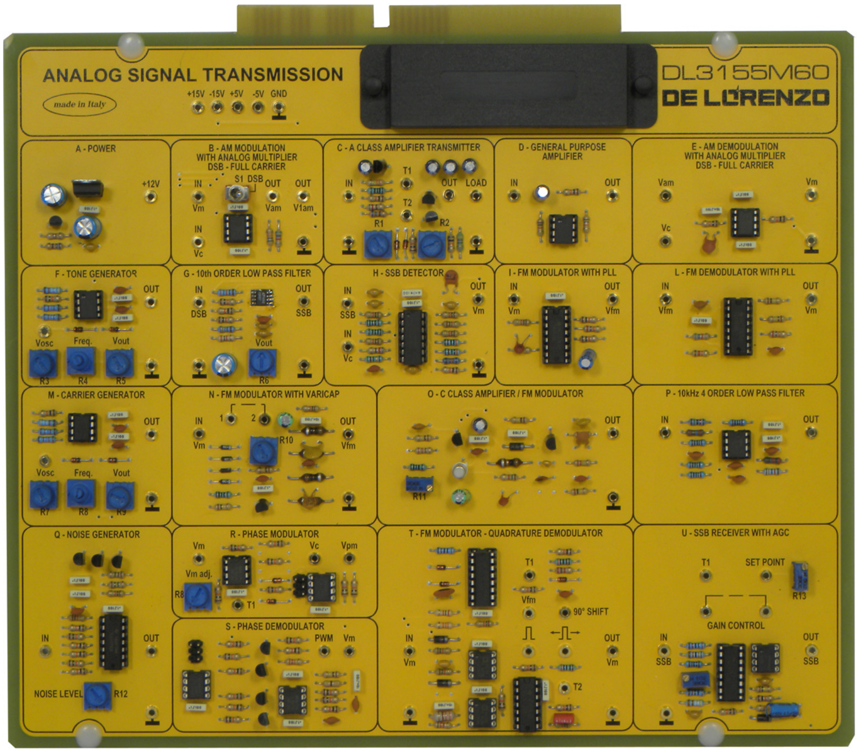
DE LORENZO

Engineering Training Solutions

ANALOG SIGNAL TRANSMISSION



STUDENT MANUAL

DL 3155M60

Rev. 20170607

DL 3155M60: Analog Signal Transmission

This Training Software analyzes the analog signal modulation principles:

THEORETICAL GUIDE
Lessons

- Introduction to Analog Signal Transmission
- Amplitude modulation (AM)
- Amplitude demodulation
- Frequency modulation (FM) and phase modulation (PM); Angular and demodualtion
- Filters

PRACTICAL GUIDE
Units:

- Introduction
- Understanding oscillators
- Understanding filters
- Understanding amplitude modulation (AM)
- Understanding amplitude demodulation (AM demodulation)
- Understanding frequency modulation/ demodulation
- Understanding PLL frequency modulation (PLL FM)
- Understanding FM modulator- quadrature demodulator
- Understanding PM- phase modulator/ demodulator. Individual work

THEORETICAL GUIDE

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Lesson N.1: Introduction to Analog Signal Transmission

Objectives:

To know:

- **the principle of Transmission and Modulation**
- **the advantages of the filters and Oscillators in Analog Transmission**
- **and identify filters and oscillators circuits used in Analog Signal Modulation**

Requisites:

- **Characteristics of the analog signals**
- **LC and RC circuits**

Arguments:

- **Frequency spectrum and Harmonics**
- **Mixing the signals**
- **Frequency response and Filtering (Filter types)**
- **Oscillators**
- **Hartley and Colpitts oscillators**
- **Crystal oscillator**
- **Microprocessor based oscillator**

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1.1 Why?

Why signal transmission?

Any signal is subject to losses along transmission environment, which interfere with the receiver's ability to interpret the information.
Everywhere a signal is driven along a wire, optical environment, or via radio communication. In any situation, the transmission environment interferes with the signal in a bad way.
At lower frequencies the signals remain within data characterization and the system performs as designed. But as system speeds increase, the higher frequency impact on the entire system.

Why radio communication?

Electromagnetic radiation travels by means of oscillating electromagnetic fields that pass through the air and the vacuum of space. Information is carried by controlled changing (modulating) some property of the radiated waves, such as amplitude, frequency, phase, or pulse width. When radio waves pass an electrical conductor, the oscillating fields induce an alternating current in the conductor. This can be detected and transformed into original signals that carry information.
Radio communication is the process of transmission of signals over by modulation of electromagnetic waves with high frequencies.

What is a modulation?

The process of adding information to a carrier signal is a process called modulation (controlled changing of one of the radiated wave's properties).
Signals with data or information (or over long wires or when stored on magnetic media) must be modulated , when they are sent by radio, with some method that prevents their shape and content from degrading before the signals can be received.
A transmitter and receiver devices must use the same process of modulation to successfully understand each other.
The process where the information is extracted from the radio signal and reconstituted in its original format is called demodulation process.

1.2 Frequency spectrum

Here, there is the table summarizing the most used classification of the range frequency, and names.

Frequency Range	Classification
3 - 30 kilohertz	Very low frequencies (VLF)
30 - 300 kilohertz	The long wave band (LW)
300 - 3000 kilohertz (3 megahertz)	The medium wave band (MW)
3 - 30 megahertz	The short wave band (SW)
30 - 300 megahertz	Very high frequency band (VHF)
300 - 3000 (3 gigahertz)	Ultra high frequency band (UHF)
3 gigahertz - 30 gigahertz	Super high frequency band (SHF)
300 - 3000 gigahertz	Microwave frequencies

Higher in frequency than this are infra red, visible light, ultra violet, X rays etc. which are all forms of Electro Magnetic radiation.

1.3 Harmonics

Let's imagine a situation when the same note, is played on different instruments, the musical notes produced sound different.
This is because that as well as producing the fundamental frequency of middle note, they also produce multiples of this frequency called harmonics.

The fundamental is a sine wave.

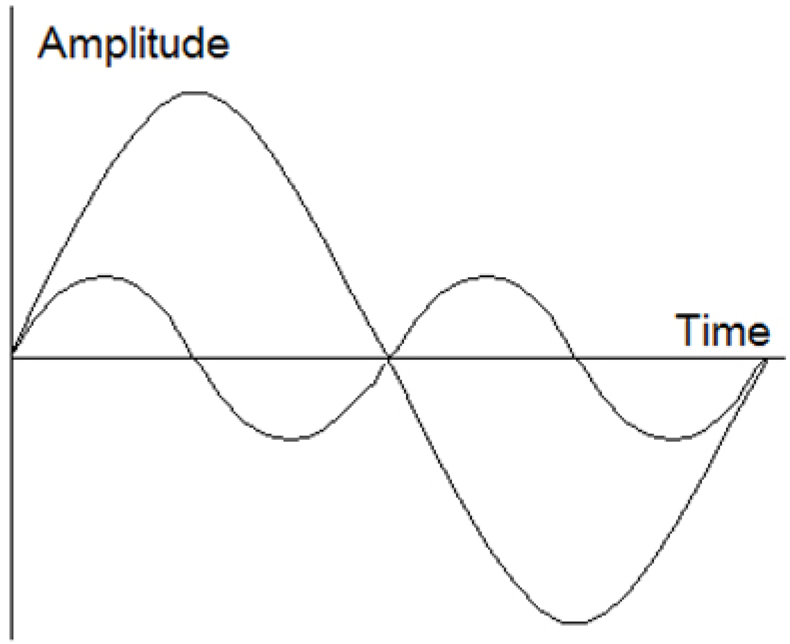

Figure 1.1 - A sine signal covered by its harmonic

For harmonic signal, there are two descriptive parameters: the amplitude of it, and the number. For upper example, the number and amplitude of the harmonics determines the characteristic sound of the instrument.

The harmonic which is twice the fundamental frequency, as in the diagram, is called the 2nd harmonic.
The frequency which is three times the fundamental is named the 3rd harmonic. The 3rd, 5th, 7th etc are called ODD harmonics.

The 2nd, 4th, 6th, 8th, etc, is called EVEN harmonics.
For some application, a square wave is made up from a sine wave fundamental frequency and an infinite number of odd harmonics.
If a sine wave is injected into an amplifier the output wave form may be distorted.

This may be due to harmonics being generated by the amplifier. This is the reason why we are studying the harmonics in transmission systems.

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1.4 Mixer the signals

The mixer is an information block that has two input signals of different frequencies, f1 and f2.

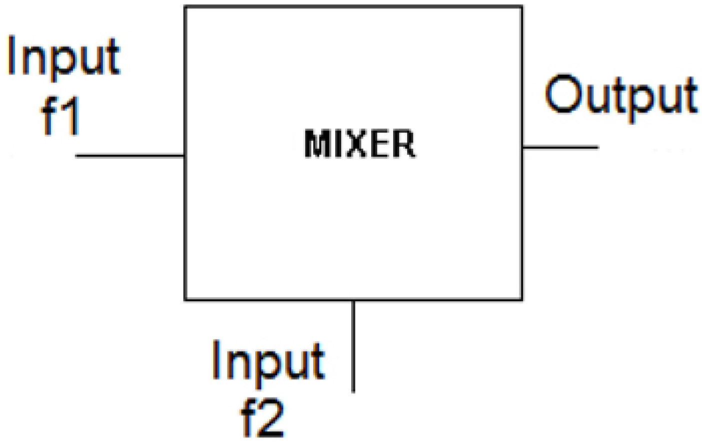

Figure 1.2 - The meaning of a signal mixer

These f1, and f2 inputs are mixed together in the mixer- some books say "beaten" together, others say "heterodyned").
One of the output frequencies is the sum of the two inputs, f1 + f2.

