اسم المادة: الاتصالات COMMUNICATIONS

المرحلة : الثانية

القسم: التقنيات الالكترونية

الساعات المقررة في السنة: 60 ساعة نظري و 60 ساعة عملي

لغة التدريس: اللغة الانكليزية

اعداد المادة:

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المعهد التقني

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وزارة التعليم العالي والبحث العلمي
هيئة التعليم التقني
الجامعة التقنية الشمالية
التخصصات الهندسية القسم التقنيات الالكترونية

اسم المادة	السنة الدراسية	الساعات الأسبوعية		
الاتصالات	الثانية	ن	ع	م
لغة التدريس	الإنكليزية	2	2	4

الهدف العام: تزويد الطالب بالمعلومات الأساسية لأنظمة الاتصالات السلكية واللاسلكية .
الهدف الخاص : تزويد الطالب بالمعلومات الكاملة عن :

- نظم وتراكيب المنظومات الإذاعية والتلفازية والهاتفية.
- طرق نقل المعلومات في نظم الاتصالات ومواصفاتها ومميزاتها والعمليات التي تجري عليها.

المفردات النظرية

الأسبوع	تفاصيل المفردات النظرية
الأول	المرشحات– مرشحات(RC):(BSF)- (LPF)-(HPF)-(BPF)
الثاني	المرشحات الفعالة: (BSF)- (LPF)-(HPF)-(BPF)
الثالث	التضمين –معناه –أنواعه—تضمين (AM) تحليل الموجه.
الرابع	الطيف الترددي-توزيع القدرة – حساب معامل التضمين المكافىء.
الخامس	أنواع التضمين الاتساع(AM) مع طيفها الترددي
السادس	أنواع المضمنات المستخدمة لتوليد (AM) المضمن المتوازن –المضمن الحلقي-مضمن كوين-مضمنات أخرى.
السابع	كشف تضمين(AM) –Envelope Detector – Synchronous Detector التشويه في دوائر الكشف – (AGC)
الثامن	مخطط كتلي لجهاز إرسال واستقبال الموجه المضمنه اتساعيا –معاملات مقارنة اتساعية أجهزة الاستقبال (الحساسية- الانتقائية-الجودة-التشويه).
التاسع	التضمين الترددي (FM) تضمين (PM)-التحليل الرياضي للموجات المضمنه – نسبة التضمين –الانحراف الترددي.
العاشر	عرض النطاق الترددي للإرسال والطيف الترددي لتضمين (PM) و(FM) .
الحادي عشر	طرق تضمين (FM) وتوليدها –الطريقة المباشرة , الطريقة الغير مباشرة التضمين الترددي المضخم (Secttreo FM)- Stereo
الثاني عشر	الكشف لإشارة (FM) –الكاشف النسبي –طريقة فوسترسلي.
الثالث عشر	الترميز-نظرية العينات (Quantization) –ترميز التحويل .
الرابع عشر	تضمين (PM) –مميزات التضمين النبضي –الأنواع (PCM)-(PPM)-(PDM)-(PAM).
الخامس عشر	التوزيع (Multiplexing) –(FDM) –(TDM).
السادس عشر	التضمين الرقمي PSK-FSK-ASK.
السابع عشر	معلومات الإرسال وسعة المنظومة –الخطأ (SNR) نسبة الإشارة للضوضاء

الثامن عشر	الهواتف الخليوية-الترددات المستخدمه –التقنيات المستخدمة (FDMA)-(TDMA) –(CDMA) .
التاسع عشر	دوائر التلغراف –(Teleprinters) –مرسلات التلغراف الراديوية.
العشرون	(FaximileTransmission) –(Fas-Receiver)-(Telex) المرسلة المستقبلة.
الحادي والعشرون	الألياف البصرية- أنواعها- صفاتها- المرسلة والمستقبلة .
الثاني والعشرون	أنواع الهوائيات –أساسيات الهوائيات-معاملات الهوائيات.
الثالث والعشرون	انتشار الموجات الراديوية(الأرضية –السماوية-موجات خط البصر.
الرابع والعشرون	الهوائيات العمودية-هوائيات قضيب الفرايت-هوائيات UHF الهوائيات المايكروية والبوقية.
الخامس والعشرون	استخدام المايكروويف في الاتصالات.
السادس والعشرون	الاتصالات بالأقمار الصناعية-المميزات والخواص-الإرسال والاستقبال-المحطات الأرضية.-مدارات الأقمار الصناعية –الدخول المتعدد Multiple Access.
السابع والعشرون	الموجات المايكروية-توليدها-الطيف الترددي..
الثامن والعشرون	الموبايل-مقدمة –التقنيات المستخدمة-أهم الاعتبارات في النقل-الظل-التداخل-الضوضاء.- نقل الإشارات لاسلكياً –لاسلكياً (ولاسلكياً-سلكياً)
التاسع والعشرون	شبكات GSM ; الوظائف والهيكلية
الثلاثون	الثريا-خدمات الثريا-خصائص الثريا-SMS-استخدامات الثريا-المناطق الجغرافية لتغطية خدمة الشبكات.

Introduction to Communication System

- A *Communication system* conveys information from its source to a destination some distance away. Fig. 1 shows the basic functional blocks of a communication system. Thus, the overall purpose of this system is to transfer information from one point in space and time called the *"Source"* to another point in space and time called *"User"* or *"Destination"*.

- As a rule, the message produced by the source is not electrical. Hence an input; a • *"transducer"*, is required for converting the message to a time - varying electrical quantity called a *"message signal"*. At the destination point, another transducer is used to convert the electrical waveform to the appropriate message.

- Signal processing operations performed by the *"transmitter"* include *"Amplification"*, *"Filtering"*, and *"Modulation"*. *"Modulation"* is the process designed to match the properties of the transmitted signals to the channel through the use of a carrier wave. Thus the modulation is the variation of one of the parameters of the carrier wave such as amplitude, frequency, or phase in accordance with the message signal.

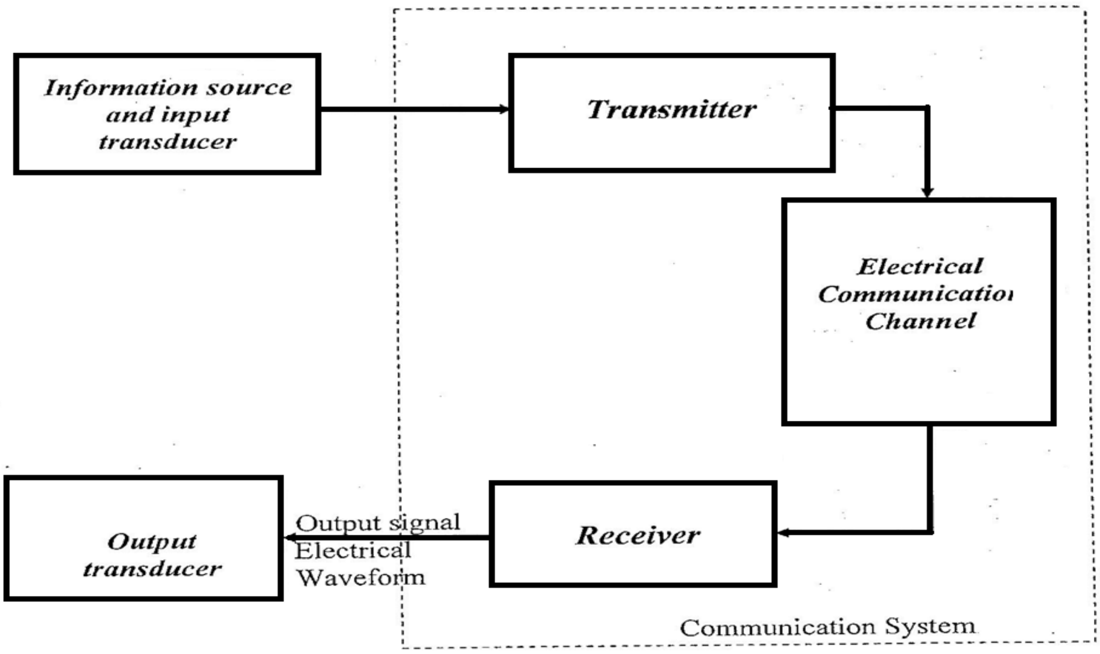


Fig. (1) Model of Electrical Communication System

- It is possible to identify two basic types of modulation; the continuous carrier-wave (CW) modulation and pulse modulation.

"Noise" and *"signal distortion"* are two basic problems of electrical communication. The transmitter and the receiver are carefully designed to avoid signal distortion and minimize the effects of noise at the receiver.

Electrical signals may have one of the following forms shown in Fig. (2).

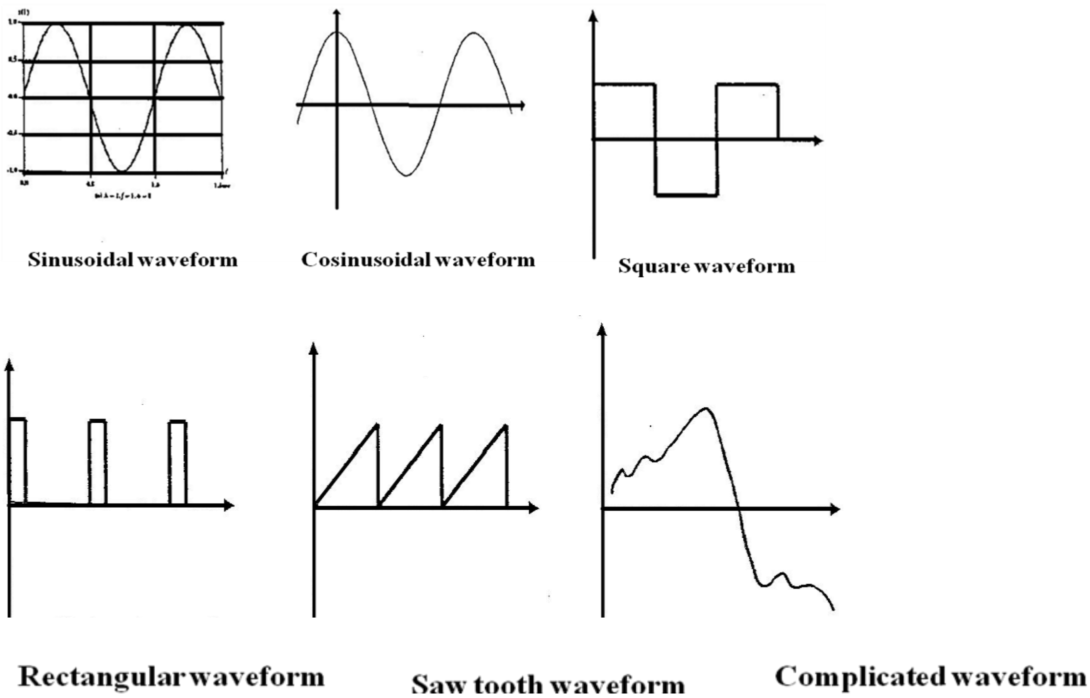
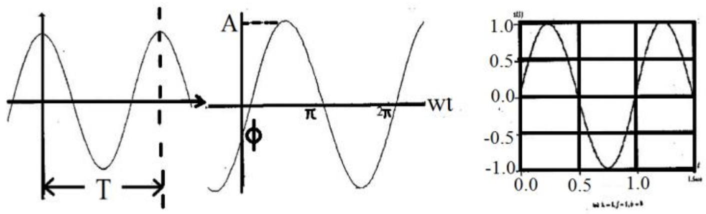


Fig. (2) Some forms of Electrical Signals

Signals and Line Spectrum

- The most familiar one is the cosine waveform which is expressed as (see Fig (3):

$$v(t) = A\cos(w_0 t + \theta)............(1)$$



$\phi = 0$ $\quad 0 < \phi < \frac{\pi}{2}$ $\quad \phi = \frac{\pi}{2}$

Fig. (3) Time- domain representation for

Signals and Line Spectrum

$$v(t) = A\cos(w_0 t + \theta)$$

Equation (1) is a time-domain mathematical representation for cosine wave. The independent variable in this equation is the time "t". The parameters of this signal are:

1) The *amplitude* "A" in volts or amps.

2) The *radian frequency wo* in radians /sec, it is equal to *wo* = *2* f_0. The time of one cycle or period T is (see Fig. (3):

$$f_o = \frac{1}{T_o} \text{............}(2)$$

3) The *phase angle* *"θ"* measured in radian or degrees, is the phase angle of the signal at t=0. When θ = 0, the expression of(1) is reduced to

$$v(t) = A\cos(w_0 t)$$

And when $\phi = \frac{\pi}{2}$, the signal will be sine wave, i.e.:

$$v(t) = A\sin(w_0 t)$$

(WEEKS 1-2)
Filters

Types of Filters

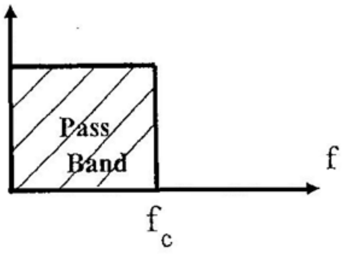
The filters networks are divided into the following four main classes:

1. Low Pass Filter (LPF)
2. High Pass Filter (HPF)
3. Band Pass Filter (BPF)
4. Band Stop Filter (BSF)

Low Pass Filter (LPF)

These filters are designed to transmit signals of all frequencies from zero to certain critical or *cut-off frequency* "fc " without attenuation (Fig, 4a). All frequencies higher than "fc " are attenuated.

Fig. 4a

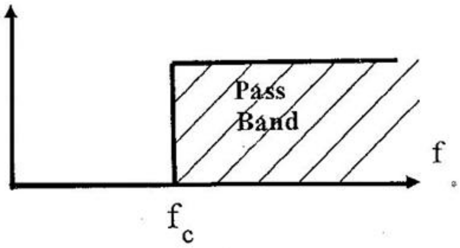


(a)

High Pass Filter (HPF)

These filters transmit signals of all frequencies lying between infinity and the *cut-off frequency* "fc " without attenuation. All other frequencies below "fc " are attenuated (Fig. 4b).

Fig. 4b

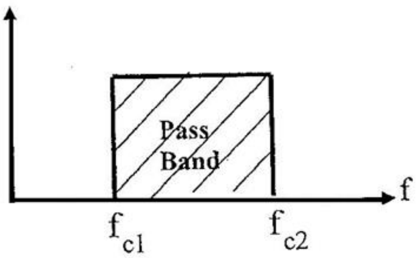


(b)

Band Pass Filter (BPF)

These filters are designed to transmit a specified band of frequencies lying between two *cut-off frequencies* "f_{c1}" and "f_{c2}" and to attenuate all other frequencies outside this band (Fig 4c).

Fig. 4c

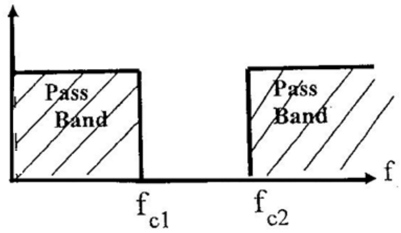


(c)

Band Stop Filter (BSF)

These filters transmit all frequencies except those frequencies lying between *two cut-off frequencies* " f_{c1} " and " f_{c2} " (Fig. 4d).

Fig. 4d



(d)

Filter Classification

1) Passive Filters
2) Active Filters

Passive filters use passive elements such as resisters, inductors, and capacitors connected in a manner so that they perform filtering. These elements need no power supply to work. Active filters use passive and active elements such as *transistors or operational amplifier (OP AMP).* Active dements need power supply to get functioning.

R-L Passive Filters

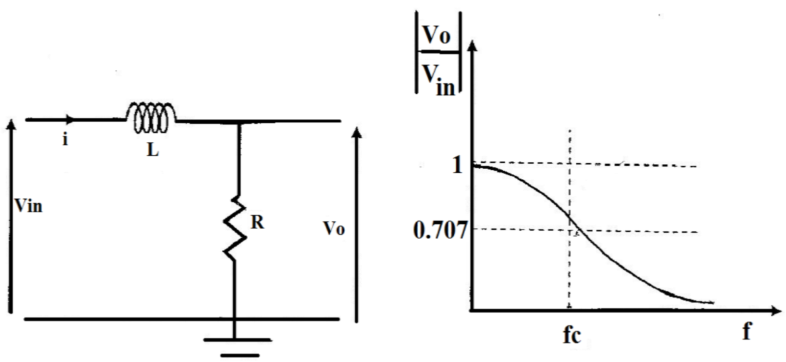
The first order of these passive filters consists of an inductor having inductance of "L" Henry connected in series with a resistor having resistance of "R" ohm.

Consider the network shown in Fig. 5. The impedance of this network (as a magnitude) which is seen by the input voltage is:

$$|Z| = \sqrt{R^2 + (X_L)^2} = \sqrt{R^2 + (2\pi f L)^2}\,\Omega$$

$$v_o = iR$$

and $v_{in} = iZ$



(a) Circuit

(b) Characteristic

Fig.(5) R-L Passive Low Pass Filter

R-L Low Pass Filter

The magnitude of transfer function $\frac{v_o}{v_{in}} = \left|\frac{v_o}{v_{in}}\right| \angle\theta^o = \left|\frac{v_o}{v_{in}}\right| \exp^{j\theta}$ equal to :

$$\left|\frac{v_o}{v_{in}}\right| = \frac{iR}{i|Z|} = \frac{R}{\sqrt{R^2 + (2\pi fL)^2}} = \frac{1}{\sqrt{1+\left[\frac{2\pi fL}{R}\right]^2}} \text{..........(5)}h$$

When $f = f_c$, the output power will be half the input power or :

$$v_o = \frac{v_{in}}{\sqrt{2}} = 0.707\, v_{in} \quad \text{or} \quad f_c = \frac{R}{2\pi L} \text{............(6)}$$

$$\therefore 1 + \left[\frac{2\pi f_c L}{R}\right]^2 = 2$$

is the *cut – off frequency*, and

$$\left|\frac{V_o}{V_{in}}\right| = \frac{1}{\sqrt{1+\left[\frac{f}{f_c}\right]^2}} \text{..........(7)}$$

At low frequencies, (2πfL), the reactance of the inductor will have small values, therefore V = Vin approximately. As the frequency of the input signals increases, (2πfL) increases and therefore signals with frequencies larger than fc will be attenuated, i.e. V =0 approximately.
Accordingly this network is *R-L Low Pass Filter (LPF).* The phase shift between the input and the output:

$$\theta = -\tan^{-1}\left[\frac{f}{f_c}\right] \text{.............(8)}$$

R-L Passive High Pass Filter

If the elements in the network of *R-L Low Pass Filter* are interchanged, the resulting network will behave as *R-L High Pass Filter (HPF).* The circuit and magnitude characteristic of the *transfer function* of such filter is shown in Fig.(6). To deduce the transfer function of this filter, we can write:

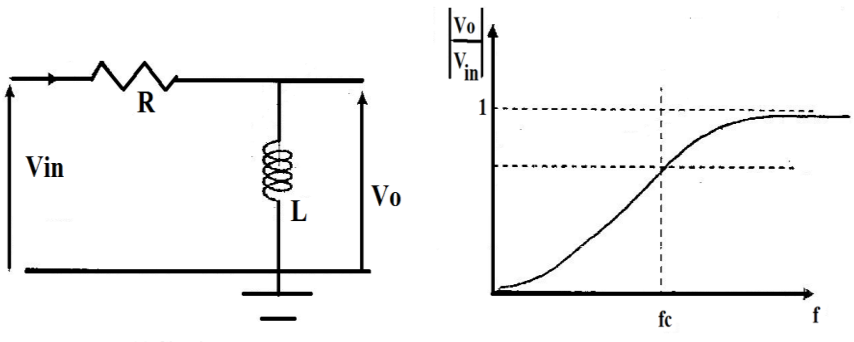


Fig.(6) R-L Passive High Pass Filter

$$v_O = i * Z_L = i * (jX_L) = i * j2\pi fL$$
$$v_{in} = i * Z = i * (R + j2\pi fL)$$
$$\frac{v_O}{v_{in}} = \frac{jwL}{R + jwL} \quad \text{or}$$
$$\left|\frac{v_O}{v_{in}}\right| = \frac{2\pi fL}{\sqrt{R^2 + (2\pi fL)^2}} = \frac{f}{\sqrt{f^2 + f_c^2}} = \frac{1}{\sqrt{1 + \left[\frac{f_c}{f}\right]^2}} \text{.........(9)}$$

Also

$$f_C = \frac{R}{2\pi L} \text{ is the } \textit{cut-off frequency} \text{ of this filter}$$

At low frequencies, the inductor acts as short circuit due to its low reactance, therefore Vo=0 approximately. When signals with frequencies larger than the cut-off frequency of this filter are applied at the input of this filter, the reactance (2 f L) starts to increase and the input voltage will appear across it, i.e. Vo = Vin approximately. Accordingly this network acts as *high pass filter*. The phase shift between the input voltage and output voltage is:

$$\theta = 90 - \tan^{-1}\left[\frac{f}{f_c}\right] \text{.............(10)}$$

<u>R-C Passive Filters</u>

Filters using only capacitors and resistors are very convenient from a circuit design viewpoint since they are cheap and compact and have a lower space-to-weight ratio than do inductors. Moreover they have good characteristics and lend themselves to integrated circuits techniques.

R-C Passive Low Pass Filter

The network of *R-C Low Pass Filter (LPF)* together with its characteristic is shown in Fig.(7). Again, the transfer function can be written as:

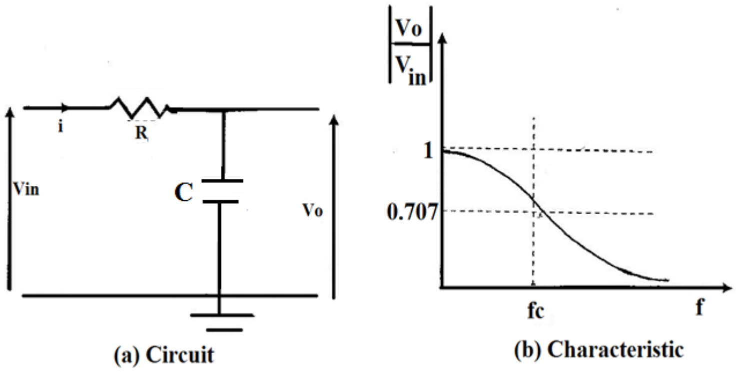


Fig.(7) R-C Passive Low Pass Filter

$$v_o = i * Z_c = i * (jX_c) = i * \frac{1}{j2\pi fC}$$

$$v_{in} = i * Z = i * (R + \frac{1}{j2\pi fC})$$

$$\frac{v_o}{v_{in}} = \frac{1}{1 + j2\pi fCR} \quad \text{or}$$

$$\left|\frac{v_o}{v_{in}}\right| = \frac{2\pi fL}{\sqrt{1 + (2\pi fCR)^2}} = \frac{1}{\sqrt{1 + \left[\frac{f}{f_c}\right]^2}} \text{..........(11)}$$

Also

$f_c = \dfrac{1}{2\pi RC}$ is the *cut-off frequency* of this filter at which

$$\text{Vo} = \frac{\text{Vin}}{\sqrt{2}} = 0.707Vin \text{ (i.e. when } 2\pi f_c RC = 1)$$

At low frequencies the term $\dfrac{1}{j2\pi fC}$ has very large values, therefore Vo= Vin approximately. (i.e. (Vo/Vin)=l), while it has very small values as frequency increases more than (fc), i.e. Vo= 0 approximately. Accordingly this network acts as *Low Pass Filter*. The phase shift is:

$$\theta = -\tan^{-1}\left[\frac{f}{f_c}\right] \text{.............(12)}$$

R-C Passive High Pass Filter

When the elements in the network of Fig. (8) are interchanged, the resulting network will be *R-C High Pass Filter (HPF)* which is depicted together with its characteristic in Fig. (8-b). As before the *transfer function* of this filter can be written as:

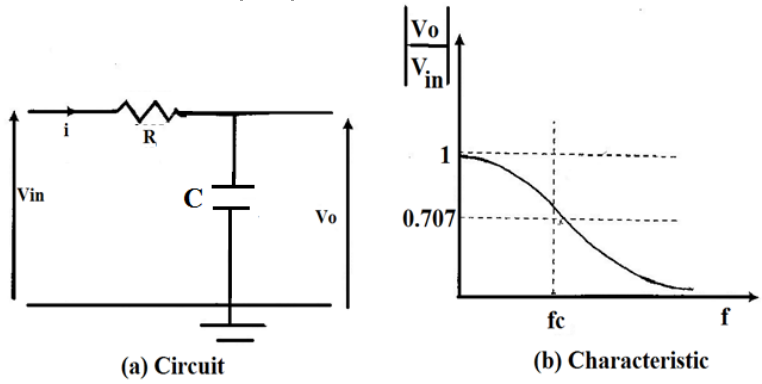


Fig.(8) R-C Passive Low Pass Filter

$$\frac{v_o}{v_{in}} = \frac{R}{R + \frac{1}{j2\pi fC}} = \frac{j2\pi fCR}{1 + j2\pi fCR}$$

$$\left|\frac{v_o}{v_{in}}\right| = \frac{2\pi fCR}{\sqrt{1 + (2\pi fCR)^2}} = \frac{1}{\sqrt{1 + \left[\frac{f_c}{f}\right]^2}} \text{..........(12)}$$

$$At\ \mathrm{f} = \mathrm{f_c} \Rightarrow V_o = 0.707Vin \text{ or}$$

$$1 + \frac{1}{2\pi f_c RC} = 2$$

$$f_c = \frac{1}{2\pi RC}$$

That is the expression of the cut-off frequency of the *R-C High Pass Filter* is the same as for the cut-off frequency expression of *R-C Low Pass Filter*. The phase shift is:

$$\theta = 90^{\mathrm{o}} - \tan^{-1}\left[\frac{f}{f_c}\right] \text{..............(13)}$$

Solved Problem

P.1) What is the type and cut-off frequency of each of the following filters

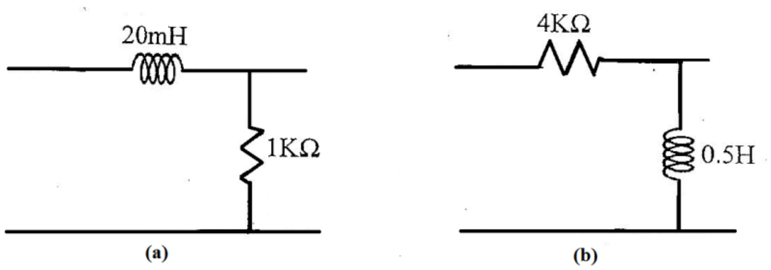


Fig. 9

Solution

- Network a) is R-L Low Pass Filter. Its cut-off frequency is:

$$fc = \frac{R}{2\pi L} = \frac{1000}{2*3.14*20*10^{-3}} \cong 7960\,Hz = 7.96\,KHz$$

- Network b) is R-L High Pass Filter with a cut-off frequency:

$$fc = \frac{R}{2\pi L} = \frac{4000}{2*3.14*0.5} \cong 1247\,Hz = 1{,}247\,KHz$$

Home Works

Calculate the cut-off frequency for each of the following filters:

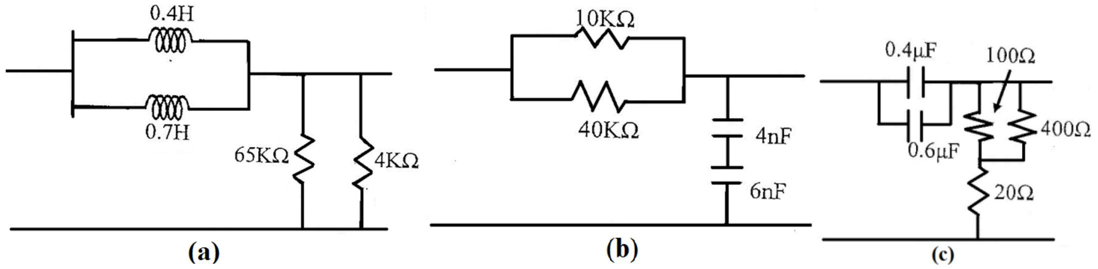


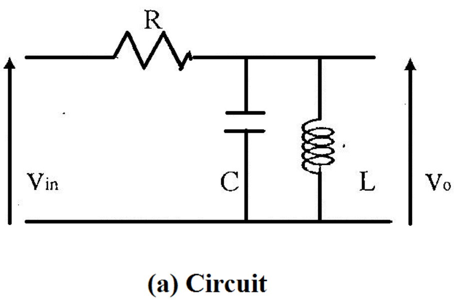
Fig. 10

R-L-C Passive Filters

Filters are constructed from passive elements such as resistors, inductors, and capacitors. They are used to transmit or reject a specific intermediate band of frequencies.

