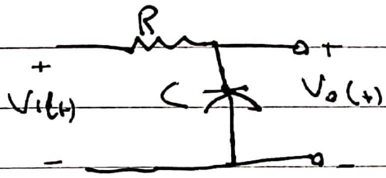
Low Pass filter. LPF.

symbols :

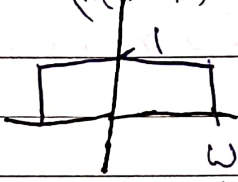
1- LPF

2- 

3- 



(H(w) of lps)



$W_c = W_1$ = Band Width , gain = 1

cutoff Frequency

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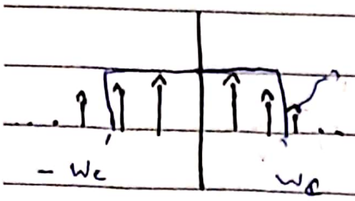


does not pass.

$= B_{LPF} = \frac{1}{2\pi RC}$

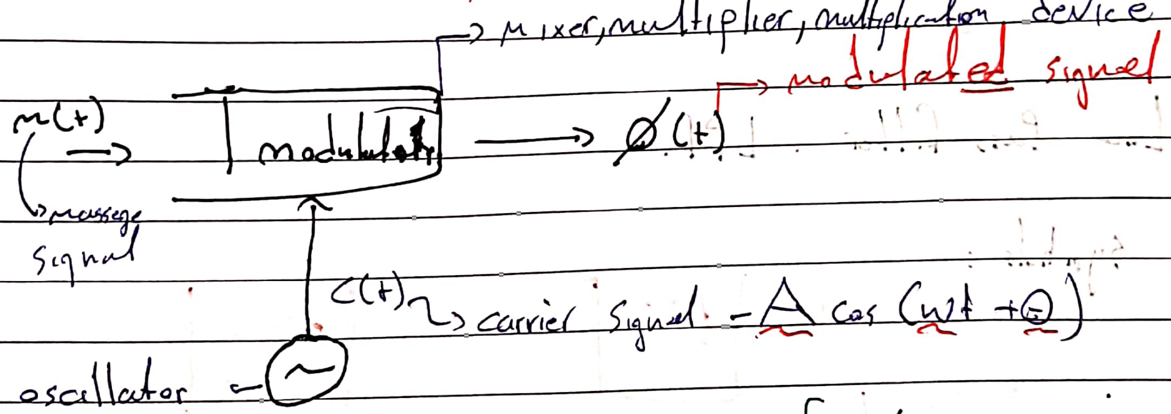
1st order low pass filter.

Amplitude modulation:

what is modulation? Process that causes a shift of frequency of signal ($m(t)$) from Baseband to carrier.

its ~~can~~ achieved by changing amplitude or frequency or shift of a high frequency periodic signal called carrier signal ($c(t)$) in proportion to the base band msg signal $m(t)$.

3 components → A, ω, θ



$c(t)$ → carrier signal $= A \cos(\omega t + \theta)$

if $c(t)$ = cosine its called Continuous wave modulation (CW modulation) communication 1

16 - 10 - 2019

There are 3 modulating methods

1- $A \propto m(t)$, W_c = constant , Θ = constant

↳ its called → Amplitude modulation (AM) C signals
↳ Amplitude shift keying (ASK) Discrete signals "Digital"

2- A = constant , $W_c \propto m(t)$, Θ = constant

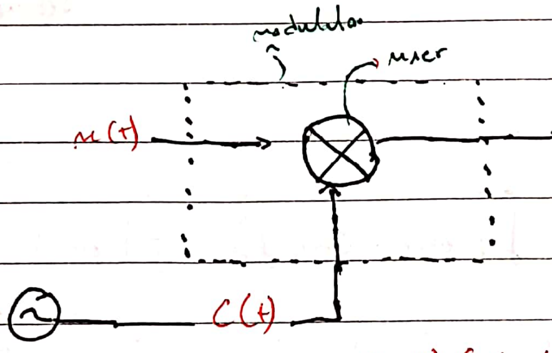
↳ its called → frequency modulating (FM) C signals
↳ frequency shift keying (FSK) Discrete signals "Digital"

3. A = constant , W_c = constant , $\Theta \propto m(t)$

↳ its called → phase modulating (PM) C signals
↳ phase shift keying (PSK) Discrete signal "Digital"

note AM = DSB - LC = Double side band ~~suppressed~~ Large carrier

- Mixer



$\phi f = m(t) \cos(w_c t)$

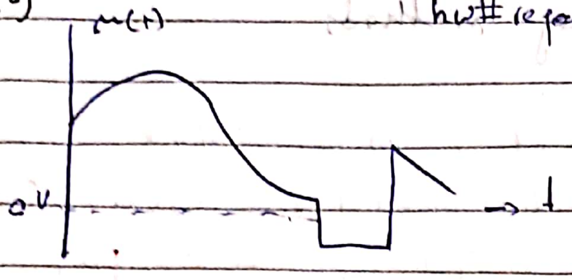
$c(t) = \cos(w_c t)$ $(A = 1, \Theta = 0)$

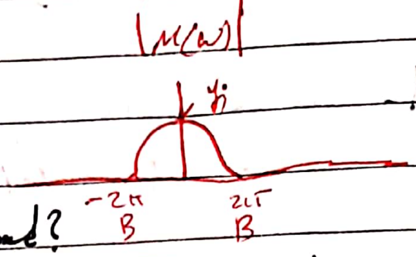
Amplitude modulation

1- DSB - SC
2- DSB - LC
3- SSB - SC
4 SSB - LC
5 - VSB - LC
6 - VSB - SC
7 - QAm

21-10-2019

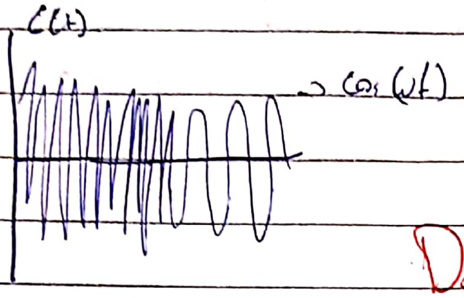
C-9 hw# report slide 5

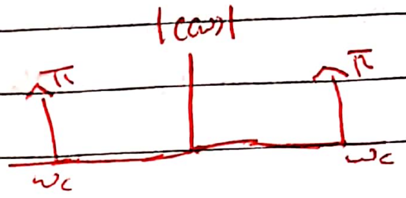




Baseband?