The other is the difference between the two inputs, f1 - f2. For example, if the inputs are 1 Mhz and 1.47 MHz then the sum frequency is 2.47 MHz.
The difference frequency is 0.47 MHz (470 kHz). A mathematical model proves very simple this result.

In transmission process, on the radio, two adjacent stations will produce an interfering signal.
This is because their frequencies are close enough to be mixed together.

Because of this situation, the difference between their frequencies is in the audio range. For this phenomenon, the mixers are used as part of the frequency changer in radio communication.
So, understanding mixers will help us to understand the modulation process in transmitters.

1.5 Frequency response

It happens that some signals are very low. In order to use and to process them, it is necessary to apply a process of amplification of them.
Because of reactive behavior of amplification block, some of signals penetrate very easy the amplification block, and other not.
The behavior of amplification from the frequencies response point of view is called frequency response.

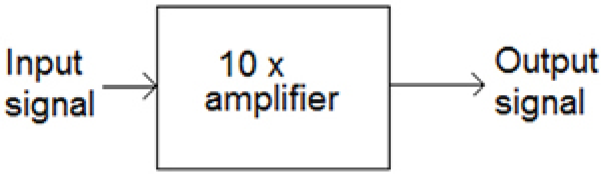

Figure 1.3 - Amplifying input signals ten times

An ideal amplifier with an amplification of times 10, as shown in upper figure, would give an output 10 times greater than the input signal, without influences in frequency domain.
If the input signal is 10 mV then the output signal would be 100 mV, as next figure shows.

This graph is known as a frequency response diagram.

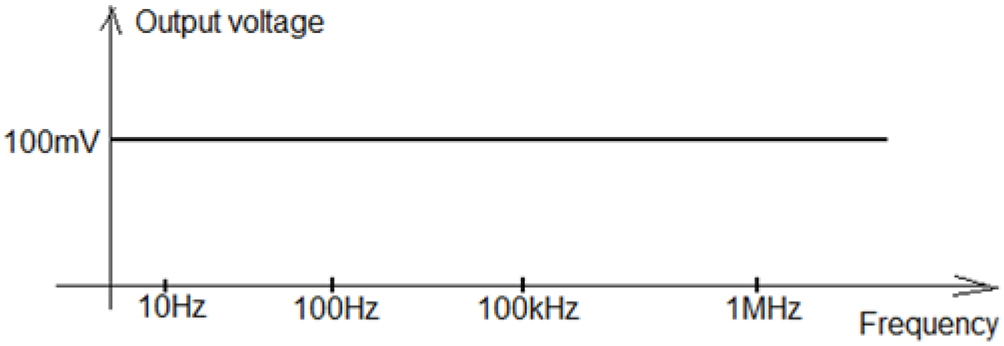

Figure 1.4 - Frequency response diagram in accordance with input signal frequencies

In a practical amplifier it is not possible to obtain such perfectly flat response curve.
This is due to limitations of electronic components and circuitry. Usually there is a fall of response at sides of some range.

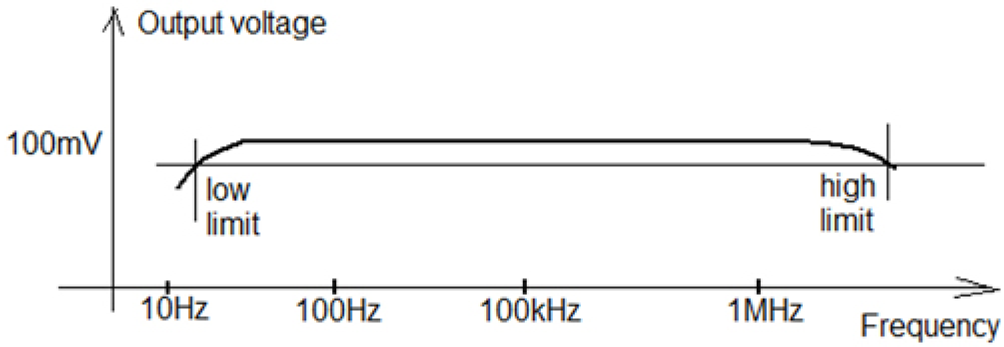

Figure 1.5 - The limits of frequency response of the amplifier

The two points (low limit and high limit) from the response curve mark where the output of the amplifier has fallen to 70.7 % of the maximum output.
This means that the 100mV output has fallen to 70.7 mV at these frequencies. In transmission process, these are called the -3 dB points. One is at about 25 kHz (f1).

The other is at about 300 MHz (f2). By subtracting f1 from f2 we get the bandwidth of the amplifier. In this case it is just under 275 MHz wide- an extremely large band amplifier!
In some cases the bandwidth is tailored to pass some frequencies and not others. This is called filtering process.

1.6 Filtering process

A filter allows some things through and holds back others. In this case we are talking about AC frequencies.
Some frequencies pass through the filter while others are rejected.

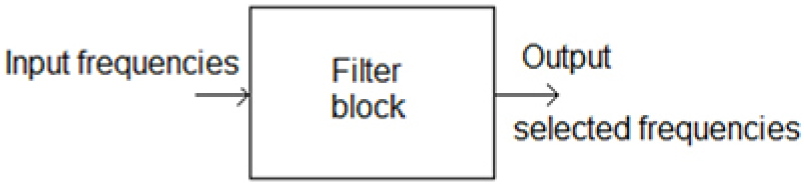

Figure 1.6 - The behavior of filter block in accordance with input frequencies

We hope that everybody accepts that the characteristics of a filter can be shown on a graph called a frequency response curve.

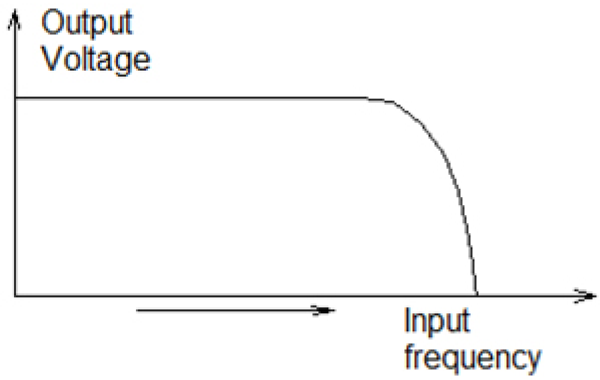

Figure 1.7 - The frequency response of a filtering block

The upper figure shows a low pass filter response curve giving output at low frequencies but none at higher frequencies.

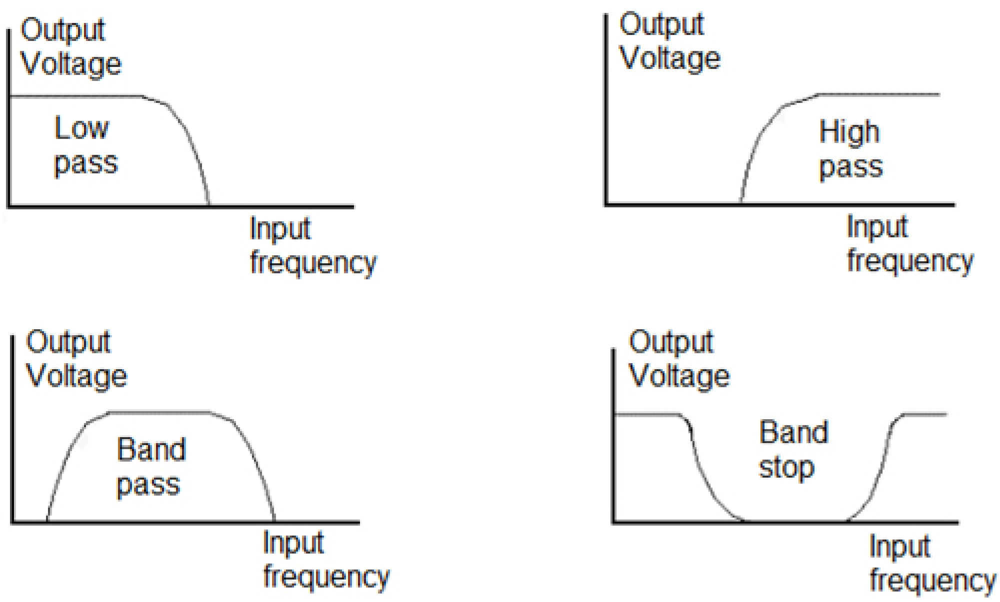

Figure 1.8 - The frequency response in different filtering processes

Following the same understanding approach, we should accept some behaviors of different responses of filters to the input frequencies.
The filters have many applications. Here they are some filtering symbols together with application meanings.