R-L-C Band Pass Filter (BPF)

The network of this filter is shown in Fig.(11). It consists of a resistor connected in series with a parallel combination of inductor and capacitor. This filter is designed to transmit the signals with frequencies that are lying between *two cut-off frequencies fc1* (or $f_{l)}$ and fc2 (or fu).



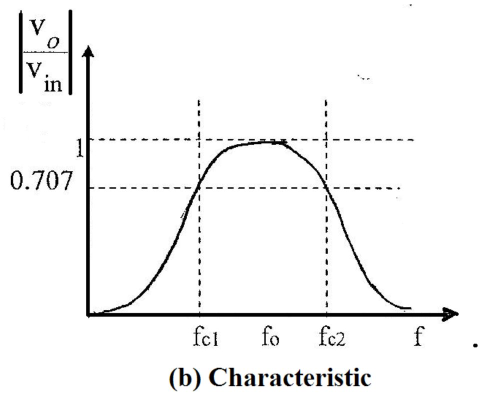


Fig.(11) R-L-C Band Pass Filter (BPF)

fc1 and fc2 are cut-off frequencies at which($V_o = \frac{V_{in}}{\sqrt{2}} = 0.707V_{in}$) and the difference between them is called the Bandwidth (B.W) of this filter.

At low frequencies less than fc1 the inductor short circuited them because of its low reactance. The capacitor will do so at frequencies greater than fc2 for the same reason. In between fc1 and fc2 both reactance's have finite values, so that $V_o \approx V_{in}$.

In addition to the cut-off frequencies and the bandwidth of these filters, there are other two important parameters that characterize these filters. First of them is the *center frequency* (fs), defined as the mid frequency of the bandwidth of the filter at which the reactance of the network vanishes, i.e. the impedance of the network is pure resistive. At this frequency the output voltage is exactly equal to the input voltage. The second parameter is the *Quality Factor.* It is defined as the ratio of the center frequency (fs) to the bandwidth, thus it measures the circuit selectivity.

R-L-C Band Stop Filter (BSF)

If the resistor and the parallel combination in the Fig.12 are interchanged, the resulting network acts as *Band Stop Filter* or *Band Reject Filter.* This filter is designed to transmit all frequencies except those frequencies lying between two cut-off frequencies f_{cl} andf_{c2} as illustrated in Fig.12.

As homework, the student must explain how this network acts as BSF according to the situation of its impedance with the variation of the frequency of the input signal.

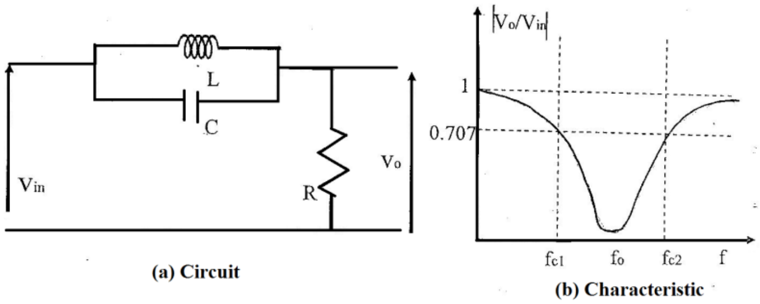


Fig.12 R-L-C Band Stop Filter (BSF)

Solved Problems

Compute the values of L, and C to give a Band Pass Filter shown in Fig.10 with a center frequency (fo) of 2KHz and bandwidth of 500Hz. Use a 250Ω resistor.

Solution

$$Q = \frac{fo}{B.W} = \frac{2000}{500} = 4$$

also

$$Q = R\sqrt{\frac{C}{L}}$$

$$4 = 250 * \sqrt{\frac{C}{L}}$$

$$\therefore \sqrt{C} = 0.016 * \sqrt{L}$$

or

$$C = 2.56 * 10^{-4} L$$

Now

$$fo = \frac{1}{2\pi\sqrt{LC}}$$

or

$$2 * 10^{3} = \frac{1}{2 * 3.14 * \sqrt{L} * 0.016 * \sqrt{L}} = \frac{1}{0.1L}$$

$$\therefore L = \frac{1}{0.1 * 2 * 10^{3}} = 5mH$$

and

$$C = 2.56 * 10^{-4} * 5 * 10^{-3} = 1.28\mu F$$

The circuit will be

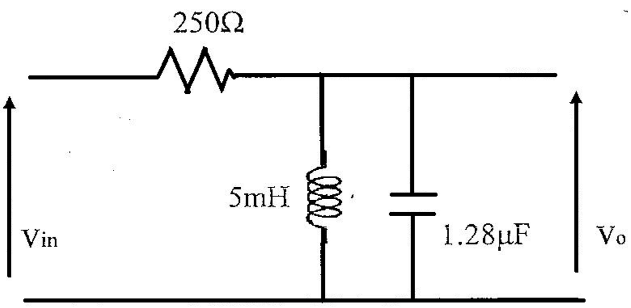


Fig. 13

Active Filters

Active Circuits can produce *Band Pass* and *Band Stop Filters* without using inductors. This desirable because inductors are usually large, heavy, costly, and they may introduce *electromagnetic field* affects that compromise the desired frequency. Besides that, we have noticed from the characteristic of *Passive Filters* that the magnitude of the output does not exceed the magnitude of the input, since the maximum magnitude of the *transfer function* does not exceed one, i.e. *Passive Filters* are incapable of *Amplification. Active Filters* provide control over amplification not available in *Passive Filters* as:

1. Reduction in size, weight, and cost.
2. Easy to design.
3. Increase circuit reliability.
4. Can provide voltage or current gain.
5. Improved performance.
6. Can provide excellent isolation capability

The Operational Amplifier (OP-AMP)

The *OP-AMP* is *Integrated Circuits.* Usually, it has a very high open-loop gain (Ao), Fig. 14 shows the circuit symbol and top-view of the package of an eight-lead DIP OP-AMP. It has two main inputs, inverting input labeled (-)(pin no. 2) and non-inverting input labeled (+)(pin no. 3). The *OP-AMP* is powered from positive and negative D.C voltages through its terminals (+Vcc)(pin no,7) & (-Vcc)(pin no. 4) respectively. The output is usually one terminal. This OP-AMP is the 741 series.

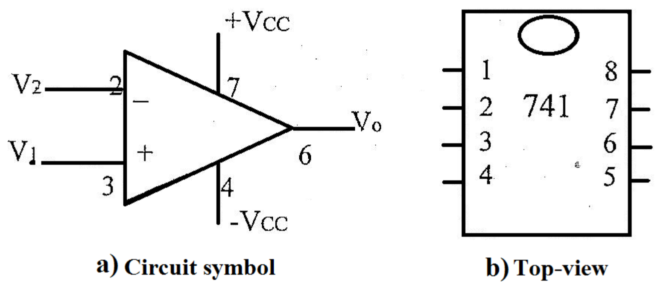


Fig.14. The Operational Amplifier (OP-AMP)

The OP-AMP with difference of the input voltages (V2 - VI) has very high open gain (Ao). i.e.:

$$Vo = -Ao(V_2 - V_1)...............(14)$$

$$\text{With} \quad -Vcc \le Vo \le +Vcc$$

$$\therefore V_2 - V_1 = \frac{-Ao}{Vo}$$

If the OP-AMP is designed to have a very large open loop gain then :

$$V_2 - V_1 \to 0 \quad \text{as} \quad Ao \to \infty$$

$$\text{i.e } V_2 \cong V_1..................(15)$$

Thus, we can conclude from (15) that the current taken by the input impedance (Zin) of the OP-AMP (Zin exist between the two input terminal of the OP-AMP) is almost zero, i.e.
As an application of the OP-AMP, consider the circuit shown in Fig. 15(a).
Since V1 = 0 earth connection V2 ≈ 0 according to (15)
Applying Kirchhoff's current law at node #

$$\therefore \frac{Vin}{Zin} = \frac{Vo}{Z_f}$$

or

$$\frac{Vo}{Vin} = A = \frac{-Z_f}{Zin} \text{..............(16)}$$

Where (A) is the closed-loop gain. If Z_f & Zin are pure resistors then :

$$A = \frac{-R_f}{Rin} \text{........................(17)}$$

Thus, equation (17) illustrates that the closed-loop gain of the amplifier shown in Fig.15 (a) is a ratio of pure resistances and can be controlled by changing either one of these resistors within a limit.

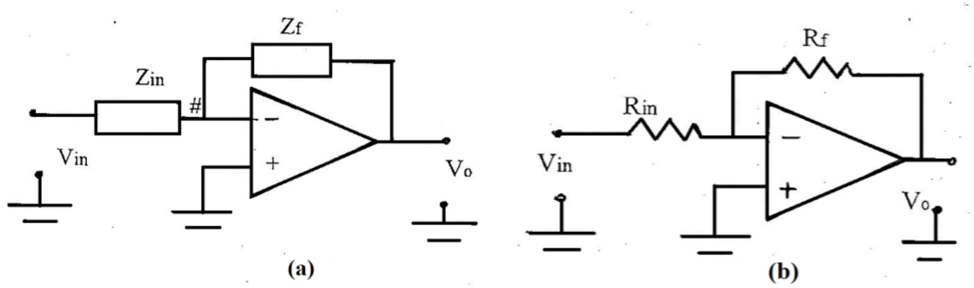


Fig.15 An Amplifier using (OP-AMP)

R-C Active Low Pass Filter

If the resistance (R*f*) in Fig. 15(b) is shunted by a capacitance, the resulting circuit acts as *Active Low Pass Filter*. At low frequencies less than the *cut-off frequency* of this filter, the capacitor will be as an open circuit because of its high reactance at these frequencies, therefore the circuit will transmit these signal to its output with a gain of

$$Z_f = \frac{ZcR_f}{R_f + Zc} = \frac{1}{Rf + \frac{1}{j2\pi fC}} = \frac{R_f}{1 + j2\pi fCR_f}$$

$$Zin = R$$

The closed loopgain(A(jf))

$$\therefore A(jf) = |A(jf)|\angle\theta^o = \frac{-Z_f}{Zin} = \frac{-R_f}{Rin} * \frac{1}{1 + j2\pi fCR_f}$$

$$\therefore |A(jf)| = \frac{A}{\sqrt{1 + (2\pi CfR_f)^2}} \text{............(18)}$$

Where (A) is defined in (17)

$$At\ \mathrm{f} = \mathrm{f_c}, \quad |\mathrm{A(jf)}| = \frac{A}{\sqrt{2}} = 0.707A$$

$$\therefore 2\pi C R_f f_c = 1$$

$$or$$

$$f_c = \frac{1}{2\pi C R_f} \text{..................(19)}$$

$$and$$

$$\theta = -\tan^{-1}\left(\frac{f}{f_c}\right) \text{...............(20)}$$

A variable (C) can be used to change the value of the cut-off frequency of this filter, while for changing the gain (A), (Rin) can be made variable.

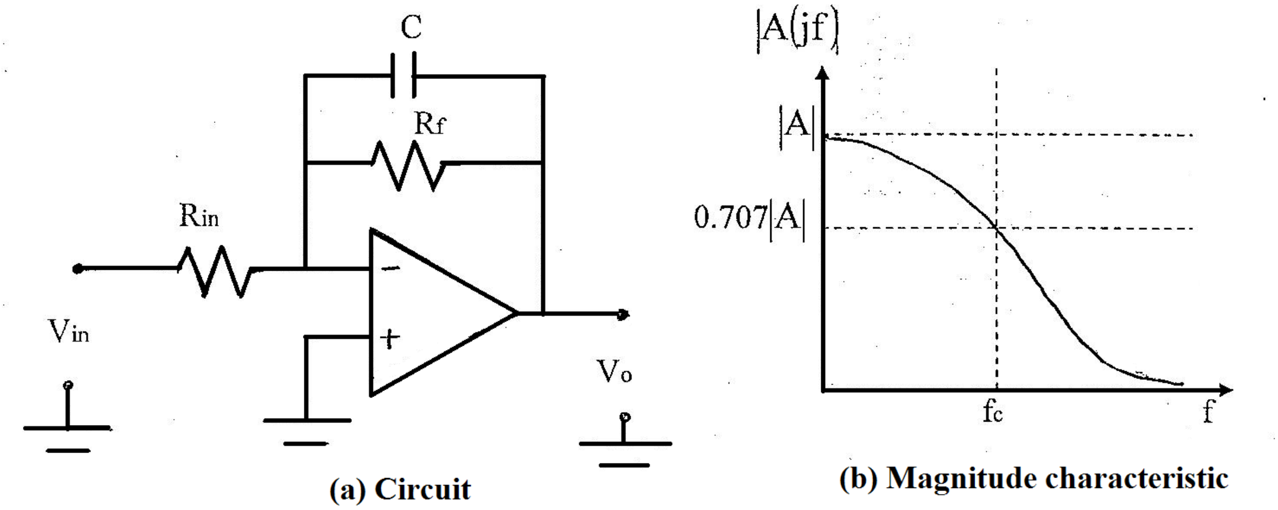


Fig.16 R-C Active Low Pass Filter

R-C Active High Pass Filter

Fig. 17 shows *R-C Active High Pass Filter*. When the frequencies of the input signals arc less than the *cut-off frequency* of this filter, the capacitor acts as an open circuit, therefore these signals will not be transmitted to the output of this filter, i.e. Vo=0. As soon as the frequency of the input signals being more than the *cut-off frequency*, the capacitor starts to pass these signals. These signals are then amplified by the OP-AMP with a gain of (-Rf / Rin). Now:

$$Z_f = R_f$$

$$Zin = Rin + \frac{1}{j2\pi Cf}$$

$$A(jf) = \frac{-Rf}{Rin + \frac{1}{j2\pi Cf}} = \frac{-Rf}{Rin} * \frac{1}{1 + \frac{1}{j2\pi CfRin}}$$

$$\therefore |A(jf)| = \frac{A}{\sqrt{1 + \left[\frac{1}{2\pi CfRin}\right]^2}} \text{.................}(21)$$

$$At \text{ f} = \text{fc}, |\text{A(jf)}| = \frac{A}{\sqrt{2}}$$

$$\therefore \frac{1}{2\pi CfRin} = 1$$

$$\therefore fc = \frac{1}{2\pi RinC} \text{..........................}(22)$$

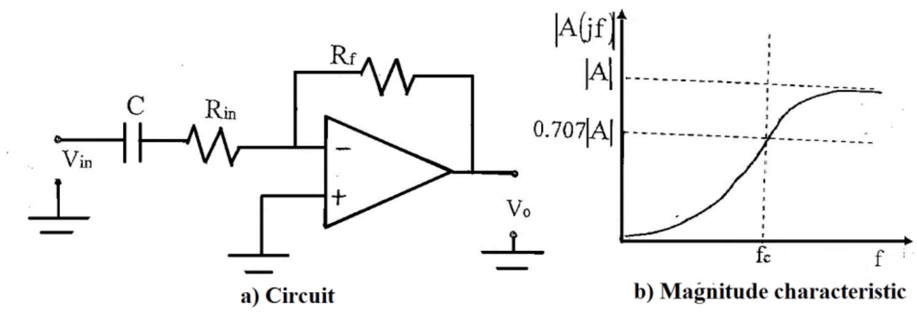


Fig. 17 R-C Active High Pass Filter

R-C Active Band Pass and Active Band Stop Filters

R-C Active Band Pass Filter circuit can be constructed by cascading a unity gain *R-C Active Low Pass Filter*, a unity gain *R-C Active High Pass Filter*, and *an inverting amplifier* with a gain (- Rf / Rin), as shown in Fig.16, provided that the *cut-off frequency* of the first stage *(Active LPF)* f_{el} is greater than the *cut-off frequency* of the next stage *(Active HPF)* f_{c2}. i.e. :

$$\frac{1}{2\pi R_1 C_1} \rangle \frac{1}{2\pi R_2 C_2}$$

and the bandwidth of this filter is :

$$\text{B.W} = \text{f}_\text{u} - \text{f}_\text{l} = \frac{1}{2\pi R_1 C_1} - \frac{1}{2\pi R_2 C_2}$$

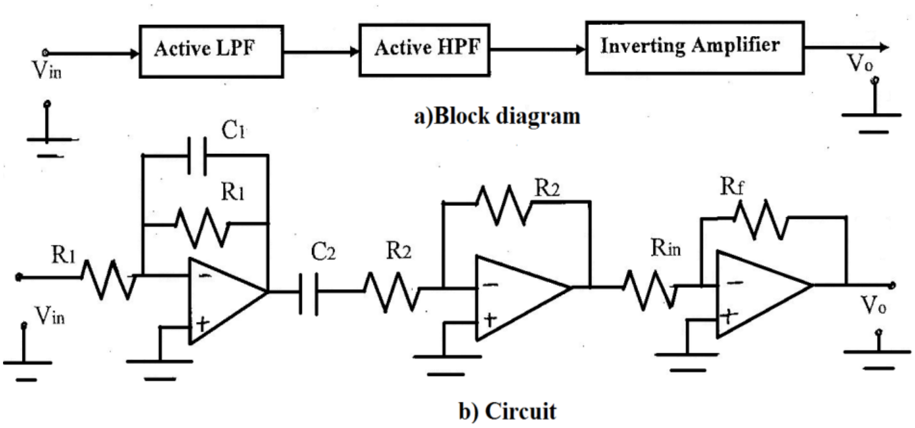


Fig.18 R-C Active Band Pass Filter

The variation of magnitude of the transfer characteristic of this filter versus frequency is depicted in Fig.19.

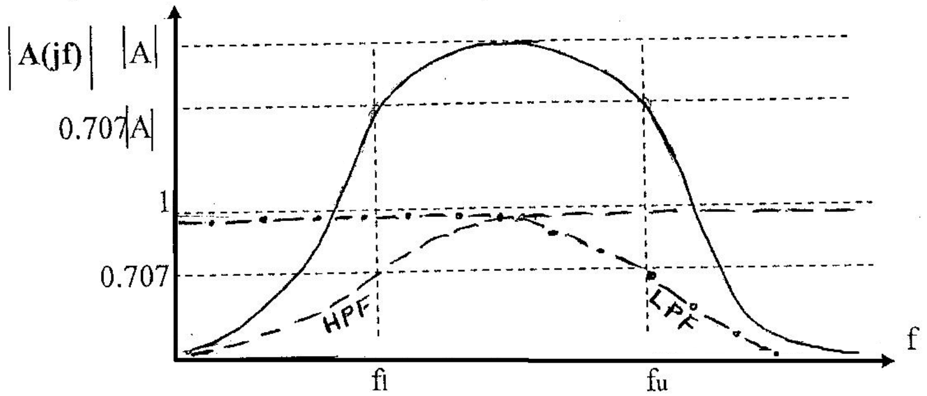


Fig.19 R-C Active Band Pass Filter Characteristic

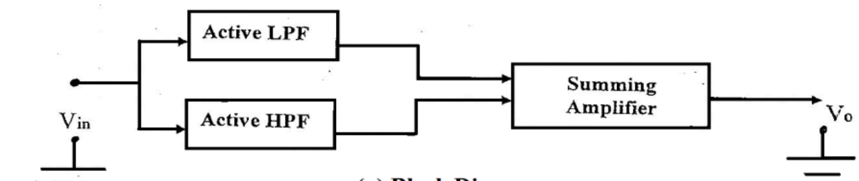
To construct *R-C Active Band Stop Filter*, a parallel combination of *R-C Active Low Pass Filter* and *R-C Active High P-ass Filter* is used instead of cascading them and a summing amplifier is then used as shown in Fig.20. The *C/Cs* of such filter is shown in Fig.21. The cut-off frequencies are:

$$f_l = \frac{1}{2\pi R_1 C_1}$$

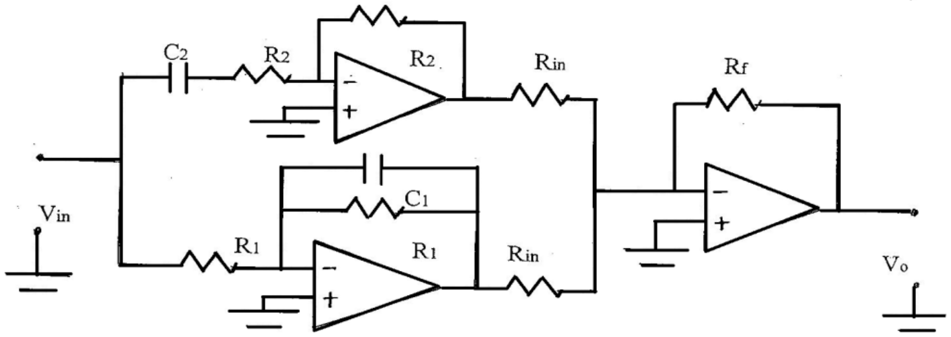
$$f_u = \frac{1}{2\pi R_2 C_2}$$

and the bandwidth

$$\mathrm{B.W} = \mathrm{f}_2 - \mathrm{f}_1$$



(a) Block Diagram



(b) Circuit

Fig.20 R-C Active Band STOP Filter (Band Reject Filter)

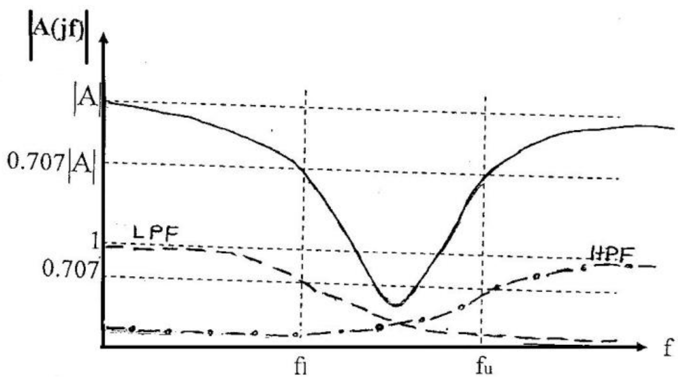


Fig.21 C/Cs of R-C Active Band Pass Filter (Band Reject Filter)

R-C Active Band Pass and Active Band Stop Filters

To have an accurate numerical plots of |A(jw)| and θ(jw) versus *(w)* of a filter, the *"Bode plots"* are used. They are two separate plots; one shows how *the magnitude* of |A(jw)| in dB ,varies with (w) . And the other plot shows how the *phase angle* of |A(jw)| in degrees, varies with *(w)*. Both of these plots are plotted in semi log paper, for greater accuracy in representing the wide range of frequency values. Fig.20 shows *Bode plot* of |A(jw)| versus *(w)* for *R-C Active Low Pass Filter*.

The plot designated *"first-order"* belongs to the filter shown in Fig.22(a). This plot illustrates that the transition between pass band and stop band has a slop of 20 dB/decade. This slop is not so sharp. To make this C/Cs more sharpness, another identical filter is followed the first one and the slop will increase to 40 dB/ decade. The resulting filter is *second-order R-C Active Low Pass Filter*. Thus, as more

identical filters are cascaded together as more is the closing the overall C/Cs to the ideal filter. Fig.23 shows such a filter.

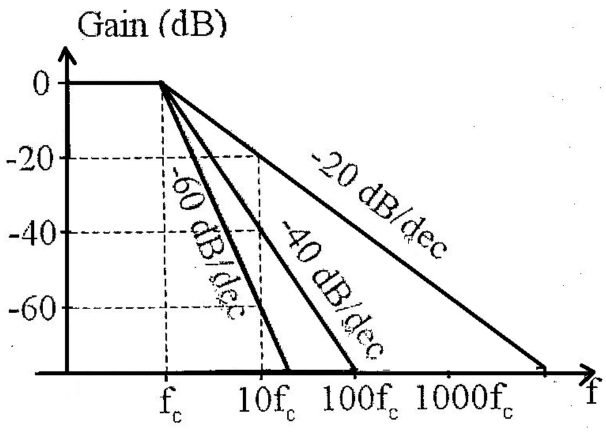


Fig. 22: Bode plot of the transfer C/Cs of Low Pass Filter

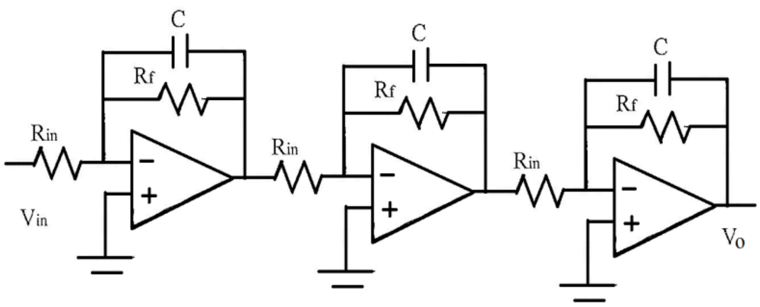


Fig. 23: Identical Active Low Pass Filters connected in cascade

Finally, the loading has no a large effect on the characteristic of *Active Filters* because of the low output impedance of the OP-AMP used in these filters. Thus, *Active Filters* can derive small values of loads.

Solved Problems

Design R-C Active Low Pass Filter to have cut-off frequency of 3KHZ and pass band gain of (-10).Sketch the circuit and the variation of |A(jf)| versus frequency.