$\boxed{B}$ in Hz or $\boxed{2\pi B}$ rad/s

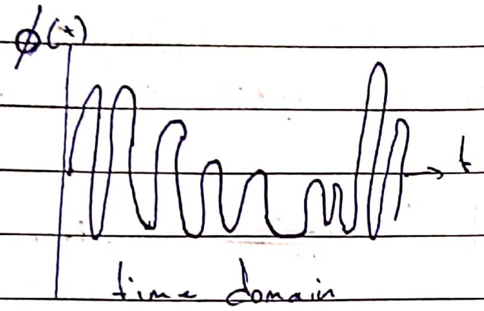




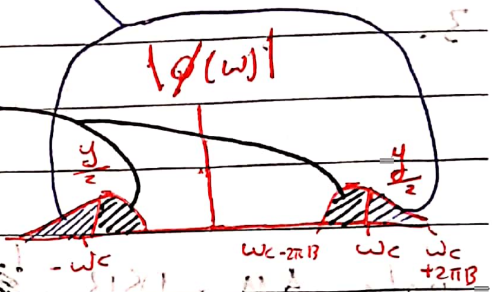
Double side band

upper side band (USB)

Lower Side band (LSB)



time domain



Frequency domain

Soln math.

$$\Phi(\omega) = \mathcal{F}\{\phi(t)\} = \mathcal{F}\{m(t)\cdot\cos(\omega_c t)\} = \mathcal{F}\left\{ m(t)\,\frac{e^{j\omega_c t}+e^{-j\omega_c t}}{2}\right\}$$

$$= \mathcal{F}\Big\{\underbrace{\tfrac{1}{2}m(t)e^{j\omega_c t} + \tfrac{1}{2}m(t)e^{-j\omega_c t}}_{\text{distributive}}\Big\} = \mathcal{F}\Big\{\underbrace{\tfrac{1}{2}m(t)e^{j\omega_c t}}_{\text{Freq. shift}}\Big\} + \mathcal{F}\Big\{\underbrace{\tfrac{1}{2}m(t)e^{-j\omega_c t}}_{\text{Freq. shift}}\Big\}$$

$$= \frac{1}{2}M(\omega-\omega_c) + \frac{1}{2}M(\omega+\omega_c)$$

21-10-2019

Soln graphical

$$\phi(\omega) = \mathcal{F}\{\phi(t)\} = \mathcal{F}\{m(t)\cdot c(t)\} = \frac{1}{2\pi} M(\omega) \circledast C(\omega)$$

$$C(\omega) = \pi\,\delta(\omega-\omega_c) + \pi\,\delta(\omega+\omega_c)$$

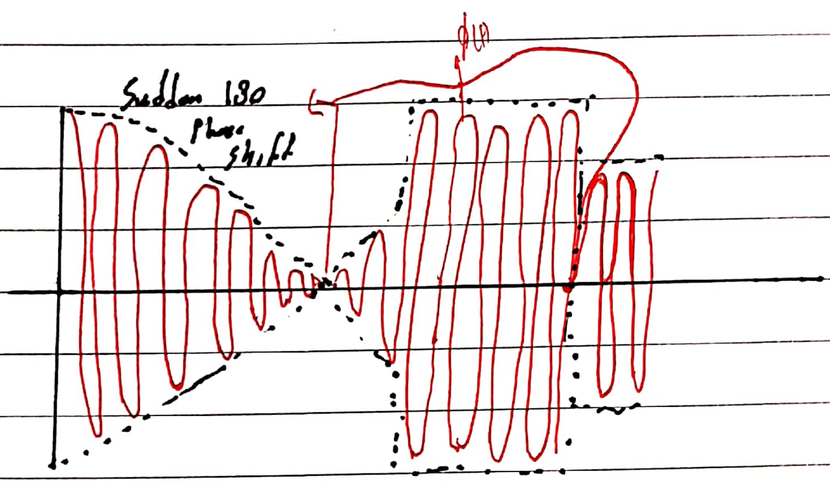
note: any Signal $F(\omega)$ $\circledast$ with $\delta(\omega)$ resaults in shift and scaling in frequency domain

: when multiplying by $\cos(\omega_c t)$ in time domain the following happens in freq. domain.

1- shift right by ω_c
2- shift left by ω_c
3- multiplying by $\frac{1}{2}$

why suppressed? → cos doesn't show as impulses in $\phi(\omega)$ or in time domain, but you can see its effect.

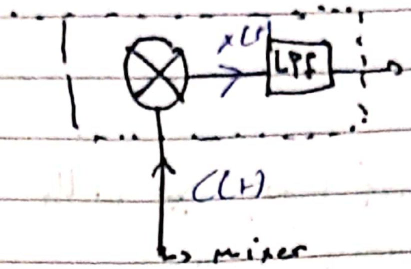
DSB - SC → Bandwidth = twice the originel (Bad)
└→ Power = half the originel



21-10

DSB-SC Demodulator → Mixer again

channel → $\phi(t)$



math.

$$\mathcal{F}\{x(t)\} = \mathcal{F}\{\phi(t)\cdot c(t)\} = \mathcal{F}\{m(t)\cos(\omega_c t)\cdot\cos(\omega_c t)\}$$

$$= \mathcal{F}\{m(t)\cos^2\omega_c t\} = \mathcal{F}\{\tfrac{1}{2}m(t) + \tfrac{1}{2}m(t)\cos(2\omega_c t)\}$$

$$= \mathcal{F}\{\tfrac{1}{2}m(t) + \tfrac{1}{4}m(t)\, e^{j2\omega_c t} + e^{-j\omega_c 2t}\}$$

$$= \tfrac{1}{2}M(\omega) + \tfrac{1}{4}M(\cancel{\omega - 2\omega_c}) + \tfrac{1}{2}M(\cancel{\omega + 2\omega_c})$$

if LPF → $= \tfrac{1}{2}M(\omega)$

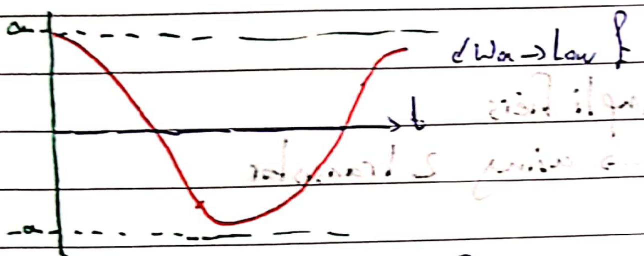
to demodulate: cosine freq should be the same as $\phi(t)$.

