College of engineering & technology
Electrical Engineering Department

Graduation project

Frequency Division Multiplexing

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Hebron – Palestine
2014/2015

جامعة بوليتكنك فلسطين

الخليل – فلسطين

كليه الهندسة و التكنولوجيا

دائرة الهندسة الكهربائية

اسم المشروع

Frequency division multiplexing

اسم الطالب
خالد محمود أبوشرخ

بناء على نظام كلية الهندسة والتكنولوجيا و إشراف ومتابعة المشرف المباشر على المشروع و موافقة أعضاء اللجة الممتحنة ، تم تقديم هذا المشروع إلى دائرة الهندسة الكهربائية و ذلك للوفاء بمتطلبات درجة البكالوريوس في الهندسة تخصص الاتصالات و الالكترونيات

توقيع المشرف

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توقيع اللجنة الممتحنة

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توقيع رئيس الدائرة

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Abstract

Frequency-division multiplexing (FDM) is inherently an analog technology. FDM achieves the combining of several signals into one medium by sending signals in several distinct frequency ranges over a single medium.

One of FDM's most common applications is the old traditional radio and television broadcasting from terrestrial, mobile or satellite stations, using the natural atmosphere of Earth, or the cable television. Only one cable reaches a customer's residential area, but the service provider can send multiple television channels or signals simultaneously over that cable to all subscribers without interference. Receivers must tune to the appropriate frequency (channel) to access the desired signal.

In this project I will use the FDM method to transmit a two different voice signals with different frequencies using one channel , and separate this signals by using a filters in the receiver side .

الملخص

تقسيم التردد (FDM) هو بطبيعته تكنولوجيا تناظرية . FDM يعمل على الجمع بين العديد من الإشارات في إشارة واحده، و من ثم نقل هذه الإشارة عبر قناة واحده إلى الطرف المستقبل ، حيث يتم هنالك فصل هذه الاشاراة .
واحد من التطبيقات الأكثر شيوعا (FDM) هو البث الإذاعي والتلفزيوني التقليدي القديم من المحطات الأرضية والمتنقلة أو الأقمار الصناعية ، وذلك باستخدام الكيبل التلفزيوني. كابل واحد فقط يصل إلى منطقة سكنية ، ولكن يمكن لمزود الخدمة أن يرسل قنوات تلفزيونية متعددة في وقت واحد أو إشارات على نفس الكابل لجميع المشتركين دون حدوث أي تداخل. مشروعنا يتمثل ببناء هذا النظام و لكن ليس على العديد من الاشاراة إنما على إرسل إشارتين صوتيتين عبر قناة واحده و من ثم فصل الإشارتين على الطرف المستقبل .

Dedication

To the candles of my life …. Mom and dad
Brothers ,sisters and friends

To my dear teacher
Dr . murad abu sbih

Acknowledgment

First of all , a great thanks for Allah for giving me the strength and knowledge to accomplish my project .

I would like to convey my very special cordial thanks to my supervisor Dr. Morad Abu Sbih for his support and for guiding the work .

Great credit is attributed to the stuff of Electrical Engineering Department in Palestine Polytechnic University .

Finally , very special thanks to all our classmates in department .

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Chapter one

Introduction

1.1 overview :

in this chapter I will talk about the general description of project, motivation ,project objective ,project scheduling for my system , and report content .

1.2 General description of the project

Frequency division multiplexing (FDM) means that the total bandwidth available to the system is divided into a series of non overlapping frequency sub-bands that are then assigned to each communicating source and user pair. Figures 1.1a and 1.1b show how this division is accomplished for a case of three sources at one end of a system that are communicating with three separate users at the other end. Note that each transmitter modulates its source's information into a signal that lies in a different frequency sub-band (Transmitter 1 generates a signal in the frequency sub-band between 92.0 MHz and 92.2 MHz, Transmitter 2 generates a signal in the sub-band between 92.2 MHz and 92.4 MHz, and Transmitter 3 generates a signal in the sub-band between 92.4 MHz and 92.6 MHz). The signals are then transmitted across a common channel.

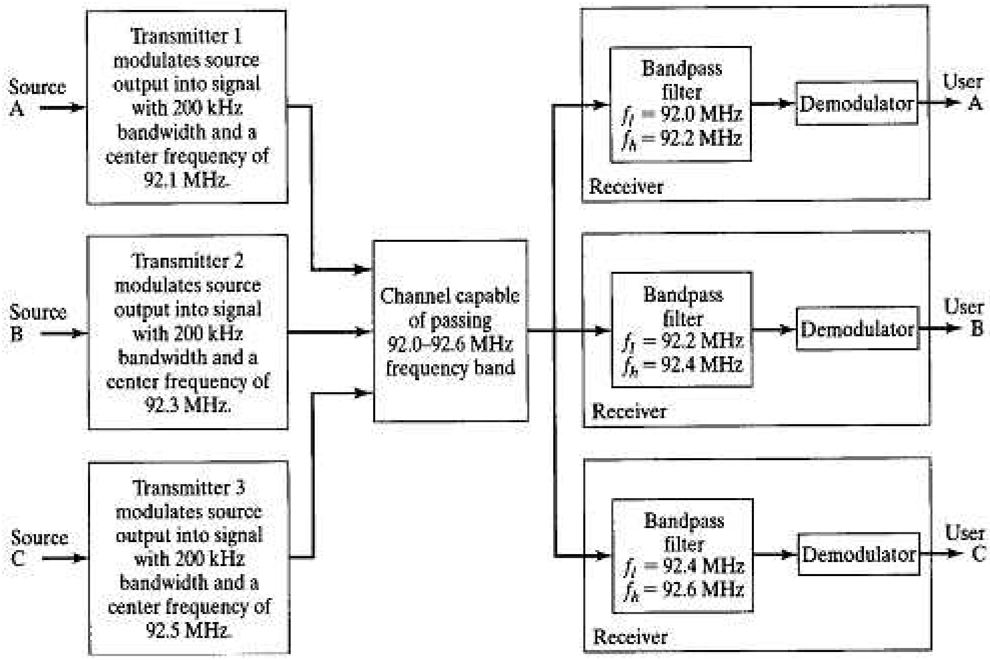

Figure 1.1a—A system using frequency division multiplexing.

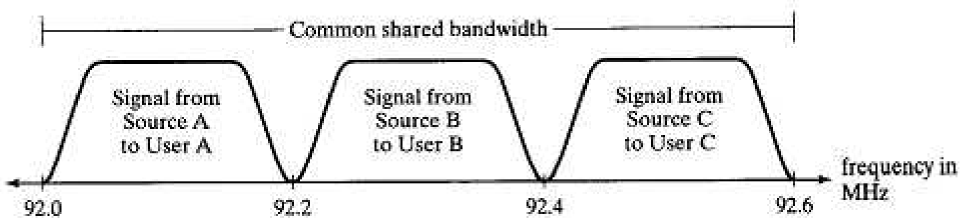

Figure 1.1b—Spectral occupancy of signals in an FDM system.

At the receiving end of the system, band pass filters are used to pass the desired signal (the signal lying in the appropriate frequency sub-band) to the appropriate user and to block all the unwanted signals. To ensure that the transmitted signals do not stray outside their assigned sub-bands, it is also common to place appropriate pass band filters at the output stage of each transmitter. It is also appropriate to design an FDM system so that the bandwidth allocated to each sub-band is slightly larger than the bandwidth needed by each source. This extra bandwidth, called a guard band, allows systems to use less expensive filters (i.e., filters with fewer poles and therefore less steep roll offs).

FDM has both advantages and disadvantages relative to TDM. The main advantage is that unlike TDM, FDM is not sensitive to propagation delays. Channel equalization techniques needed for FDM systems are therefore not as complex as those for TDM systems. Disadvantages of FDM include the need for band pass filters, which are relatively expensive and complicated to construct and design (remember that these filters are usually used in the transmitters as well as the receivers). TDM, on the other hand, uses relatively simple and less costly digital logic circuits. Another disadvantage of FDM is that in many practical communication systems, the power amplifier in the transmitter has nonlinear characteristics (linear amplifiers are more complex to build), and nonlinear amplification leads to the creation of out-of-band spectral components that may interfere with other FDM channels. Thus, it is necessary to use more complex linear amplifiers in FDM systems.