Solution

Let us choose C=0.01 µF.

$$f_c = \frac{1}{2\pi C R_f}$$

$$\therefore R_f = \frac{1}{2\pi C f_c} = \frac{1}{2*3.14*0.01*10^{-6}*3*10^{3}} = 5305\Omega$$

$$\text{since} \quad A = 10 = \frac{R_f}{Rin}$$

$$\therefore Rin = \frac{5305}{10} \cong 530\Omega$$

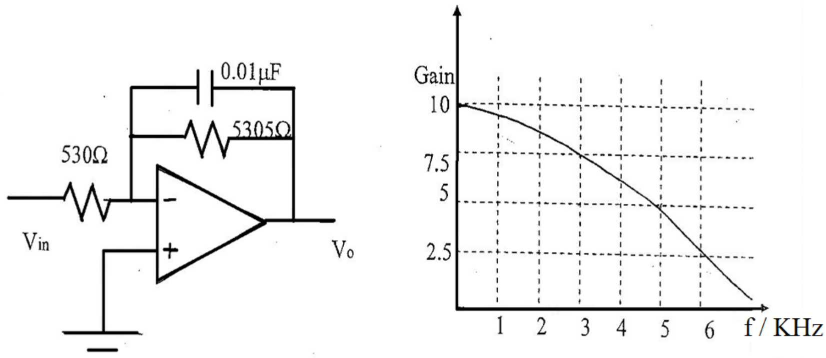


Fig. 24

Home Works

Find the maximum and minimum values of both; gain and cut-off frequency of the filters shown below:

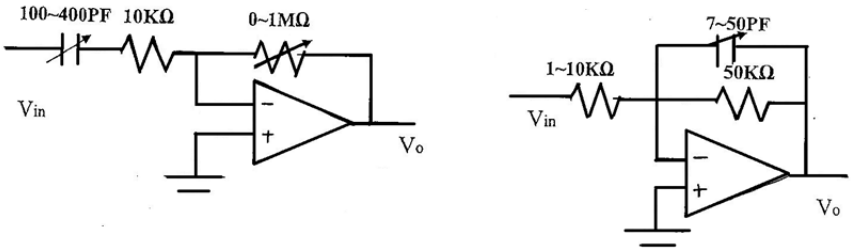


Fig. 25

(WEEKS 2-6)

Modulation

Modulation is the addition of information to an electronic or optical carrier signal. A carrier signal is one with a steady waveform -- constant height (amplitude) and frequency. Information can be added to the carrier by varying its:-

1-amplitude, 2-frequency,
3-phase, 4-polarization

Why Modulation?

1. *For sufficient transmission and ease of radiation.*
2. *To reduce noise and interference.*
3. *For frequency assignment*
4. *For Multiplexing*
5. To overcome equipment limitations.

Type of modulation

1. Amplitude modulation (AM),
2. Frequency modulation (FM)
3. Phase modulation (PM),
4. Polarization modulation(POM)
5. Pulse-code modulation(PCM

Amplitude modulation

As stated previously, the carrier in *Amplitude Modulation* is sine wave. Therefore it is expressed as:

$V_c = A\cos w_c t$ ……………… (1)

and the modulating signal as:

$V_m = B\cos w_m t$ ……………....(2)

Where $w_c \gg w_m$. The modulated carrier Of the AM signal will be:

$V_{AM} == (A + B\cos w_m t)\cos w_c t$

Because m=B/A or B=mA then

$V_{AM} = A(1 + m\cos w_m t)\cos w_c t$ …. (3)

Where "m" is the Modulation Index or Modulation Depth and it is equal to

$$m = \frac{\text{Peak value of modulating signal}}{\text{Peak value of unmodulated carrier signal}} = \frac{B}{A}.$$

The percent Modulation index "M" is

$$M = \frac{B}{A} * 100\%.$$

- Fig. 1 shows unmodulated carrier and a carrier which is modulated by a single tone information signal.
- "M" varies between 0<M< 100% without introducing distortion and the best value is 100% where A=B and the maximum peak of *AM* signal is 2A. When M > 100%, the *AM* signal will be distorted and have a phase reversal (see fig.2 below

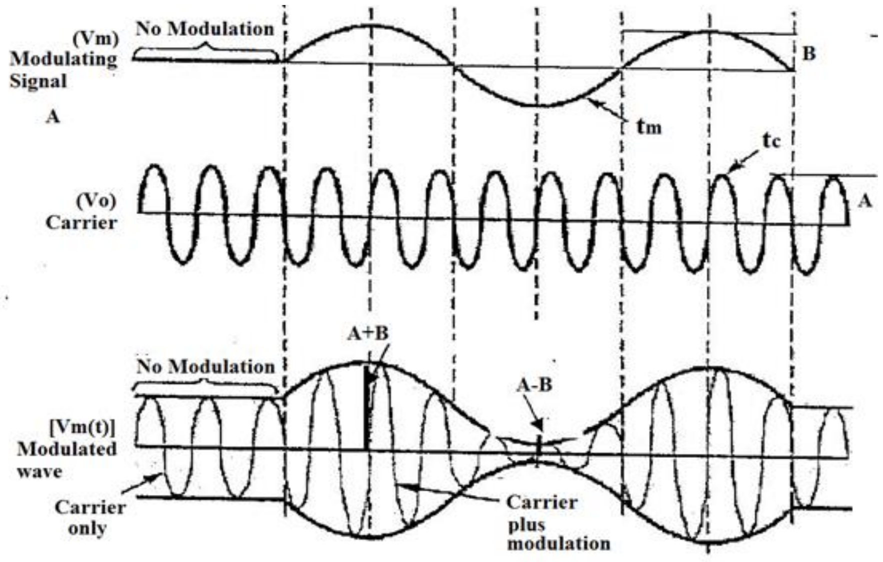
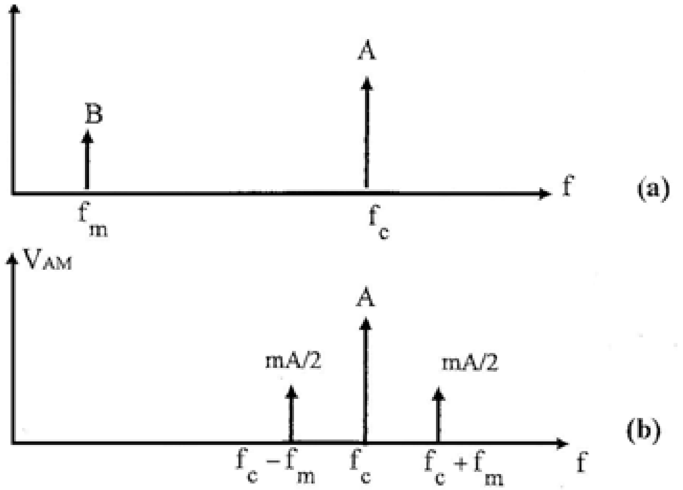


Fig. 1

<u>Thus, AM signal consists of three components:</u>

- The un-modulated carrier.
- The Lower Side Frequency (LSF) with frequency of (fc -fm) and peak of mA/2
- The Upper Side Frequency (USF) with frequency of (fc+ fm) and peak of mA/2



a) Unmodulated carrier and modulating signal (i.e. before modulation).

b) AM signal

Fig.2 Amplitude frequency spectrum for:

Amplitude- frequency spectrum for AM signal:

The carrier signal is amplitude modulated by a complex waveform.

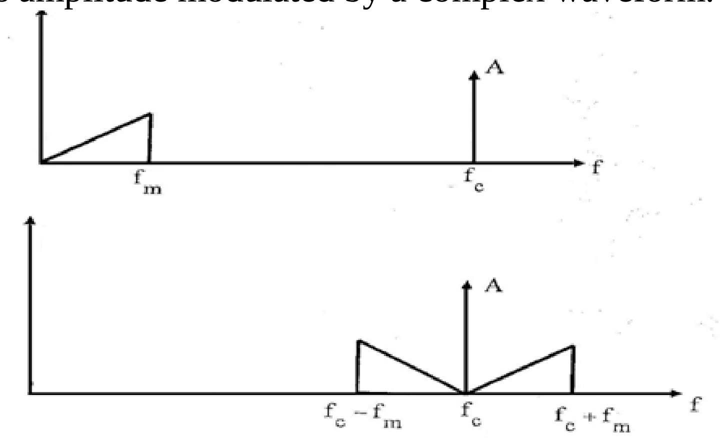


Fig.3 The carrier signal is amplitude modulated

The bandwidth in this case is twice of the maximum of the modulating signal (the message), i.e.:

$$B.W=2(f_m)_{max}$$

and the overall modulation index (m): $$m_t = \sqrt[2]{m_1^2 + m_2^2 + m_3^2 + \cdots}$$

Provided that $m_t \leq 1$. And m_1, m_2, m_3,,, are the modulation index of the individual sine and cosine waves of the complex modulating signal

Power contents of AM signal:

The total power of *AM* signal that is produced by modulating a carrier signal by a single tone modulating signal is:

$$= \frac{A}{2} + \frac{m^2 A^2}{8} + \frac{m^2 A^2}{8} = \frac{A^2}{2} + \frac{m^2 A^2}{4}$$

$$\therefore P_t = \frac{A^2}{2}\left(1 + \frac{m^2}{2}\right).$$

When m=1 (i.e. 100% modulation), the total power of the side bands ($P_{USB} + P_{LSB}$) is equal to 50% of the carrier signal power. The *power efficiency* η is the ratio of the side bands power to the total power, i.e.:

$$\eta = \frac{P_{side\ bands}}{P_t} = \frac{\frac{m^2 A^2}{4}}{\frac{A^2}{2}\left(1 + \frac{m^2}{2}\right)} = \frac{m^2}{2 + m^2}$$

Amplitude Modulation

Amplitude modulation basics

When an amplitude modulated signal is created, the amplitude of the signal is varied in line with the variations in intensity of the sound wave. In this way the overall amplitude or envelope of the carrier is modulated to carry the audio signal. Here the envelope of the carrier can be seen to change in line with the modulating signal Fig.4.

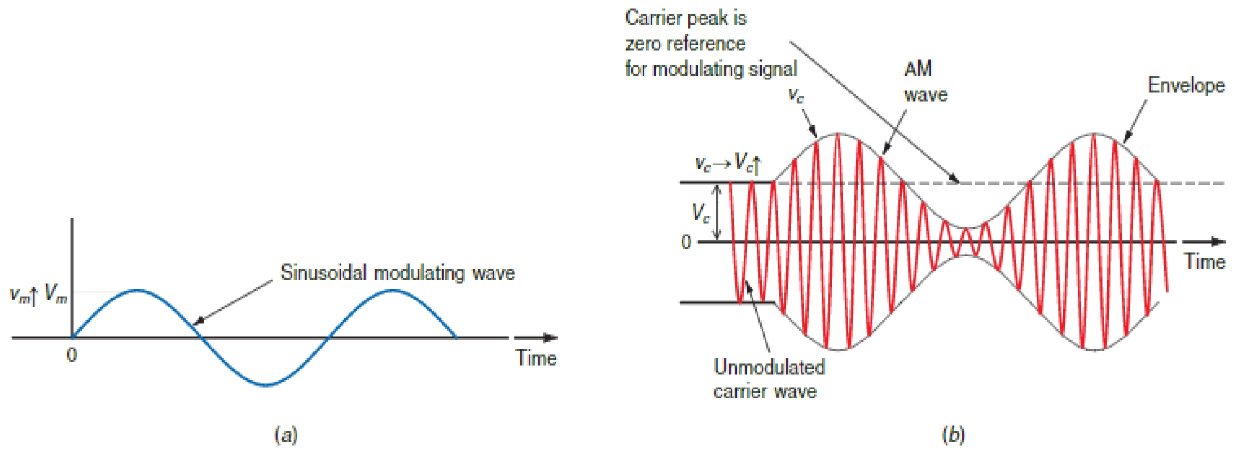


Fig.4

ADVANTAGES

1. It is simple to implement
2. It can be demodulated using a circuit consisting of very few components
3. AM receivers are very cheap as no specialized components are needed.

DISADVANTAGES

1. An amplitude modulation signal is not efficient in terms of its power usage
2. It is not efficient in terms of its use of bandwidth, requiring a bandwidth equal to twice that of the highest audio frequency
3. An amplitude modulation signal is prone to high levels of noise because most noise is amplitude based and obviously AM detectors are sensitive to it.

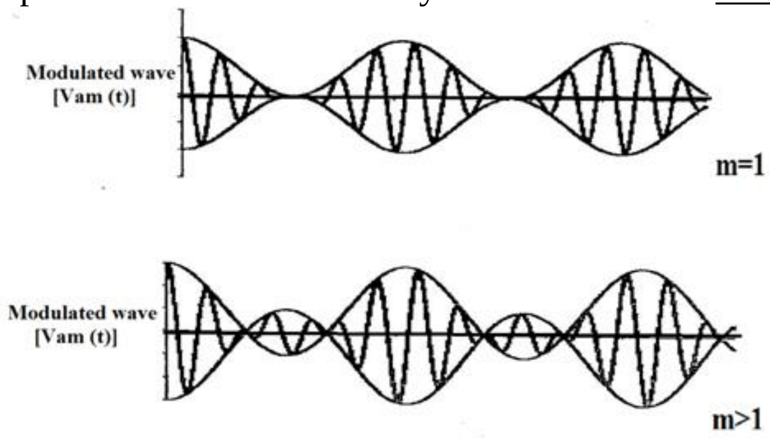


Fig. 5 AM signal for different states of modulation index

From the AM signal shown in figure above, the *Modulation Index* "m" can be determined from:

$$m = \frac{(A+B)-(A-B)}{(A+B)+(A-B)} = \frac{2B}{2A} = \frac{B}{A} \quad \textit{as before}$$

Example

Suppose that on an AM signal, the $V_{max(p-p)}$ value read from the graticule on the oscilloscope screen is 5.9 divisions and $V_{min(p-p)}$ is 1.2 divisions.

a. What is the modulation index?

$$m \frac{V_{max} - V_{min}}{V_{max} + V_{min}} = \frac{5.9 - 1.2}{5.9 + 1.2} = \frac{4.7}{7.1} = 0.662$$

b. Calculate V_c, V_m, and m if the vertical scale is 2 V per division. (*Hint:* Sketch the signal.)

$$V_c = \frac{V_{max} + V_{min}}{2} = \frac{5.9 + 1.2}{2} = \frac{7.1}{2} = 3.55 \text{ @ } \frac{2\text{ V}}{\text{div}}$$

$$V_c = 3.55 \times 2\text{ V} = 7.1\text{ V}$$

$$V_m = \frac{V_{max} - V_{min}}{2} = \frac{5.9 - 1.2}{2} = \frac{4.7}{2}$$

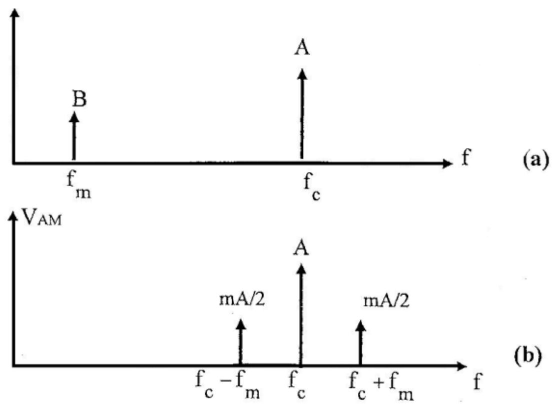
$$= 2.35 \text{ @ } \frac{2\text{ V}}{\text{div}}$$

$$V_m = 2.35 \times 2\text{ V} = 4.7\text{ V}$$

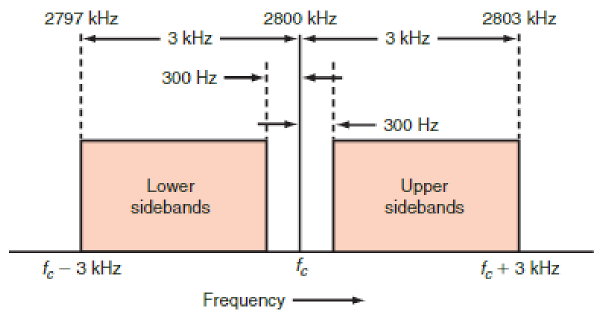
$$m = \frac{V_m}{V_c} = \frac{4.7}{7.1} = 0.662$$

The Lower Side Frequency (LSF) with frequency of (fc -fm) and peak of $\frac{mA}{2}$

The Upper Side Frequency (USF) with frequency of (fc+ fm) and peak of $\frac{mA}{2}$



The upper and lower sidebands of a voice modulator AM signal.



Amplitude frequency spectrum for:
a)Unmodulated carrier and modulating signal (i.e. before modulation).
b) AM signal.

Fig. 6 From figure above, the Bandwidth of AM signal is:

$$B.W = (f_c + f_m) - (f_c - f_m) = 2f_m \text{ Hz}$$

Example

A standard AM broadcast station is allowed to transmit modulating frequencies up to 5 kHz. If the AM station is transmitting on a frequency of 980 kHz, compute the maximum and minimum upper and lower sidebands and the total bandwidth occupied by the AM station.

$$f_{USB} = 980 + 5 = 985 \text{ kHz}$$
$$f_{LSB} = 980 - 5 = 975 \text{ kHz}$$
$$BW = f_{USB} - f_{LSB} = 985 - 975 = 10 \text{ kHz} \quad \text{or}$$
$$BW = 2(5 \text{ kHz}) = 10 \text{ kHz}$$

Suppressed-Carrier, Double-Side Band, and Single-Side Band Transmission

- If the carrier signal is suppressed, only the side bands power remains. As this is only (m^2P_c /2). Upon suppressing the carrier signal, the AM signal is called *Double-Side Band Suppressed Carrier (DSB-SC) (the AM signal is called Double Side Band Large or Full Carrier DSB-LC or DSB-FC).The* bandwidth of *DSB-SC* is still twice the modulating signal band.
- If one of the side bands is also suppressed, the remaining power is (m^2P_c /4), This new signal is called *Single Side Band-Suppressed Carrier (SSB-SC).* An additional advantage of *(SSB-SC)* scheme is that only half as much bandwidth is required for the transmission, and therefore twice as many stations can be transmitted simultaneously.
- The major disadvantage of *(DSB)* & *(SSB)* schemes is that they require more expensive receiver, since those receivers must be able to restore the eliminated carrier signal before demodulating can occur. Figure below shows the amplitude-frequency spectrum *for (DSB-SC)* and *(SSB-SC)* schemes fig. 7**.**

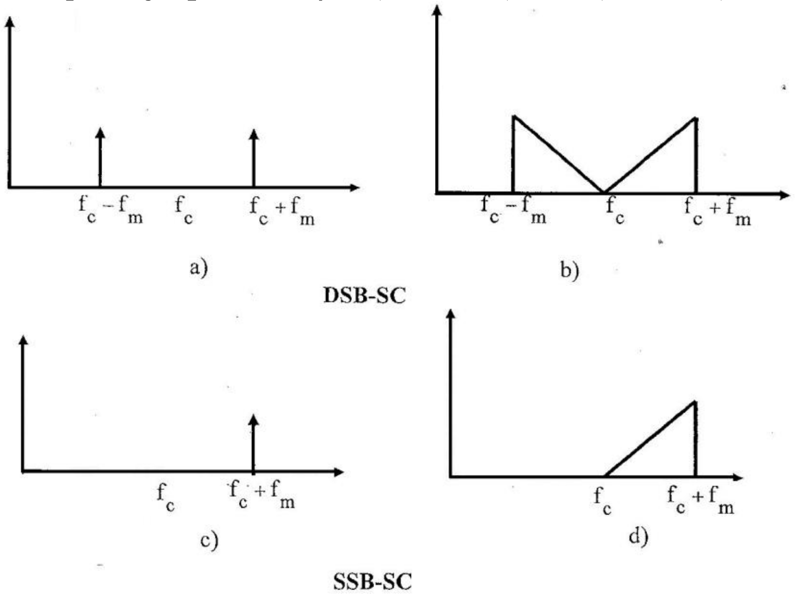


Fig. 7 Amplitude-frequency spectrum for DSB-SC & SSB-SC

From figure above, it is clear that:
$(B.W)_{DSB\text{-}SC} = 2f_m$ (For single tone modulating signal)
$(B.W)_{DSB\text{-}SC} = 2(f_m)_{max}$ (For complex modulating signal)
$(B.W)_{SSB\text{-}SC} = f_m$ (For single tone modulating signal)
$(B.W)_{SSB\text{-}SC} = (f_m)_{max}$ (For complex modulating signal)

Solved problems

1) An amplitude modulated signal is expressed as:

$V_{AM}=5(1+0.5\cos 3140t)\cos 2\pi 10^5 t$ volts

Find:

a) The modulation depth.
b) The modulating frequency.
c) The carrier frequency.
d) The peak value of the signal.
e) If the signal is to be transmitted, what will be its power?

Solution:

Comparing the given expression with standard AM equation which is rewritten here:
$V_{AM}=A(1+m\cos w_m t)\cos w_c t$

$a)\ m = 0.5,\ \ M=50\%$

$$b)\ f_m = \frac{3140}{2\pi} \cong 500Hz$$

$$c)\ f_c = \frac{2\pi * 10^5}{2\pi} = 10^5\,Hz = 100KHz$$

$d)$ The peak value $= (A + B)$

$A = 5\text{ v}, B = mA = 0.5 * 5 = 2.5\text{ v}$

Peak value $= 5 + 2.5 = 7.5\text{ v}$

Or

The peak value $= A(1 + m)$ (This is because the peak value of cosine wave occurs when $wt = 0, \pi, 2\pi, 3\pi$)

$= 5(1 + 0.5) = 7.5v$ as before

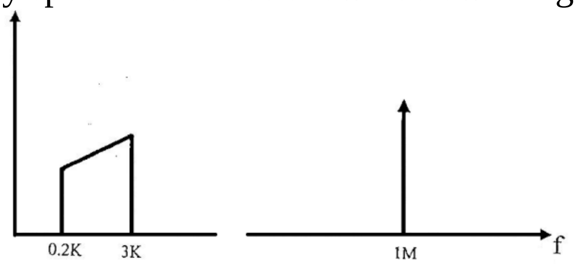
$$\text{e) The signal power} = P_t = \frac{A^2}{2}\left(1 + \frac{m^2}{2}\right)$$

$$= \frac{5^2}{2}\left(1 + \frac{0.5^2}{2}\right) = 14.0625\ \text{watt}$$

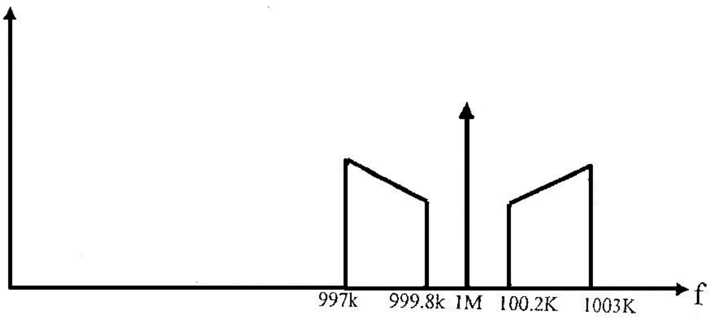
2) a carrier signal of frequency 1MHZ is amplitude modulated by audio signals ranging from 200HZ to 3KHZ. What side bands are generated? sketch the amplitude frequency spectrum.

Solution:
The modulating signal in this problem is complex, therefore, after modulation, we have:
USB is (fc +fm), it is ranging from:
1000K + 0.2K == 100.2 KHz to
1000K +3K == 1003 KHz
LSB is (fc - fm), it is ranging from :
1000K - 0.2K == 999.8 KHz to
1000K - 3K = 997 KHz

The amplitude-frequency spectrums are as shown below in fig.8:



Before modulation



After modulation
Fig.8

3) AM wave have the following information:
50v at 750 KHz
20v at 748 KHz ,
20v at 752 KHz
a) What is the carrier frequency?
b) What are the frequencies of USB and LSB?
c) What is the frequency of the modulating signal?
d) Calculate the modulation index.

Solution:

a) The carrier frequency is 750 KHZ.
b) USB frequency is 752 KHZ, while LSB frequency is 748 KHZ.
c) The modulating frequency (fm) is determined from:

$$f_m = f_c - (f_c - f_m) = 750 - 748 = 2KHz$$

or

$$f_m = (f_c + f_m) - f_c = 752 - 750 = 2KHz \text{ as before}$$

$$\text{d) } \frac{mA}{2} = 20$$

or

$$m = \frac{2*20}{A} = \frac{2*20}{50} = 0.8$$

$$\therefore M = 80\%$$

4) Sketch the amplitude-frequency spectrum at the output of each stage for the system shown below in fig 9.:

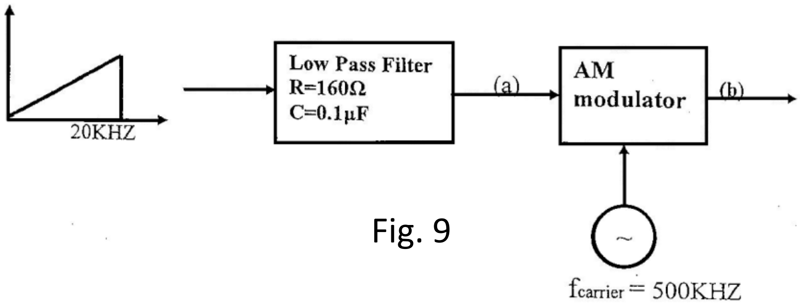


Fig. 9

Solution:
At (a), the low pass filter will transmit frequencies from 0 HZ to ($f_{\text{cut- off}}$) of this filter which is equal to**:**

$$f_{cut-off} = \frac{1}{2\pi RC} = \frac{1}{2*3.14*160*0.1*10^{-6}} \cong 10KHz$$

The amplitude-frequency spectrum at (a) will be:

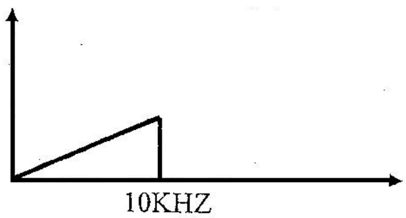


Fig.10 a

The filtered wave at (a) will amplitude modulate the carrier at the AM modulator. Therefore the amplitude-frequency spectrum at (b) will be:

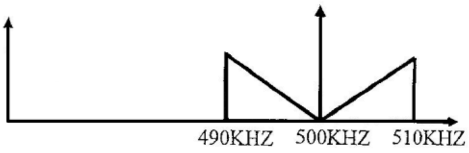


Fig.10 b

Home works:

1) an AM voltage is expressed as:

   $V_{AM}=5\cos 10\pi 10^5 t+1.5\cos 5\pi 10^5 t+1.5\cos 15\pi 10^5 t$

Find:

   a) The modulation index (m).
   b) The modulating frequency.
   c) The peak value of the modulated wave.
   d) The signal power.

2) A 2500 HZ pure sine wave amplitude modulates a 650 KHZ carrier signal. If the carrier power is 1KW, determine the power in the side bands for m=50%. What will be the transmitted power if the carrier and one side band are both suppressed?
3) Sketch the amplitude-frequency spectrum at the output of each stager for the system shown below:

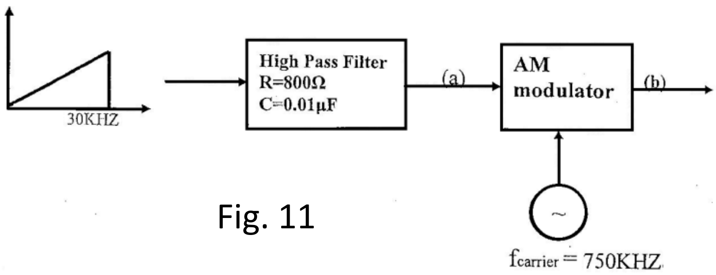


Fig. 11

AM Transmitters

Amplitude-modulated (AM) transmitters are mostly used for the broadcast of speech or music. These transmitters either operate on the broadcast band (535- 1605 KHZ) or medium-wave band. When used for long distance transmission such as overseas broadcast, these transmitters operate on high frequencies between 3 and 30 MHZ and are known as ***"HF"*** transmitters or ***"Short-Wave (SF/!)"*** transmitters. Fig.4.15 is a' block diagram of an amplitude-modulated transmitter.