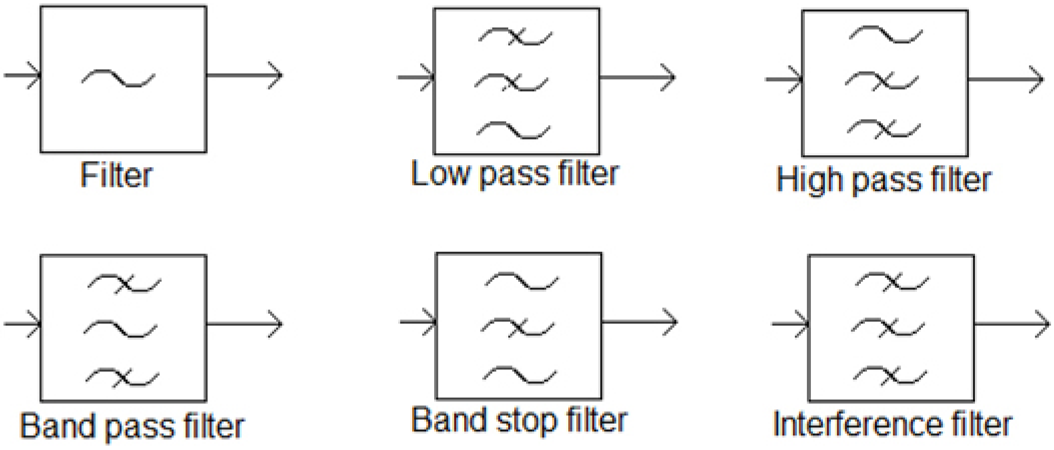

Figure 1.9 - Filters block diagram and symbols

1.7 Oscillators

Oscillators, as electronic blocks, are amplifiers with positive feedback, so they produce an output signal with no signal applied to the input.

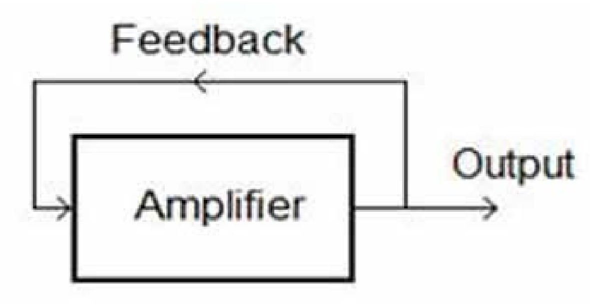

Figure 1.10 - The block diagram of the oscillators

As it is figured, the output amplitude is determined by the gain of the amplifier and the feedback circuit.
They are characterized by frequency of oscillation and amplitude of the output signals. Usually, the oscillators can produce sine waves, with the frequency of which is determined by the designed circuit. The designed circuit, as a tuned circuits consist of a capacitor and inductance.
Square wave oscillators use resistors and capacitors to determine the frequency of oscillation.
Ideally the frequency of an oscillator should be stable, but in practical applications, due to temperature variations and mechanical behaviors this may not be so.
Another important parameter that defines the quality of the oscillator is frequency drift.

The resonance of the oscillators
As we know, at high frequencies of the input signal, the reactance of a capacitor is very low acting as a short circuit while the reactance of the inductor is high acting as an open circuit.
At opposite side, at low frequencies the reverse is true, the reactance of the capacitor acts as an open circuit and the reactance of the inductor acts as a short circuit.
In between these two extremes, the combination of the inductor and capacitor produces a "tuned" or "resonant" circuit that has a resonant Frequency (fr), in which the capacitive and inductive reactance's are equal, and they are compensating each other, by leaving only the resistance of the circuit to establish the flow of current.

Using oscillators
The oscillators are circuits that generate a continuous voltage output waveform at a required frequency, and amplitude, with the values of the inductors, capacitors or resistors forming a frequency selective LC resonant feedback network.

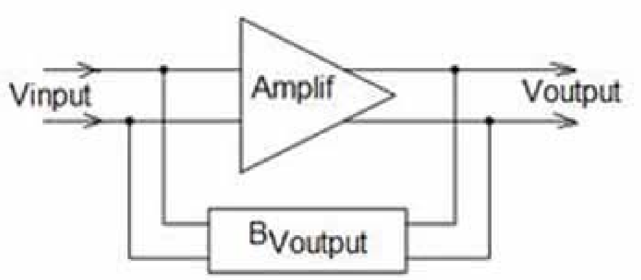

Figure 1.11 - The basic block diagram of amplification module with feedback

If in a normal amplification module, the output voltage is calculated as the ratio between Voutput and Vinput:

$$A_V = V_{Output} / V_{Input}$$

when there is intercalated a feedback module, the calculus is proper adjusted:

$$V_{Output}(1 - A_V \beta) = A_V V_{Input}$$

So, the output amplified voltage is corrected with amplification factor (β) of the feedback module.
The oscillator's frequency is controlled using an adapted or resonant inductive/capacitive (LC) circuit with the resulting output frequency being known as the oscillation
Frequency. It is done by making the oscillators feedback as a reactive network for that, the phase angle of the feedback will vary as a function of frequency and this is called Phase-shift process.
Taking into account the resonant frequency (when capacitive reactance is equalized by inductive reactance), we are analyzing now the resonant LC circuit.
We know that for Resonance to occur both the capacitive, X_C and inductive, X_L reactance's must be equal, $X_L = X_C$ and opposite to cancel out each other out, and leaving only the resistance in the circuit to establish the flow of current. So, the frequency at which this will happen is given as:

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

This basic equation shows that if L or C has a variation, then the resonant frequency has opposite variation.
As we know, the essence of oscillators working is to keep constant the output variable voltage.
It means that we must keep the oscillations going in an LC circuit we have to replace the energy lost in each oscillation and also to maintain the amplitude of the oscillations at a constant level. The simplest way of replacing this energy is to take part of the output from the LC circuit, amplify it and then feed it back into the LC circuit again and this can be achieved using a voltage amplifier. The methods we are using for implementing this requirement are many, but in transmission process some of them are well known.

1.8 Hartley Oscillator

In the previous section, the basic LC oscillator circuit there was not presented methods of controlling the amplitude of the oscillations.
It might happen, if the electromagnetic coupling between L and L2 is too small there would be insufficient feedback and the oscillations would eventually die away to zero.

Also, if the feedback was too strong the oscillations would continue to increase in amplitude until they produce distortions or damages. So, if we feed back more than is necessary the amplitude of the oscillations can be controlled by biasing the amplifier in such a way that if the oscillations increase in amplitude, the bias is increased and the gain of the amplifier is reduced.

If the amplitude of the oscillations decreases the bias decreases and the gain of the amplifier increases, thus increasing the feedback.
In this way the amplitude of the oscillations are kept constant and this is known as Automatic Base Bias.

This procedure could be implemented by oscillators made by providing a Class-B bias or even a Class-C bias as the Collector current flows during only part of the cycle and the quiescent Collector current is very small. So, there is a "self-tuning" base oscillator circuit. This forms the basic configuration for the Hartley Oscillator circuit.

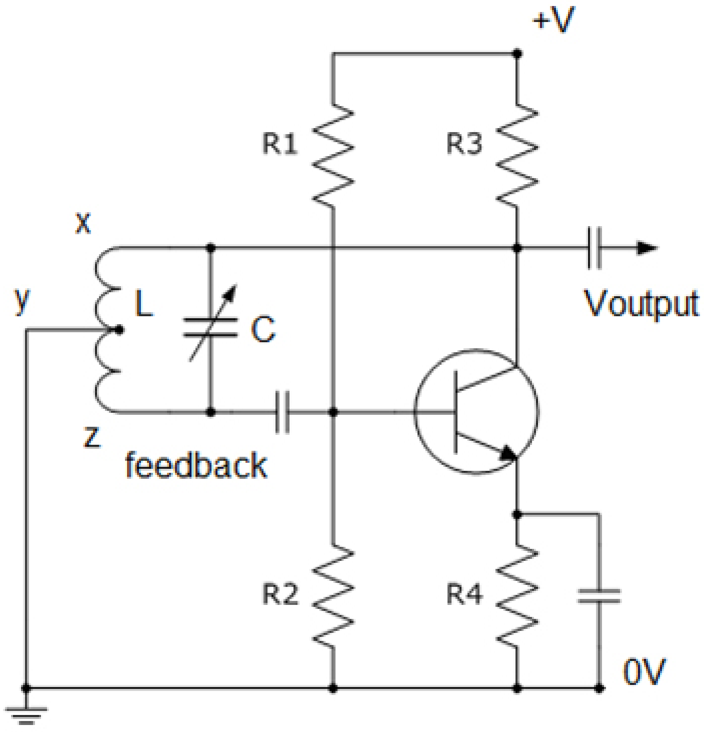

Figure 1.12 - Basic schematic of Class-B amplification for Hartley oscillator

n this diagram, the adjusted LC circuit is connected between the Collector and the Base of the transistor amplifier and, as far as the oscillatory voltage is concerned, the emitter is connected to a tapping point on the tuned circuit coil. The feedback of the tuned resonant circuit is taken from the centre tap of the inductor coil or two separate coils in series which are in parallel with a variable capacitor, C.

When the circuit is oscillating, the voltage at point X (collector terminal), relative to point Y (emitter terminal), is 1800 out-of-phase with the voltage at point Z (base terminal) relative to point Y. At the frequency of oscillation, the impedance of the Collector load is resistive and any increase in Base voltage causes a decrease in the Collector voltage.

Then there is an 1800 phase change in the voltage between the Base and Collector and this along with the original 1800 phase shift in the feedback loop provides the correct phase relationship of positive feedback for oscillations to be maintained.

$$f_r = \frac{1}{2\pi\sqrt{(L1+L2)C}}$$

The L1, L2 inductances form the total inductance that is used in building the resonant circuit (Lxyz). In this diagram it can be seen that the resonant frequency could be adjusted by variation of the capacitor value (C).

The limits of the capacitor value will determine the upper, and the lower resonant frequency, and also the oscillator bandwidth as the difference between the two resonant frequencies.

1.9 Colpitts Oscillator

Like in the Hartley oscillator, the tuned resonant circuit consists of an LC resonance circuit connected between the Collector and Base of the transistor amplifier.
The basic configuration of the Colpitts Oscillator is that the centre tapping is now made from a "Capacitive Voltage Divider" network instead of a tapped double inductor.