I am planning to build a circuit with a transmitter and a receiver , in which the transmitter will transmit two voice signal over one channel, and at the other side (receiver) the signals will be separated by a filters to have two different output signals .

1.3 Motivation

FDM's most common applications is the old traditional radio and television broadcasting from terrestrial, mobile or satellite stations, using the natural atmosphere of Earth, or the cable television. Only one cable reaches a customer's residential area, but the service provider can send multiple television channels or signals simultaneously over that cable to all subscribers without interference. Receivers must tune to the appropriate frequency (channel) to access the desired signal; so I planning to built such system to see how its work .

1.4 Project objective

the main objective of this project is to do multiplexing and de-multiplexing of the signals .

1.5 Time plane / project schedule

Task/weak	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
Project determination	■															
Data collection		■	■	■	■	■	■	■	■	■	■					
History			■	■	■	■										
Design and analysis							■	■	■	■	■	■	■	■	■	
Documentation			■	■	■	■	■	■	■	■	■	■	■	■	■	■
Implementation and testing			■	■	■	■	■	■	■	■	■	■	■	■	■	■

Table 1.1 time planning

CHAPTER 2

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Chapter two

Theoretical background

2.1 Introduction

It has been observed that most of the individual data-communicating devices typically require modest data rate. But, communication media usually have much higher bandwidth. As a consequence, two communicating stations do not utilize the full capacity of a data link. Moreover, when many nodes compete to access the network, some efficient techniques for utilizing the data link are very essential. When the bandwidth of a medium is greater than individual signals to be transmitted through the channel, a medium can be shared by more than one channel of signals. The process of making the most effective use of the available channel capacity is called Multiplexing. For efficiency, the channel capacity can be shared among a number of communicating stations just like a large water pipe can carry water to several separate houses at once. Most common use of multiplexing is in long-haul communication using coaxial cable, microwave and optical fiber.

Figure 2.1 depicts the functioning of multiplexing functions in general. The multiplexer is connected to the de-multiplexer by a single data link. The multiplexer combines (multiplexes) data from these 'n' input lines and transmits them through the high capacity data link, which is being de-multiplexed at the other end and is delivered to the appropriate output lines. Thus, Multiplexing can also be defined as a technique that allows simultaneous transmission of multiple signals across a single data link.

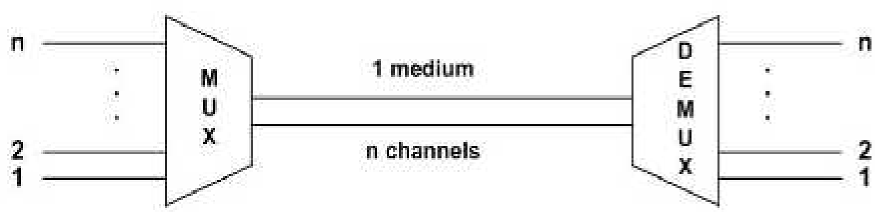


Figure 2.1 Basic concept of multiplexing

2.2 Frequency-Division Multiplexing (FDM)

In frequency division multiplexing, the available bandwidth of a single physical medium is subdivided into several independent frequency channels. Independent message signals are translated into different frequency bands using modulation techniques, which are combined by a linear summing circuit in the multiplexer, to a composite signal. The resulting signal is then transmitted along the single channel by electromagnetic means as shown in Fig.2.2 Basic approach is to divide the available bandwidth of a single physical medium into a number of smaller, independent frequency channels. Using modulation, independent message signals are translated into different frequency bands. All the modulated signals are combined in a linear summing circuit to form a composite signal for transmission. The carriers used to modulate the individual message signals are
called sub-carriers, shown as f1, f2, …, f n in Fig. 2.3 (a).

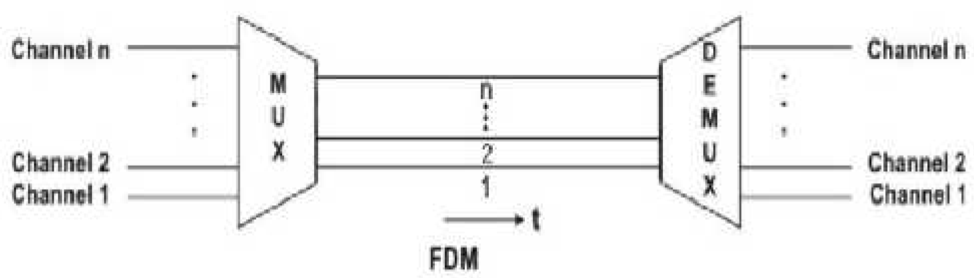
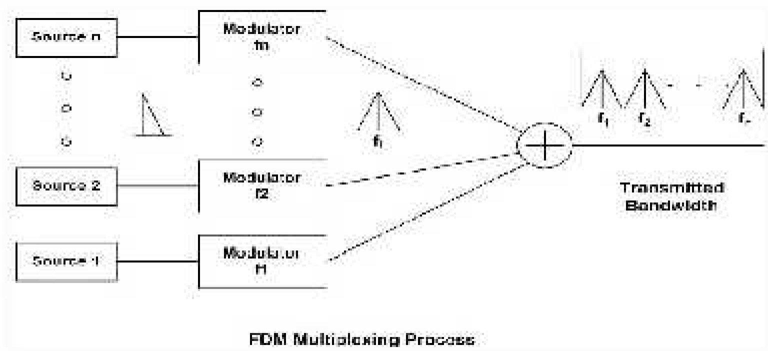


Figure 2.2 Basic concept of FDM

At the receiving end the signal is applied to a bank of band-pass filters, which separates individual frequency channels. The band pass filter outputs are then demodulated and distributed to different output channels as shown in Fig. 2.3(b).



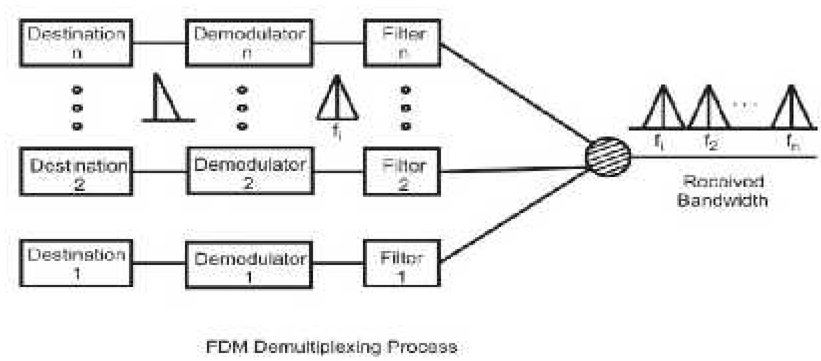


Figure 2.3 (a) FDM multiplexing process, (b) FDM de multiplexing process

If the channels are very close to one other, it leads to inter-channel cross talk. Channels must be separated by strips of unused bandwidth to prevent inter-channel cross talk.

These unused channels between each successive channel are known as guard band s as shown in Figure 2.4 .

FDM are commonly used in radio broadcasts and TV networks. Since, the frequency band used for voice transmission in a telephone network is 4000 Hz, for a particular cable of 48 KHz bandwidth, in the 70 to 108 KHz range, twelve

separate 4 KHz sub channels
could be used for transmitting twelve different messages simultaneously. Each radio and TV station, in a certain broadcast area, is allotted a specific broadcast frequency, so that independent channels can be sent simultaneously in different broadcast area. For example, the AM radio uses 540 to 1600 KHz frequency bands while the FM radio uses 88 to 108 MHz frequency bands.