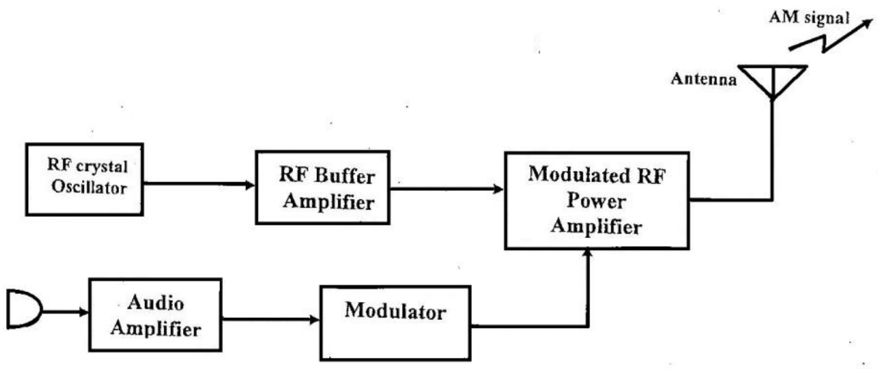


Fig. 12: Block diagram of an AM transmitter

Notice that the modulator unit of the *AM* transmitter shown in fig.12 is not the unit in which the modulation process occurs; it is the last stage of audio amplification. The modulation process Occurs in the stage ***"Modulated RF Power Amplifier"*** where the amplified audio signal is impressed together with the carrier signal generated and amplified by the ***"RF Crystal Oscillator"*** **and** ***"RF Buffer Amplifier"*** respectively. The output of the last stage ***"Modulated RF Power Amplifier"*** is fed to the antenna. Fig.13 is a block diagram of ***SSB*** transmitter.

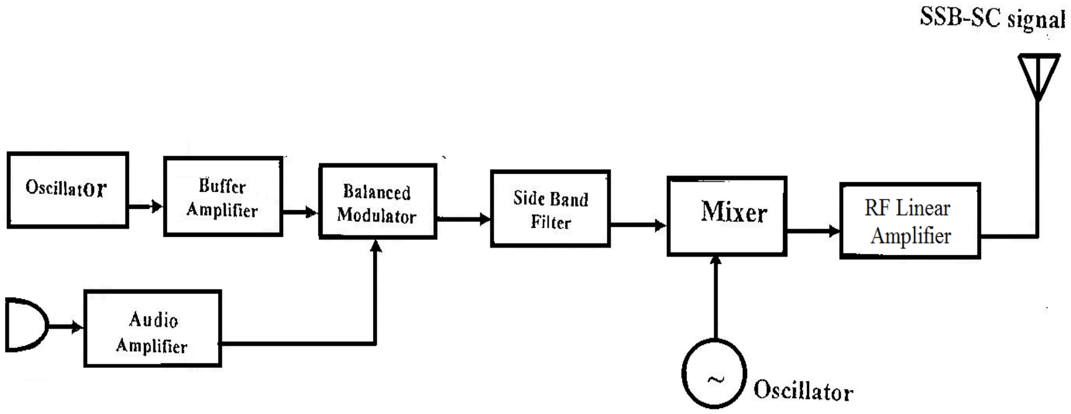


Fig.13: Block diagram of SSB transmitter

In commercial broadcasting each broadcasting station is assigned a 5KHZ for each of the Upper and lower side bands, i.e, the bandwidth is 10KHZ. The 5KHZ band is sufficient to cover the information content of speech and music frequencies which range between 0 and 304KHZ.

Homework:

1) Write the expression of the output signal for the system shown below.

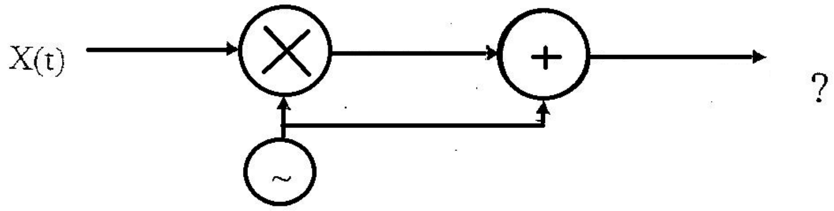


Fig.14 Where X(t) is the modulating signal.

2) calculate the percentage Modulation index (M) for the following wave.

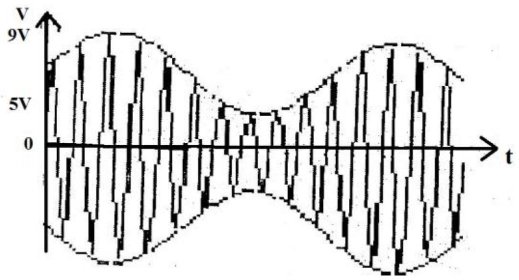


Fig.15: Circuit diagram of simple AM transmitter

(WEEKS 7-8)

AM Demodulators (Detectors) and Receivers

AM Demodulators (Detectors)

In the transmitter the modulation process takes place so that the message (the modulating signal) is suited properly to the communication channel. After receiving the modulated wave by the receiver the ***Demodulation*** *(Detection)* process must be performed in the receiver to recover the message. **Thus the *Demodulation* (*Detection*) is the inverse process of modulation.**

Demodulators (Detectors) fall into the two broad categories of: (*Envelope (Asynchronous) detectors*, and *Synchronous detectors*)

1) Envelope (Asynchronous) detectors:

Envelope detectors are the simplest and the most popular used detectors. Examination of the amplitude- modulated carrier in fig. 16 indicates that we can recover the information from the modulated carrier by rectification and then low pass filtering the modulated carrier wave as illustrated in fig. 4.18. The effect of the capacitor on the output is to bridge from the peak of one pulse to the next pulse. With proper choice of RC time constant, we can get an approximation of the original information signal.

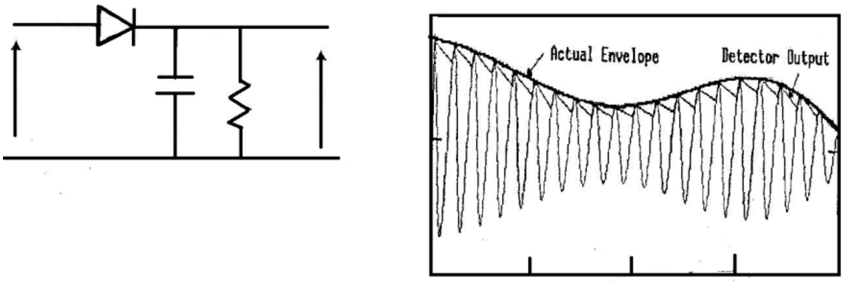


Fig. 16: Envelope detector

The output of the *Envelope detector* has a D.C level in addition to the approximated information wave. This D.C voltage is useful. It can be used in a feedback loop to control the gain of the preceding stages. This system of feeding back a part of the output of the detector is called *"Automatic Gain Control"* or *"AGC"*. Fig. 17 illustrates the circuit of *AGC*. In this circuit the *AGC* voltage is developed across C_1 by filtering out the modulating signal by a low pass filter R_1 & C_1

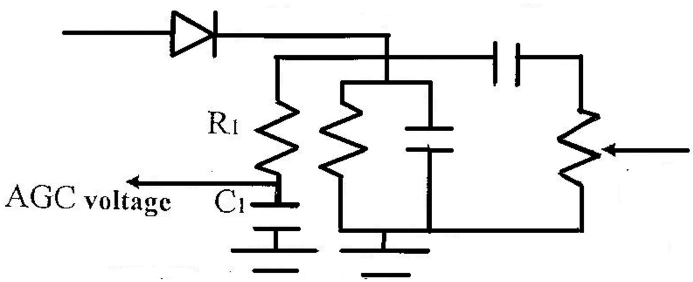


Fig.17: Envelope detector with AGC circuit

2) Synchronous detector:

FIg. 18 shows the block diagram of the *Synchronous (Product) detector*. This detector detects the information from the DSB signal. First, the incoming signal is multiplied by a Local Oscillator (L.O). This oscillator must have the same frequency and in phase with the transmitted carrier, i.e. it is synchronized with the transmitted carrier. The output of the multiplier is then low pass filtered to develop the approximated information at the output. Mathematically, we can write:

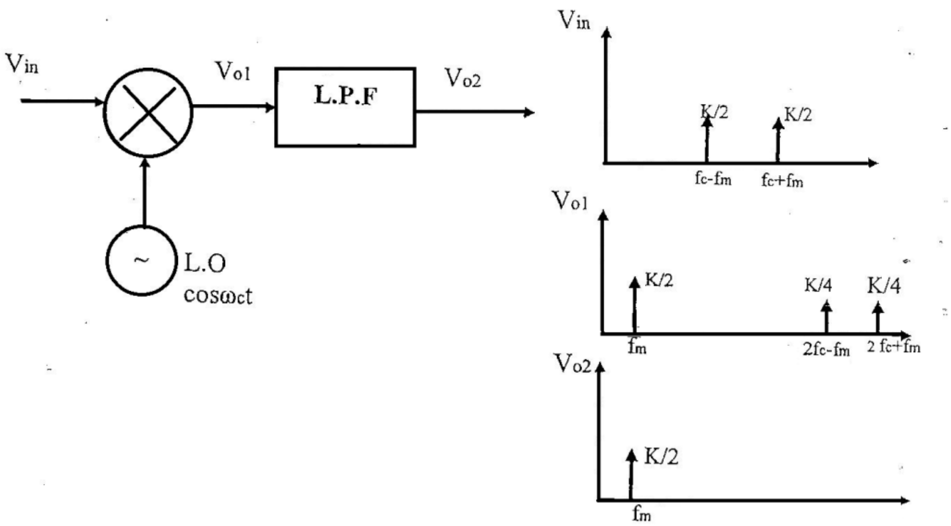


Fig.18: AM Synchronous detector

AM Receivers

A very fundamental *AM receiver* system is shown in fig.19. The essential parts of this system are: I) *An antenna* to pick up the necessary Radio-frequency signals from the atmosphere, 2) *A selector* capable of passing the carrier and the associated upper and lower side bands of the desired station as well as capable of eliminating all other signals, 3) *A demodulator* and 4) *An audio amplifier* which drives the headphone or loudspeaker.

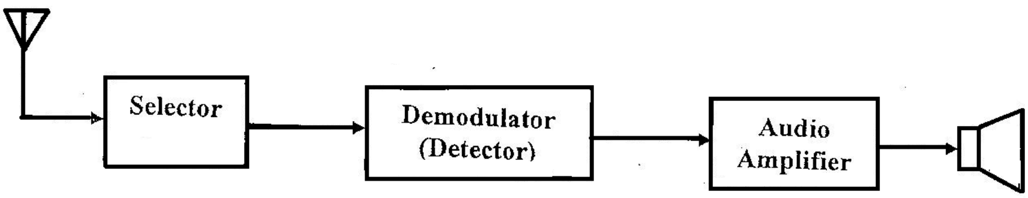


Fig. 19: Most Fundamental AM Receiver

Tuned Radio Frequency Receiver(TRF):

The *TRF* receiver was popular in the early days of radios. It has been replaced for the most parts by the *"Superhetrodyne"* receivers. In order to upgrade the *AM* receiver

shown in figA.21 so that it can **receive lower-power signal satisfactory**, amplifier stages close to the input of the receiver are added, see fig 20. These amplifiers have tuning circuits which are tuned to the same frequencies tuned by the selector and detector stages.

The main disadvantage of this receiver is that the bandwidth of the tuned circuits (selector, detector, and amplifiers) will be increased as the *RF* carrier frequency is increased. Hence the undesired adjacent frequencies will not be rejected at higher *RF* frequencies.

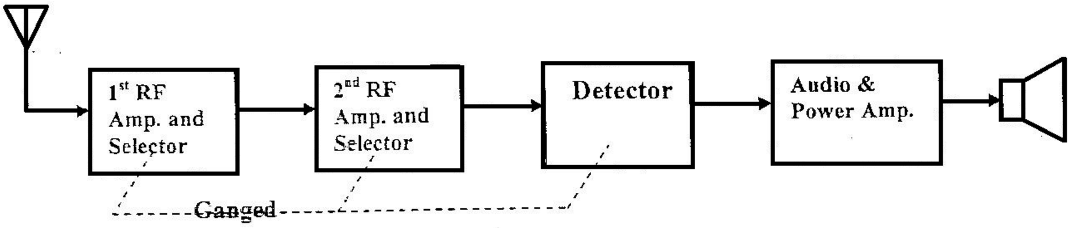


Fig.20: Block diagram of TRF receiver

The Superheterodyne receiver:

The *superheterodyne* receiver is the most popular type of radio receiver. In this receiver, see fig. 21, the incoming signal from the *RF* stage which is the tuned desired station captured by the antenna is combined with the local oscillator(L.O) frequency in the *"Mixer"* stage and normally converted into a new modulated signal of a lower fixed frequency. This signal is called the *"Intermediate Frequency (IF) "*, It contains the same information as the original modulated carrier. This signal is then amplified in the *"IF Amplifier"* stage and the detected by the *"Detector"* stage to produce the original information. Thus the *IF* signal will have a fixed bandwidth (lOKHZ) when passed through the two stages; *IF Amplifier* and *Detector*. The value of the IF frequency in *AM Superhetrodyne* receiver is always:

$$f_{LO}-f_{\text{carrier of station}} = 455 \text{ KHz}$$

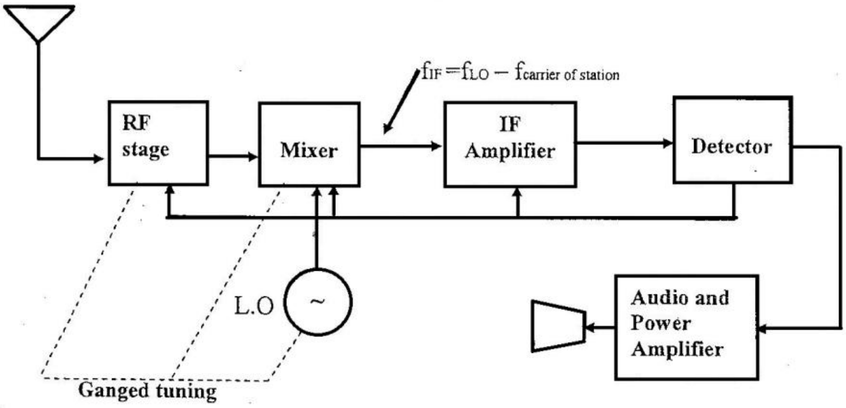


Fig. 21 shows a detailed circuit diagram of AM Superhetrodyne receiver

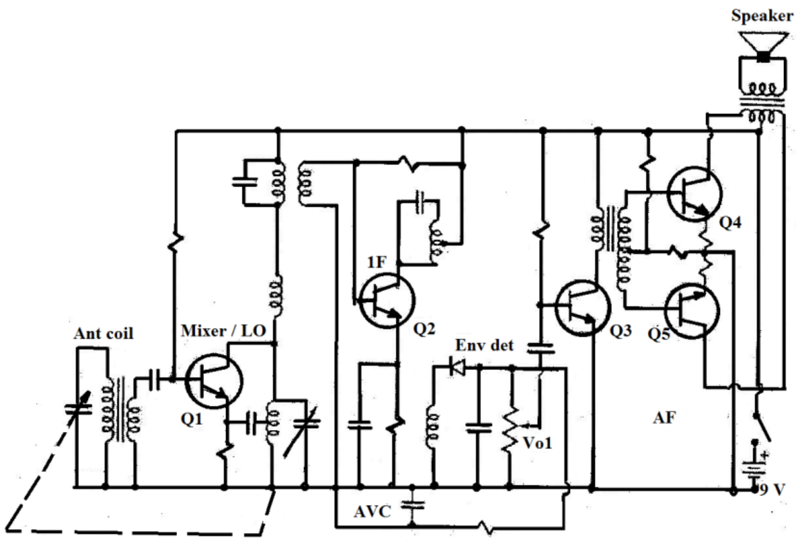


Fig. 22: Schematic diagram of AM superheterodyne receiver

(WEEKS 9-14)

Frequency Modulation (FM)

Frequency Modulation (FM):

Frequency Modulation is the process by which the frequency of the carrier is modulated (**varied**) by the modulating signal (information). In contrast with *AM* modulation, the amplitude of the carrier in *FM* is kept constant.

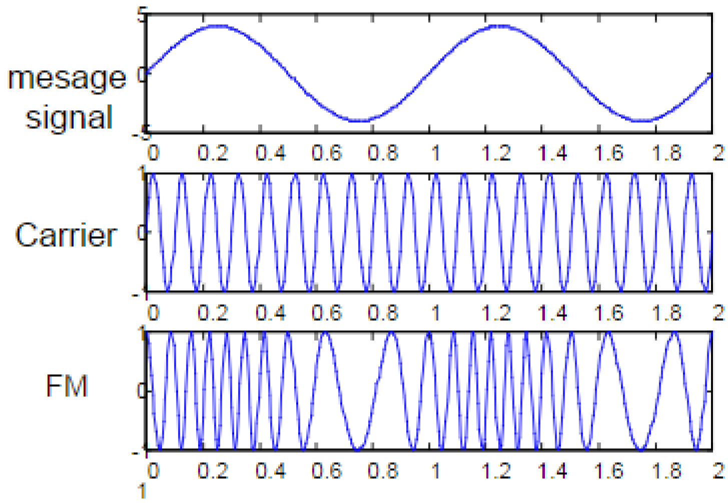


Fig. 23 FM Modulation

Mathematical representation of FM waves:

If the carrier signal has the following expression:

$$V_c = A\cos 2\pi f_c\, t$$

and its frequency is varied by the amplitude of a single tone modulating frequency which has the expression:

$$V_m = B\cos 2\pi f_m\, t$$

Then the frequency of the FM signal will be:

$f_{modulated\ carrier} = f_c + (\Delta f)\cos 2\pi f_m t$ ………………. (4.20)

Where Δf is the peak frequency deviation in the frequency of the frequency modulated carrier and it equals:

$$(\Delta f) = KB \text{ ……………………… } (4.21)$$

Where "K" is a constant and its unit is *HZ/ Volt.* Then:

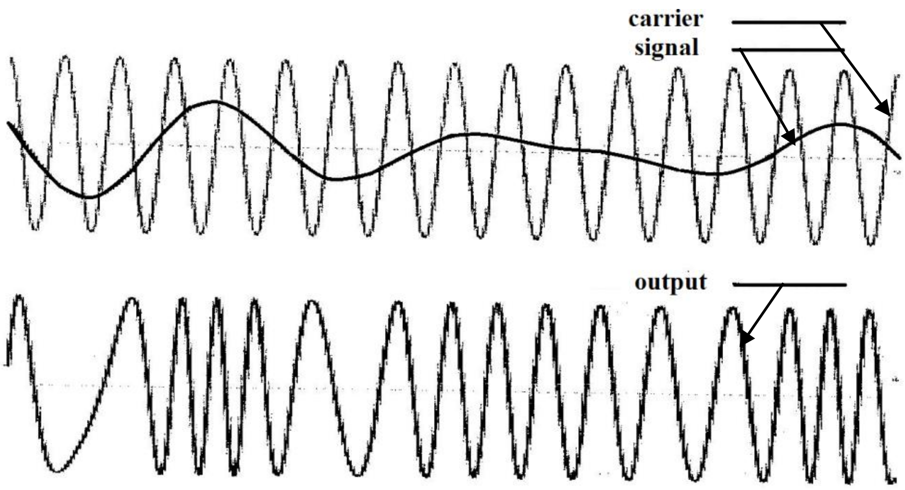
$$V_{FM} = A\cos(2\pi f_c t + \beta \sin 2\pi f_m t)\text{....................}(4.22)$$

Where $\beta = \dfrac{\Delta f}{f_m}$ is the frequency modulation index

The waveforms of the unmodulated carrier, the modulating signal, and the FM signal are shown in figs. 23, 24.

The bandwidth of *FM* wave depends on the modulation index β. Depending on the value of β the bandwidth of FM wave has two types:

a) Narrow Band FM (NBFM):



FM wave with β « (π/2) is called Narrow Band FM (NBFM).

Fig. 24: FM signal

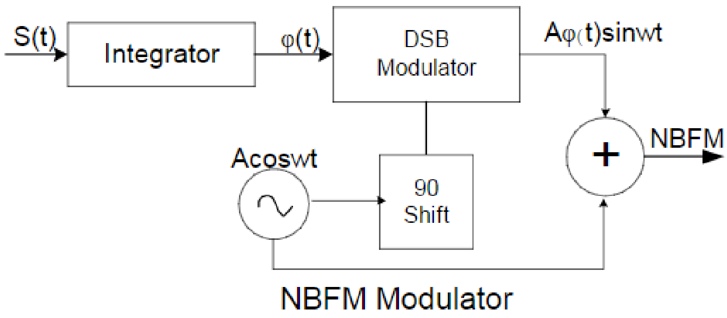


Fig. 25 Modulator

Equation 4.23 states that the bandwidth of *NBFM* is twice the modulating signal band, i.e.

$$(B.W)_{NBFM}=2f_m \text{ ....(23)}$$

b) <u>Wide Band FM (WBFM):</u>

When $\beta >> \pi/2$, the bandwidth of the *FM wave* will be very wide.

$$(B.W)_{WBFM}=2\Delta f \quad \text{.................................... (4.24)}$$

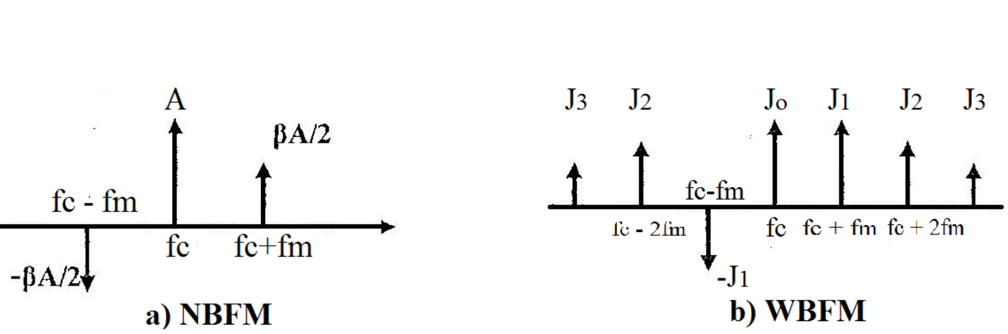


Fig.4.26: Bandwidth of FM wave

In summary:

For $\beta \ll (\pi/2)$ the ***FM*** signal is ***NBFM*** and the bandwidth is $(B.W)_{NBFM} = 2f_m$. For $\beta \gg (\pi/2)$ the ***FM*** signal is ***WBFM*** and the bandwidth is $(B.\ W)_{WBFM} = 2\Delta f$ The Carson's rule states that as a good approximation:

$$(B.W)=2(\Delta f + 2f_m) \quad \text{............................... (4.25)}$$

This rule is used for $\beta > 6$.

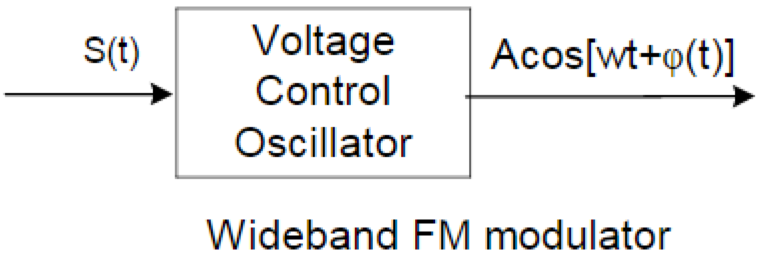


Fig.27

In Commercial FM broadcast,

1- A maximum of frequency deviation Δf_{max}=75KHz is allowed to use.
2-For TV transmission Δf_{max} = 25KHZ.
3-The maximum frequency of the modulating signal used for commercial ***FM*** broadcast and TV transmission is f = 15KHZ
4-(The band 88~ 108 MHZ is assigned for FM commercial broadcasting).

There is another type of modulation in which the phase of the carrier is varied according to the modulating signal. This process is called ***Phase Modulation (PM).*** If the carrier wave is expressed as:

$V_c = A\cos(2\pi f_c t + \Phi)$

and

$V_m = B\cos 2\pi f_m t$ is the modulating signal Then

$V_{PM} = A\cos(2\pi f_c t + \beta \cos 2\pi f_m t)$ (4.26)

and

$\beta = KB$ where K is a constant with unit radian / Volt.

Solved problems

5) FM signal is defined as

$V_{FM}=10\cos(2\pi 10^6 t+8\sin 2\pi 10^3 t)$

Determine:
The frequency of the carrier & modulating signal.
The modulation index & peak frequency deviation.
The total average power of this signal.

Solution:
a) Comparing the given expression with the one in 4.22, we can conclude:

$$f_c = \frac{2\pi * 10^6}{2\pi} = 10^6 Hz = 1MHz$$

$$f_m = \frac{2\pi * 10^3}{2\pi} = 1000Hz = 1KHz$$

b) The modulation index β =8. The peak frequency deviation is:
$\Delta f = \beta * f_m = 8 * 1 * 10^3 = 8$ KHz

c) $$\text{The total power} = \frac{A^2}{2} = \frac{10^2}{2} = 50W$$

6) when a 10MHZ carrier is frequency modulated, the frequency deviation is to be 50KHZ. Determine the bandwidth for the following information (modulating signal) frequencies:

a) f_m =500KHz b) f_m =500Hz c) f_m=5KHz

Solution:

$a)$

$$\beta = \frac{\Delta f}{f_m} = \frac{50KHz}{500KHz} = 0.1$$

$$\because \beta \langle\langle (\pi/2), \text{therefore the signal is NBFM}$$

$$\therefore \text{B.W} = 2\text{f}_\text{m} = 2*500\text{KHz} = 1000\text{KHz} = 1\text{MHz}$$

b)

$$\beta = \frac{\Delta f}{f_m} = \frac{50KHz}{500Hz} = 100$$

$$\because \beta \rangle\rangle (\pi/2), \text{therefore the signal is WBFM}$$

$$\therefore \text{B.W} = 2\Delta\text{f} = 2*50\text{KHz} = 100\text{KHz}$$

c)

$$\beta = \frac{\Delta f}{f_m} = \frac{50KHz}{5KHz} = 10$$

For this case we can use carson's rule to determine the bandwidth B.W=2(f_m+Δf)=2(5K +50K) =110 KHZ

Home work:

1) Find the carrier frequency, modulating frequency, the modulation index, and the peak deviation of the FM wave represented by the expression:

$$V_{FM} = 12\cos(6*10^8 t + 5\sin 1250t)$$

What will be the power of this signal?

2) Determine the modulation index for both commercial FM broadcast and TV transmission.

FM modulators and transmitters

FM Modulators

In this section we will discuss two methods for generating FM signal.