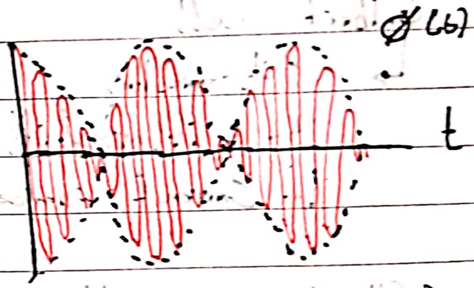
23 - 10 - 2019

L-10 tone modulation → signal = $a\cos(\omega_m t)$

ex: assume we perform DSB-SC modulation for the base band signal $m(t) = a\cos(\omega_m t)$

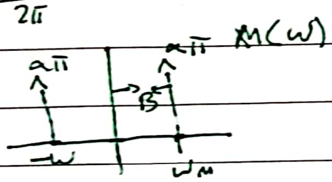
1- sketch time domain of $\phi(t)$
2- sketch the F.T of $\phi(t) \to \phi(\omega)$
3- find the bandwidth of $m(t)$ and $\phi(t)$
4- find the average power in both $m(t)$ and $\phi(t)$
5- show the demodulator hardware
6- sketch $\underbrace{x(t)}_{\text{Before LPF}}$ at $\underbrace{y(t)}_{\text{after LPF}}$ in the demodulator

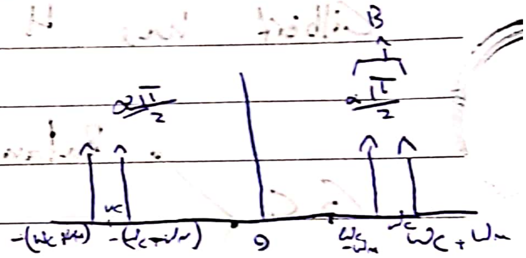




Bandwidth of $m(t) = ?$

$\frac{\omega_m}{2\pi}$ Hz





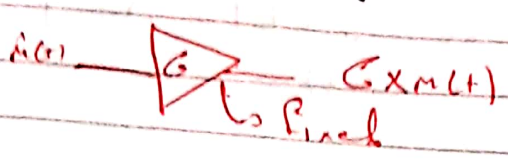
Bandwidth of $\phi(\omega)$

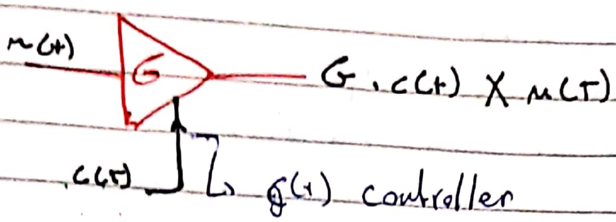
$(\omega_c+\omega_m)-(\omega_c-\omega_m) = \frac{2\omega_m}{2\pi}$ Hz

examples at tamin slides

how to build a mixer ?

1- Variable gain amplifier
→ gain is not constant
→ make it c(t)





eg of Variable gain amplifiers → Gilbert cell ic (MC 1496)

Gilbert cell
→ 8 transistors
3- Differential amplifiers
1- Current source → using 2 transistor

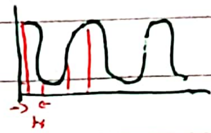
Gilbert has 4 modes

G.C
- Balanced → Linear mode gives $m(t)\cos(\omega_c t)$
- unbalanced → gives $m(t)\cos(\omega_c t) + A\cos(\omega_c t)$ → DSD-LC

26-10

Digitization
1- sampling
2- Quantization
3- Mapping
4- coding
5- pulse shaping

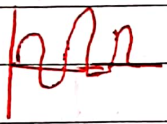
Sampling :- analog signal to sample it (subset of Point)
(Discrete signal.)



• ts = sampling time
• in real life we can't use pulses
impulses → ideal sampling

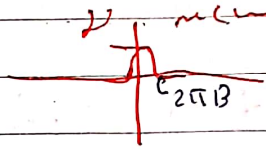
ideal sampling

$m(t)$



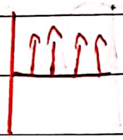
$W_s = 2\pi f_s$
$= \frac{2\pi}{T_s}$

$M(\omega)$

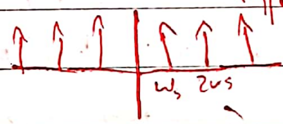


X

$\delta(t)$

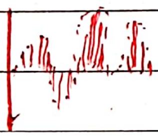


$Rep\{\delta(\omega)\}$



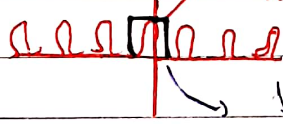
=

$x(t)$



$\frac{1}{2\pi}\delta \cdot \frac{2\pi}{T_s} = \frac{\delta}{T_s}$

$X(\omega)$



to regain put LPF
(Connecting the dots)

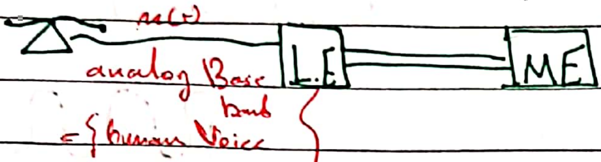
If bad decision of $T_s\uparrow$
aliasing happens
(over Lapping)

Solution to avoid aliasing

① use nyquist rate → $f_s \geq 2B$ in ideal sampling

usually a lit more than nyquist rate

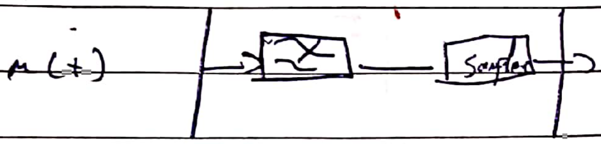
example → in telephony.