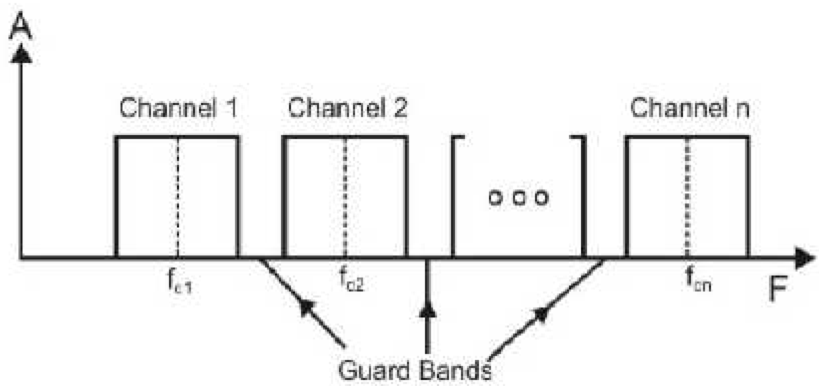


Figure 2.4 Use of guard bands in FDM

2.3The voice signal

A voice frequency (VF) or voice band is one of the frequencies, within part of the audio range, that is used for the transmission of speech.

the bandwidth of a voice signal is taken to be 4kHz, with an effective spectrum of 300 to 3400Hz as shown in figure 2.5.

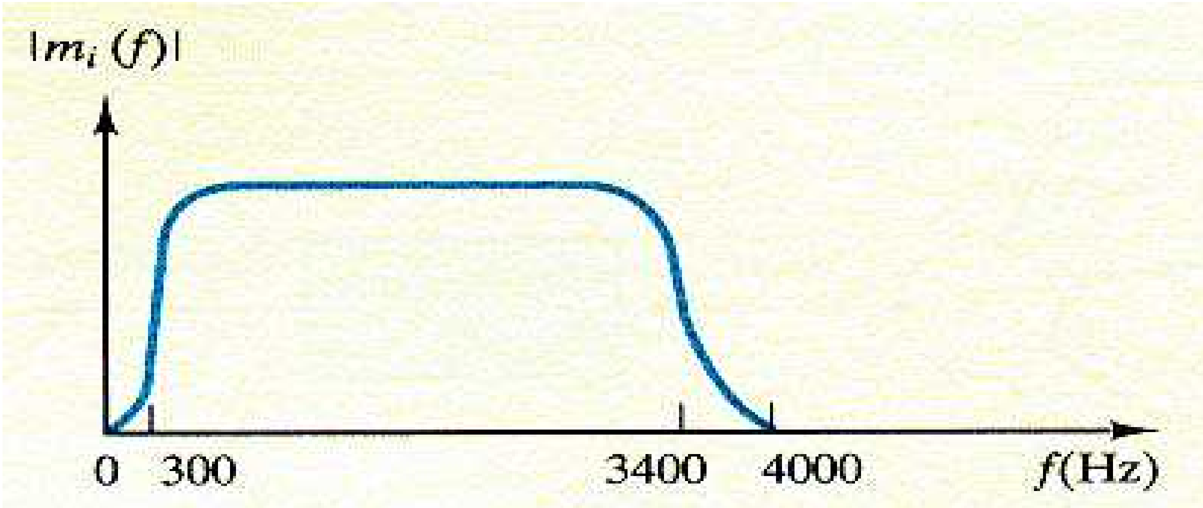


Figure 2.5 effective spectrum of the voice signal

If this signal is used to modulate at 64 kHz carrier, The modulated signal has a bandwidth of 8 kHz, extending from 60 to 68kHz.as shown in figure 2.6.

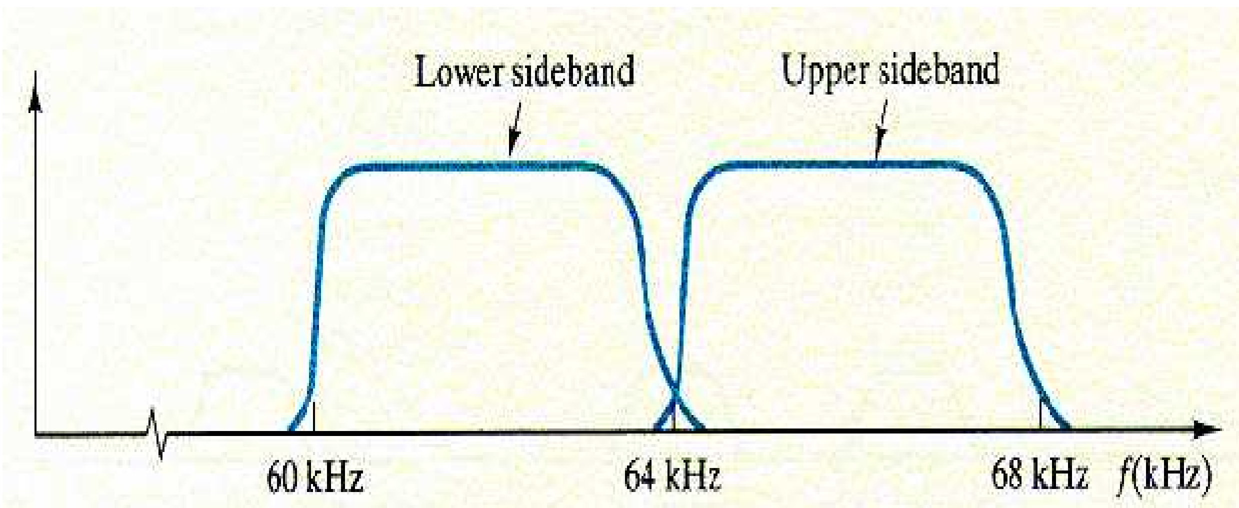


Figure 2.6 the modulated signal at 64khz carrier

To make efficient use of bandwidth, we transmit only the lower sideband. If three voice signals are used to modulate, carriers at 64, 68, and 72 kHz, the spectrum is as shown in figure 2.7.

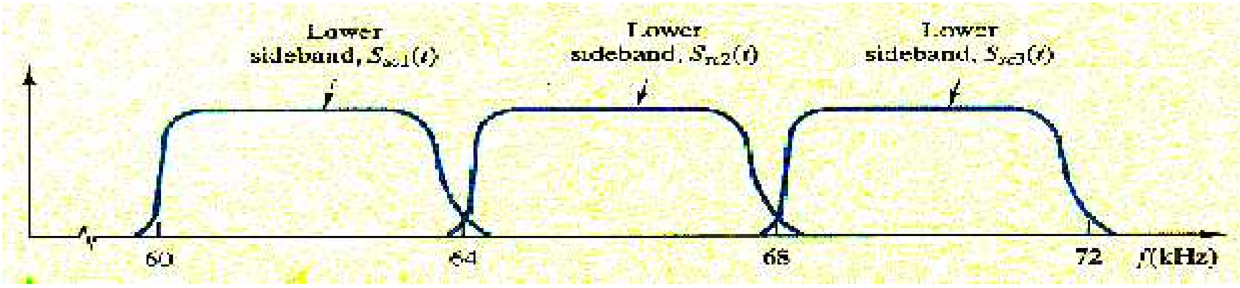


Figure 2.7 a transmission of three voice signals

2.4 Frequency division multiplexing of the voice signal

FDM is possible when the useful bandwidth of the medium excess the required bandwidth of signal to be transmission .

A number of signals are transmitted simultaneously if each signal is modulated onto a different carrier frequency, and the carrier frequencies are sufficiently separated that the bandwidths of the signals do not overlap.

2.4.1 Multiplexing

N sources are fed into a multiplexer the multiplexer modulates each signal onto a different Frequency each modulated signal requires a certain bandwidth. centered around its carrier frequency, referred to as a channel ,The modulated signals are summed to produce a composite signal figure2.8.