1) Direct method:

In this method, the modulating signal directly controls the carrier frequency. This process can be done by means of a device whose reactance can be varied by the

amplitude of the modulating signal. The *FM* signal is generated when this variable controlled reactance device is placed across the frequency-determining tank circuit of an oscillator. Thus, this system is ***Variable Controlled Oscillator (VCO)*.**

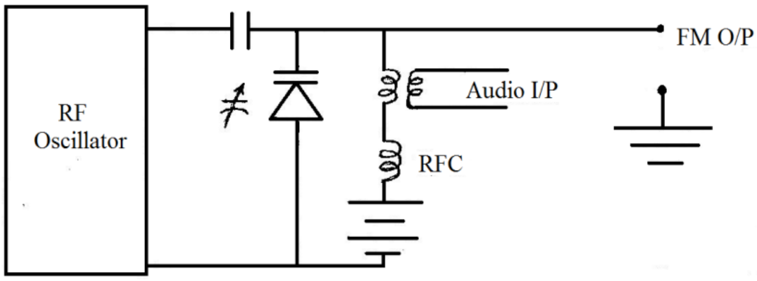
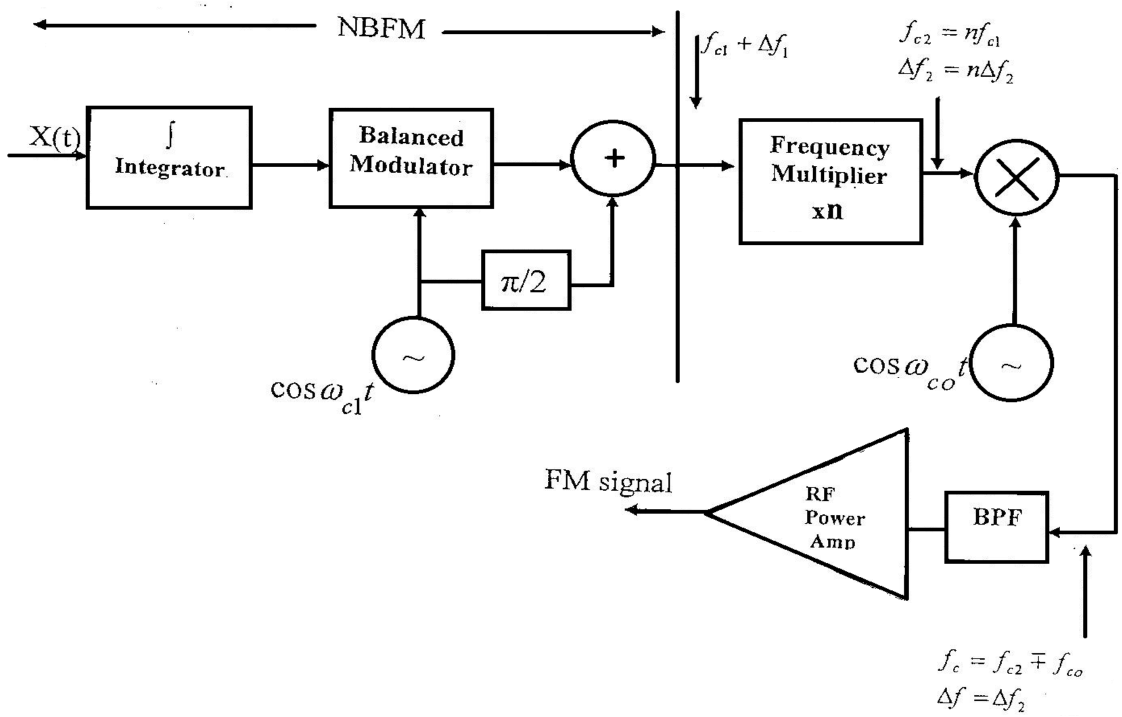


Fig. 28: The circuit diagram of "Direct method" (VCO circuit)

2) **Indirect method:**
In this method, the input data is first integrated and then phase-modulated to produce the *Narrowband FM signal (NBFM)*. The next step is to increase the modulation index to the desired values by means of *Frequency Multiplication unit* (see fig. 4.28).

Fig.29: The block diagram of "Indirect method"



FM transmitters:

Fig. 30 shows a block diagram of a simplified *FM transmitter*. The FM signal is generated at the Exciter unit by one of the two methods discussed previously. This unit is preceded by a unit called *"Preemphasis Network "* in which the higher frequencies of the information signal boosted and correspondingly cut by the

"Deemphasis Network" at the receiver. Hence, improvement <u>immunity</u> could be expected, see fig. 31.

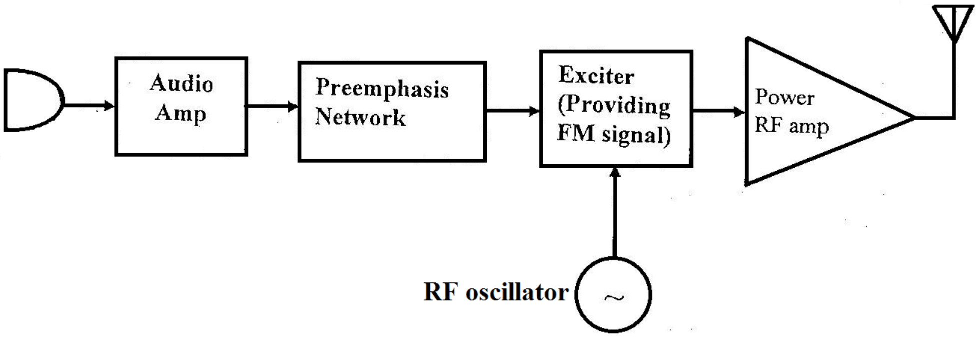
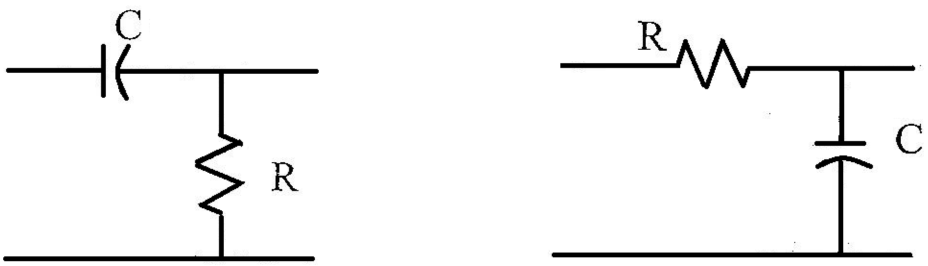


Fig.4.30: Simplified block diagram of FM transmitter



$\tau = RC = 75\mu s$

a) Preemphasis network　　b) Deemphasis Network

Fig. 31: Emphasis circuits

FM Demodulators and Receivers

A block diagram of a typical *FM receiver* covering the broadcast range of 88 *to 108MHZ* is shown in fig.32. Note that except for the *Limiter* and *Deemphasis* circuits, the form of the receiver is similar to that of a conventional *AM receiver*, since both of them employ heterodyning process. The *IF amplifier* is tuned to the center frequency of lO.7MHZ. The *Limiter* unit is a double-ended clipping circuit (see fig 4.33). <u>It squares-off the lower and the upper extremities of the signal to remove the amplitude variations caused by the noise</u>. The *Deemphasis* unit, in conjunction to the *Preempliasis* at the transmitter, provides additional discrimination against noise and interference.

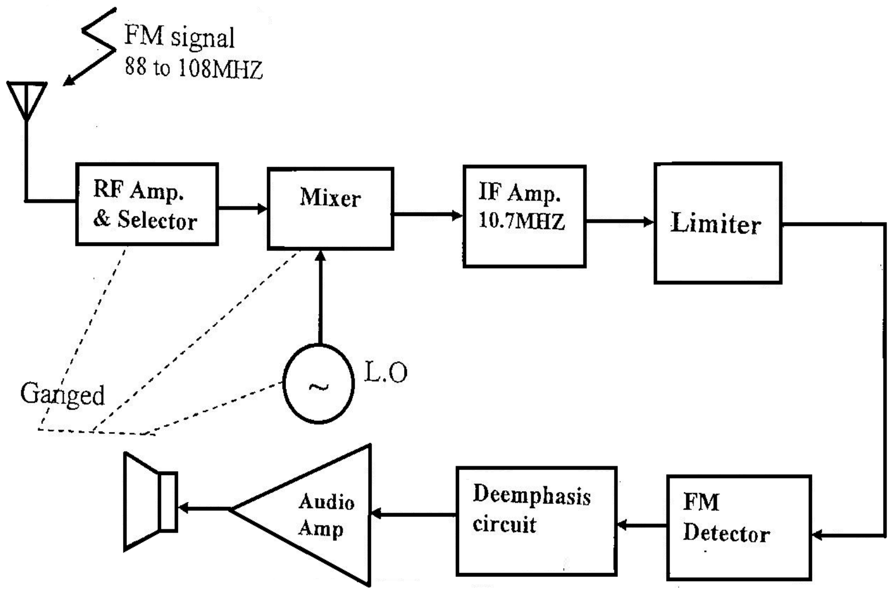


Fig. 32: The block diagram of FM Super heterodyne receiver.

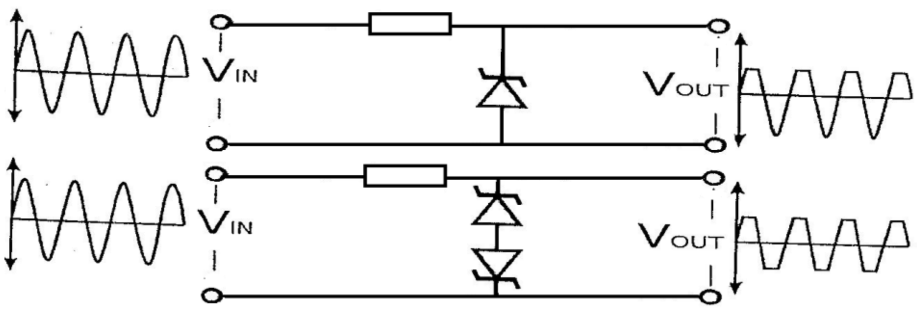


Fig.33: Circuit diagram and waveforms of the Limiter (Clipper) unit.

The *Detector (Demodulator)* unit detects the modulating signal from the *frequency modulated IF* signal. This process can be carried out in a variety of methods. In general, a frequency detector, often called a ***Discriminator***, produces an output voltage that should vary linearly with the instantaneous frequency of the input. We will discuss three methods for detecting the modulating signal from FM signal:

1) *The Foster-Seeley Discriminator (Phase Discriminator).*
2) *The Ratio Detector.*
3) *Phase-Locked Loop detector (PLL)*

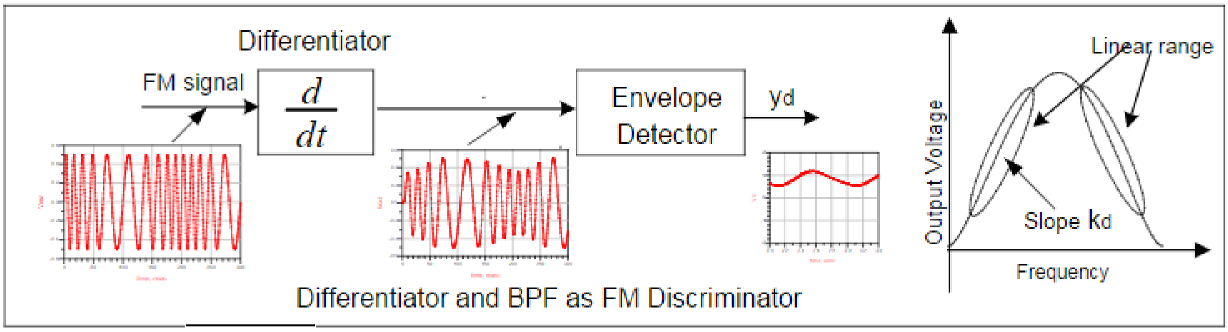


Fig.34 Ideal and real frequency demodulator

The circuit of this detector is shown in detail as in fig.4.33.

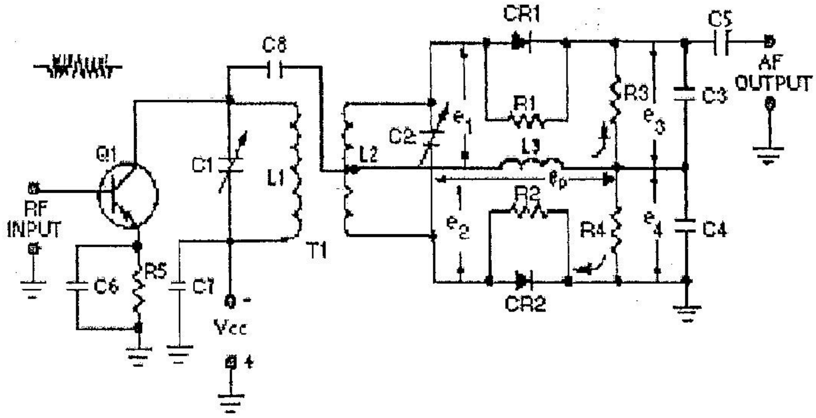


Fig.35 Detailed circuit diagram of Foster-Seeley Discriminator

The IF signal is directly-applied at both coil Ll and choke L3. The transformer Tl phase-shifts the IF signal at its secondary coil L2. L2 together with the capacitor C2 form a series resonance circuit. This circuit produces a frequency sensitive phase-shifting voltages el and e2. These voltages are combined with the original IF signal applied at L3 to form two AM signals applied at the anodes of diode CRI and diode CR2. These AM signals are detected by the envelope detectors (CRI, R3, and C3) and (CR2, R4, C4) to produce the AF output.

2) <u>The Ratio Detector:</u>

By making a few changes in the ***Foster-Seeley Discriminator***, it is possible to have FM detector which has built in capability of limiting process (Fig. 36) depicts the circuit diagram of *Ratio Detector*. L3 is tertiary coil of Tl which replaces the choke L3 in Foster-Seeley Discriminator. Notice that the diode CRl has been reversed because of L3 being added. Because of these changes a capacitor, C5, can be added to serve as a Limiter.

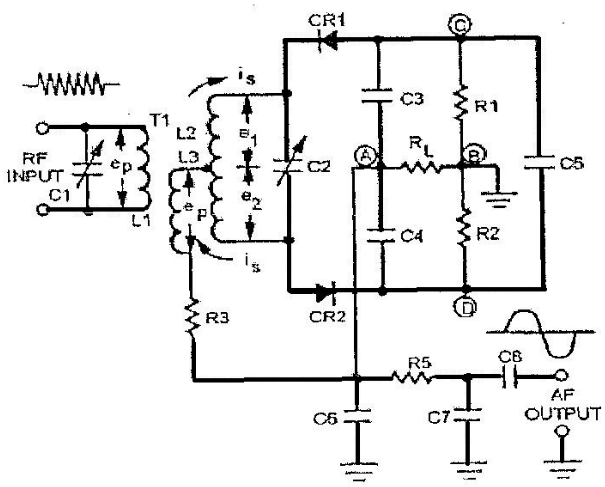


Fig.36: Circuit diagram of Ratio detector

3) Phase Locked Loop Detector (PLL):

The *Phase Locked Loop (PLL)* is a feedback system which may extract a baseband signal from a frequency-modulated carrier. The basic aim of a PLL is to lock or synchronize the instantaneous angle (i.e. phase and frequency) of a VCO (Voltage Control Oscillator)output to the instantaneous angle of external bandpass signal that may have some type of CW modulation. A block diagram of *PLL* system is shown in fig.37.

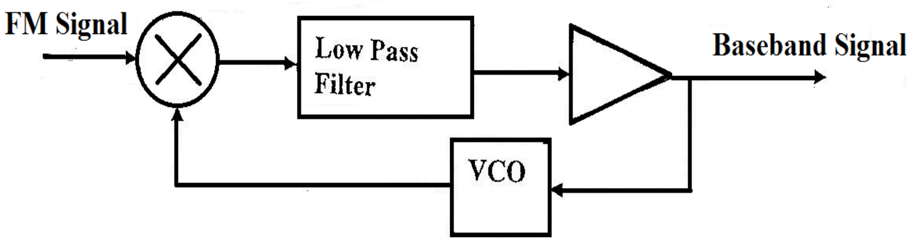


Fig. 37: A block diagram of Phased Locked Loop (PLL)

Q: Sketch the block diagram of Stereophonic PM transmitter

Solution:

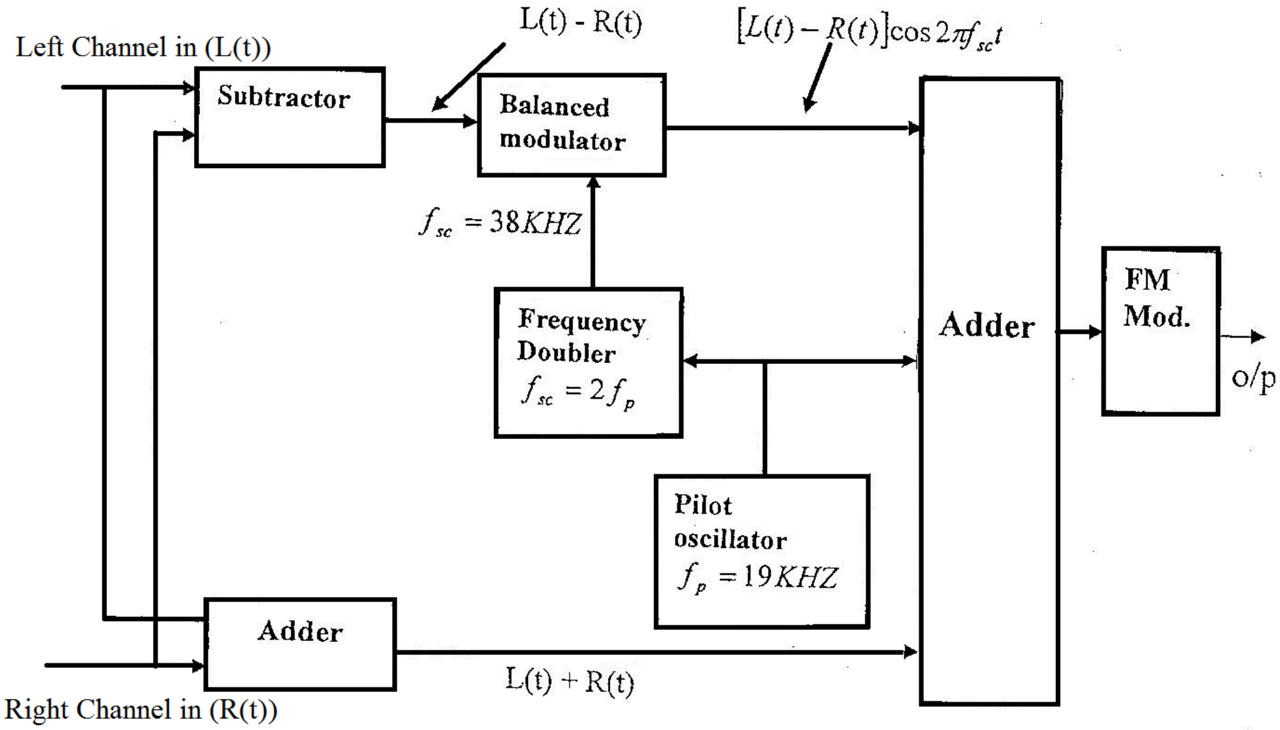


Fig.38

Compare between AM and FM systems.

No.	FM	AM
1	The frequency of the carrier is varied according to the modulating signal.	The amplitude of the carrier is varied according to the modulating signal.
2	It is a wide band system because large number of sidebands is produced.	It is narrowband system because only two sidebands are generated
3	It is with good *SIN* ratio	It is with *SIN* ratio less than that of FM system
4	AFT circuit is used in FM receivers.	AGC circuit is used in AM receivers
5	AFT circuit is used in FM	The transmitting and the receiving equipments are simple
6	Interference and noise have almost no effect on the performance FM systems.	Interference and noise degrade the the performance of AM systems.

(WEEKS 14-17)

Pulse Modulation:

In previous sections, we have discussed *CW* modulation in which the carrier is continuously modulated by the modulating signal. The carrier was sine or cosine wave. In this section, we will discuss another type of modulation; *Pulse Modulation'*, in which a discrete carrier signals, is modulated by the modulating signal. Hence, the carrier is in the form of train of pulses. ***Pulse modulation*, therefore, employs *sampling technique***. Thus, in sampling process, continuous waveforms are sampled at regular intervals. Hence, instead of transmitting continuous modulated carrier, modulated pulses or samples are transmitted.

Pulse modulation can be classified into two types:

1) *Analogue Pulse modulation*, which is similar to *CW* modulation. In this type, one of the pulse parameters, such as amplitude, width, or position is varied by the modulating signal.
2) *Coded Pulse modulation*, which has no CW equivalent. In this type, the pulses are quantized and encoded to binary bits (bits).

Pulse modulation offers two main advantages over *CW* modulation. They are:
1) The transmitted power can be concentrated into short bursts of pulses.
2) The time intervals between pulses (the off time) can be filled with samples values from other messages.

Analogue Pulse Modulation, PAM, PDM and PPM

The three types of *Analogue Pulse Modulation* are designated as:

1) PAM; Pulse Amplitude Modulation.
2) PDM or PWM; Pulse Duration or Width Modulation.
3) PPM; Pulse Position Modulation.

Fig. 39 shows waveforms of a typical message and its three types of *Analogue Pulse Modulation.* The methods of creating these types are also illustrated. As it is clear, the amplitude, the duration, and the position of the pulses is varied in direct proportion to the message signal.

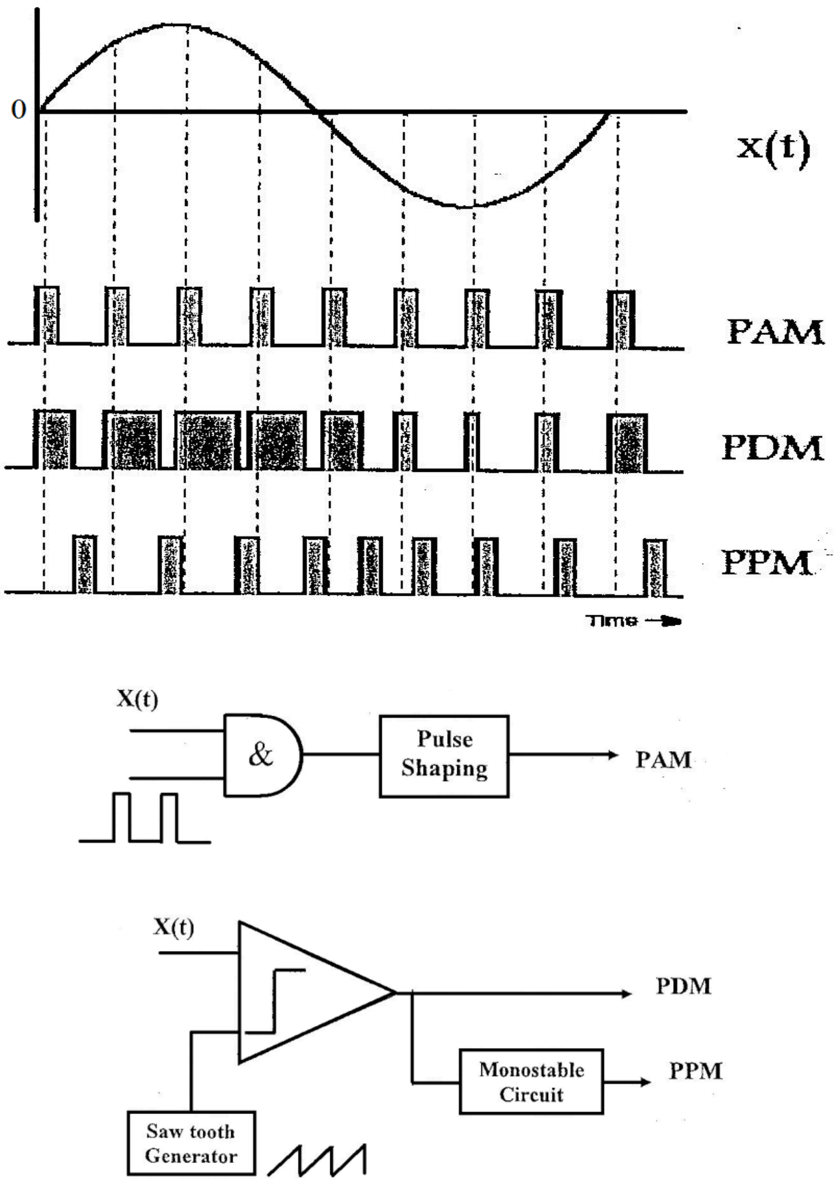


Fig. 39: Types of Analogue Pulse Modulation and their corresponding modulators

Note that the PDM, PPM modulator employs a comparator and a saw tooth wave generator with period *T* , The output is zero except when the value of the message waveform *X(t)* exceeds the value of the saw tooth wave at which the output goes to a positive value; A, see Fig.40.

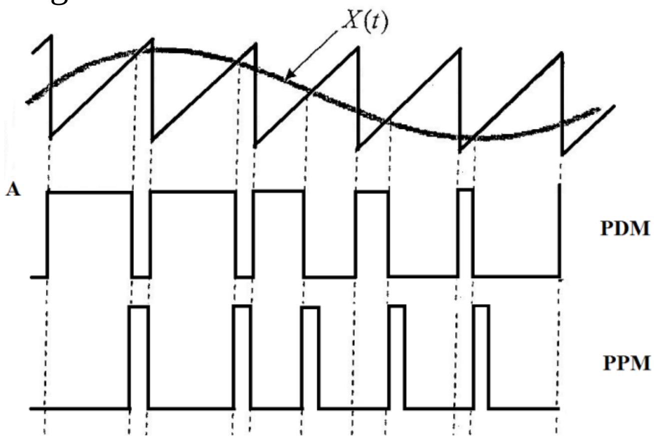


Fig. 40: PDM & PPM waveforms

Note that, from the waveforms in fig. 40, in *PDM,* the pulse duration or width is varied according to the modulating *signal X(t).* In *PPM,* the position of the pulse is shifted left or right according to the' modulating signal. In both types, the amplitude of the pulse is kept unchanged.
Recovering *X(t)* from *PAM* signal is shown as block diagram in fig. 4.40a. It consists of *Low Pass Filter* stage followed by an *Equalizer*. *Fig.4.40b* shows the demodulation process of *X(t)* from *PDM by Low pass Filter* with *a DC block.*
Finally, the *PPM* signal is first converted to *PDM* signal then this signal is *Low Pass Filtering* this signal to produce *X(t)* as shown in fig.41c.

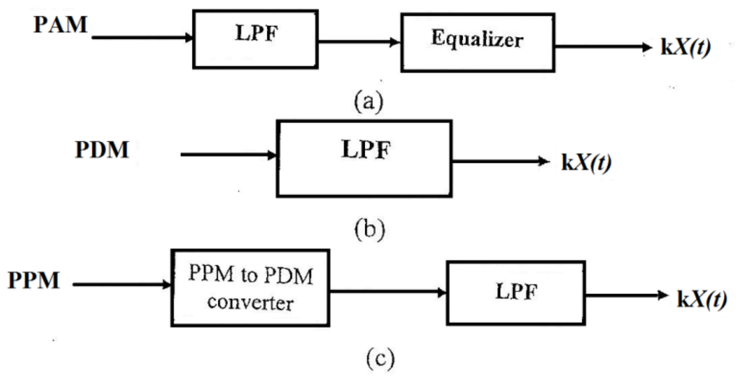


Fig.41: Demodulators of PAM, PDM, and PPM.

The ability to use constant-amplitude pulses is a major advantage of pulse modulation, and since *PAM* does not utilize constant-amplitude pulses, it is in frequently used. When it is used, the pulses frequency modulates the carrier.

PDM and *PPM* have the same advantage over *.PAM* as frequency modulation has over amplitude modulation. In *PDM* and *PPM,* the pulse amplitude remains constant, so that *Amplitude Limiter* can be used to provide a good degree of noise immunity.