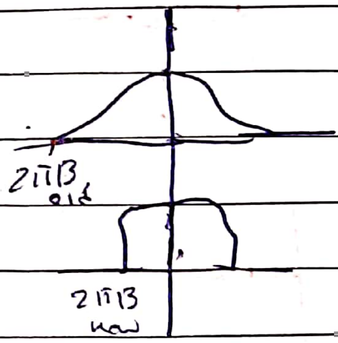


$f_s = 8KHz = \left\{ 8000 \frac{\text{samples}}{s} \right\}$ → very costly

anti-alasing Before ← Digitizer
Sampling

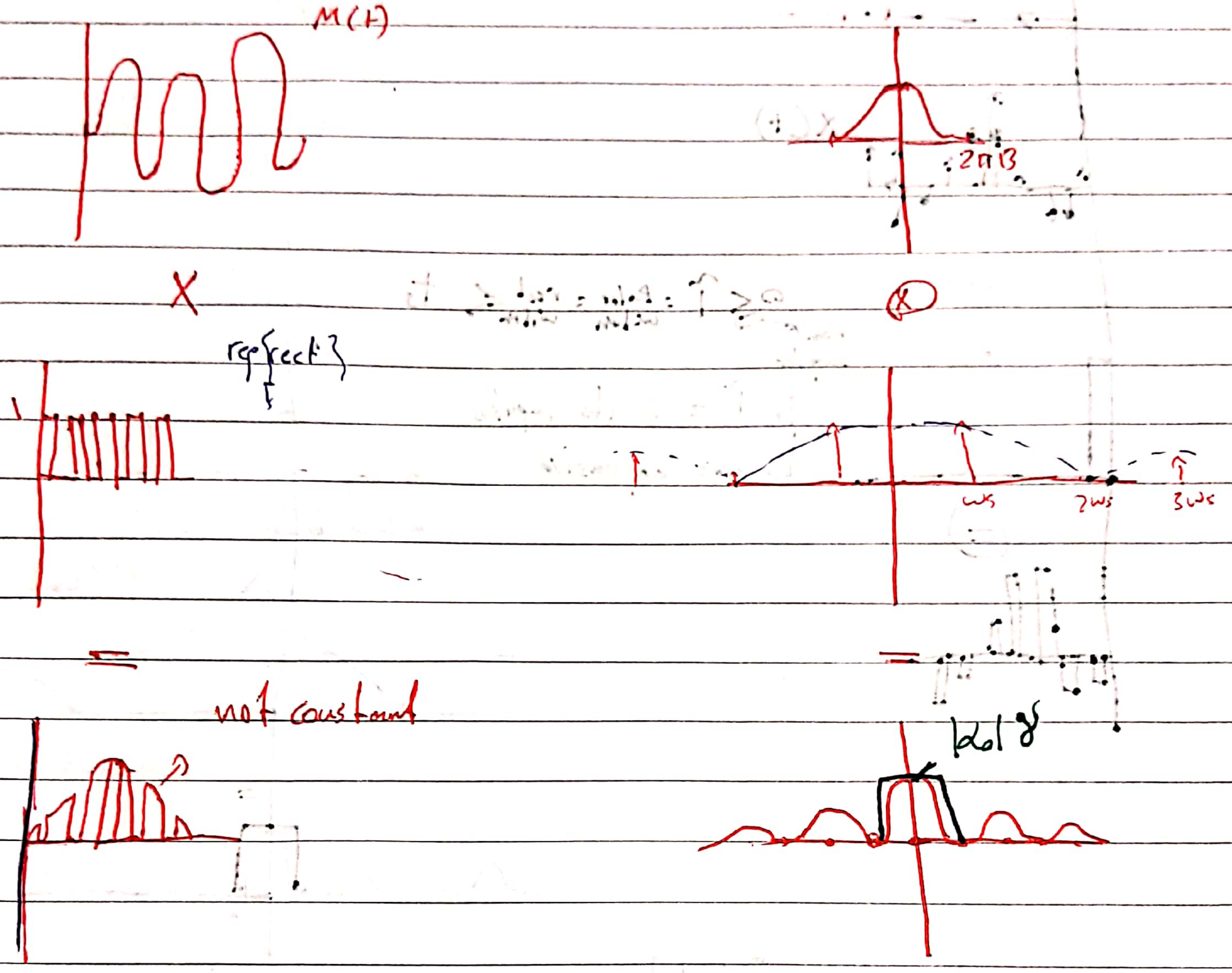
② Limit your Bandwidth of $m(t)$ with fixed





If you send music on telephony system it sounds horrible Because system is not designed for music natural sampling (switching on/off)

sends parts of the signal

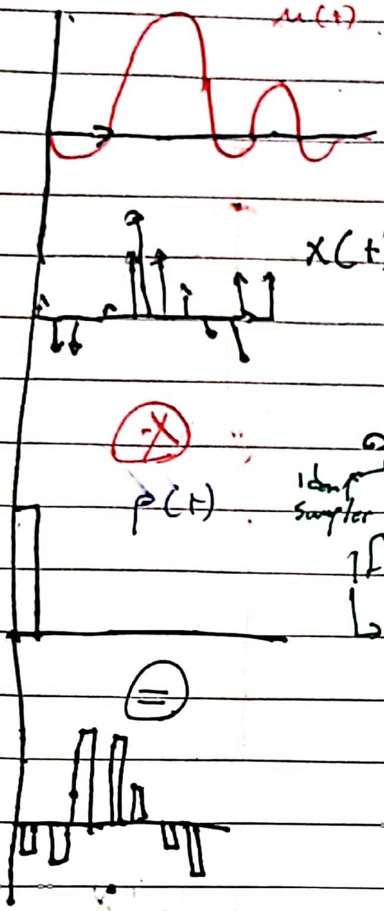


after LPF

you get $|d_0|\, m(t)$

Practical Sampling (PAM)

PAM is what we use in real life



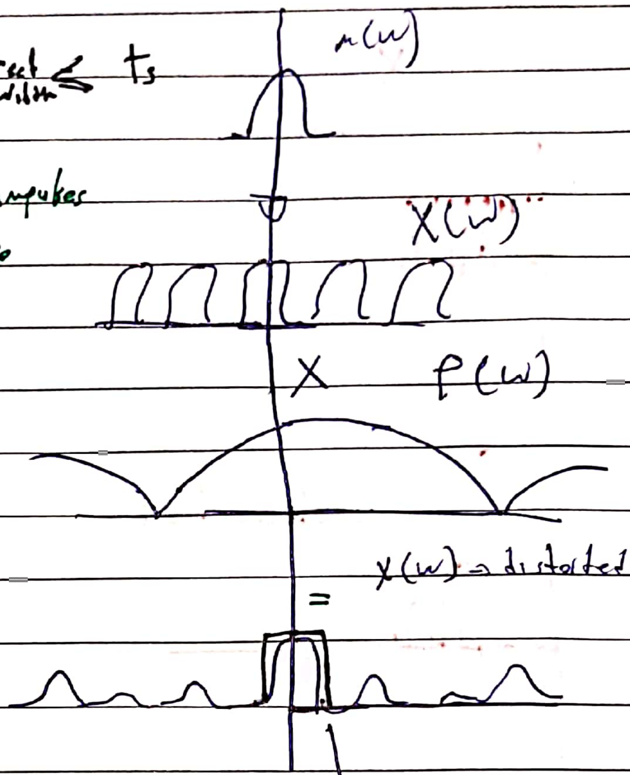
conv in time – multiply in freq

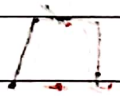
$0 \le \tau$ = Pulse width = rect width $\le T_s$

Ideal Sampler

If $\tau = 0$ its impulses

↳ more expensive





$x(t) \rightarrow \boxed{\Sigma} \rightarrow \boxed{P^{-1}(w)} \rightarrow m(t)$

this part unfortunately is distorted

↳ soln to this is equilizer

26-10

Summary

Tx $m(t) \rightarrow \boxed{LPF} \rightarrow \boxed{Sampler} \rightarrow x(t)$

Prevent aliasing

Rx $x(t) \rightarrow \boxed{LPF} \rightarrow \boxed{P(\omega)} \rightarrow m(t)$

recover from sampled $m(t)$ — reduce distortion

Quantization

Limit the value of signals $\Delta V = 2m_p/L$, m_p = peak of $m(t)$

Quantization error ΔV — if $L \uparrow$ $\Delta V \downarrow$

Quantization level L — $L=6$ → ΔV

$L=4$		$L=8$		$L=16$
00 ⇐ 0 V	$\Delta V = 1$	111	3.5	$\Delta V = 0.5$
01 ⇐ 1 V			3	If sent 2.6
10 ⇐ 2 V			2.5	↳ truncate to 2.5
11 ⇐ 3 V			2	so 0.1 is noise
if sent 2.6			1.5	L (smaller Quantization noise)
↳ truncate to		↑	1	
2V so 0.6 is like		001	0.5	
noise		000	0	