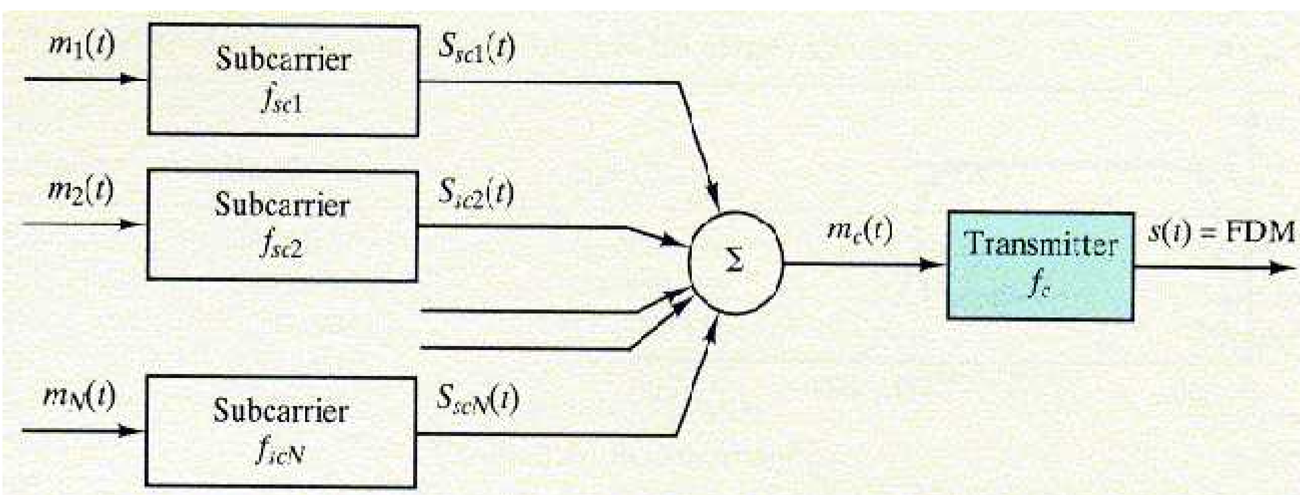


Figure 2.8 general multiplexing

The composite signal transmitted across the medium is analog, The input signals may be either digital or analog, A digital signal must be passed through a modem.
to prevent interference, the channels are separated by guard bands, which are unused portions of the spectrum figure 2.9.

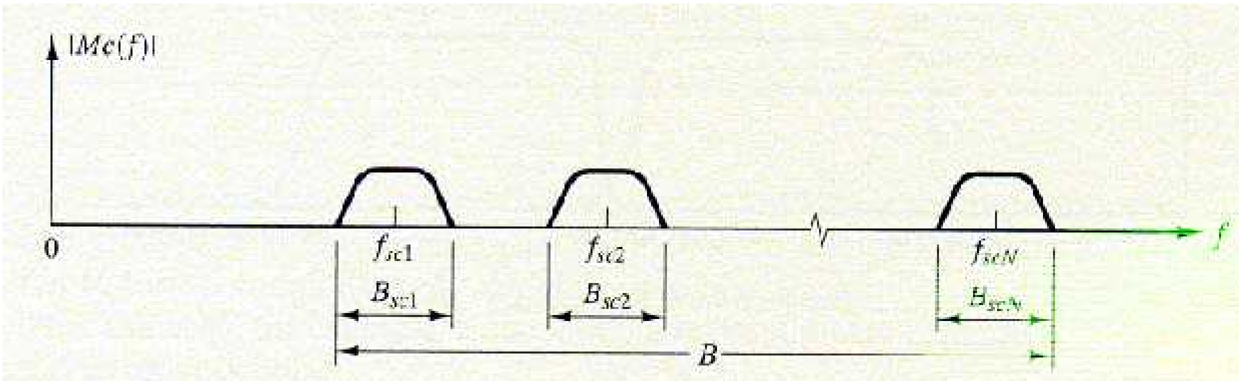


Figure 2.9 grand band

2.4.2 De-multiplexing

At the receiving end, the composite signal is passed through N band pass filters, each filter centered at fsci and having a bandwidth Bsci ,Each component is then demodulated to recover the original signal. Figure 2.10

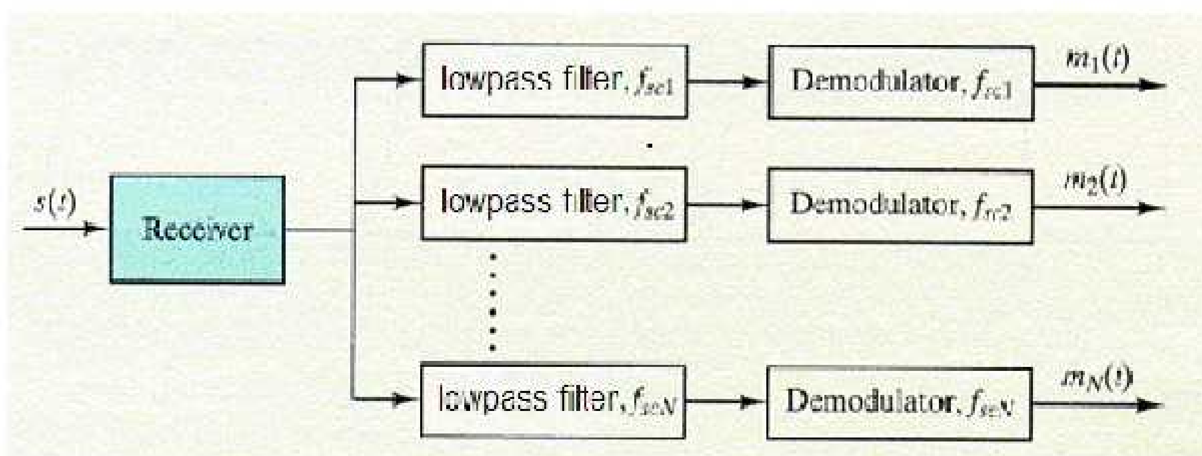


Figure 2.10 de-multiplexing

2.5 Summing Amplifier

Summing amplifier or an adder is used to sum two signal voltages. Voltage adder circuit is a simple circuit that enables you to add several signals together. It has wide variety of applications in electronic circuits. For example, on a precision amplifier, you may need to add a small voltage to cancel the offset error of the op amp itself. An audio mixer is another good example of adding waveforms (sounds) together from different channels (vocals, instruments) before sending the combined signal to a recorder.

You can change the gain or add another input without messing up with the gains of other inputs. Just remember that the inverting summing amplifier circuit inverts the input signals. That's not a big deal. If you need the opposite polarity, all you have to do is to put an inverting stage before or after the summer.

Here the input voltages V1, V2, V3 are given in to adder circuit.This is an inverting summing amplifier because output is the sum of inputs with a sign change.To construct a non inverting adder, you can cascade one 'Inverting amplifier' with unity gain along with this circuit. Output of this adder circuit is given by – (V1+V2+V3).Consider the current flowing through the input resistors are

$$I_1 = \frac{V_1}{R},$$

$$I_2 = \frac{V_2}{R},$$

$$I_3 = \frac{V_3}{R}$$

Then by Kirchof's current law, the current flowing through feedback resistor Rf is given by the sum of these 3 currents.

$$I_K = I_1 + I_2 + I_3$$

$$= \frac{V_1}{R} + \frac{V_2}{R} + \frac{V_3}{R}$$

$$= \frac{1}{R}(V_1 + V_2 + V_3)$$

This current will flows through the feedback resistor Rf, because the point ‘K’ acts as virtual ground point. So the voltage drop at Rf is given by

$$V_O = -\, I_K\, R$$

$$= -\frac{1}{R}(V_1 + V_2 + V_3)\, R$$

$$= -(V_1 + V_2 + V_3)$$

-ve sign is due to the op amp connected in inverting mode. This circuit is called a summing ‘amplifier’ (figure 2.11) because it can provide gain. By adjusting the value of Rf the gain can be changed .Then the output becomes