PDM has a disadvantage when compared with *PPM* in that its pulses are of varying width and therefore of varying power content. On the other hand, *PDM works* if synchronization between transmitter and receiver fails whereas *PPM* does not.

Pulse-Code Modulation(PCM):

In common with the other forms of *Pulse Modulation, Pulse-Code Modulation; PCM,* also uses the sampling technique, but it differs from the others in that it is a *digital process.* That is, instead of sending a pulse train capable of continuously varying one of the parameters, the *PCM* generator produces a series of numbers, or *digits.*

In PCM the message is represented by coded group of digital pulses. The modulating signal *X(t)* is first Low Pass Filtered (why?) and sampled to give Xs*(t).* The samples are then quantized to the nearest predetermined discrete values. The quantized samples X_{sq} (t) are then encoded to digital words. The elements-of *PCM* generation are diagrammed in fig. 42 which, viewed in another light, is *Analogue-to-Digital converter (A/D).*

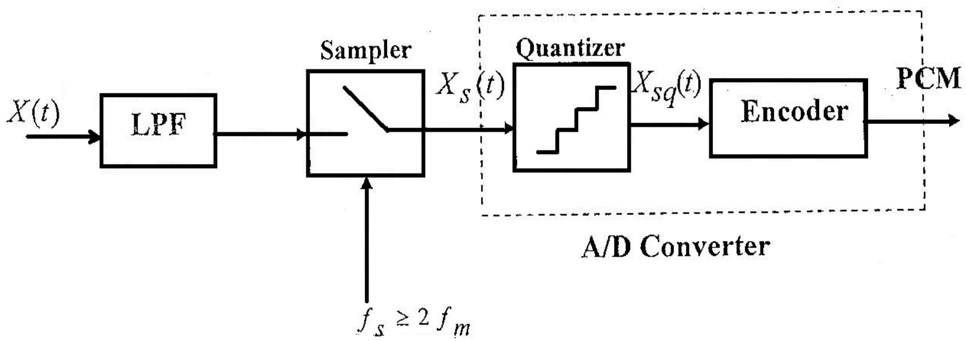


Fig. 42: Block diagram of PCM generation system

In the *Quantizer* unit, the total amplitude range which the signal may occupy is divided into a number of standard levels. Since these levels are transmitted in binary code, the actual number of levels is *a power of 2.* Thus, if the *Quantizer* has *Q* levels and the number of bits in each digital word is *v,* then for binary *PCM:*

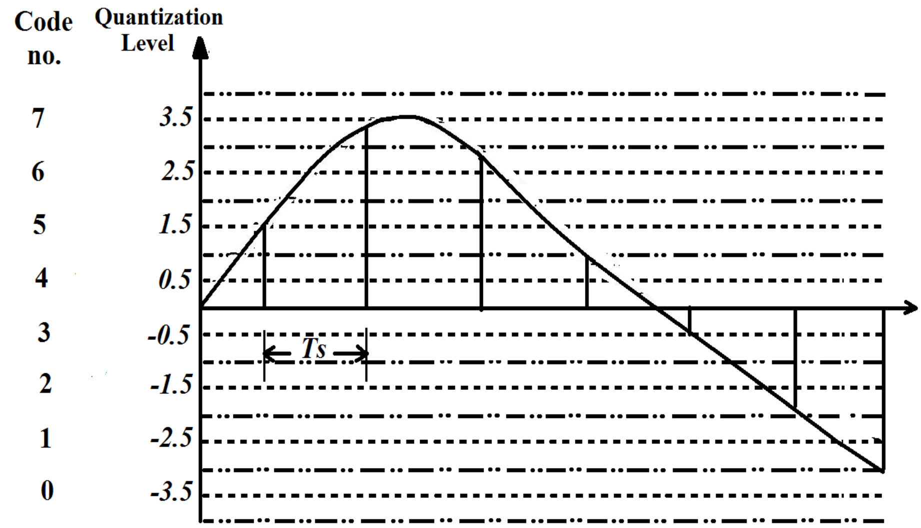
$$Q = 2^v \text{.........................}(29)$$

And the PCM bandwidth will be :

$$(B.W)_{PCM} = vf_m \text{..............................}(30)$$

For the analogue signal shown in fig. 43 which is limited in its excursions to the range of -4 to +4 volts, a Quantizer with eight levels must be used in order that the sampled values can be converted to digital words each one consists of three bits. The

quantization levels are located -3.5, -2.5, -1.5 ... and +3.5 volts and the code numbers 0, 1,2 ... and 7 are assigned to these levels respectively.



Sample values	1.3	3.3	2.4	1	-0.25	-1.25	-2.4
Nearest quantization level	1.5	3.5	2.5	1.5	-0.5	-1.5	-2.5
Code Number	5	7	6	5	3	2	1
Binary Representation	101	111	110	101	011	010	001

Fig.43: A message X(t) is sampled, quantized, and then encoded to digital words to produce PCM signal.

Thus the transmitted *PCM* for the signal *X(t)* shown in fig. 44 will be:

101111110101011010001

For speech signal, it is normally used that *v*= 7 and Q=128 while for video signals v= 8 and Q=256 are used.

Fig.4.43 depicts the receiver system of *PCM.* Note that the input to the receiver is PCM plus quantization noise and channel noise. These two types of noise are removed in the PCM reconstruction unit, then the coded- pulses are decoded to the original sampled pulses in the *Digital-to-Analogue D/A* unit The output of *D/A* stage is then Low Pass Filtered to recover the original message X(t).

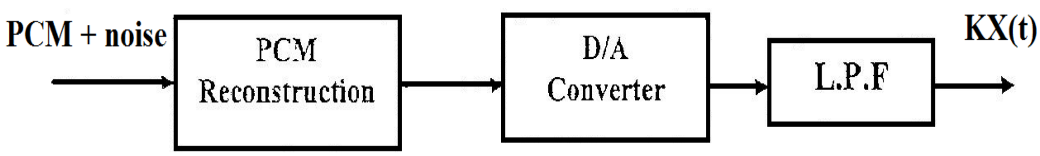


Fig.44: A block diagram of PCM receiver

Solved problems

4,9) Find the bandwidth of a PCM system used to transmit speech signal sampled at f_s==8000Hz and each sample is encoded into 7 binary digits.

Solution:

$$v = 7 \quad \text{as given}$$

$$\therefore Q = 27 = 128 \text{ Levels}$$

$$\therefore (\text{B.W})_{PCM} = v * f_m = v * \frac{f_s}{2} = 7 * \frac{8000}{2} = 28KHz$$

4.10) A speech signal is transmitted using PCM with sampling rate f_s==8000Hz and quantization levels Q =32. Find the bit rate in bits/sec. & bandwidth required to transmit the code over AM channel.

Solution:

$$\text{Since } Q = 32 \text{ as given, then}$$

$$Q = 2^{v} \Rightarrow 32 = 2^{v} \Rightarrow 2^{5} = 2^{v}$$

$$\therefore v = 5 \quad \text{i.e. each pulse is encoded into5bits. Then}$$

$$T_s = \frac{1}{f_s} = \frac{1}{8000} = 125\mu s$$

$$\therefore \text{The time occupied by each bit is } t_b = \frac{125*10^{-6}}{5} = 25\mu s$$

$$\therefore \text{Bit rate} = \frac{1}{t_b} = \frac{1}{25*10^{-6}} = 40000 bits/\sec$$

or

$$\text{Bit rate} = (\text{no. of samples/sa mple}) * f_s$$

Now

$$(B.W)_{PCM} = v * \frac{f_s}{2} = 5 * \frac{8000}{2} = 20KHz$$

$$\therefore (B.W)_{AM} = 2 * (B.W)_{PCM} = 2 * 20KHz = 40KHz$$

Homework

1) If the quantization *levels* Q in the solved problem () is changed to 128 and f_s is unchanged. What will be the new bit rate? And what will be the bandwidth required to transmit the speech signal over the same AM channel?

Multiplexing

Multiplexing is the process of transmitting several 'messages on one transmission media. There are two basic multiplexing techniques:

Frequency-Division Multiplexing (FDM)

Frequency-Division Multiplexing (FDM) is produced by superposition (addition) of "N" messages separated in frequency (see fig. 45). Thus, the available channel bandwidth is divided into a number of non-overlapped "Slots" and each message is assigned a slot of frequencies. ***FDM*** is used in long distance *telephone*, FM stereo and TV broadcasting, space probe telemetry, and other applications.

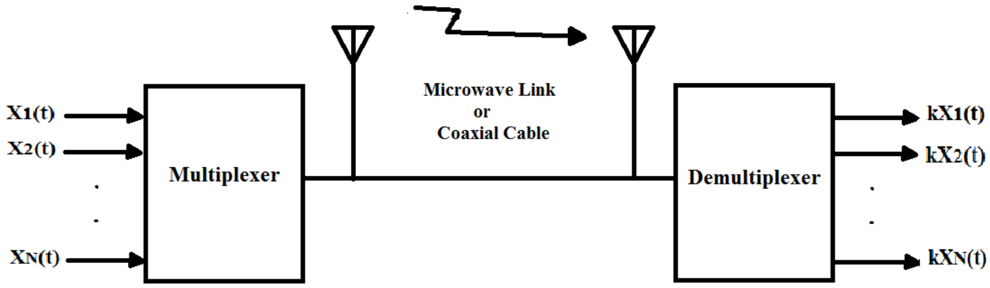


Fig.45: Illustration of FDM

The block diagram of FDM technique is shown in fig.45 , where several input messages (X_1, X_2 ... X_N) individually modulate sub-carriers f_{c1} ,f_{c2}, ... and, f_{cn} after being passed through Low Pass Filter. The sub-carriers are modulated as ***DSB* or *SSB*,** then the outputs of the modulators are summed to produce the base band signal; Xb(t). Finally, Xb(t) modulates a master oscillator.

Demodulation of ***FDM*** is accomplished in three steps, see fig.4.46 The first demodulator recovers the base band signal Xb(t) from the modulated master carrier, .. Then this base band signal is passed through bank of Band Pass Filters to retrieve the modulated sub-carriers. Finally, each message is recovered from the corresponding modulated sub-carrier by using product demodulator or phase shift demodulators.

The major practical *problem* of ***FDM*** is the ***Crosstalk;*** the unwanted coupling of one message into another. To reduce this coupling, the modulated messages spectra are spaced out in frequency by the ***Guard band*** *"W"*.

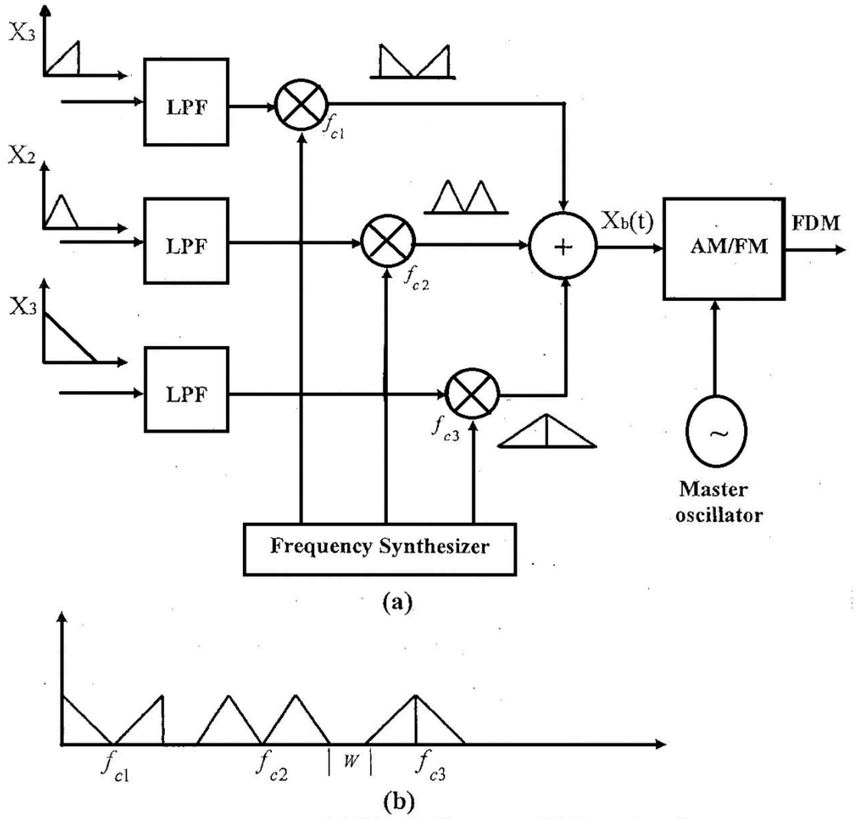


Fig.46: Typical FDM transmitter a) Block diagram b) Baseband spectrum

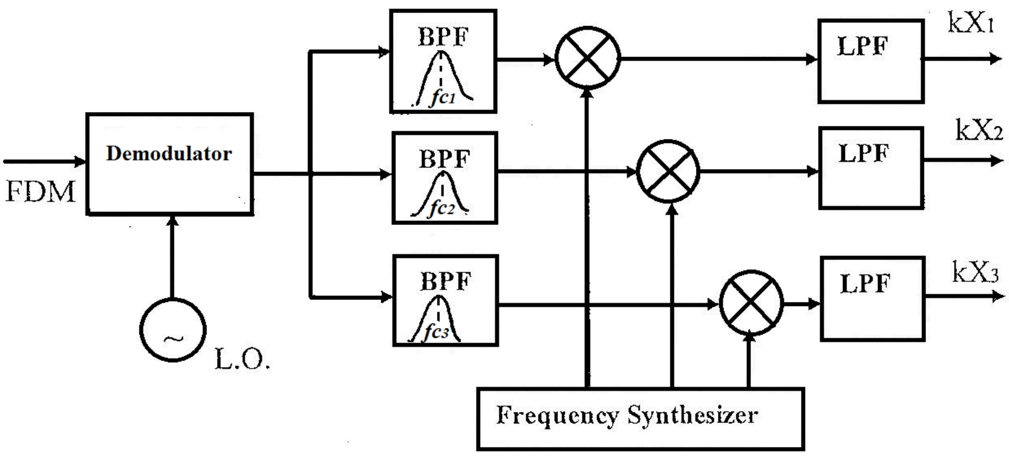


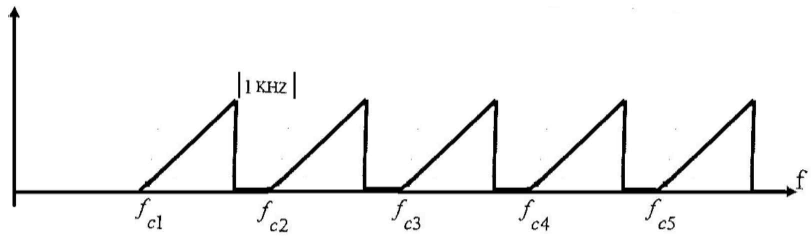
Fig.47: Block diagram of FDM Receiver

Example:

Five signal each is band limited to 3KHz are to be FDM -ed with 1KHz guard band (W==l KHz). The subscriber modulation is SSB and f_{c1} = 100KHz. Sketch the spectrum of base band signal and calculate the required bandwidth.

Solution

$(B.W)_T = (Bo + W)*N = (3+1)*5 == 20KHz$



f_{c1} =100 KHz, f_{c2} =104KHz, f_{c3}=108KHz, f_{c4}=112KHz, f_{c5}= 116KHz

The sub carriers f_{c1} , f_{c2} , f_{c3}, f_{c4}, f_{c5} are generated by the *Frequency Synthesizer unit.*

Time-Division Multiplexing (TDM):

In *Time Division Multiplexing (TDM),* each message signal occupies the channel (e.g. a *transmission line)* for a short period of time. The *TDM* system is illustrated in fig. 48. Several input signals are filtered by the bank: of *LPFs* and sampled sequentially. The rotating sampling switch or *commutator* at the transmitter extracts one sample of each input per one revolution. The time of one revolution is $T_s \leq (1/2B)$ seconds where ***B*** is the pass band of the *LPFs*. Hence, the output of the *commutator* is a *PAM* waveform that contains the individual samples periodically interlaced in time. A similar rotary switch at the receiver; a *decommutator* or *distributor*, separates the samples and distributes them to another bank of *LPFs* for the reconstruction of individual messages.
The time interval T_s containing one sample from each input is called *e* ***frame***, If there are *M* input channels, the pulse-to-pulse spacing within *a frame* is $T_s/M = 1/Mf$, (see fig. 48) . Thus, the total number of pulses per second will be:

$$R = Mf_s \text{...} \quad (4.31)$$

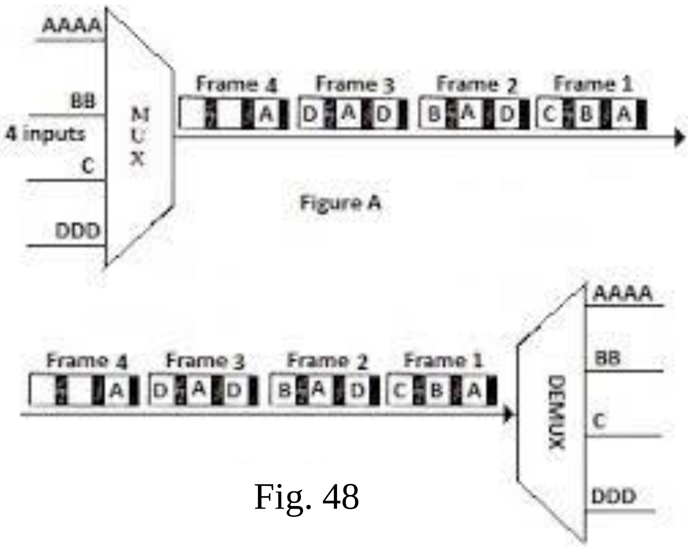


Fig. 48

Example:

If a carrier telephone system has to transmit l2/4KHZ channels. Find the number of pulses/sec if TDM-P AM system is used. An extra pulse required for a frame.

Solution:

Number of pulses / frame = 12+ 1 = 13 *pulse*

$f_s = 2f_m = 2 * 4 = 8KHz$

:. *Number of pulses* / sec *or pulse rate of TDM – PAM*

$r = Mf = 13 \text{x} 8 = 104 Kpulse/sec$

Digital Modulation:

Data communications becomes important with the expansion of the use of computers and data processing and have continued to develop into a major industry providing the interconnection of computer networks and transmission of data between distant sites.

Digital information can modulate the amplitude, frequency, and phase of a sinusoidal carrier wave. Therefore, in digital modulation, the information has a digital form while the carrier is a continuous form.

4.4.1 Amplitude Shift Keying (ASK) or ON-OFF Keying (OOK):

This method, the binary information switches the carrier between two states; *ON* or *OFF,* see fig.4.50. The resultant waveform consists of *"ON'* mark pulses representing binary "1" and "OFF' space pulses representing binary "0". The *ASK* modulator and demodulator is shown in fig. 49.

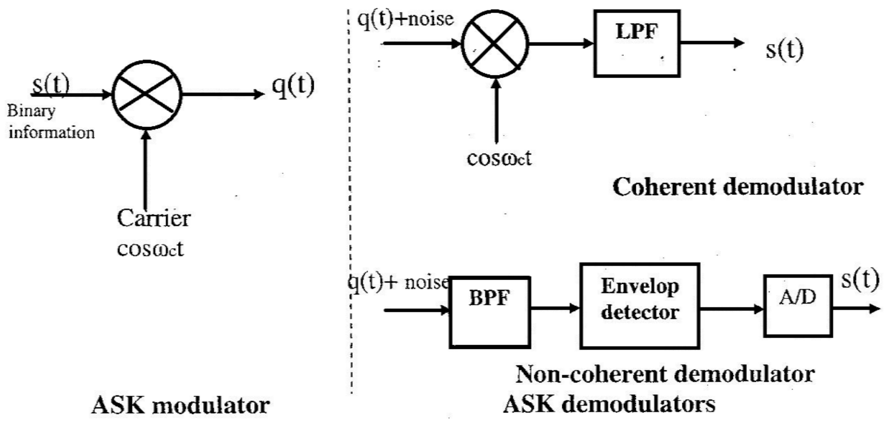


Fig. 49: ASK modulator and demodulator

The ASK signal; q(t), is expressed mathematically as:

$$q(t) = S(t) * \cos w_c t = \begin{cases} \cos w_c t & \text{logic } 1 \\ 0 & \text{logic } 0 \end{cases} \ldots\ldots\ldots\ldots\ldots(4.31)$$

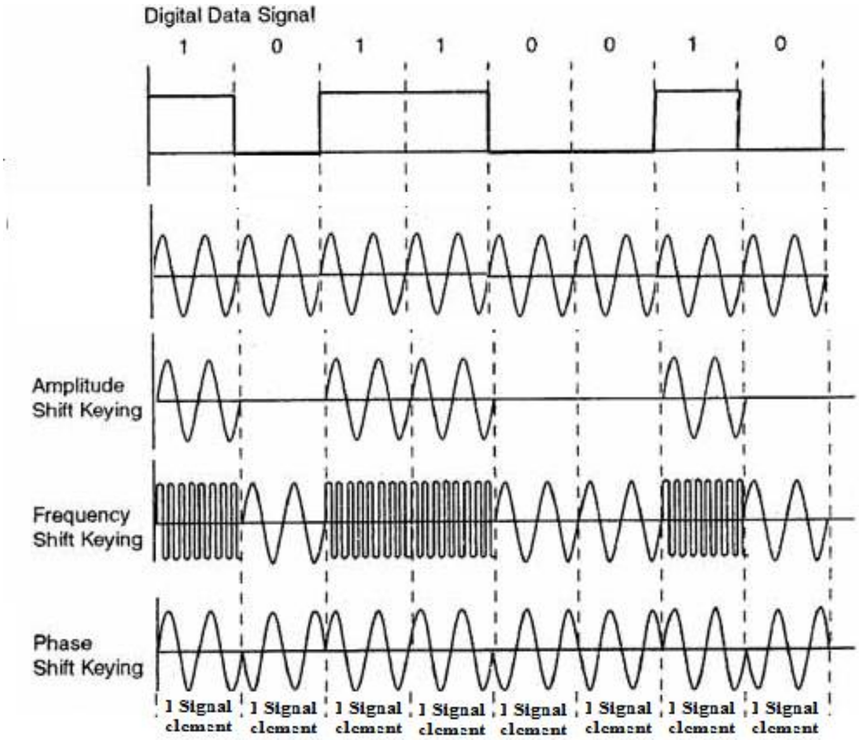


Fig.50: Waveforms of ASK, FSK, and PSK.

Frequency Shift Keying (FSK):

In this modulation type, the binary information deviate the frequency of the carrier into two values; $f_1=f_c+\Delta f$ when the information is logic" 1" and $f_1=f_c-\Delta f$ when the information is logic "0". Thus, the ***FSK*** is considered as the addition of two ***ASK*** signals, see fig. 50). The expression for ***FSK*** is:

$$q(t) = \cos w_i t = \begin{cases} \cos w_1 t = \cos(w_c + \Delta w)t & \text{logic } 1 \\ \cos w_2 t = \cos(w_c - \Delta w)t & \text{logic } 0 \end{cases} \quad \ldots\ldots\ldots\ldots\ldots(4.32)$$

$$i = 1, 2$$

The modulator and demodulator of ***FSK*** are shown in fig. 51.

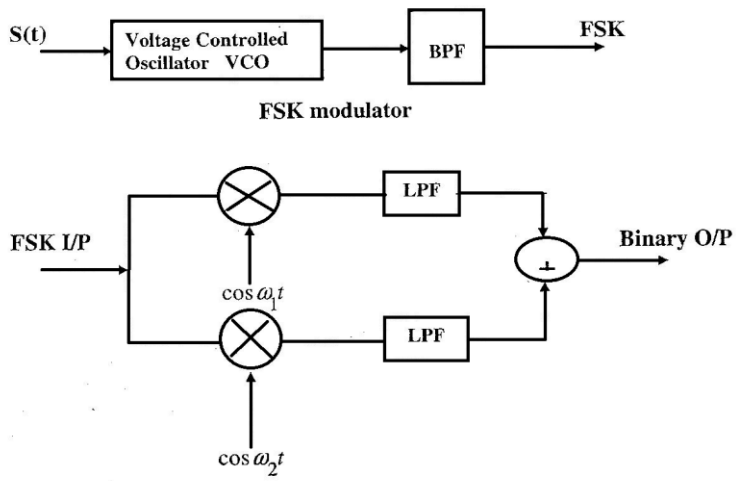


Fig. 51: Modulator and demodulator of FSK signal.

Phase Shift Keying (PSK):

In this method of digital modulation, the binary information alters the phase of the carrier to 180° (the negative version) when the information is binary "1", while the logic"0" information keeps the phase of the carrier unchanged, see fig. 50. The ***PSK*** expression is:

$$q(t) = \cos(w_c t + \phi) = \begin{cases} \cos(w_c t + 180°) = -\cos w_c t & \text{logic } 1 \\ \cos(w_c t + 0°) = +\cos w_c t & \text{logic } 0 \end{cases} \quad \ldots\ldots\ldots\ldots\ldots(4.33)$$

The modulator and demodulator of ***PSK*** signal is depicted in fig. 52. Note that the binary information S(t) is first changed to **S′(t)** before applying to the modulator. The output of the modulator is S′ (t) and must be re-changed to S(t) .

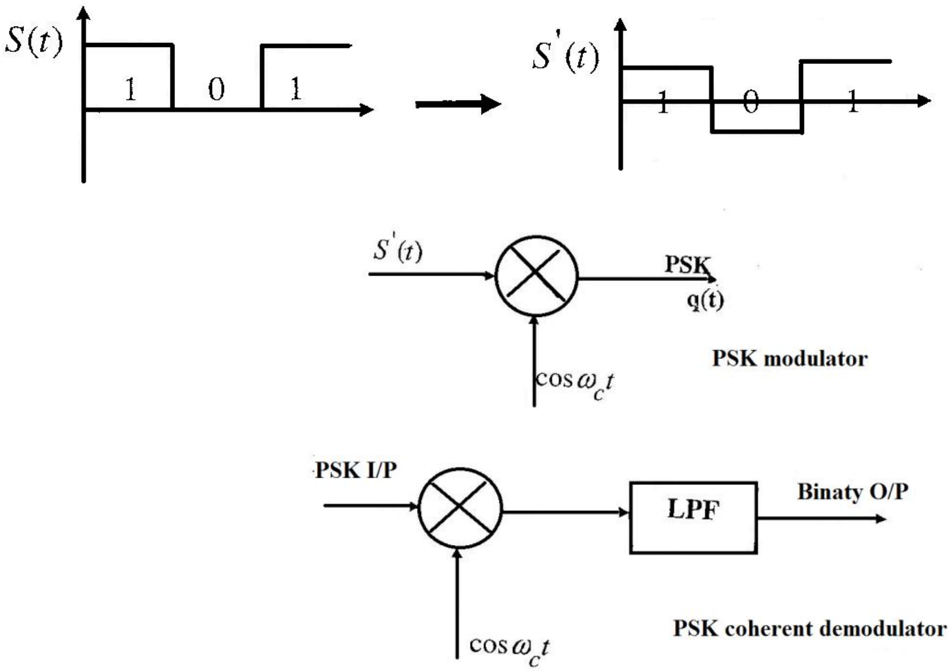


Fig. 52: Modulation and demodulation methods of PSK signal

For conserving the bandwidth, combined with the constant amplitude property, multilevel ***PSK*** has considerable practical value. In four-phase ***PSK*** (also called *Quaternary* ***PSK, QPSK),*** one of four possible waveforms is transmitted during each signaling interval. These waveforms are:

$$q_1(t) = A\cos w_c t \qquad \phi = 0°$$
$$q_2(t) = -A\sin w_c t \qquad \phi = 90°$$
$$q_3(t) = -A\cos w_c t \qquad \phi = 180°$$
$$q_4(t) = A\sin w_c t \qquad \phi = 270°$$

Information Theory

Information theory is a mathematical subject dealing with three basic concepts:

1) The measure of information.
2) The capacity of a communication channel to transfer information.
3) Coding as a means of utilizing channel at full capacity.