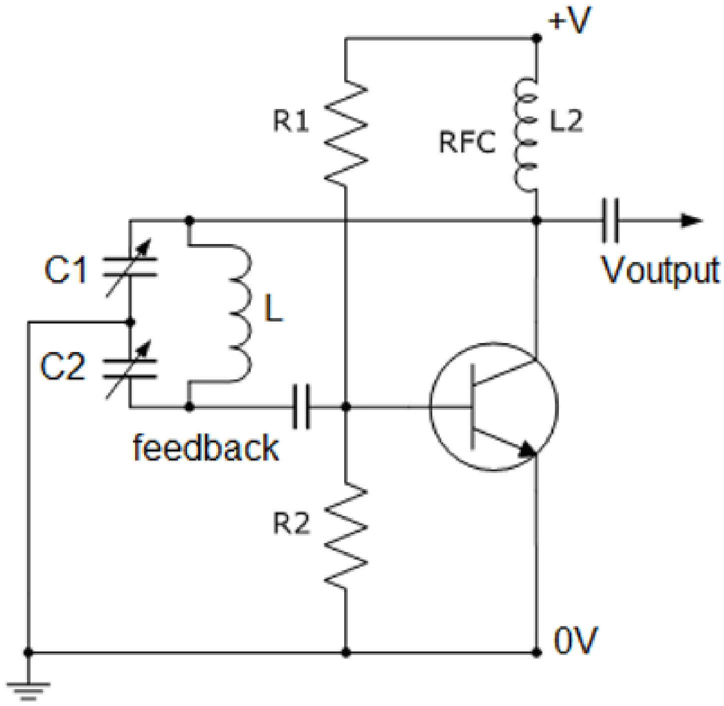

Figure 1.13 - The new method of centre tapping with capacitors (Colpitts oscillator)

The transistor Emitter is connected to the junction of capacitors, C1 and C2 which are connected in series with the required external phase shift is obtained in a similar manner to that in the Hartley Oscillator. The amount of feedback is determined by the ratio of C1 and C2 which are engaged together to provide a constant amount of feedback.
The frequency of oscillations for a Colpitts Oscillator is determined by the resonant frequency of the LC resonant circuit and is calculated as:

$$f_r = \frac{1}{2\pi\sqrt{LC_T}} = \frac{1}{2\pi\sqrt{L\dfrac{C1C2}{C1+C2}}}$$

One more remark: the configuration of the transistor amplifier is of a Common Emitter Amplifier with the output signal 1800 out of phase with regards to the input signal.
The additional 1800 phase shift require for oscillation is achieved by the fact that the two capacitors are connected together in series but in parallel with the inductive coil resulting in overall phase shift of the circuit being zero or 3600.

The C1, C2 capacitances form the total capacitance that is used in building the resonant circuit. In this diagram it can be seen that the resonant frequency could be adjusted by variation of the capacitors value (C1, C2).
The limits of the capacitors values will determine the upper, and the lower resonant frequency, and also the oscillator bandwidth as the difference between the two resonant frequencies.

1.10 Crystal Oscillator

Everybody says that one of the most important features of an oscillator is its frequency stability. Some factors that affect the frequency stability of an oscillator are: temperature, variations in the load and changes in the power supply parameters.
For very high stability a quartz crystal is generally used as the frequency generator device.

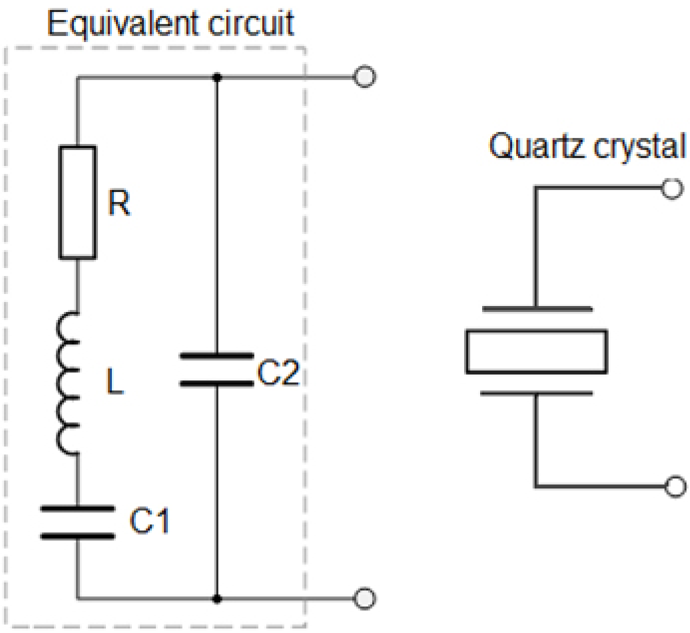

Figure 1.14 - The representation of crystal equivalent circuit

When a voltage source is applied to a small thin piece of crystal quartz, it begins to change shape producing a characteristic known as the Piezo-electric Effect. A mechanically vibrating crystal can be represented by an equivalent electrical circuit consisting of low resistance, large inductance and small capacitance.

As we can see, a quartz crystal has a resonant frequency similar to that of an electrically adjusted circuit but with a much higher Q factor due to its low resistance. It also might have two or more resonant frequencies having both a fundamental frequency and harmonics such as second or third harmonics.

In a Crystal Oscillator circuit the oscillator will oscillate at the crystals fundamental series resonant frequency as the crystal always wants to oscillate when a voltage source is applied to it.
It is also possible to "tune" a crystal oscillator to any even harmonic of the fundamental frequency, (2nd, 4th, 8th etc.) and these are known generally as harmonic oscillators while overtone oscillators vibrate at odd multiples of the fundamental frequency, 3rd, 5th, 11th etc).

1.11 Microprocessor based oscillator

One typical application today is microprocessor based oscillator. Most microprocessors, microcontrollers, and PICs, have two oscillator pins labeled OSC1 and OSC2 to connect to an external quartz crystal, or a Ceramic resonator.

In this application the crystal oscillator produces a train of continuous square wave pulses whose frequency is controlled by the crystal.

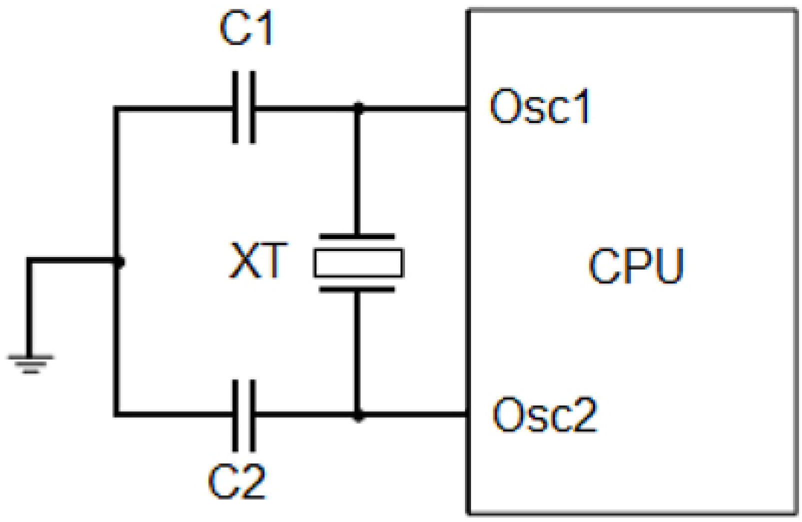

Figure 1.15 - Typical diagram of microprocessor- based oscillator

In order cu calculate the resonant frequency of the designed circuit, for each crystal device, from the catalog we are taking the specific values for the cut crystal.
For instance, 20Mhz crystal might have R=1 kΩ, C=0.05pF, L=3H. For such parameters, with shown formula, the fundamental frequency of oscillation is 411 kHz.

Lesson N.2: Amplitude Modulation (AM)

Objectives: **To know:**

- **the characteristics of the amplitude modulation**
- **the mathematical model of the amplitude modulation**
- **some circuits used in the AM modulation**

Requisites:

- **Signal transmission**
- **Lesson 1 of this Theoretical guide**

Arguments:

- **Modulation and carrier frequency**
- **Amplitude modulation**
- **Mathematical model**
- **AM modulation**
- **Types of AM**

2.1 Modulation and carrier frequency

Signals in transmission process

High frequency radio signals can be used to carry information.
The information is used to modify (modulate) a single frequency known as the carrier. The information superimposed onto the carrier forms a radio signal which is transmitted to the receiving device- called as receiver. Here, the information is extracted from the radio signal and reconstituted in its original format in a process known as demodulation.

There are many different varieties of modulation: amplitude modulation, frequency modulation and phase modulation. Each type has its own advantages and disadvantages. By reviewing all the techniques, a greater understanding of the advantages and disadvantages can be gained.

Radio carrier
The basis of any radio signal or transmission is the carrier. This consists of a bipolar alternating waveform.

This is generated in the transmitter. The pair device- receiver must have ability to receive it.
So, when we speak about transmission we must imagine two processes (modulation and demodulation) and to devices (transceiver and receiver).

Why modulation
For sending signal, the radiating elements (antennas) are in accordance with signal wavelength. For instance, to send 1 kHz signal, the antenna has to be 75 km length.
If we send signal included (or modulated) in 630 kHz, the radiating antenna should be only 119 m length.