$$V_O = -\frac{R_F}{R}(V_1 + V_2 + V_3)$$

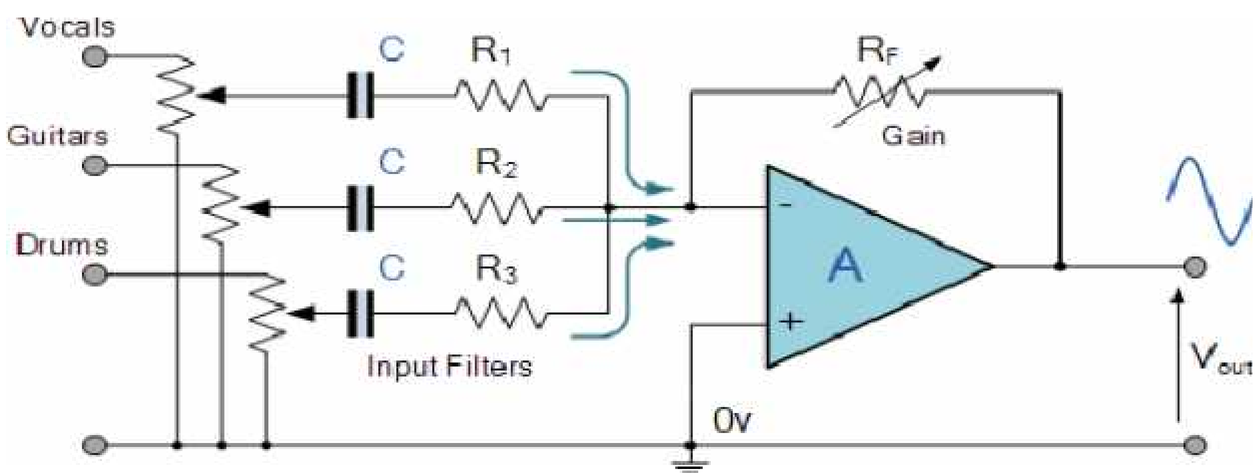


Figure 2.11 summing amplifier

2.6 Band Pass Filters

The cut-off frequency or fc point in a simple RC passive filter can be accurately controlled using just a single resistor in series with a non-polarized capacitor, and depending upon which way around they are connected, we have seen that either a Low Pass or a High Pass filter is obtained.

One simple use for these types of Passive Filters is in audio amplifier applications or circuits such as in loudspeaker crossover filters or pre-amplifier tone controls. Sometimes it is necessary to only pass a certain range of frequencies that do not begin at 0Hz, (DC) or end at some upper high frequency point but are within a certain range or band of frequencies, either narrow or wide.

By connecting or "cascading" together a single Low Pass Filter circuit with a High Pass Filter circuit, we can produce another type of passive RC filter that passes a selected range or "band" of frequencies that can be either narrow or wide while attenuating all those outside of this range. This new type of passive filter arrangement produces a frequency selective filter known commonly as a Band Pass Filter or BPF for short .

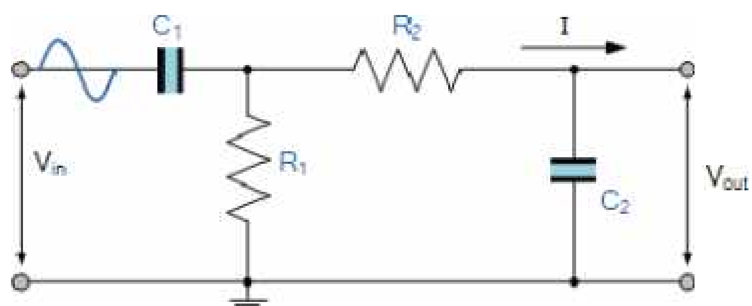


Figure 2.12 band pass filter

Unlike a low pass filter that only pass signals of a low frequency range or a high pass filter which pass signals of a higher frequency range, a Band Pass Filters passes signals within a certain "band" or "spread" of frequencies without distorting the input signal or introducing extra noise. This band of frequencies can be any width and is commonly known as the filters Bandwidth as shown in figure 2.12.

Bandwidth is commonly defined as the frequency range that exists between two specified frequency cut-off points (fc), that are 3dB below the maximum centre or resonant peak while attenuating or weakening the others outside of these two points.

Then for widely spread frequencies, we can simply define the term "bandwidth", BW as being the difference between the lower cut-off frequency (fcLOWER) and the higher cut-off frequency (fcHIGHER) points. In other words, BW = fH – fL. Clearly for a pass band filter to function correctly, the cut-off frequency of the low pass filter must be higher than the cut-off frequency for the high pass filter.

The “ideal” Band Pass Filter can also be used to isolate or filter out certain frequencies that lie within a particular band of frequencies, for example, noise cancellation. Band pass filters are known generally as second-order filters, (two-pole) because they have “two” reactive component, the capacitors, within their circuit design. One capacitor in the low pass circuit and another capacitor in the high pass circuit.

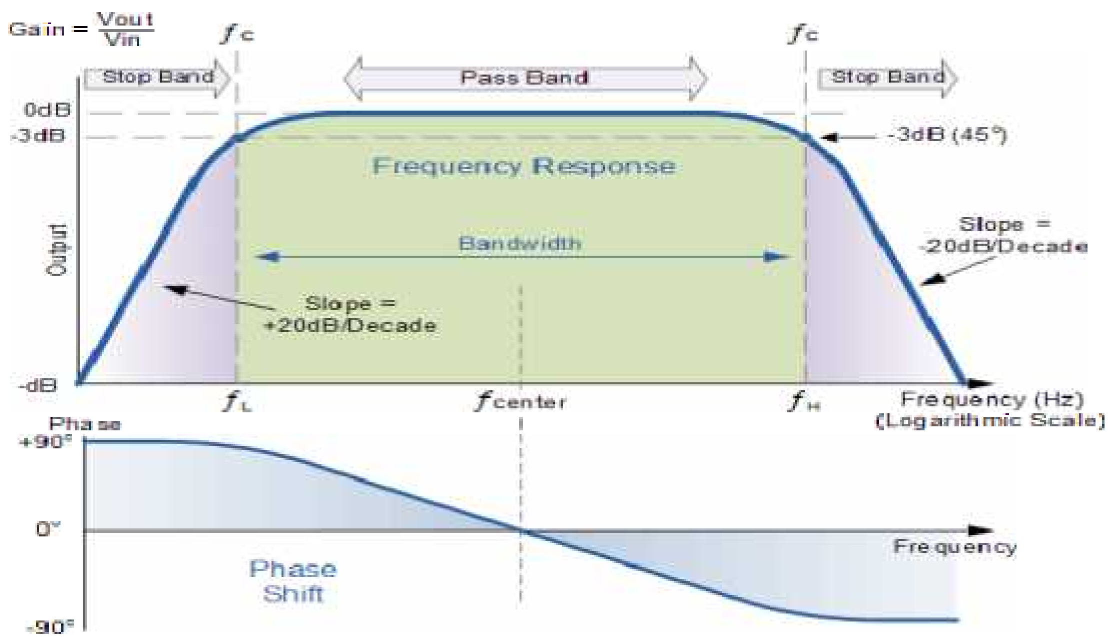


Figure 2.13 frequency response curve

The Bode Plot or frequency response curve above (figure 2.13) shows the characteristics of the band pass filter. Here the signal is attenuated at low frequencies with the output increasing at a slope of +20dB/Decade (6dB/Octave) until the frequency reaches the “lower cut-off” point fL. At this frequency the output voltage is again $1/\sqrt{2}$ = 70.7% of the input signal value or -3dB (20 log (Vout/Vin)) of the input.

The output continues at maximum gain until it reaches the “upper cut-off” point fH where the output decreases at a rate of -20dB/Decade (6dB/Octave) attenuating any high frequency signals. The point of maximum output gain is generally the geometric mean of the two -3dB value between the lower and upper cut-off points and is called the “Centre Frequency” or “Resonant Peak” value$f$r. This geometric mean value is calculated as being fr 2 = f(UPPER) x f(LOWER).

A band pass filter is regarded as a second-order (two-pole) type filter because it has “two” reactive components within its circuit structure, then the phase angle will be twice that of the previously seen first-order filters, ie, 180o. The phase angle of the output signal LEADS that of the input by+90o up to the centre or resonant frequency, ƒr point were it becomes “zero” degrees (0o) or “in-phase” and then changes to LAG the input by -90o as the output frequency increases.

The upper and lower cut-off frequency points for a band pass filter can be found using the same formula as that for both the low and high pass filters, For example.

$$fc = \frac{1}{2\pi RC}\ Hz$$

Then clearly, the width of the pass band of the filter can be controlled by the positioning of the two cut-off frequency points of the two filters.

A second-order band pass filter is to be constructed using RC components that will only allow a range of frequencies to pass above 1kHz (1,000Hz) and below 30kHz (30,000Hz). Assuming that both the resistors have values of 10kΩ´s, calculate the values of the two capacitors required.