Thus, the fundamental concept of *information theory* is as follows:
((Given an information source and a communication channel, there exists a coding technique such that the information can be transmitted over the channel at any rate less than the channel capacity and with small frequency of errors despite the presence of noise)).

Information measure:

Information is simply that which is produced by the source for transfer to the user. This implies that before transmission, the information was not available at the destination; otherwise the transfer would be zero. Thus, the information measure is the choice of one message of a finite set of messages. The less likely the messages, the more information it conveys to the user. At the transmitting end of a communication system, information measure is can indication of the freedom of choice exercised by the source in selecting a message. Thus, the information measure involves probabilities. The rate of information transfer is called *Information rate* or *Entropy* and designated as *"R"*. It is measured in bits/sec (binary digits per second).

Channel capacity:

The communication channel is a model representing the vehicle of transmission plus all phenomena that tend to restrict transmission. Just as information rate measures the amount of information produced by a source in a given time, *Capacity* is a measure of the amount of information a channel can transfer per unit time. Channel capacity is symbolized by *"C"* and its unit is bits/sec. According to the fundamental concept of the information theory mentioned above, the information rate is related to the channel capacity as:

$$R \le C \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots (1)$$

If $R > C$, it is not possible to transmit without *errors*.

Discrete Noiseless channel

A discrete channel is one that transmits information by successively assuming various disjoint electrical states, voltage levels ... etc. Let" μ" be the number of possible states and "r" is the signaling rate in states per unit time and if the signal-to-noise ratio is sufficiently large, the probability of the error at the receiver can be extremely small, then the capacity of a discrete channel will be:

$$C = r \log_2 \mu \qquad bits/\sec \ldots\ldots\ldots\ldots\ldots (2)$$

For a binary channel (μ =2) the capacity is equal to the signaling rate, that is C = *r* . Thus, for a given "r", the capacity of noiseless channel increases as μ increases.

Continuous channels

A continuous channel is one in which messages are represented as waveforms i.e. continuous functions of time, and the relevant parameters are *Bandwidth "B"* and *Signal to Noise ratio "SIN"*. Then the capacity of continuous channel will be:

$$C = B\log_2\left(1+\frac{S}{N}\right) \qquad bits/\sec\ ....3)$$

This famous equation is called the *Hartley-Shannon* law. This equation indicates that a noiseless channel (S / N)= ∞ has an infinity capacity. On the other hand, the channel capacity does not become infinite as the bandwidth becomes infinite because with an increase in bandwidth, the noise power also increases. This equation also indicates that we may trade off bandwidth for signal to noise ratio and vice-versa.

Solved Problems:

P5.1) A noiseless discrete channel is transmitting information at a rate of 900 bits/sec. Determine the minimum of channel states if r =200 & r = I 000.

Solution:

$$C = 900\ \ bits/\sec \qquad \text{as given}$$

$$C = r\log_2 \mu$$

$$or \quad \log_2 \mu = \frac{C}{r}$$

$$i.e. \quad \mu = 2^{\frac{C}{r}}$$

$$for \quad r = 200$$

$$\mu = 2^{\frac{900}{200}} = 2^{4.5} \cong 23$$

$$for \quad r = 1000$$

$$\mu = 2^{\frac{900}{1000}} = 2^{0.9} \cong 2$$

P5.2) calculate the capacity of standard 4KHZ telephone channel with Signal to Noise power ratio of 32dB.

Solution:

$$\left(\frac{S}{N}\right)_{db} = 32 = 10\log_{10}\frac{S}{N}$$

Note :

$$\text{If } x = \text{power ratio in dB} = \left(\frac{P_1}{P_2}\right)_{db}$$

$$then \;\; x = 10\log_{10}\frac{P_1}{P_2}$$

or

$$\frac{P_1}{P_2} = 10^{\frac{x}{10}}$$

and

$$\frac{P_1}{P_2} = 10^{\frac{y}{20}} \quad \text{if Y is a voltage ratio in dB}$$

$$\text{also} \quad \log_2 Z = 3.32\log_2 Z$$

$$\therefore \frac{S}{N} = 10^{\frac{32}{10}} = 1585$$

$$\text{and} \;\; C = B\text{Iog}2\left(1+\frac{S}{N}\right) = 4*10^3 * 3.32 * \text{Iog}_{10}(1+1585)$$

$$= 42500 \text{ bits/sec} = 42.5 \text{ kbits / sec.}$$

3) A system has a bandwidth of 4KHZ and a Signal to Noise ratio of 28dB at the input of the receiver. Calculate:

a) Its information-carrying capacity.
b) The capacity of the channel if its bandwidth is doubled, while the transmitted power remains unchanged.

$$a)\left(\frac{S}{N}\right)db = 28 \qquad \text{as given}$$

$$\frac{S}{N} = 10^{\frac{28}{10}} = 631 = 631:1$$

$$\therefore C = B\log_2\left(1+\frac{S}{N}\right) = 4*10^3 * \log(1+631) = 37.193 \text{ kbits/sec}$$

(WEEK 18)

Mobile Phone

A **mobile phone** is a telephone that can make and receive calls over a radio frequency carrier while the user is moving within a telephone service area. The radio frequency link establishes a connection to the switching systems of a mobile phone operator, which provides access to the public switched telephone network (PSTN). Most modern mobile telephone services use cellular network architecture, and therefore mobile telephones are often also called *cellular telephones* or *cell phones*. In addition to telephony, modern mobile phones support a variety of other services, such as text messaging, MMS, email, Internet access, short-range wireless communications (infrared, Bluetooth), business applications, gaming, and photography. Mobile phones which offer these and more general computing capabilities are referred to as smart phones.

Advantages of Mobile Phones

1-Easy Communication
2-Always Connected
3-Multiple Uses
4-Emergency Situations

Disadvantages of Mobile Phones

1-Constant Interruption

Since you're always connected when you have your mobile phone, it becomes harder to ignore interruptions. People are always on their phones receiving calls, checking out their friends on Facebook, chatting on WhatsApp, checking emails and listening to music. It has become virtually impossible to avoid unneeded interruption

2-Possibility of Privacy Leak

Having all your information on your device is very convenient. However, it's also dangerous because there's a possibility of someone else accessing your phone. Mobile phone theft is quite common and it could leave you exposed.

3-Distraction

The accidents caused by usage of mobile phones when driving is innumerable. The temptation to pick an important call when driving is huge. While you may have done it successfully in the past, it is not a good idea. The distraction could easily cause you to lose control of the car and consequently cause an accident.

4-Affect Real Interaction

Today, socializing that involves real interaction is very rare. People have been reduced to interacting on social platforms such as Facebook and Twitter, or chat applications such as Viber and WhatsApp. While there's nothing expressly wrong with chatting with your friends on these platforms, it can be a problem if it is done at the expense of face to face interaction. It can easily take you away from the real life activities and you will find it hard interacting with real people. In a family where every family member has a mobile phone and uses it every time they are together, it might cause breakdown of relationships and families.

Frequency mixing between GSM 900/1800 and GSM 850/1900

Some countries in Central and South America have allocated spectrum in the **900 MHz and 1800 MHz bands for GSM** in addition to the common **GSM deployments at 850 MHz and 1900 MHz for ITU-Region 2 (Americas)**. The result hereof is a mixture of usage in the Americas that requires travelers to confirm that the phones they have are compatible with the band of the networks at their destinations. Frequency compatibility problems can be avoided through the use of multi-band (tri-band or, especially, quad-band) phones. See fig. 1,2 and 3.

Frequency Division Multiple Access (FDMA)

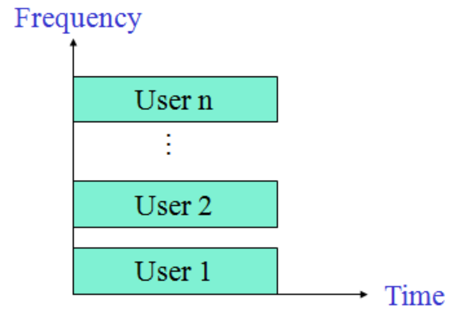


Fig. 1

- Single channel per carrier
- All first generation systems use FDMA

Time Division Multiple Access (TDMA)

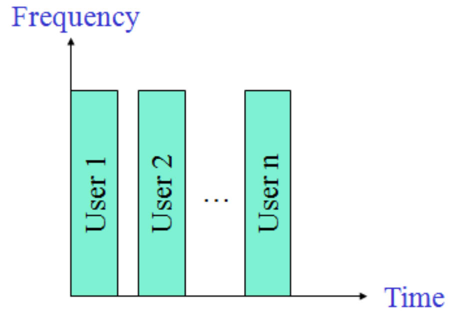


Fig. 2

- Multiple channels per carrier
- Most of second generation systems use TDMA

Code Division Multiple Access (CDMA)

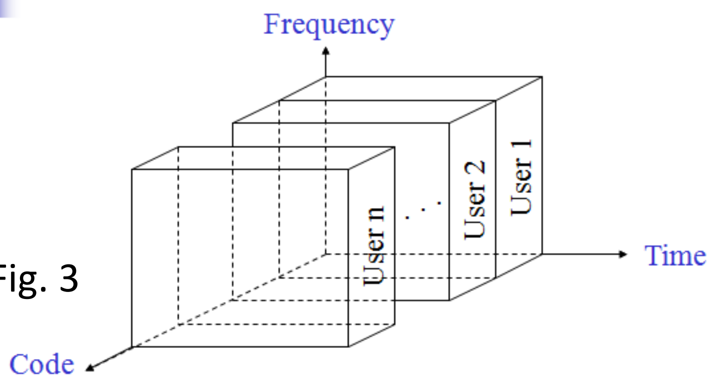


Fig. 3

- Users share bandwidth by using code sequences that are orthogonal to each other
- Some second generation systems use CDMA
- Most of third generation systems use CDMA

(WEEK 19)

TELEGRAPH

BACKGROUND, THEORY, & CONSTRUCTION OF THE "ELECTRIC TELEGRAPH"

Ever since the beginnings of time, people have been trying to communicate over distances greater than the human voice could reach. Early attempts included the use of smoke signals, signal fires, waving flags, and the moving arms of semaphores. Mirrors were also used to flash the image of the sun to distant observers.

After the discovery of electricity, wires were stretched from one point to another and an electric current was either allowed to flow through the wires or broken by a switch called a telegraph key. The electric current was first used to make marks on a paper tape and later, it was used activate a "sounder" which made clicking sounds. The short and long times between the clicks could be decoded into letters from the alphabet.

This revolutionary discovery allowed people to communicate instantly over distances that had required days or weeks for horse or train-carried messages. Telegraph stations were set up along railroads first because the right-of-way had already been cleared and it was easy to set up poles to carry the telegraph wires. Railroad dispatchers sent messages via telegraph to control the movement of trains and the wires also began to carry messages telling of news events and business transactions.

HOW LAND-LINE TELEGRAPH WORKS:

A telegraph system is basically an electrical circuit consisting of 3 parts, all hooked together by a WIRE.

The circuit is shown below: (The lines indicate the wires and the arrowheads show the path of the electrical current as it flows through the wires.)

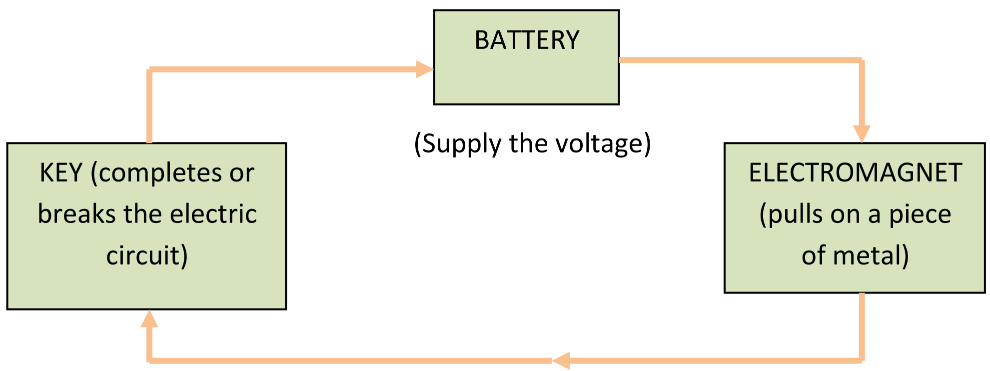


Fig.4

The WIRES were usually made of copper because it conducted electricity better than other metals.

It consist also **battery**, **key** and **electromagnet**.

TYPES OF "DETECTORS"

First, it was found that the electromagnet could move a compass needle and the "Needle Telegraph" began to be used.

1. When the key was closed for a short time and then a longer time, the pencil marked the paper with a dot followed by a dash and this signified the letter "A". This paper tape writing device was called a "**REGISTER**".
2. In the 1850's telegraph operators began to realize that they could recognize the different sounds made by the register as dots and dashes and a new detector mechanism called a "**SOUNDER**" was invented. This device used an electromagnet to pull on a piece of iron and make a clicking sound.
3. A dot was a **CLUNK** followed a short time later by a **CLICK**. A dash was a **CLUNK** followed, a long time later by a **CLICK**. This method of copying the code by ear persisted well into the 1950's.
4. Sounders continued to be improved and the most important improvement was to place them in a small wooden partial-enclosure called a "**RESONATOR**" which had the effect of amplifying the sound by bouncing the echoes of the sounder out the front of the resonator along with the original sound. Sounders in resonators became an integral part of every telegraph system.
5. The original "**Morse code**" (Also called the "American Morse Code") was used on the land-lines in this country but a slightly different code called the "Continental" or "International" code was used in Europe and on the radio waves.

The Basic Circuit of Telegram

The diagram shown below is representative of the simplest early Morse telegraph circuit.

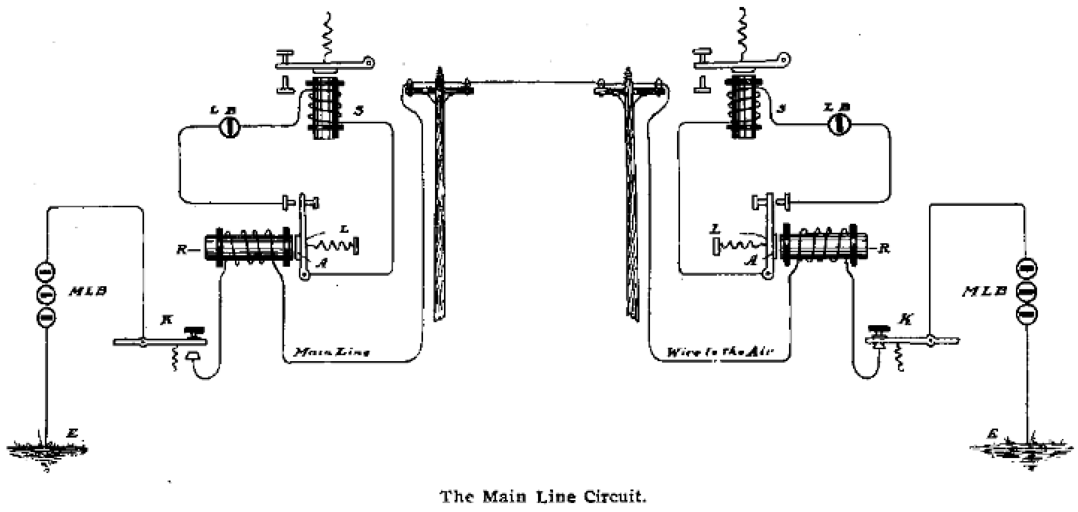


Fig. 5

This diagram shows a Morse telegraph circuit connected between to distant points. The point at which telegraph messages are sent and received is called an "office". In this case, each office is equipped with three instruments and two batteries.

(WEEK 20)

Faximile transmission

Fax (short for **facsimile**), sometimes called **telecopying** or **telefax** (the latter short for **telefacsimile**), is the telephonic transmission of scanned printed material (both text and images), normally to a telephone number connected to a printer or other output device. The original document is scanned with a **fax machine** (or a **telecopier**), which processes the contents (text or images) as a single fixed graphic image, converting it into a bitmap, and then transmitting it through the telephone system in the form of audio-frequency tones. The receiving fax machine interprets the tones and reconstructs the image, printing a paper copy.

Telephone transmission

By the late 1970s, many companies around the world (especially Japan), entered the fax market. Very shortly after a new wave of more compact, faster and efficient fax machines would hit the market. Xerox continued to refine the fax machine for years after their ground-breaking first machine. In later years it would be combined with copier equipment to create the hybrid machines we have today that copy, scan and

fax. Some of the lesser known capabilities of the Xerox fax technologies included their Ethernet enabled Fax Services on their 8000 workstations in the early 1980s.

The reflected light, varying in intensity according to the light and dark areas of the document, was focused on a photocell so that the current in a circuit varied with the amount of light. This current was used to control a tone generator (a modulator), the current determining the frequency of the tone produced. This audio tone was then transmitted using an acoustic coupler (a speaker, in this case) attached to the microphone of a common telephone handset. At the receiving end, a handset's speaker was attached to an acoustic coupler (a microphone), and a demodulator converted the varying tone into a variable current that controlled the mechanical movement of a pen or pencil to reproduce the image on a blank sheet of paper on an identical drum rotating at the same rate.

Computer facsimile interface

In many corporate environments, freestanding fax machines have been replaced by fax servers and other computerized systems capable of receiving and storing incoming faxes electronically, and then routing them to users on paper or via an email (which may be secured). Such systems have the advantage of reducing costs by eliminating unnecessary printouts and reducing the number of inbound analog phone lines needed by an office.

The once ubiquitous fax machine has also begun to disappear from the small office and home office environments. Remotely hosted fax-server services are widely available from VoIP and e-mail providers allowing users to send and receive faxes using their existing e-mail accounts without the need for any hardware or dedicated fax lines. Personal computers have also long been able to handle incoming and outgoing faxes using analogue modems or ISDN, eliminating the need for a stand-alone fax machine. These solutions are often ideally suited for users who only very occasionally need to use fax services. There are 17 million fax machines in the US, about one every 4.47 square miles

A major breakthrough in the development of the modern facsimile system was the result of digital technology, where the analog signal from scanners was digitized and then compressed, resulting in the ability to transmit high rates of data across standard phone lines. The first digital fax machine was the Dacom Rapidfax first sold in late 1960s, which incorporated digital data compression technology developed by Lockheed for transmission of images from satellites.

(WEEK 21)

Fiber optics

What is fiber optics?

We're used to the idea of information traveling in different ways. When we speak into a landline telephone, a wire cable carries the sounds from our voice into a socket in the wall, where another cable takes it to the local telephone exchange. Cell phones work a different way: they send and receive information using invisible radio waves, a technology called **wireless** because it uses no cables. **Fiber optics** works a third way. It sends information coded in a beam of light down a glass or plastic pipe. It was originally developed for endoscopes in the 1950s to help doctors see inside the human body without having to cut it open first. In the 1960s, engineers found a way of using the same technology to transmit telephone calls at the speed of light (normally that's 186,000 miles or 300,000 km per second in a vacuum, but slows to about two thirds this speed in a fiber-optic cable).

Optical technology

Fig. 6

A fiber-optic cable is made up of incredibly thin strands of glass or plastic known as optical fibers; one cable can have as few as two strands or as many as several hundred. Each strand is less than a tenth as thick as a human hair and can carry something like 25,000 telephone calls, so an entire fiber-optic cable can easily carry several million calls.

Fiber-optic cables carry information between two places using entirely optical (light-based) technology. Suppose you wanted to send information from your computer to a friend's house down the street using fiber optics. You could hook your computer up

to a laser, which would convert electrical information from the computer into a series of light pulses. Then you'd fire the laser down the fiber-optic cable. After traveling down the cable, the light beams would emerge at the other end. Your friend would need a photoelectric cell (light-detecting component) to turn the pulses of light back into electrical information his or her computer could understand. So the whole apparatus would be like a really neat, hi-tech version of the kind of telephone you can make out of two baked-bean cans and a length of string!

How fiber-optics works

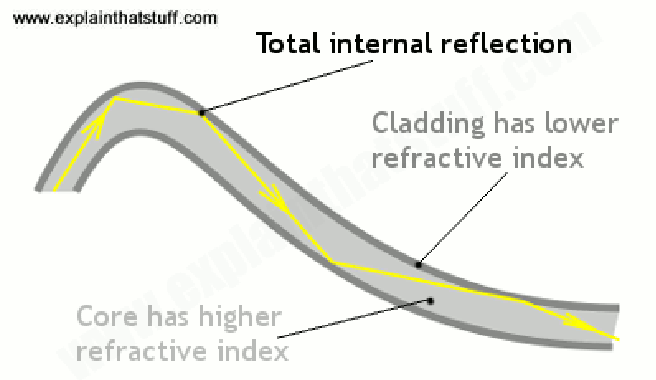


Fig. 7

Light travels down a fiber-optic cable by bouncing repeatedly off the walls. Each tiny **photon** (particle of light) bounces down the pipe like a bobsleigh going down an ice run. Now you might expect a beam of light, traveling in a clear glass pipe, simply to leak out of the edges. But if light hits glass at a really shallow angle (less than 42 degrees), it reflects back in again, as though the glass were really a mirror. This phenomenon is called total internal reflection. It's one of the things that keeps light inside the pipe.

The other thing that keeps light in the pipe is the structure of the cable, which is made up of two separate parts. The main part of the cable—in the middle—is called the **core** and that's the bit the light travels through. Wrapped around the outside of the core is another layer of glass called the **cladding**. The cladding's job is to keep the light signals inside the core. It can do this because it is made of a different type of glass to the core. (More technically, the cladding has a lower refractive index.)

Types of fiber-optic cables:

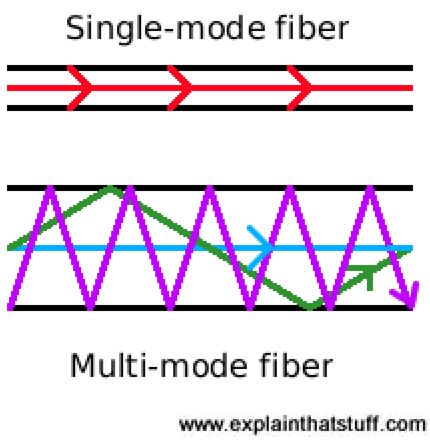


Fig. 8

Optical fibers carry light signals down them in what are called **modes**. That sounds technical but it just means different ways of traveling: a mode is simply the path that a light beam follows down the fiber. One mode is to go straight down the middle of the fiber. Another is to bounce down the fiber at a shallow angle. Other modes involve bouncing down the fiber at other angles, more or less steep.

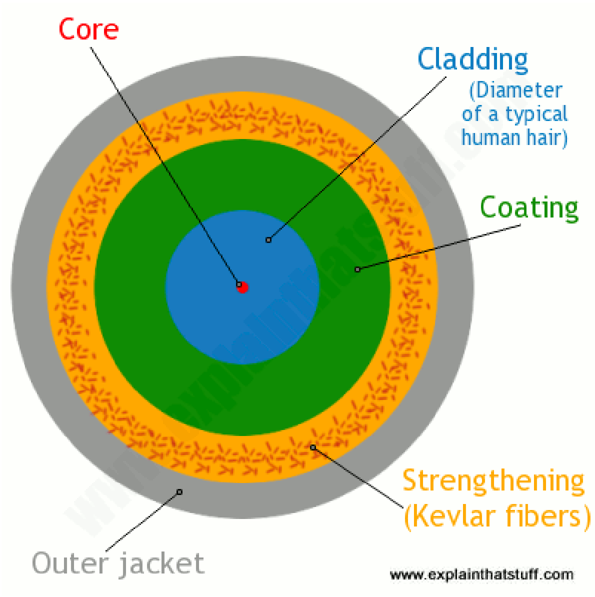


Fig. 9

The simplest type of optical fiber is called **single-mode**. It has a very thin core about 5-10 microns (millionths of a meter) in diameter. In a single-mode fiber, all signals travel straight down the middle without bouncing off the edges (red line in diagram). Cable TV, Internet, and telephone signals are generally carried by single-mode fibers, wrapped together into a huge bundle. Cables like this can send information over 100 km (60 miles).

Another type of fiber-optic cable is called **multi-mode**. Each optical fiber in a multi-mode cable is about 10 times bigger than one in a single-mode cable. This means light beams can travel through the core by following a variety of different paths (purple, green, and blue lines)—in other words, in multiple different modes. Multi-mode cables can send information only over relatively short distances and are used (among other things) to link computer together.

Even thicker fibers are used in a medical tool called a **gastroscope** (a type of endoscope), which doctors poke down someone's throat for detecting illnesses inside their stomach. A gastroscope is a thick fiber-optic cable consisting of many optical fibers. At the top end of a gastroscope, there is an eyepiece and a lamp. The lamp shines its light down one part of the cable into the patient's stomach. When the light reaches the stomach, it reflects off the stomach walls into a lens at the bottom of the cable. Then it travels back up another part of the cable into the doctor's eyepiece. Other types of endoscopes work the same way and can be used to inspect different parts of the body. There is also an industrial version of the tool, called a fiberscope, which can be used to examine things like inaccessible pieces of machinery in airplane engines.

(WEEKS 22-24)

Antenna

Basics, antenna types, antenna functions

Antennas are the means for coupling the transmitter to the medium, in this case, free space. An antenna is an electromagnetic radiator; it creates an electromagnetic field that proceeds out from the transmitting antenna to the receiver's antenna, which then converts the electromagnetic wave into electrical signals that are applied to the receiver's input stages.

There are several different types of antennas in three broad categories: Omni-directional, directional, and semi-directional.

- Omni-directional antennas propagate in all directions.
- Semi-directional antennas propagate in a constricted fashion, defined by a specific angle.

- Directional antennas have a narrow “beam” that allows highly directional propagation; familiar types are the parabolic and Yagi. Each has unique characteristics and applications.

Propagation patterns are shown on a polar chart, the angle of propagation being limited to where the power level drops by 3 dB. Figure 1 shows a polar plot for different antennas; the half-power beamwidth for a Yagi antenna is shown.

Passive gain amplifies the signal

All antennas exhibit passive gain, which serves to amplify the signal. Passive gain is measured by the quantity dBi, which is the gain referenced to a theoretical isotropic antenna; an isotropic antenna transmits energy equally in all directions, and does not exist in nature. The gain of an ideal half-wave dipole antenna is 2.15 dBi.

EIRP, or equivalent (or effective) isotropic radiated power, is the measure of the maximum power a theoretical isotropic antenna would emit in the direction of maximum antenna gain. EIRP accounts for losses from transmission lines and connectors, and includes actual antenna gain. EIRP allows calculation of real power output and field strength values, if actual antenna gains and transmitter output power are known.