# of bits = 4

example - telephony $L = 256$ ∴ 8 bits

2- cd's, $L = 65{,}536$ ∴ 16 bits

2- $L\uparrow$ more expensive because more bits to send

(better SQNR ⇒ better quality but expensive) in PTSN

$2^{\text{no of bits}} = L$

SQNR → Signal to Quantization noise ratio

$\boxed{\text{LPF}} \longrightarrow \boxed{\text{Sample}} \longrightarrow \boxed{\text{Quantization}}$

$f_s = \frac{8K\,\text{sample}}{s}$

$L = 256$

Bits $= 8$ bit/sample

Data rate $= f_0 = \frac{64000 \text{ bits}}{s} = \boxed{64\text{ KbPS}}$ → PCM stream: Pulse coded modulation

$$SQNR = \frac{\overbrace{\overline{m^2(t)}}^{\text{average Power of signal}}}{\underbrace{\overline{q^2(t)}}_{\text{avg. quantization noise power}}} = \frac{\overline{m^2(t)}}{\left(\frac{m_p^2}{3L^2}\right)}$$

$$= 3L^2 \frac{\overline{m^2(t)}}{m_p^2} = \frac{3L^2}{K_m^2}$$

L = Level of Quantization ⇒ L↑ SQNR↑

$$K_m^2 = \text{crest factor} = \frac{\text{Peak}^2}{\text{rms}^2} = \frac{\text{Peak}^2}{\text{Power}}$$

example: find crest factor for $m(t) = \alpha \cos(\omega_m t)$

sol $K_m^2 = \frac{\text{Peak}^2}{\text{Power}} = \frac{\alpha}{\left(\frac{\alpha^2}{2}\right)} = 2$

another way to increase SQNR

Companding = non-uniform Quantization

ΔV is not constant

in telephony

US system is called μ-Law

europe system " " A-Law

Purpose improve SQNR without adding levels

Mapping → converting Voltage level to bits

RLE
↳ used in Video to eleminate redundancy

huffman coding
↳ Popular bits (colors) are assigned with Less bits so the Average is low.

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28 - 10

L-11

there is no Perfect line Code.

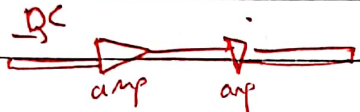
5.b → Summery

Clock recovery in digital

↳ freq of tx = freq of Rx

↳ like manchester line code

1- why DC not ~~used~~ wanted



If you send DC you might make amp ~~[illegible]~~ malfunction

M-ary code

you send 2 bit / symbol

00 → -10 V

01 → -5 V

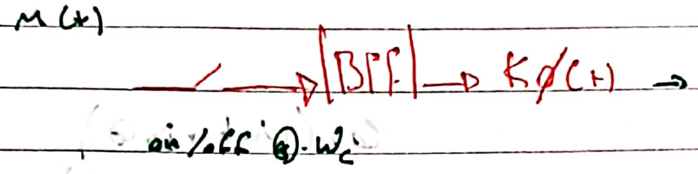
10 → 5 V

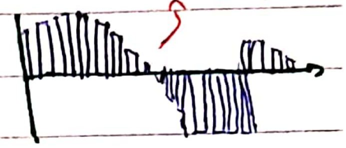
11 → 10 V

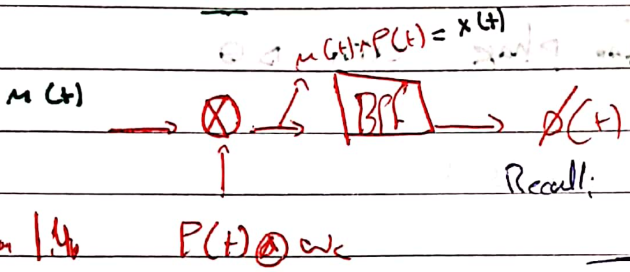
Mixer can use any VGAmp but gilbert used Differential VGA

the 3rd way to build mixer
uses diodes

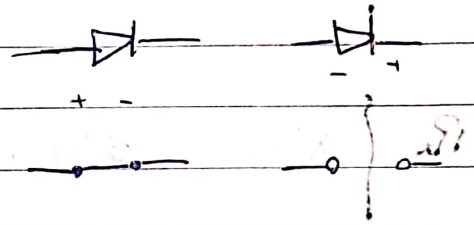
Switched signal = Natural Sampling



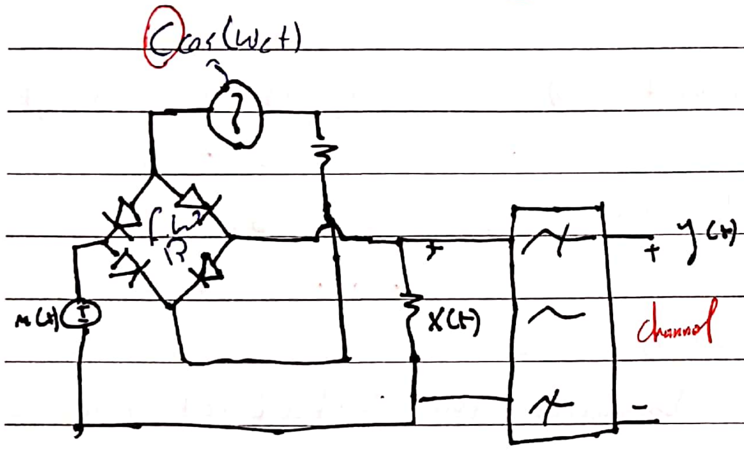




Recall: forward reverse



more than 1.4v



during +ve half cycle all diodes are forward biased , $V_{D} = 0V$ $x(t) = m(t)$

" -ve " " " reverse biased a and b open crkt

$x(t) = 0V$

30 - 10

L-12 not in (st) Synchronizing Oscillator at the Rx

- the Oscilator at Rx should have the same W_c of the Tx
  - ↳ hard to do in real life
  - ↳ components are not perfect.

2 - errors in Rx oscillator.