2.6.1 The High Pass Filter Stage.

The value of the capacitor C1 required to give a cut-off frequency ƒL of 1kHz with a resistor value of10kΩ is calculated as:

$$C = \frac{1}{2\pi fc.R} = \frac{1}{2\pi x 1{,}000 x 10{,}000} = 15.8\,nF$$

Then, the values of R1 and C1 required for the high pass stage to give a cut-off frequency of 1.0kHz are: R1 = 10kΩ´s and C1 = 15nF.

2.6.2The Low Pass Filter Stage.

The value of the capacitor C2 required to give a cut-off frequency ƒH of 30kHz with a resistor value of 10kΩ is calculated as:

$$C = \frac{1}{2\pi fc.R} = \frac{1}{2\pi x 30{,}000 x 10{,}000} = 510\,pF$$

Then, the values of R2 and C2 required for the low pass stage to give a cut-off frequency of 30kHz are, R = 10kΩ´s and C = 510pF. However, the nearest preferred value of the calculated capacitor value of 510pF is 560pF so this is used instead.

With the values of both the resistances R1 and R2 given as 10kΩ, and the two values of the capacitors C1 and C2 found for both the high pass and low pass filters as 15nF and 560pFrespectively, then the circuit for our simple passive Band Pass Filter is given as.

2.6.3 Completed Band Pass Filter Circuit

Figure 2.14 shows Completed Band Pass Filter Circuit

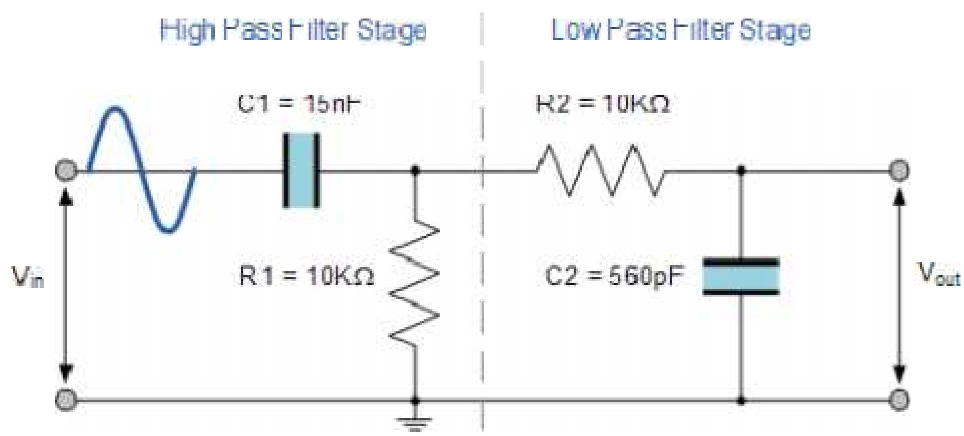


Figure 2.14 Completed Band Pass Filter Circuit

2.6.4 Band Pass Filter Resonant Frequency

We can also calculate the "Resonant" or "Centre Frequency" (ƒr) point of the band pass filter were the output gain is at its maximum or peak value. This peak value is not the arithmetic average of the upper and lower -3dB cut-off points as you might expect but is in fact the "geometric" or mean value. This geometric mean value is calculated as being ƒr 2 = ƒc(UPPER) x ƒc(LOWER) for example:

$$fr = \sqrt{f_L \; x \; f_H}$$

Where, fr is the resonant or centre frequency

fL is the lower -3dB cut-off frequency point

fH is the upper -3db cut-off frequency point

and in our simple example above, the calculated cut-off frequencies were found to be fL = 1,060 Hz and fH = 28,420 Hz using the filter values.

Then by substituting these values into the above equation gives a central resonant frequency of:

$$fr = \sqrt{f_L \times f_H} = \sqrt{1{,}060 \times 28{,}420} = 5{,}48\,kHz$$

2.6.5 Band Pass Filter Summary

A simple passive Band Pass Filter can be made by cascading together a single Low Pass Filter with a High Pass Filter. The frequency range, in Hertz, between the lower and upper -3dB cut-off points of the RC combination is know as the filters "Bandwidth".

The width or frequency range of the filters bandwidth can be very small and selective, or very wide and non-selective depending upon the values of R and C used.

The centre or resonant frequency point is the geometric mean of the lower and upper cut-off points. At this centre frequency the output signal is at its maximum and the phase shift of the output signal is the same as the input signal.

The amplitude of the output signal from a band pass filter or any passive RC filter for that matter, will always be less than that of the input signal. In other words a passive filter is also an attenuator giving a voltage gain of less than 1 (Unity). To provide an output signal with a voltage gain greater than unity, some form of amplification is required within the design of the circuit.

A Passive Band Pass Filter is classed as a second-order type filter because it has two reactive components within its design, the capacitors. It is made up from two single RC filter circuits that are each first-order filters themselves.

If more filters are cascaded together the resulting circuit will be known as an "nth-order" filter where the "n" stands for the number of individual reactive components and therefore poles within the filter circuit. For example, filters can be a 2nd-order, 4th-order, 10th-order, etc.

The higher the filters order the steeper will be the slope at n times -20dB/decade. However, a single capacitor value made by combining together two or more individual capacitors is still one capacitor.

Our example above shows the output frequency response curve for an “ideal” band pass filter with constant gain in the pass band and zero gain in the stop bands. In practice the frequency response of this Band Pass Filter circuit would not be the same as the input reactance of the high pass circuit would affect the frequency response of the low pass circuit (components connected in series or parallel) and vice versa. One way of overcoming this would be to provide some form of electrical isolation between the two filter circuits as shown below.

2.6.6 Buffering Individual Filter Stages

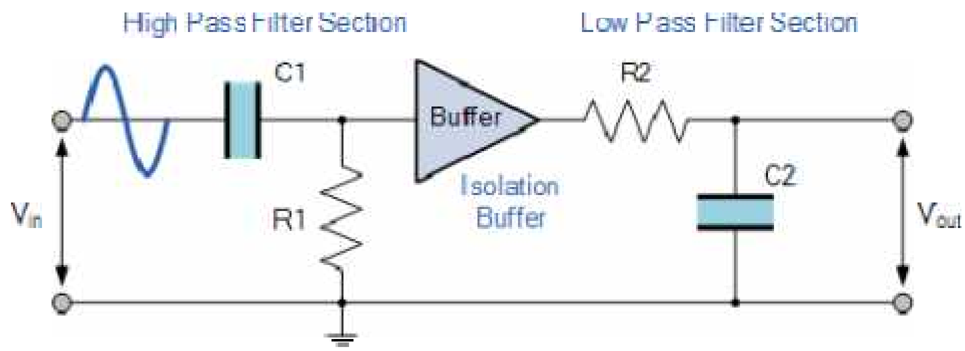


Figure 2.15 Active filter

One way of combining amplification and filtering into the same circuit would be to use an Operational Amplifier or Op-amp, and examples of these are given in the Operational Amplifier section. In the next tutorial we will look at filter circuits which use an operational amplifier within their design to not only to introduce gain but provide isolation between stages. These types of filter arrangements are generally known as Active Filters.

CHAPTER 3

Project Conceptual And Technical Design

3.1 Overview

3.2 Introduction

3.3 Subsystem detailed design

3.3.1 Balance Modulator / De-modulator MC1496

3.3.2 Summing amplifier LM741

3.3.3 Band pass filter TL084

Chapter three
Project conceptual and technical design

3.1 Overview
This chapter describes the detailed subsystems , like op-amp (741) , Balance modulator / demodulator (MC1496) , TL084.

3.2 Introduction

Data transfer process require two main parts , the transmitter and the receiver , the transmitter main function is to transmit the data as a signal and the receiver to receive it .