On the other hand, if we consider simultaneous transmission of different audio signals for instance, we could send only one signal a time.
Otherwise, a mixture, an overlap of signals could be received, without possibility to separate them.
Through the modulation, we can transmit many signals simultaneously by shifting their spectra using different carrier frequencies.

2.2 Amplitude modulation

Possibly the most obvious method of modulating a carrier is to change its amplitude in line with the modulating signal.
The simplest form of amplitude modulation is to employ a system known as 'on–off keying' (OOK), where the carrier is simply turned on and off. This is a very elementary form of digital modulation and was the method used to carry Morse transmissions, which were widely used especially in the early days of 'wireless'.

Here, the length of the on and off periods defined the different characters.
More generally, the amplitude of the overall signal is varied in line with the incoming audio or other modulating signal, as shown in Figure 3-2.
Here, the envelope of the carrier can be seen to change in line with the modulating signal. This is known as Amplitude Modulation (AM).

Definitions

The modulation is a process by which some of characteristics of a high-frequency carrier signal are altered to convey information contained in a lower-frequency message.
There are a variety of modulation schemes available for both analog and digital modulation.

Amplitude Modulation (AM) is an analog modulation scheme where the amplitude (A) of a fixed-frequency carrier signal is continuously modified to represent information in a message.
The carrier signal is generally a high frequency sine wave used to “carry” the information on the envelope of the message.
The result is a double-sideband signal, centered on the carrier frequency, with twice the bandwidth of the original signal.

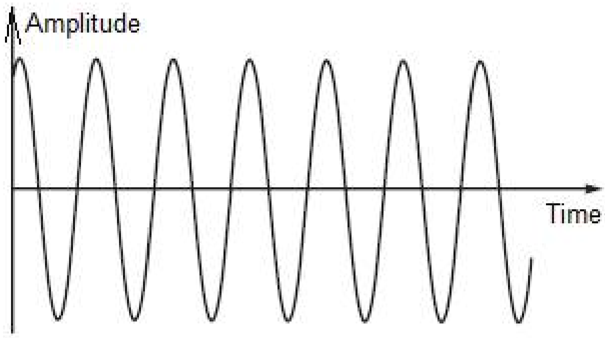

Figure 2.1 - This is the carrier, with the frequency fc

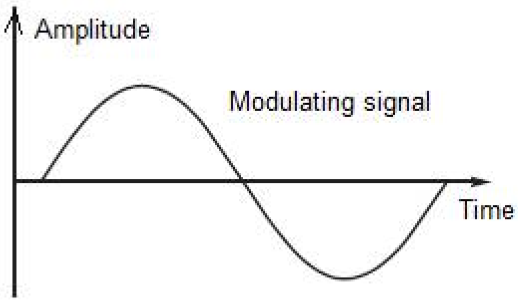

Figure 2.2 - This is modulating signal - m(t)

2.3 Mathematical model

The carrier, and signal models are:

$$x_c(t) = A_c \cos(2\pi f_c t) \qquad m(t) = M \cos(2\pi f t)$$

The general equation that support and explain amplitude modulation is:

$$y(t) = C\sin(\omega_c t) + M\frac{\cos(\varphi - (\omega_m - \omega_c)t)}{2} - M\frac{\cos(\varphi + (\omega_m + \omega_c)t)}{2},$$

and after simplifications:

$$y(t) = A_c[1 - k_a m(t)]\cos(2\pi f_c t)$$

where m(t) is baseband signal (carrier), fc =carrier frequency, Ac = carrier amplitude, ka = modulation index. As we could see, in general practice, a DC offset is added for commercial purpose. Taking into account these parameters values, there are some types of modulations: DSB-SC (double sideband suppressed carrier) AM, and SSB (single sideband) AM.
With no DC offset, mathematical model of amplitude modulation becomes:

$$y(t) = m(t)\cos(2\pi f_c t)$$

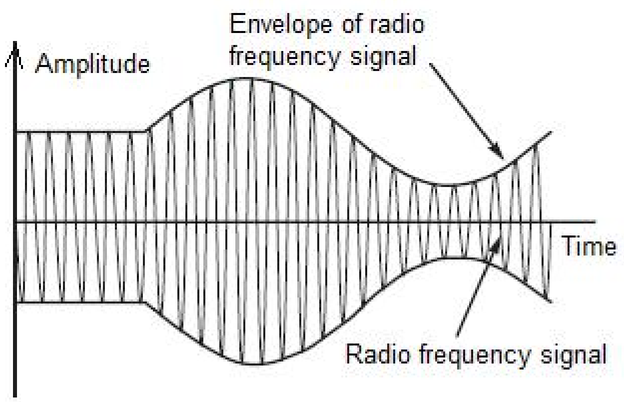

Figure 2.3 - These are changes in carrier, because of modulating signal

2.4 AM parameters

In our study and experiments, we are concerned with all these types of modulations. For that matter, the Fourier transform of the AM signal y(t) have to be applied.
The signal spectra are defined next.

In order to have a god analyze, we should think to the AM signals, by considering some descriptive parameters.

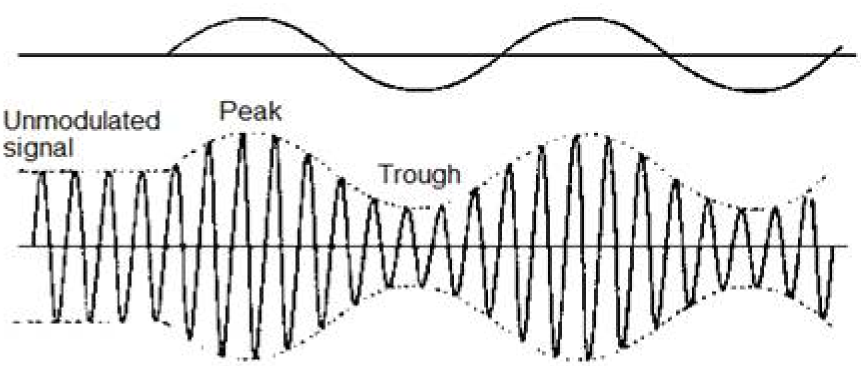

Figure 2.4 - The peak and trough values of modulated signal

$$S(f) = \frac{A_c}{2}[\delta(f - f_c) + \delta(f + f_c)] + \frac{k_a A_c}{2}[M(f - f_c) + M(f + f_c)]$$

They are two spectrum signals, corresponding $f - f_c$, and $f + f_c$, for that matter, the bandwidth of the spectra is calculated as:

$$B_{MA} = 2\pi f$$

Where, M(f) is the transformer of the information carrying useful information m(t). It is important to imagine any particular cases (ka =1, ka <1, ka >1).
Modulation index is usually expressed as percentage values [%]:

$$\beta = \left| k_a m(t) \right|_{\max} 100$$

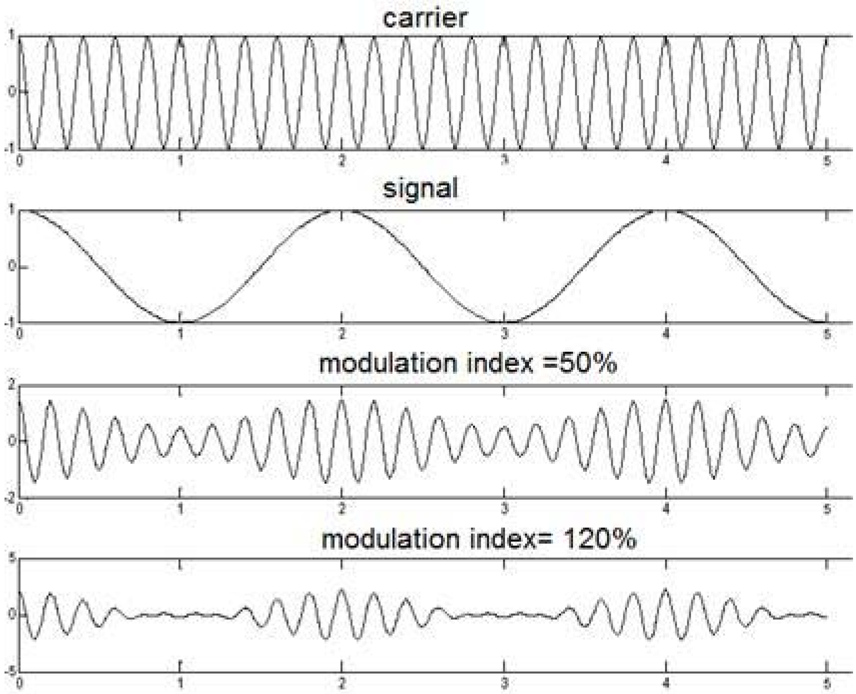

Figure 2.5 - Here, they are figured, for comparison, carrier, signal, modulated signals 50% (normal modulated), 120% (over modulation, with distortions)

2.5 Types of AM

By analyzing the spectra and modulation index, there are many different types of amplitude modulation, such as commercial amplitude modulation, DSB-SC (double sideband suppressed carrier) AM, and SSB (single sideband) AM. We are mainly concerned with the commercial amplitude modulation and the DSB-SC techniques for the purposes of this lab.