Fig. 10

Dipole antennas, rubber ducky

Dipole antennas are the most common type of antenna used and are omni-directional, propagating radio frequency (RF) energy 360 degrees in the horizontal plane. These devices are constructed to be resonant at a half or quarter wavelength of the frequency being applied. This antenna can be as simple as two pieces of wire cut to the proper length or can be encapsulated as shown in the illustration; this configuration is commonly referred to as a “rubber ducky” antenna. The dipole is used in many enterprise and small office and home office (SOHO) Wi-Fi deployments.

Directional antenna

Directional and semi-directional antennas focus radiated power into narrow beams, adding a significant amount of gain in the process. Antenna properties are also reciprocal. This type of antenna is frequently used for long distance links.

Fig. 11

Patch antenna, microstrip antenna

A patch antenna is a semi-directional radiator using a flat metal strip mounted above a ground plane. Radia1tion from the back of the antenna is effectively cut off by the ground plane, enhancing forward directionality. This type of antenna is also known as a microstrip antenna. It is typically rectangular and enclosed in a plastic enclosure. This type of antenna lends itself to being manufactured by standard printed circuit board methods. Patch antennas are widely used semi-directionals; a patch antenna can have a beamwidth of between 30 to 180 degrees and a typical gain of 9 dB.

Sector antenna

Sector antennas are another type of semi-directional antenna. Sector antennas provide a pie-shaped (sector) radiation pattern and are usually installed in what is known as a sectorized array. Beamwidth for a sector antenna can be between 60 to 180 degrees, with 120 degrees being typical. In a sectorized array, antennas are mounted back-to-back to provide full 360-degree coverage. Sector antennas are used extensively for cellular communication.

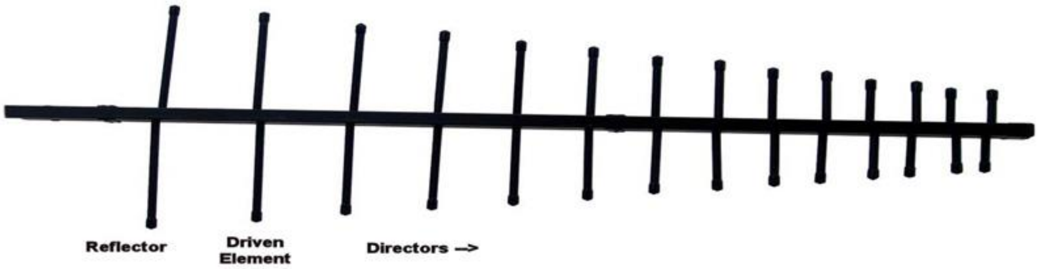


Fig. 12

Yagi antenna

A commonly used directional antenna is the Yagi-Uda Array, usually just called a Yagi. It was invented by Shintaro Uda and his colleague, Hidetsugu Yagi, in 1926. A Yagi antenna uses several elements to form a directional array. A single driven element, typically a dipole, propagates RF energy; elements placed immediately in front of and behind the driven element re-radiate RF energy in phase and out of phase, enhancing and retarding the signal, respectively. The elements are called parasitic elements; the element behind the driven element is called the reflector, while the elements in front of the driven element are called directors. Yagi antennas have beamwidths in the range of 30 to 80 degrees and can provide well in excess of 10 dBi passive gain. A multi-element high-gain Yagi is shown in Figure 4.

Fig. 13

Parabolic or dish antenna

Parabolic, or dish, antennas are the most familiar type of directional antenna. A parabola is a symmetric curve; a parabolic reflector is a surface that describes that curve throughout a 360-degree rotation, a dish or, to use the technical term, a paraboloid. A parabolic reflector has a high degree of directivity and has the ability to focus RF energy into a beam, much like a flashlight. Parabolic antennas have a very narrow beamwidth, usually not exceeding 25 degrees. Gain is dependent on diameter

and frequency; at 2.4 GHz, a 1 meter dish will provide about 26 dBi gain, while a 10 meter antenna will provide 46 dBi gain at the same frequency. The antenna is “fed” by either a half wave dipole antenna or a feed horn. Parabolic antennas are used for long distance communication links between buildings or over large geographic areas. Very large parabolic antennas are used for radio astronomy and can provide gain of 10 million or about 70 dBi.

Fig. 14

Grid antenna

A variation of the dish is the grid antenna. Given that a parabolic reflector will present a large solid surface to the wind, it follows that high or even moderate wind conditions will cause the dish to move out of alignment or deform. To prevent this from happening, the reflector is perforated into a grid. The spacing of the grid elements is frequency dependent; it is inversely proportional to the frequency. Gain and beam-width are similar to the parabolic antenna.

(WEEK 25)

Microwave Communication

What is Microwave Communication?

Microwave communication is method of wirelessly sending data. It is very similar to radio technology. Microwaves are right next to radio waves on the electromagnetic spectrum.

What is microwave used for?

Modern microwave systems are used in telephone networks (both wireless and wireline) and ISPs. They're used by power utilities to remotely manage the power grid. They're used by public safety agencies (ex. police, fire) for remote monitoring and management. Many industries used microwave, and they do it for one important reason:

Microwave is a powerful tool wherever data must be transmitted a long distance without physical wires. This is common in rural mountainous regions, where installing physical transmission lines is difficult and expensive. Microwave systems minimize installations and maintenance, as a signal microwave tower can transmit data across dozens of miles.

Fig. 15

Why are microwave transmitters mounted on towers?

Just like visible light, microwaves are blocked by obstacles. They need a clear path to reach their destination. Mounting transmitters and receivers high on a tower offers a clear line-of-sight to the next tower. Also, taller towers reduce the impact of the Earth's curvature. Taller towers can be spaced farther apart and still see one another.

How is digital microwave different from analog?

Microwave communication uses analog and digital formats. While digital is the most advanced form, both are useful.

Microwave communication has been around for a long time. Just like most other communication technologies, microwaves had a digital revolution. What that means for you today is that only older microwave transmitters are analog. Special workarounds are required for them to carry modern digital data. As time continues to progress, analog microwaves will grow increasingly rare.

Microwave communication is the sending of signals via radio using a series of microwave towers. It's a form of "line of sight" communication. There must be nothing obstructing transmission of data between these towers. That's why microwave towers are frequently placed on mountaintops. When positioned on a tall peak, a tower has lines of sight to valleys below on all sides and to other mountaintop towers. The increase elevation also reduces the impact of the Earth's curvature on line of sight.

You must monitor your microwave systems

Whether you use digital or analog, you have to monitor your microwave communication equipment. You need to know that your data transmission equipment is online. Don't let your customers be the first to tell you about a network outage.

You also have other concerns not directly related to microwave. Consider the risk of copper theft. Most microwave sites are very remote. Sure, a typical equipment failure is a problem, but fixing it quickly minimizes the revenue impact. A copper theft event is a much bigger problem. First, it probably takes a long time to fix. Second, replacing (and paying labor costs to reinstall) copper is expensive. This is equally true for any other kind of theft or vandalism. You lose revenue AND skyrocket costs at the same time.

Also think about the required aircraft obstruction lighting for your towers. To avoid FCC/FAA fines for non-compliance, you have to keep your lights online. Any lighting failure must be reported promptly to minimize your risk of fines and liability.

It's problems like these that necessitate a remote monitoring system to instantly notify you of microwave transmission failures.

(WEEK 26)

A communications satellite

A communications satellite is an artificial satellite that relays and amplifies radio telecommunications signals via a transponder; it creates a communication channel between a source transmitter and a receiver at different locations on Earth. Communications satellites are used for television, telephone, radio, internet, and military applications. There are over 2,000 communications satellites in Earth's orbit, used by both private and government organizations.

Wireless communication uses electromagnetic waves to carry signals. These waves require line-of-sight, and are thus obstructed by the curvature of the Earth. The purpose of communications satellites is to relay the signal around the curve of the Earth allowing communication between widely separated points. Communications satellites use a wide range of radio and microwave frequencies. To avoid signal interference, international organizations have regulations for which frequency ranges or "bands" certain organizations are allowed to use. This allocation of bands minimizes the risk of signal interference.

Satellite orbits

Communications satellites usually have one of three primary types of orbit, while other orbital classifications are used to further specify orbital details:-

1. Geostationary satellites have a geostationary orbit (GEO), which is 35,786 kilometers (22,236 mi) from Earth's surface. This orbit has the special characteristic that the apparent position of the satellite in the sky when viewed by a ground observer does not change, the satellite appears to "stand still" in the sky.
2. *Medium Earth orbit* (MEO) satellites are closer to Earth. Orbital altitudes range from 2,000 to 35,786 kilometers (1,243 to 22,236 mi) above Earth.
3. The region below medium orbits is referred to as low Earth orbit (LEO), and is about 160 to 2,000 kilometers (99 to 1,243 mi) above Earth.

As satellites in MEO and LEO orbit the Earth faster, they do not remain visible in the sky to a fixed point on Earth continually like a geostationary satellite, but appear to a ground observer to cross the sky and "set" when they go behind the Earth.

Structure

Communications Satellites are usually composed of the following subsystems:

1. Communication Payload, normally composed of transponders, antennas, and switching systems.
2. Engines used to bring the satellite to its desired orbit
3. Station Keeping Tracking and stabilization subsystem.
4. Power subsystem, solar cells, and batteries
5. Command and Control subsystem.

The bandwidth available from a satellite depends upon the number of transponders provided by the satellite. Each service (TV, Voice, Internet, radio) requires a different amount of bandwidth for transmission. This is typically known as link budgeting and a network simulator can be used to arrive at the exact value.

Frequency Allocation for satellite systems

Allocating frequencies to satellite services is a complicated process which requires international coordination and planning. This is carried out under the auspices of the International Telecommunication Union (ITU). To facilitate frequency planning, the world is divided into three regions:

1. Region 1: Europe, Africa, what was formerly the Soviet Union, and Mongolia
2. Region 2: North and South America and Greenland
3. Region 3: Asia, Australia, and the southwest Pacific

Some of the services provided by satellites are:

1. Fixed satellite service (FSS)
2. Broadcasting satellite service (BSS)
3. Mobile satellite service
4. Radionavigation-satellite service
5. Meteorological-satellite service
6. Amateur-satellite service

(WEEK 27)
Microwave and Radio Waves

Microwaves are electromagnetic waves with frequencies between 300MHz (0.3GHz) and 300GHz in the electromagnetic spectrum.
Radio waves are electromagnetic waves within the frequencies 30KHz - 300GHz, and include microwaves. Microwaves are at the higher frequency end of the radio wave band and low frequency radio waves are at the lower frequency end.
Mobile phones, phone mast antennas (base stations), DECT cordless phones, Wi-Fi, WLAN, WiMAX and Bluetooth have carrier wave frequencies within the microwave band of the electromagnetic spectrum, and are pulsed/modulated. Most Wi-Fi computers in schools use 2.45GHz (carrier wave), the same frequency as microwave ovens.
It is worth noting that the electromagnetic spectrum is divided into different bands based on frequency. But the biological effects of electromagnetic radiation do not necessarily fit into these artificial divisions.

Microwave frequency bands

The microwave spectrum is usually defined as electromagnetic energy ranging from approximately 1 GHz to 100 GHz in frequency, but older use includes lower frequencies. Most common applications are within the 1 to 40 GHz range. One set of microwave frequency bands designations by the **Radio Society of Great Britain** (**RSGB**), is tabulated below:

Microwave frequency bands

Designation	Frequency range	Wavelength range	Typical uses
L band	1 to 2 GHz	15 cm to 30 cm	military telemetry, GPS, mobile phones (GSM), amateur radio
S band	2 to 4 GHz	7.5 cm to 15 cm	weather radar, surface ship radar, and some communications satellites (microwave ovens, microwave devices/communications, radio astronomy, mobile

			phones, wireless LAN, Bluetooth, ZigBee, GPS, amateur radio)
C band	4 to 8 GHz	3.75 cm to 7.5 cm	long-distance radio telecommunications
X band	8 to 12 GHz	25 mm to 37.5 mm	satellite communications, radar, terrestrial broadband, space communications, amateur radio
K_u band	12 to 18 GHz	16.7 mm to 25 mm	satellite communications
K band	18 to 26.5 GHz	11.3 mm to 16.7 mm	radar, satellite communications, astronomical observations, automotive radar
K_a band	26.5 to 40 GHz	5.0 mm to 11.3 mm	satellite communications
Q band	33 to 50 GHz	6.0 mm to 9.0 mm	satellite communications, terrestrial microwave communications, radio astronomy, automotive radar
U band	40 to 60 GHz	5.0 mm to 7.5 mm	
V band	50 to 75 GHz	4.0 mm to 6.0 mm	millimeter wave radar research and other kinds of scientific research

W band	75 to 110 GHz	2.7 mm to 4.0 mm	satellite communications, millimeter-wave radar research, military radar targeting and tracking applications, and some non-military applications, automotive radar
F band	90 to 140 GHz	2.1 mm to 3.3 mm	SHF transmissions: Radio astronomy, microwave devices/communications, wireless LAN, most modern radars, communications satellites, satellite television broadcasting, DBS, amateur radio
D band	110 to 170 GHz	1.8 mm to 2.7 mm	EHF transmissions: Radio astronomy, high-frequency microwave radio relay, microwave remote sensing, amateur radio, directed-energy weapon, millimeter wave scanner

P band is sometimes used for K_u Band. "P" for "previous" was a radar band used in the UK ranging from 250 to 500 MHz and now obsolete per IEEE Std 521.

When radars were first developed at K band during World War II, it was not known that there was a nearby absorption band (due to water vapor and oxygen in the atmosphere). To avoid this problem, the original K band was split into a lower band, $\mathbf{K_u}$, and upper band, $\mathbf{K_a}$

(WEEK 28)

Mobile phone

A **mobile phone** is a portable telephone that can make and receive calls over a radio frequency carrier while the user is moving within a telephone service area. The radio frequency link establishes a connection to the switching systems of a mobile phone operator, which provides access to the public switched telephone network (PSTN). Most modern mobile telephone services use cellular network architecture, and therefore mobile telephones are often also called *cellular telephones* or *cell phones*. In addition to telephony, 2000s-era mobile phones support a variety of other services, such as text messaging, MMS, email, Internet access, short-range wireless communications (infrared, Bluetooth), business applications, gaming, and digital photography. Mobile phones which offer these and more general computing capabilities are referred to as smart phones.

All mobile phones have a variety of features in common, but manufacturers seek product differentiation by adding functions to attract consumers. This competition has led to great innovation in mobile phone development over the past 20 years.

The common components found on all phones are:

1. A **battery**, providing the power source for the phone functions.
2. An **input mechanism** to allow the user to interact with the phone. The most common input mechanism is a keypad, but touch screens are also found in most smart phones.
3. A **screen** which echoes the user's typing, displays text messages, contacts and more.
4. **Basic mobile phone services** to allow users to make calls and send text messages.
5. All GSM phones use a **SIM card** to allow an account to be swapped among devices. Some CDMA devices also have a similar card called a R-UIM.
6. Individual **GSM, WCDMA, IDEN** and some satellite phone devices are uniquely identified by an International Mobile Equipment Identity (IMEI) number.

Low-end mobile phones are often referred to as feature phones, and offer basic telephony. Handsets with more advanced computing ability through the use of native software applications became known as smart phones.

Using of Mobiles

Mobile phones are used for a variety of purposes, such as keeping in touch with family members, for conducting business, and in order to have access to a telephone in the event of an emergency. Some people carry more than one mobile phone for different purposes, such as for business and personal use. Multiple SIM cards may be

used to take advantage of the benefits of different calling plans. For example, a particular plan might provide for cheaper local calls, long-distance calls, international calls, or roaming.

Sound quality

In sound quality, smart phones and feature phones vary little. Some audio-quality enhancing features, such as Voice over LTE and HD Voice, have appeared and are often available on newer smart phones. Sound quality can remain a problem with both, as this depends not so much on the phone itself.

Text messaging

Main article: SMS

The most commonly used data application on mobile phones is Short Message Service (SMS) text messaging.

SIM card

Fig. 16

Typical mobile phone SIM card.

GSM feature phones require a small microchip called a **Subscriber Identity Module** or SIM card, in order to function. The SIM card is approximately the size of a small postage stamp. The SIM securely stores the service-subscriber key (IMSI) and the K_i used to identify and authenticate the user of the mobile phone. The SIM card allows users to change phones by simply removing the SIM card from one mobile phone and inserting it into another mobile phone or broadband telephony device, provided that this is not prevented by a SIM lock.

The first SIM card was made in 1991 by Munich smart card maker Giesecke & Devrient for the Finnish wireless network operator Radiolinja.

Multi-card hybrid phones

A hybrid mobile phone can hold up to four SIM cards. SIM and R-UIM cards may be mixed together to allow both GSM and CDMA networks to be accessed.

From 2010 onwards, such phones became popular in India and Indonesia and other emerging markets, and this was attributed to the desire to obtain the lowest on-net calling rate. In Q3 2011, Nokia shipped 18 million of its low-cost dual SIM phone range in an attempt to make up for lost ground in the higher-end Smartphone market.[13]

Mobile broadband

Mobile broadband uses the spectrum of 225 MHz to 3700 MHz's

Mobile broadband is the marketing term for wireless Internet access delivered through mobile phone towers to computers, mobile phones(called "cell phones" in North America and South Africa), and other digital devices using portable modems. Although broadband has a technical meaning, wireless-carrier marketing uses the phrase "mobile broadband" as a synonym for mobile Internet access. Some mobile services allow more than one device to be connected to the Internet using a single cellular connection using a process called tethering.

The bit rates available with Mobile broadband devices support voice and video as well as other data access. Devices that provide mobile broadband to mobile computers include:

- PC cards, also known as *PC data cards*, and Express cards
- USB and mobile broadband modems, also known as *connect cards*
- portable devices with built-in support for mobile broadband, such as laptop computers, netbook computers, smart phones, tablets, PDAs, and other mobile Internet devices.

Internet access subscriptions are usually sold separately from mobile phone subscriptions.

Cellular Frequencies

Cellular frequencies are the sets of frequency ranges within the ultra high frequency band that have been assigned for cellular phone use. Most cellular phone networks worldwide use portions of the radio frequency spectrum, allocated to the Mobile service, for the transmission and reception of their signals. The particular bands may

also be shared with other radio communication services, e.g. broadcasting service, and Fixed service operation.

Radio frequencies used for cellular networks differ in ITU Regions (Americas, Europe, Africa and Asia). The first commercial standard for mobile connection in the United States was AMPS, which was in the 800 MHz frequency band. In Nordic countries of Europe, the first widespread automatic mobile network was based on the NMT-450 standard, which was in the 450 MHz band. As mobile phones became more popular and affordable, mobile providers encountered a problem because they couldn't provide service to the increasing number of customers. They had to develop their existing networks and eventually introduce new standards, often based on other frequencies. Some European countries (and Japan) adopted TACS operating in 900 MHz's. The GSM standard, which appeared in Europe to replace NMT-450 and other standards, initially used the 900 MHz band too. As demand grew, carriers acquired licenses in the 1,800 MHz band. (Generally speaking, lower frequencies allow carriers to provide coverage over a larger area, while higher frequencies allow carriers to provide service to more customers in a smaller area.)

In the U.S., the analog AMPS standard that used the cellular band (800 MHz) was replaced by a number of digital systems. Initially, systems based upon the AMPS mobile phone model were popular, including IS-95 (often known as "CDMA", the air interface technology it uses) and IS-136 (often known as D-AMPS, Digital AMPS, or "TDMA", the air interface technology it uses). Eventually, IS-136 on these frequencies was replaced by most operators with GSM. GSM had already been running for some time on US PCS (1,900 MHz) frequencies.

And, some NMT-450 analog networks have been replaced with digital networks using the same frequency. In Russia and some other countries, local carriers received licenses for 450 MHz frequency to provide CDMA mobile coverage area.

Many GSM phones support three bands (900/1,800/1,900 MHz or 850/1,800/1,900 MHz) or four bands (850/900/1,800/1,900 MHz), and are usually referred to as tri-band and quad-band phones, or world phones; with such a phone one can travel internationally and use the same handset. This portability is not as extensive with IS-95 phones; however, as IS-95 networks do not exist in most of Europe.

Mobile networks based on different standards may use the same frequency range; for example, AMPS, D-AMPS, N-AMPS and IS-95 all use the 800 MHz frequency band. Moreover, one can find both AMPS and IS-95 networks in use on the same frequency in the same area that do not interfere with each other. This is achieved by the use of different channels to carry data. The actual frequency used by a particular phone can vary from place to place, depending on the settings of the carrier's base station.

(WEEK 29)

GSM

What is GSM?

If you are in Europe or Asia and using a mobile phone, then most probably you are using GSM technology in your mobile phone.

- GSM stands for **G**lobal **S**ystem for **M**obile Communication. It is a digital cellular technology used for transmitting mobile voice and data services.
- The concept of GSM emerged from a cell-based mobile radio system at Bell Laboratories in the early 1970s.
- GSM is the name of a standardization group established in 1982 to create a common European mobile telephone standard.
- GSM is the most widely accepted standard in telecommunications and it is implemented globally.
- GSM is a circuit-switched system that divides each 200 kHz channel into eight 25 kHz time-slots. GSM operates on the mobile communication bands 900 MHz and 1800 MHz in most parts of the world. In the US, GSM operates in the bands 850 MHz and 1900 MHz.
- GSM owns a market share of more than 70 percent of the world's digital cellular subscribers.
- GSM makes use of narrowband Time Division Multiple Access (TDMA) technique for transmitting signals.
- GSM was developed using digital technology. It has an ability to carry 64 kbps to 120 Mbps of data rates.
- Presently GSM supports more than one billion mobile subscribers in more than 210 countries throughout the world.
- GSM provides basic to advanced voice and data services including roaming service. Roaming is the ability to use your GSM phone number in another GSM network.
- GSM digitizes and compresses data, then sends it down through a channel with two other streams of user data, each in its own timeslot.

Why GSM?

Listed below are the features of GSM that account for its popularity and wide acceptance.

- Improved spectrum efficiency
- International roaming
- Low-cost mobile sets and base stations (BSs)
- High-quality speech
- Compatibility with Integrated Services Digital Network (ISDN) and other telephone company services

- Support for new services

A GSM network comprises of many functional units. These functions and interfaces are explained in this chapter. The GSM network can be broadly divided into:

- The Mobile Station (MS)
- The Base Station Subsystem (BSS)
- The Network Switching Subsystem (NSS)
- The Operation Support Subsystem (OSS)

Given below fig.17 is a simple pictorial view of the GSM architecture.

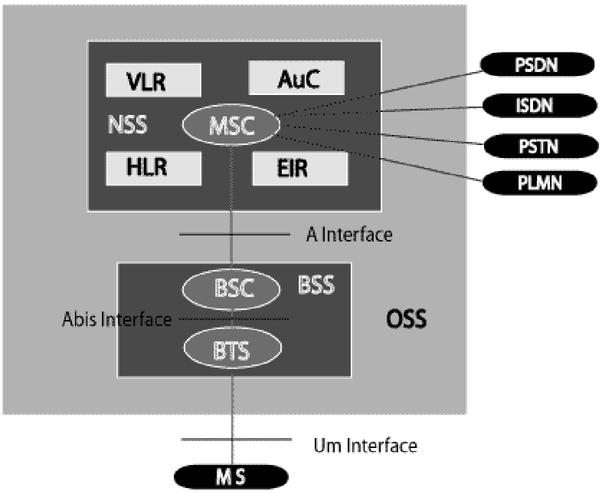


Fig. 17

The additional components of the GSM architecture comprise of databases and messaging systems functions:

- Home Location Register (HLR)
- Visitor Location Register (VLR)
- Equipment Identity Register (EIR)
- Authentication Center (AuC)
- SMS Serving Center (SMS SC)
- Gateway MSC (GMSC)
- Chargeback Center (CBC)
- Transcoder and Adaptation Unit (TRAU)

The following diagram fig. 18 shows the GSM network along with the added elements:

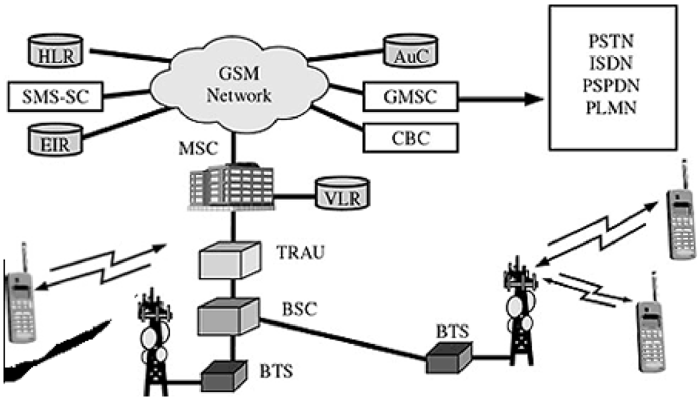


Fig. 18

The MS and the BSS communicate across the Um interface. It is also known as the *air interface* or the *radio link*. The BSS communicates with the Network Service Switching (NSS) center across the *A* interface.

GSM network areas

In a GSM network, the following areas are defined:

- **Cell** : Cell is the basic service area; one BTS covers one cell. Each cell is given a Cell Global Identity (CGI), a number that uniquely identifies the cell.
- **Location Area** : A group of cells form a Location Area (LA). This is the area that is paged when a subscriber gets an incoming call. Each LA is assigned a Location Area Identity (LAI). Each LA is served by one or more BSCs.
- **MSC/VLR Service Area** : The area covered by one MSC is called the MSC/VLR service area.
- **PLMN** : The area covered by one network operator is called the Public Land Mobile Network (PLMN). A PLMN can contain one or more MSCs.

(WEEK 30)

Thuraya

Thuraya, from the Arabic name for the constellation of the Pleiades, "Thuraya",[1] is a regional mobile satellite phone provider. The company is based in the United Arab Emirates; it provides mobile coverage to more than 162 countries in Europe, the Middle East, North, Central and East Africa, Asia and Australia. Amphenol Antenna Solutions offers antennas for virtually all applications, with product ranging from 25 MHz to 3.8 GHz.

Services

1. Voice communications with satellite phones or fixed terminals
2. Short message service
3. 60 kbit/s downlink and 15 kbit/s uplink "GMPRS" mobile data service on Thuraya satellite phones
4. 144 kbit/s high-speed data transfer via a notebook-sized terminal (ThurayaDSL)
5. A number of other services, such as call waiting, missed calls, voicemail, etc.
6. A one-way 'high power alert' capability that notifies users of an incoming call, when the signal path to the satellite is obstructed (e.g., inside a building)

Virtual country code

Thuraya's country calling code is +882 16, which is part of the ITU-T International Networks numbering group. Thuraya is not part of the +881 country calling code numbering group as this is allocated by ITU-T for networks in the Global Mobile Satellite System, of which Thuraya is not a part, being a regional rather than a global system.

Air interface

Transceivers communicate directly with the satellites using an antenna of roughly the same length as the handset and have a maximum output power of 2 Watts. QPSK modulation is used for the air interface. Thuraya SIM cards will work in regular GSM telephones and ordinary GSM SIM cards can be used on the satellite network as long as the GSM provider has a roaming agreement with Thuraya. As with all geosynchronous voice services a noticeable lag is present while making a call.

Due to the relatively high gain of the antennas contained within handsets, it is necessary to roughly aim the antenna at the satellite. As the handsets contain a GPS receiver it is possible to program the ground position of the satellites as waypoints to assist with aiming.

Thuraya operates two communications satellites built by Boeing.

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