1- Phase error $\cos(W_c t + \Delta\theta)$

2- Frequency error $\cos((W_c + \Delta W)t)$

ex1, find $y(t)$ for phase error $\Delta\theta$

Tx { $\phi(t) = m(t) \cdot \cos(W_c t)$

Rx { $x(t) = \phi(t) \cdot \cos(W_c t + \Delta\theta) = m(t)\cos(W_c t) \cdot \cos(W_c t + \Delta\theta)$

$x(t) = \frac{1}{2} m(t) \cos(2W_c t + \Delta\theta) + \frac{1}{2} m(t) \cos(\Delta\theta)$

High freq   Low freq

$y(t) = \frac{1}{2} m(t) \cos(\Delta\theta)$

↳ Lowering the value (acts like attenuation)

if $0 < \theta < 90$

L attenuation

if $\theta = 90$

↳ Loss of signal

So phase error = attenuation (can't use AMP) because θ is not constant

30-10 Communication I

Problem 2- Freq. error of L.O of Rx

$$T_x \left\{ \phi(t) = m(t) \cos(\omega_c t) \right.$$

$$R_x \left\{ \begin{aligned} x(t) &= \phi(t) \cdot c(t) \\ &= m(t) \cos(\omega_c t) \cdot \cos((\omega_c + \Delta\omega) t) \\ &= \tfrac{1}{2} m(t) \cos(2[\omega_c + \Delta\omega] t) + \tfrac{1}{2} m(t) \cos(\Delta\omega t) \end{aligned} \right.$$

LPF X h.f V. L.f

$$y(t) = \tfrac{1}{2} m(t) \cos(\Delta\omega t)$$

it creates distortion

h.w: Phase + freq errors

Sol: to Phase and freq error (95%)

1- use PLL

↳ complex and expensive

↳ adv → Power efficient synchronizer

2- let the T_x send another cos ~~wave~~ with the signal (eg: DSB-LC) (5%)

↳ asynchronous and power inefficient

↳ ~~Complex~~ simple and cheap

chapter 6 (QAM)(SSB)

QAM) - Quadrature amplitude modulation

uses orthogonality

(QAM) = DSB-SC with extra orthogonality

Average Value of $x(t) \cdot y(t) = 0$

$\therefore$ $x(t)$ and $y(t)$ are orthognal.

in modulation:

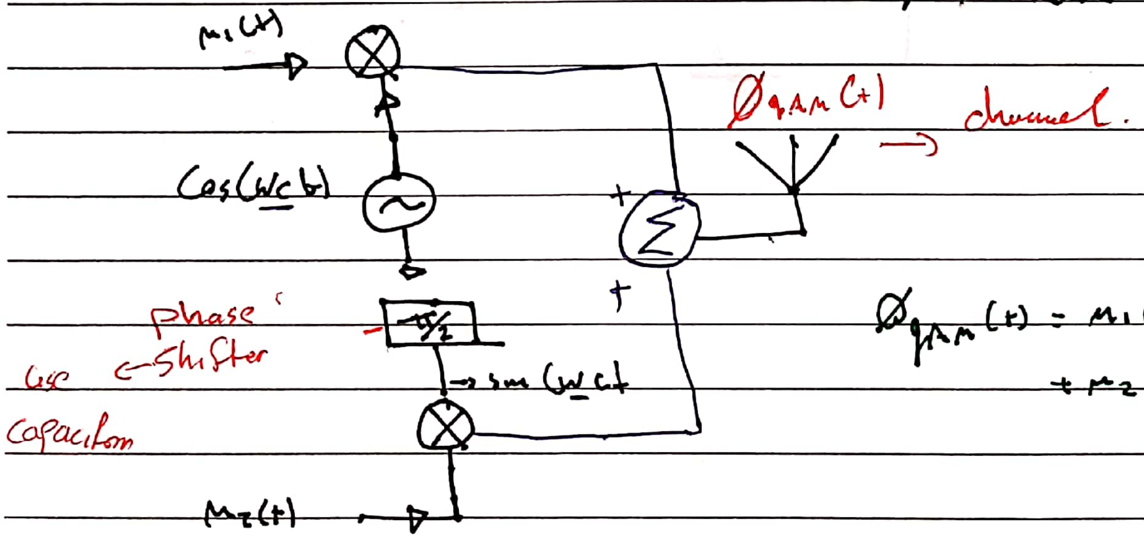
- QAM modulation (like sin/cos)
- used in 4G - 3G, wi-fi, wiMAX, DVB

in multiplexing:

- CDMA (Walsh codes, Gold codes) = code division multiple Access
- used in 3G - cellular telephony

- OFDMA (multiple cosines) = orthogonal freq devision multiple access
- used in Wi-Fi, Wimax 4G, 5G
- Very Popular.

cos and sin must have same w_c



$\phi_{QAM}(t) = m_1(t)\cos(w_c t) + m_2(t)\sin(w_c t)$

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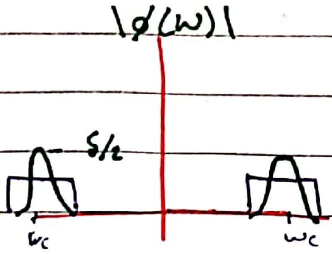
$$\phi_{QAM}(W) = F\{\phi(t)\} = F\{m_1(t)\cos(w_c t) + m_2(t)\sin(w_c t)\}$$

$$= F\{m_1(t)\cos(w_c t)\} + F\{m_2(t)\sin(w_c t)\}$$

$$\frac{1}{2}M_1(W - W_c) + \frac{1}{2}M_1(W + W_c)$$
$$+ \frac{-j}{2}M_2(W - W_c) + \frac{j}{2}M_2(W + W_c)$$



Same hardware of Tx and Rx but Rx has 2 more LPF

using orthogonality We can get our signals Back

$$\phi(t) = m_1(t)\cos(w_c t) + m_2(t)\sin(w_c t)$$

$$x(t) = \phi(t) \cdot \cos(w_c t)$$

$$= m_1(t)\cos^2(w_c t) + m_2(t)\underbrace{\sin(w_c t)\cos(w_c t)}_{\text{orthogonality} = 0}$$

$$= \underbrace{\frac{1}{2}m_1(t)}_{\text{LPF} \checkmark} + \underbrace{\frac{1}{2}m_1(t)\cos(2w_c t)}_{\text{LPF} \times} + \underbrace{\frac{1}{2}m_2(t)\sin(2w_c t)}_{\text{LPF} \times}$$

Quadrature is limited to only 2 signals.

Heterodyning (frequency conversion) * we don't go from baseband to carrier

* we go to an intermediate frequency

$W_m \to W_I \to W_c$

• its cheaper.