Commercial amplitude modulation is done by adding a dc offset to the baseband signal, m(t) and then multiplying by a sinusoid of frequency fc.

$$y(t) = A_c[1 + k_a m(t)]\cos(2\pi f_c t)$$

In DSB- SC AM, with no DC offset, the result of modulated signal becomes:

$$y(t) = m(t)\cos(2\pi f_c t)$$

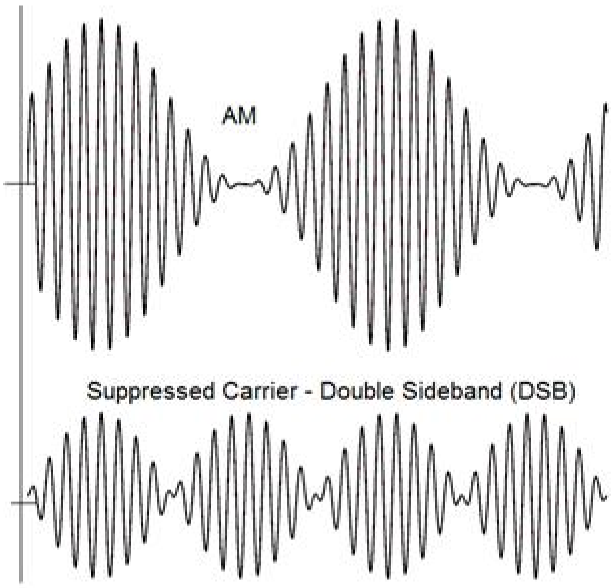

Figure 2.6 - There is a big difference between AM and DSB-SC signal power

For SSB we use again complete formula that defines amplitude modulation process:

$$y(t) = C\sin(\omega_c t) + M\frac{\cos(\varphi - (\omega_m - \omega_c)t)}{2} - M\frac{\cos(\varphi + (\omega_m + \omega_c)t)}{2}$$

Taking into account

$$y(t) = C\sin(\omega_c t) + M\frac{\cos(\varphi - (\omega_m - \omega_c)t)}{2} - M\frac{\cos(\varphi + (\omega_m + \omega_c)t)}{2}$$

we obtain lower sideband amplitude modulation, and with

$$y(t)_{LSB} = C\sin(\omega_c t) + M\frac{\cos(\varphi - (\omega_m - \omega_c)t)}{2}$$

we obtain upper sideband amplitude modulation.

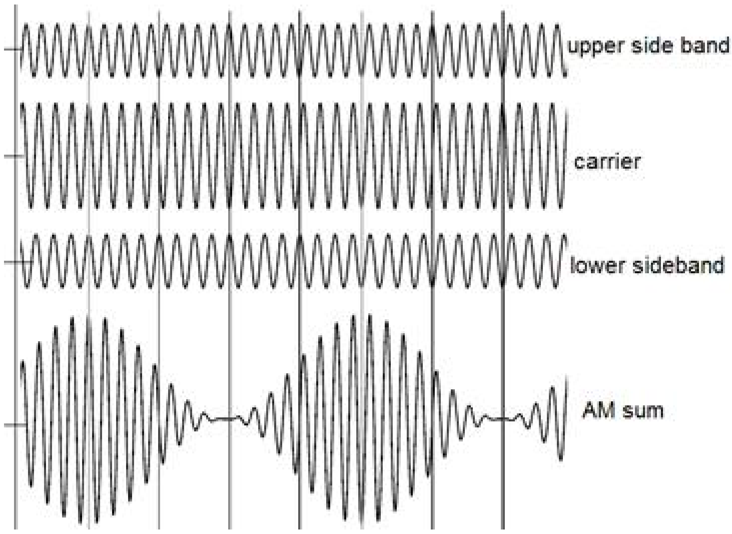

Figure 2.7 - Here, they are figured SSB amplitude modulation types.

AM advantages

As we could see, the main advantage of using AM modulation is that it has a very simple circuit implementation (especially for reception), creating widespread adoption quickly.
The carrier requires the majority of the signal power, but actually does not hold any information. As general representation we agree that AM uses twice the required bandwidth by transmitting redundant information in both the upper and lower sidebands.

2.6 A simple exercise

Any experiments must prove the modulation index together with the spectra of the modulated signal.

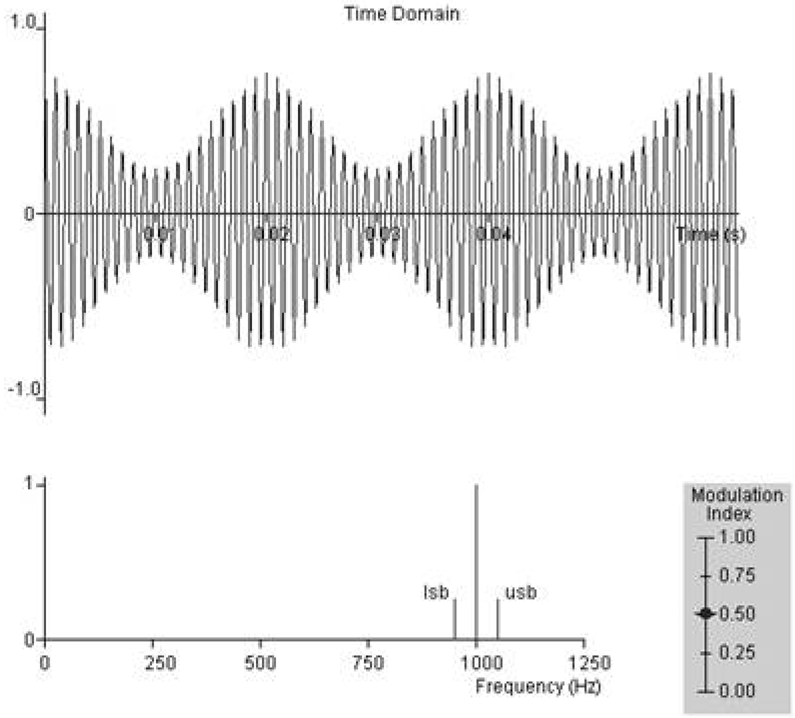

Figure 2.8 - Basic experiment panel

The upper experimental panel figures out a possible modeling interface that consists of generating AM signals related to modulation index.
It has the opportunity to understand the signal spectra related to the index modulation.

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Lesson N.3: Amplitude demodulation

Objectives: **To know:**

- **the characteristics of the amplitude demodulation**
- **the different techniques to recover the carrier signal**
- **the different SSB types and advantages**

Requisites:

- **Lessons 1 & 2 of this theoretical guide**

Arguments:

- **Amplitude demodulation**
- **Technics for recovering the carrier signal**
- **SSB demodulation**
- **SSB types and advantages**
- **Automatic Gain Control (AGC)**

3.1 Amplitude demodulation

In general practice equipment that is used to recover the original modulating signal from a modulated wave.
In radio industry, a demodulator is also known as a detector. In transmission systems the information to be transmitted is a periodic wave called a carrier. The carrier is then called modulated signal.
At reception side the original modulating signal is recovered by the process of demodulation or detection.

Amplitude demodulation

A demodulator device is an electronic circuit used to recover the information content from the modulated carrier wave.
The simple block diagram of AM demodulator is

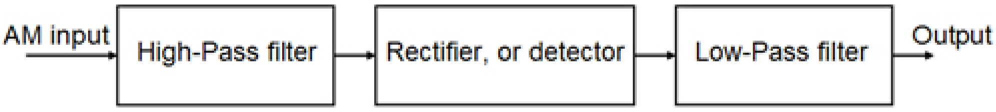

Figure 3.1 - The general block diagram of AM demodulator

They are numbers of methods that can be used to demodulate AM, but the simplest is a diode detector.
It operates by detecting the envelope of the incoming signal. It achieves this by simply rectifying the signal, then, the capacitor removes high frequencies.

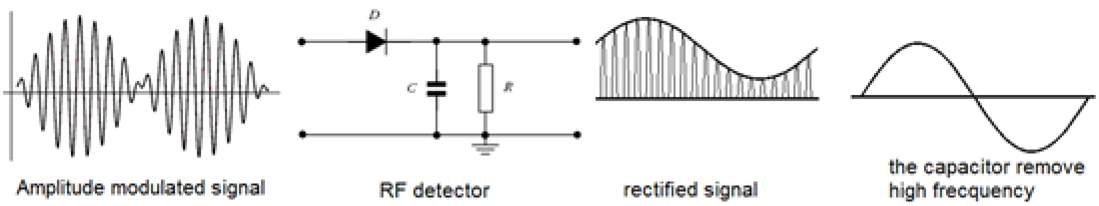

Figure 3.2 - The general principle of AM demodulation

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3.2 Practical implementation and techniques for recovering the carrier signal

They are two types of implementations:

- envelope detector (called also noncoherent detection) which uses diodes and RC filters, and it is suitable for DSB signals;
- product detector (called also coherent detector) which uses analog multipliers, and it is suitable for any AM.
- recovery of carrier by a band-pass filter

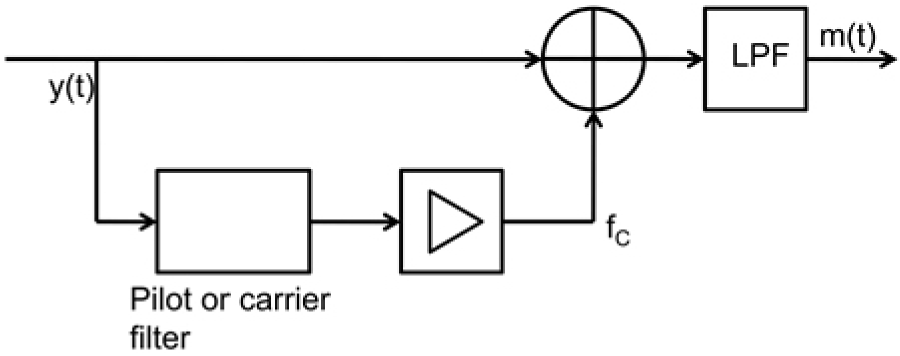

Figure 3.3 - The recovery technique with band pass filter block diagram

Where fC is recovered carrier signal.