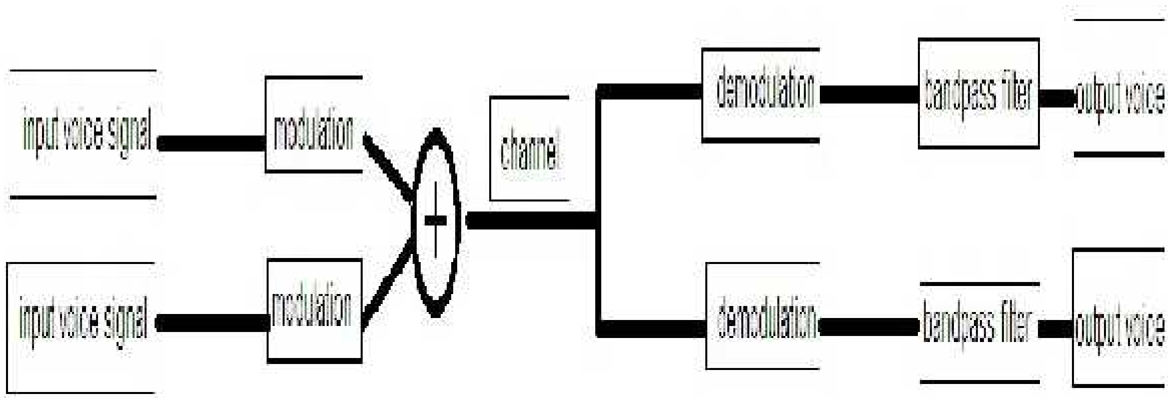


Figure 3.1: transmitter and receiver system

3.3 Subsystem detailed design
This section contains detailed description of the subsystems .

3.3.1 Balance modulator / demodulator (MC1496)
These devices were designed for use where the output voltage is a product of an input voltage (signal) and a switching function (carrier). Typical applications include suppressed carrier and amplitude modulation, synchronous detection
FM detection, phase detection, and chopper applications.

The MC1496 have 14 pins each with a special connection :

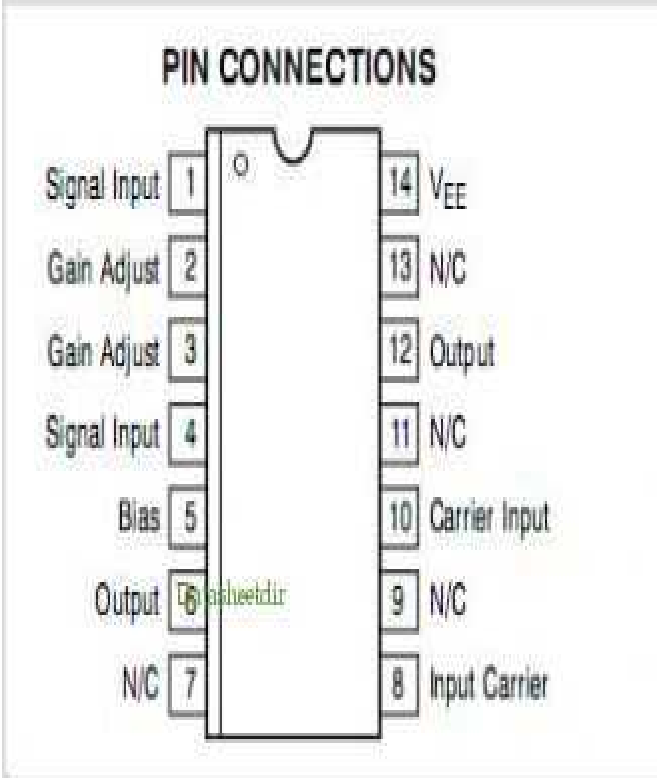


Figure 3.2 MC1496

As shown in figure 3.2 the MC1496 have input signal applied to pin 1 and input carrier applied to pin 10 and output signal applied to pin 6 . figure 3.3 shows the modulation and/de-modulation circuit .

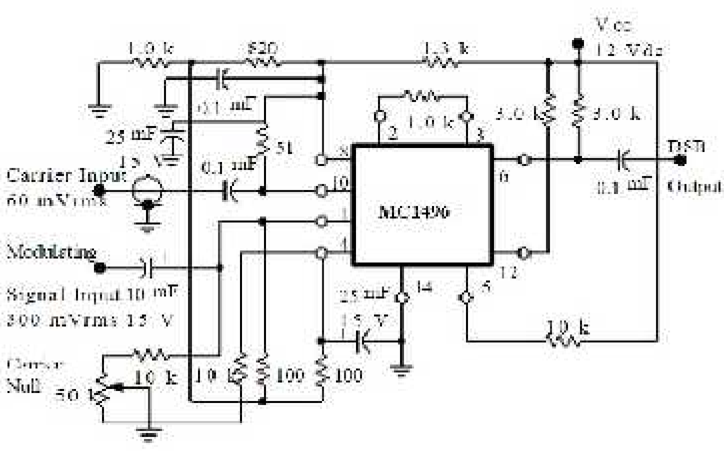


Figure 3.3modulation /de-modulation circuit

3.3.2 Summing amplifier LM741

The LM741 series are general purpose operational amplifiers.
The amplifiers offers many features which make their application nearly foolproof overload protection on the input and output .no latch up when the common mode range is exceeded .

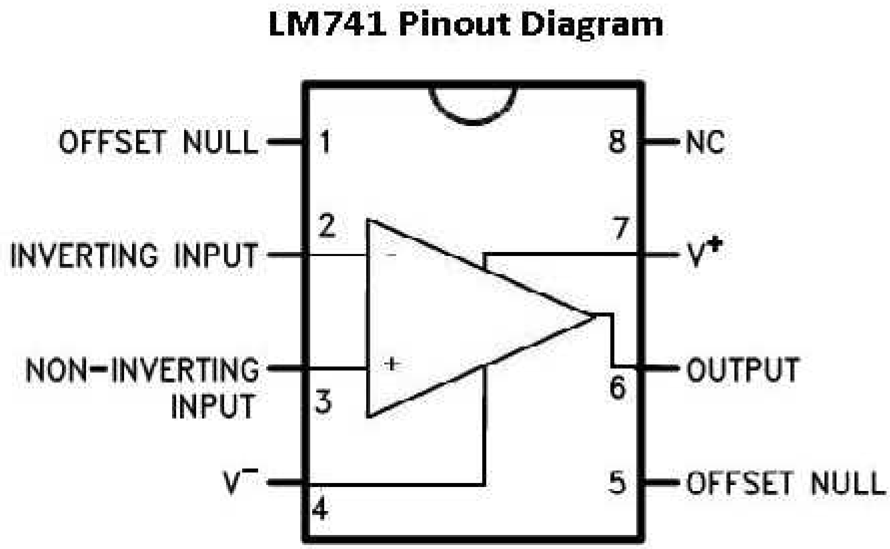


Figure 3.4 LM741

As shown in figure 3.4 LM741 have 8 pins ,each with special connection , figure 3.5 below show the LM741 as a summing amplifier .

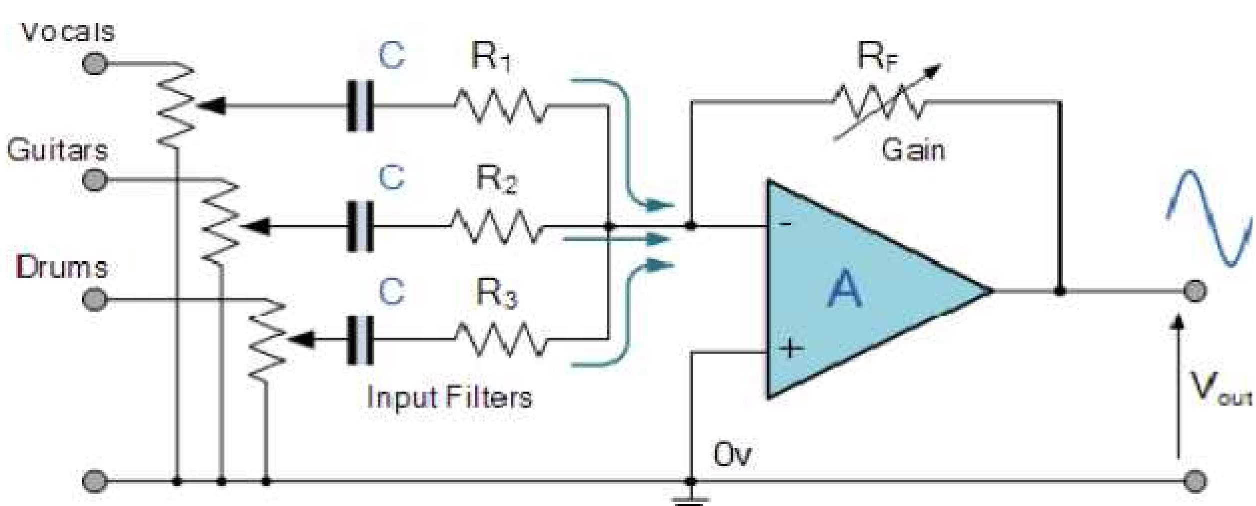


figure 3.5 summing amplifier

3.3.3 Band pass filter TL084

The TL 084 high speed J-FET input quad operational amplifiers incorporating well matched, high voltage J-FET and bipolar transistor in a monolithic integrated circuit .