Lecture 7: Heterodyning
(*Up & Down Frequency Converter*)

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Electrical Engineering Department
University of Jordan

Heterodyne: Multiple Frequencies

- Typical **transmitters** do not modulate immediately from baseband to carrier frequency ω_c. Rather, they modulate to an *intermediate frequency* ω_I, then an up-converter shifts the frequency to the higher frequency ω_c.
- Also, real-life **receivers** do not demodulate immediately from carrier frequency ω_c to baseband. Rather, they use a down-converter to shift the modulated signal to an *intermediate frequency* ω_I, then demodulate to baseband.
- This has advantages, especially in FDM systems and digital systems (see: super-heterodyne receiver).

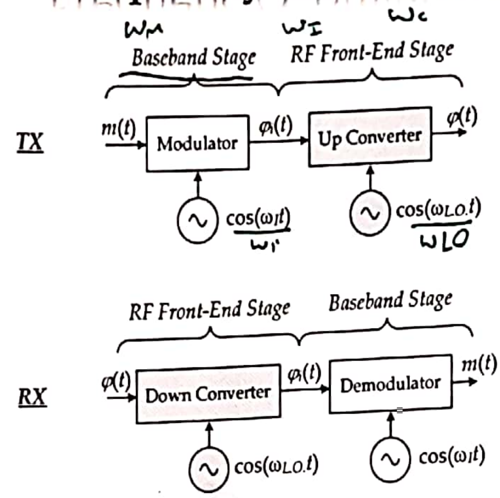
RF: radio frequency (high freq)

Transmitter ≠ modulator.

Frequency Conversion

up converter = heterodyner

heterodyner ≠ modulator



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heterodyner, modulator, demodulator might use the same type of devices

Be careful!

- A frequency converter is also commonly called a **mixer**, but do not confuse it with a multiplication device.
- A frequency converter is **not** a demodulator. ≠ demodulator
- A frequency converter is **not** a modulator.
- **Up converter** takes you from *low* input frequency to *high* output frequency.
- **Down converter** takes you from *high* input frequency to *low* output frequency.

mixer is not Gilbert cell

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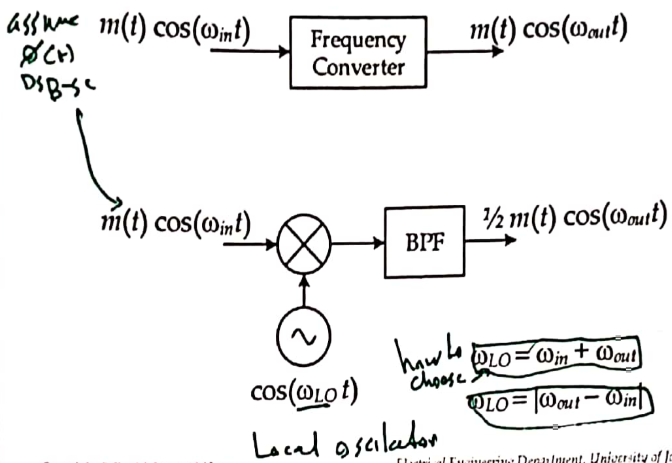
hw: repeat for QAM, SSB

Frequency Converter Hardware



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Examples

Input frequency f_{in}	Output frequency f_{out}	Device Type	L.O. frequency	
300 MHz	100 MHz	Down converter (*sum*)	400 MHz	solve
300 MHz	100 MHz	Down converter (*difference*)	200 MHz	HW
100 MHz	300 MHz	Up converter (*sum*)	400 MHz	HW
100 MHz	300 MHz	Up converter (*difference*)	200 MHz	HW

Camera Tx
300 → 100

in Tx
100 → 300

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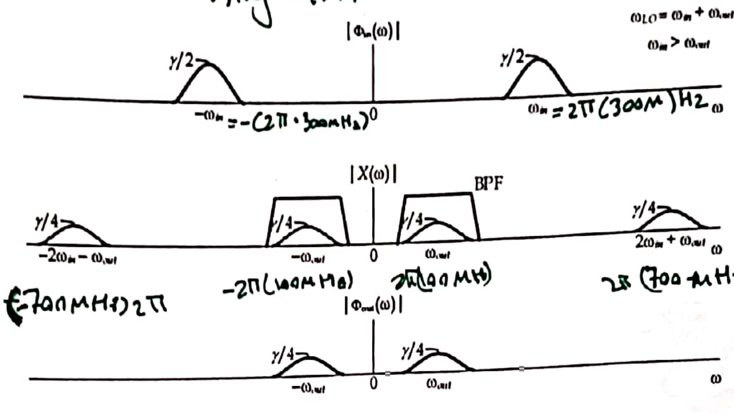
The specification of the BPF defines what the converter is doing

Frequency Converter
(Down Converter, Sum)

using 400MHz



$\Rightarrow x \cos(\omega_{LO} t)$

$\omega_{lo} = \omega_{in} + \omega_{out}$

$= (400 + 300)\, 2\pi Hz$

BPF specs

1- ω_c (60 MHz) 2π

2- $BW = 2 BW_{in}$ Hz

$= 2\pi BW_{in}$ rad/s

3- gain

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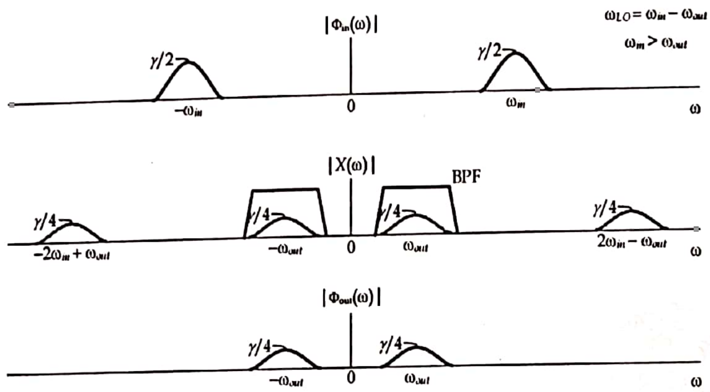
Frequency Converter
(Down Converter, Difference)



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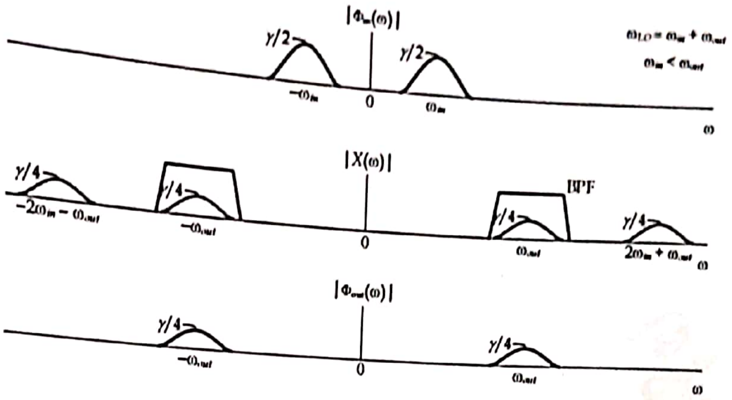
Frequency Converter
(Up Converter, Sum)