- recovery of carrier by phase-locked loop;

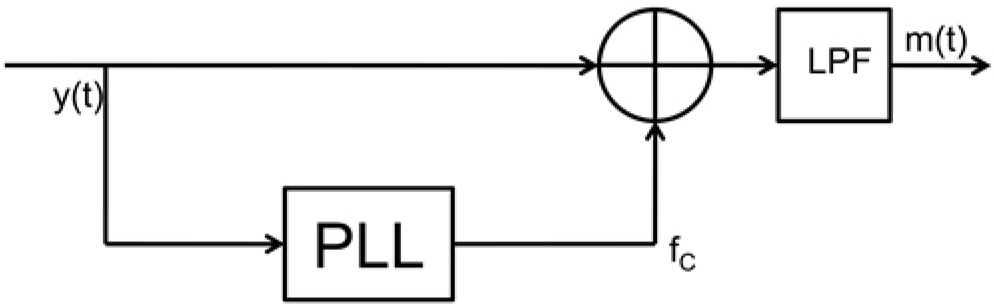

Figure 3.4 - Using PLL technique for carrier recovery

For both demodulation methods, a low pass filter is required in order to suppress the sum- frequency output.
The product detector, in any cases, is used because of high accuracy recovery of signal.

The main parameters as amplitude, frequency, or phase of a carrier could be changed in the modulation process.
The process of demodulation and the practical circuits for accomplishing it differ in each case. Anyway, the demodulators require the use a nonlinear device in order to recover the original modulating frequencies, because all these frequencies are not present in the modulated carrier and new frequencies cannot be produced by a linear device.

A simple high frequency semiconductor diode is frequently used to detect an amplitude-modulated (AM) carrier. A very simple filter consisting of capacitance and resistance is used to eliminate the carrier and other unnecessary frequencies. Another often used AM detector uses a multiplier circuit. A square-law detector is often used to demodulate single-sideband (SSB) signals. A multiplier circuit with both inputs tied together serves as a squaring circuit and may be used as a low-distortion demodulator for SSB signals.

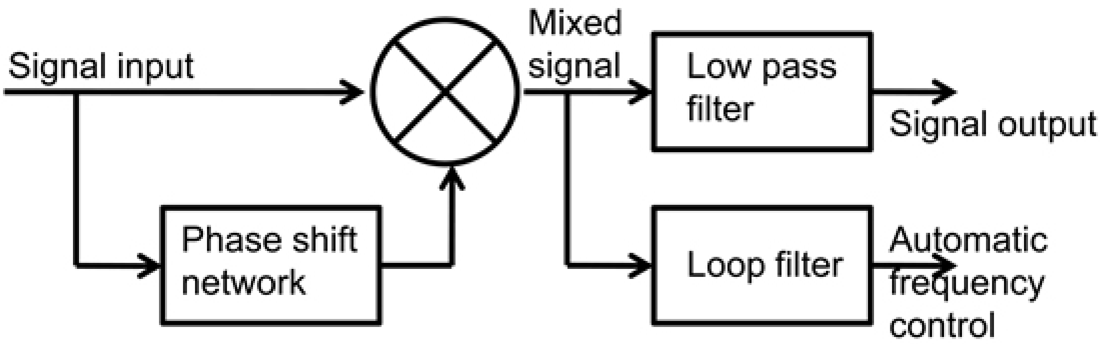

Figure 3.5 - Typical structure of modern AM demodulator

In many approaches, the amplitude modulation and demodulation may be implemented in the same device. For example, a multiplier or phase shift network performs both of these functions.
In addition, phase-locked loops incorporate all the basic circuits needed for the modulation and demodulation of AM signals.

Therefore, circuits have been designed in the way that will either modulate or demodulate FM, PM, and AM signals.
These circuits are known as modems and they are commonly used in modern communications systems.

3.3 SSB demodulation

Typically the SSB detector uses a mixer circuit to combine the SSB modulation and the LO signals. This circuit is often called a product detector because (like any RF mixer) the output is the product of the two inputs.

It is necessary to introduce the carrier using the LO on the same frequency relative to the SSB signal as the original carrier.
Any deviation from this will cause the pitch of the recovered audio to change.

Receiving SSB signal

There are several types of two way radio communication that it is possible to listen to legally. From the popularity point of view, it is necessary to know how to tune an SSB signal and receive the SSB signal in the best way to ensure that the best copy is obtained.
When receiving SSB it is necessary to have a basic understanding of how a receiver works.

Most radio receivers that will be used to receive SSB modulation will be of the super heterodyne type. The incoming signals are converted down to a fixed intermediate frequency.
It is at this stage where the LO signal is mixed with the incoming SSB signals.
It is necessary to set the LO to the correct frequency to receive the form of SSB, either LSB (low side band) or USB (upper side band), that is expected.

In practical implementation, the SSB Signal is easily detected using algebra method- Product Detector.

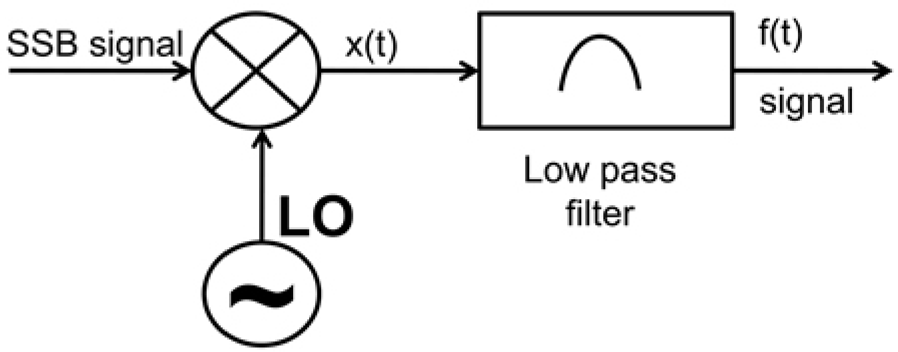

Figure 3.6 - The simple block diagram structure of SSB demodulation as product detector

The SSB signal is multiplied by LO, and is passing through LPF to eliminate de baseband signal. Next are presented some spectra of signals in different stages.

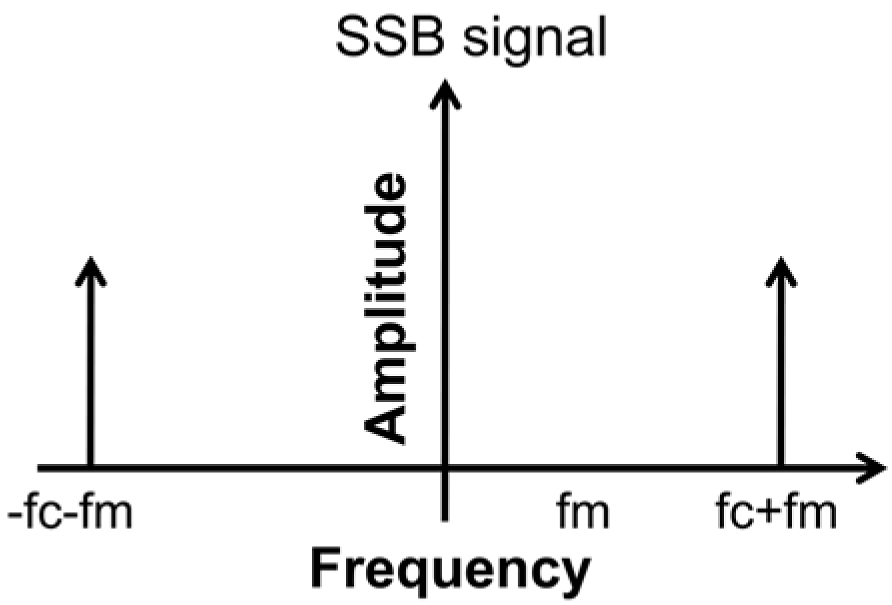

Figure 3.7 - Spectrum of SSB signal

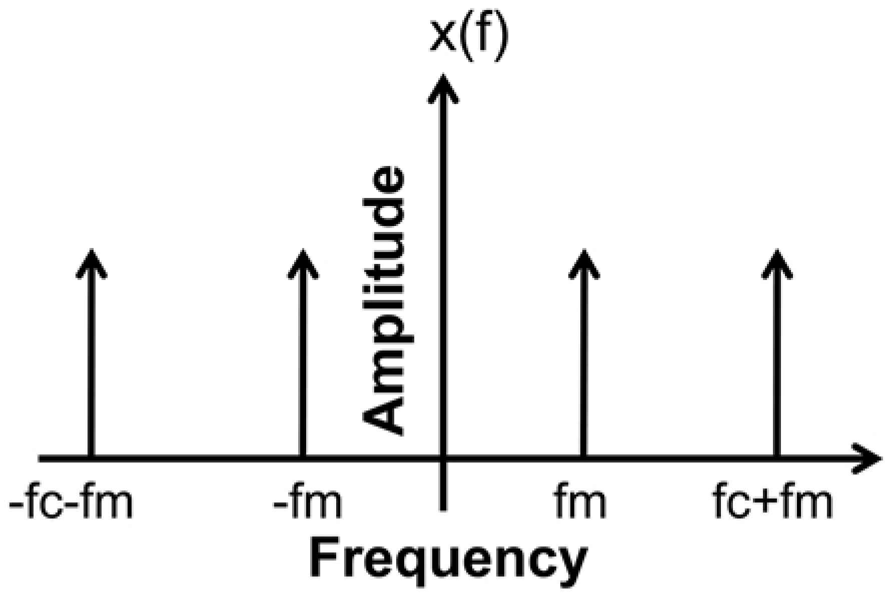

Figure 3.8 - Spectrum of x(f) that contains baseband frequency and high frequency produced by LO