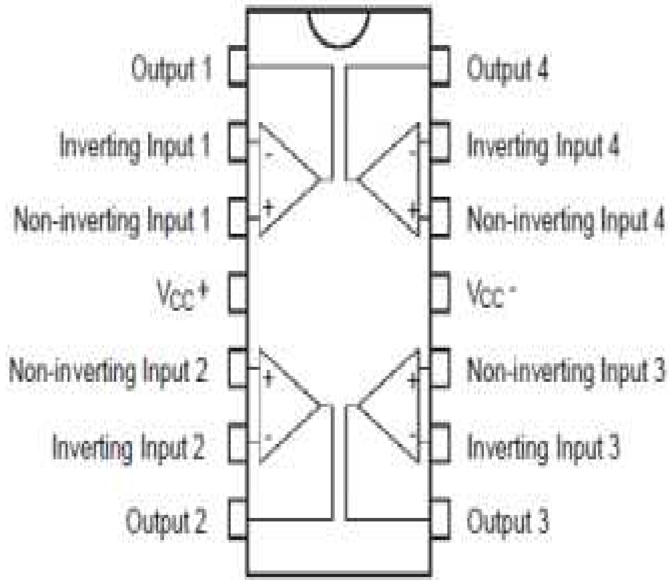


Figure 3.6 TL084

As shown in figure 3.6 TL084 have 14 pine .We use TL084 in the design of the band pass filter as in the circuit below :

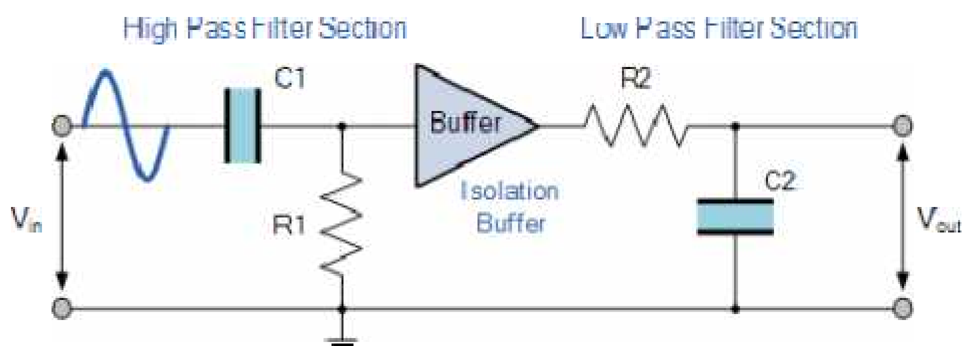


Figure 3.7 band pass filter

CHAPTER

4

Implementation and Testing

Chapter four

Implementation and Testing

5.1 Overview

In this chapter the construction and testing for the system would be shown , the construction and testing processes are very important to insure that the system work successfully .

5.2 Construction

At the beginning all the required components to build the project are provided , I built the circuit and I soldered the electronic components onto the board .

5.3 Testing

First of all I will show the testing on a single tone , as shown below

We use an 4Khz sine wave as an input signal , and a carrier frequencies of 100Khz and 120Khz

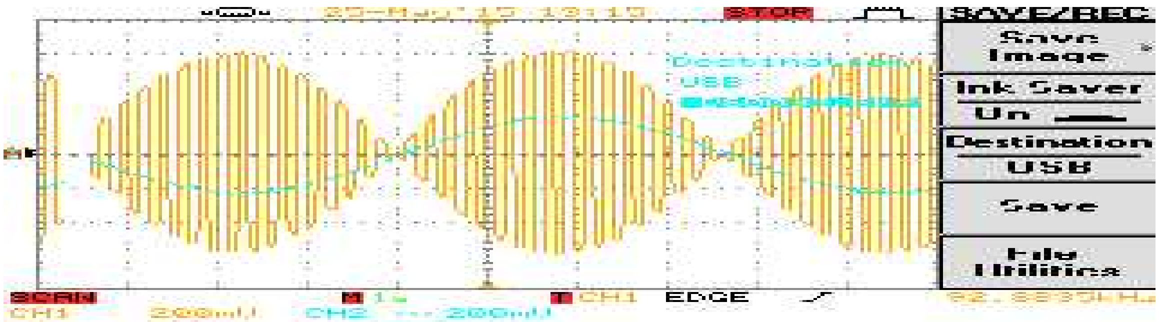


Figure 4.1 modulated signal

In this figure we can see the two modulated signal before summing

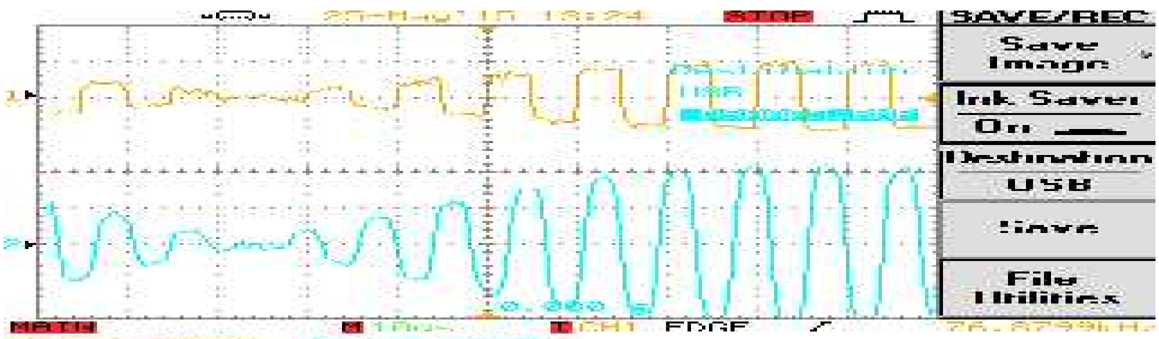


Figure 4.2 two signals modulated

And here we can see the two signal after summing (transmitted signal)

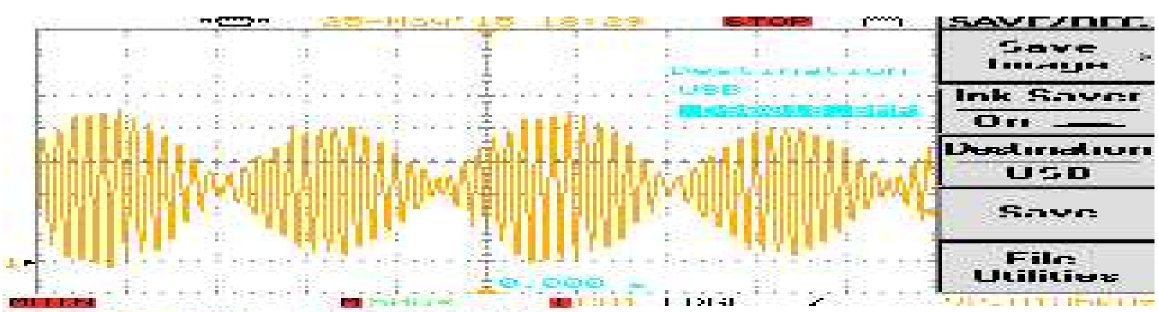


Figure 4.3 two modulated signal after summing

Reference

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[7]:MC 1496 datasheet.

[8]:LM741 datasheet .

[9]:TL084 datasheet.

Appendix