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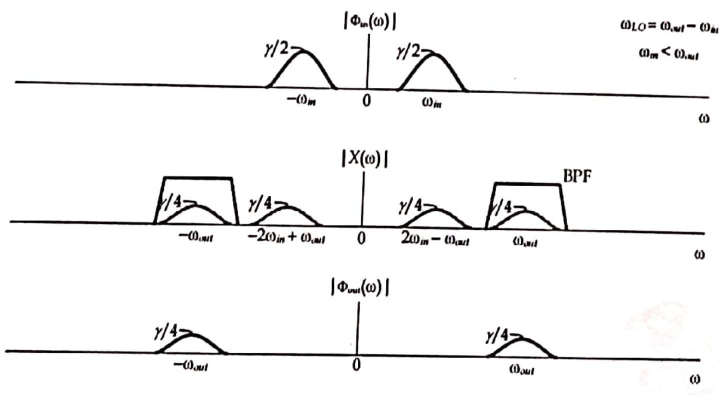
Frequency Converter
(Up Converter, Difference)



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ملاحظة

Homework 1

- Repeat the four cases above for SSB-SC (USB) input modulated signal:
  - Up converter, Sum
  - Up converter, Difference
  - Down converter, Sum
  - Down converter, Difference
- Find k in the output signal:
  $y(t) = k\, \varphi_{SSB-SC}(t)$
- Provide specifications for the BPF to be used.

Homework 2

- Does the frequency converter (mixer) work if you use $p(t)$ instead of $\cos(\omega_{LO}t)$?
- If so, what is the necessary frequency(ies) for $p(t)$?
- Find k in the output signal:
  $y(t) = k\, m(t) \cos(\omega_{out}t)$
- Provide specifications for the BPF to be used.

11/11/2019

Comm I — L-8 in slides

SNR

Chapter 8 — Signal to noise Power (Ratio)

$$SNR = \frac{\text{Signal Power}}{\text{noise Power}} = \frac{P_s}{P_n} \text{ (unitless)}$$

- noise : alteration and unpredictable signals affecting the message by the channel

- SNR decides Quality in analog system

- in digital systems SNR decides bit error rate (BER) so Quality

BER for typical digital system $BER \leq 10^{-6}$

$BER = 10^{-3}$ → Bad Quality

$$SNR = \frac{\text{Signal Power}}{\text{noise Power}} = \frac{P_s}{P_n} \text{ (unit less)} \rightarrow \text{not used}$$

$$SNR = 10 \times \log_{10}\left(\frac{P_s}{P_n}\right) \text{ db} \rightarrow \text{used}$$

$P_s = \frac{V_s^2}{R}$, $P_n = \frac{V_n^2}{R}$

in electronics $= 20 \log_{10}\left(\frac{V_s}{V_n}\right)$ not used in Comm

Five Apple

memorize the scale — db use in: 1- SNR
2- attenuation = $\frac{P_{out}}{P_{in}} < 1$
3- amplification $\frac{P_{out}}{P_{in}} > 1$

1- db . dbm = dbm
unitless × W = W

2- dB . dB = dB
unitless . unitless = unitless

3- dBm + dbW = dBW
unitless . W = W

Practical value of SNR — Performance = SNR

- for Voice signals
  - 5db < SNR < 10 db → can't hear
  - 25db < SNR < 35 db → telephony quality.
  - Minimum SNR for Voice = 30 db → good quality
- for Video

  SNR ≥ 50 db, eyes more sensitive

- for digital SNR ⇒ BER = 10^{-6} → good quality

to find P_s → 1- $\lim_{T\to\infty} \frac{1}{T} \int_{T} x^2(t)\,dt$

Power of Signal

2- $\frac{1}{2\pi} \int_{-\infty}^{\infty} S_x(\omega)$

P_n?? → we can't find P_n because its random
→ we know the Probability

the most popular noise model used in Comm is AWGN

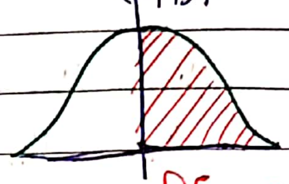
AWGN noise → easiest
→ fits well with thermal noise

additive white Gaussian noise

→ with mean $= \mu = 0 \Rightarrow$ DC
→ Variance $= \sigma^2 \Rightarrow$ avg power $=$ rms^2

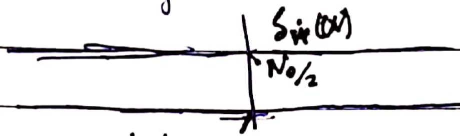
noise in time domain

$m(t) \rightarrow \bigoplus \rightarrow m(t) + n(t)$



area $= P[n > 0]$

→ why white?



Power spec density ⇒ for thermal noise

Simplest representation:

$$P_n = \frac{1}{2\pi}\int_{-\infty}^{\infty} S_n(\omega)\,d\omega$$ → every ω has noise power

$N_0 = 4KT$, $T =$ temp $K°$, $K =$ Boltzmann Constant $= 1.38 \times 10^{-23}$ J/K

$$S_n(\omega) = \frac{N_0}{2}$$

ex

Sol 1: $P_{tx} = 1000W = 30$ dbW

$\Delta tt = -5$ db

$P_{rx} = 30$ dbW $+ \underbrace{(-5db \times 10)}_{\Delta tt}$

$= -20$ dbW

Sol 2: $P_{tx} = 1000w$

$= 60$ dbm

$P_{rx} = 60 + (-5db) \times 10$

$= 10$ dbm

Sol 3

Power of channel noise $= \infty$

so SNR at the out of the channel $= 0$ always

$$SNR_{channel} = \frac{\overline{K^2 m^2(t)}}{\overline{n^2(t)}} = \frac{10\,mW}{\infty}$$

$\therefore \overline{n^2(t)} = \infty$

after placing filter most of the Power is disgarded in of the unwanted Frequency

$$\frac{1}{2\pi}\int_{-\infty}^{\infty} S_{no}(\omega)\, d\omega = \frac{1}{2\pi}\int_{-2\pi B}^{2\pi B} \frac{N_o}{2}\, d\omega = \frac{1}{2\pi}\left[\text{area } \pm ve\right]\frac{N_o}{2}$$

$$= \frac{1}{\cancel{2\pi}} \cdot \cancel{2} \cdot \cancel{2\pi} B \cdot \frac{N_o}{\cancel{2}}$$

- Amplifier at the R_x is not good because the noise will also be used and amplified

$$= B N_o = N_o B_{LPF} \text{ Watt}$$

$$= 4\times10^{-9} \frac{\text{watt}}{\cancel{Hz}} \cdot 4\times10^{3} \cancel{Hz}$$

$$= 16 \times 10^{-6} = 16\,\mu w$$