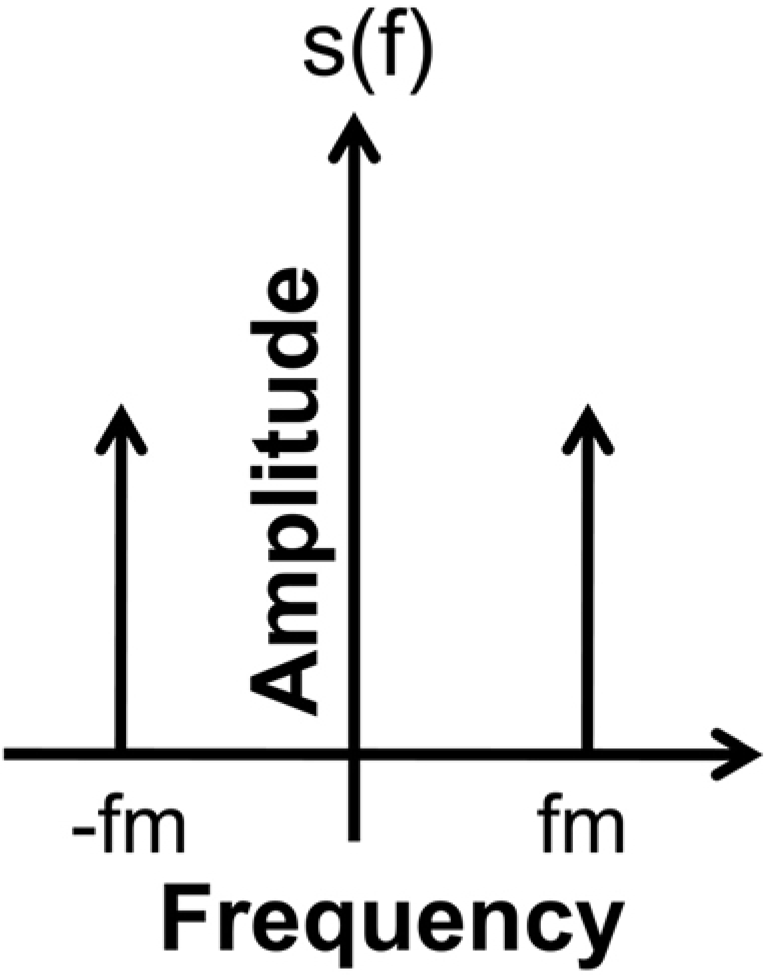

Figure 3.9 - The signal spectrum passed through LPF

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3.4 SSB types and advantages

There are some types of single sideband modulation:

• ***LSB***:	This is Lower Side Band. This form of SSB modulation is formed when the lower sideband only of the original signal is transmitted.
• ***USB***:	This is Upper Side Band. This form of SSB modulation is formed when the lower sideband only of the original signal is transmitted.
• ***DSB***:	This is Double Side Band and it is a form of modulation where an AM signal is taken and the carrier is removed to leave the two sidebands.
• ***SSB SC***:	This is Single Side Band with Suppressed Carrier. It is the form of SSB modulation where the carrier is removed completely as opposed to SSB reduced carrier where some of the carrier is left.
• ***VSB***:	This is Vestigial Side Band. It is a form is signal where one sideband is completely present, and the other sideband that has been only partly cut off or suppressed. It is widely used for analogue television transmissions.
• ***SSB reduced carrier***:	In this variant of SSB modulation one sideband is present along with a small amount of the carrier.

SSB advantages

It has some several advantages for two way radio communication:

1. This enables a 50% reduction in transmitter power level for the same level of information carrying signal.
2. As only one sideband is transmitted there is a further reduction in transmitter power.
3. As only one sideband is transmitted the receiver bandwidth can be reduced by half.

3.5 Automatic gain control (AGC)

The gain control is a method for adjusting the receiver sensitivity for the best reception of signals of widely varying amplitudes.
The complex form of automatic gain control (AGC), or instantaneous automatic gain control (IAGC) is used during normal operation.

The simplest type of AGC adjusts the RF amplifier bias (and its gain) according to the average level of the received signal. Generally, with AGC, the gain is controlled by the largest received signals. When several signals are being received simultaneously, the weakest signal may be of greatest interest.

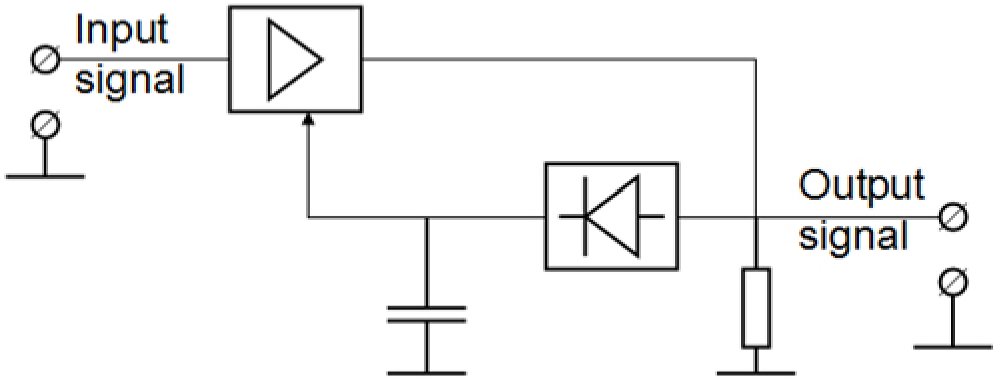

Figure 3.10 - This is the block diagram of automatic gain control circuit

The basic AGC circuit is essentially a wide-band, dc amplifier. It instantaneously controls the gain of the RF amplifier as the radio device return signal changes in amplitude.
The effect of AGC is to allow full amplification of weak signals and to decrease the amplification of strong signals.

Technically, the range of AGC is limited, by the number of RF stages in which the gain is controlled.

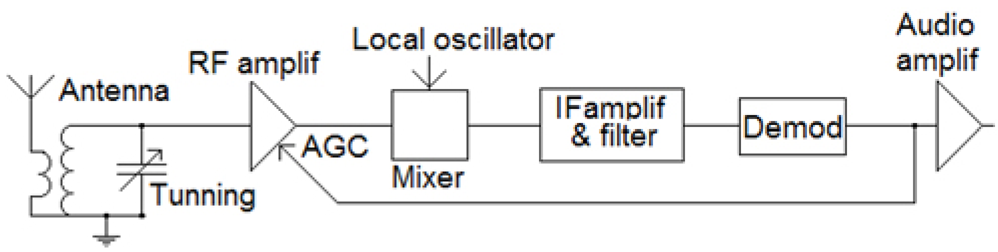

Figure 3.11 - This is typical receiver block diagram with integrated AGC

When only one RF stage is controlled, the range of AGC is technically limited to approximately 20 dB. When more than one RF stage is controlled, the AGC range can be increased to approximately 40 dB.

A lot of the parameters of the AGC loop depend on the type of modulation used inside the system. Generally speaking, if any kind of amplitude modulation (AM) is present, the AGC should not respond to any change in amplitude modulation or distortion will occur.

This is the reason, why the bandwidth of the AGC must be limited to a value lower than the lowest modulating frequency.

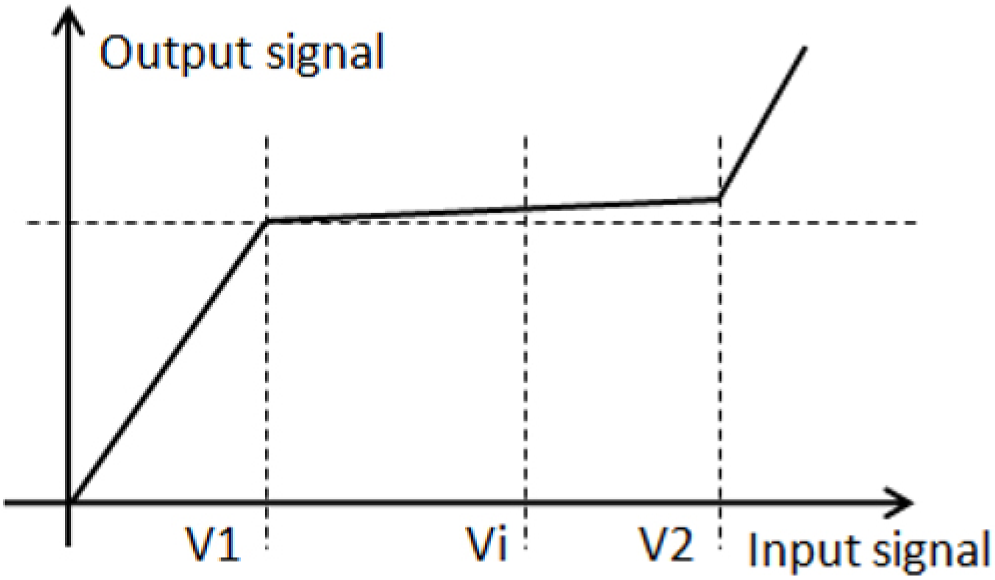

Figure 3.12 - The behavior of output signal in accordance with input variation

The upper figure (the designed AGC system) shows a linear relationship as long as input and output quantities are expressed in decibels.
From the last expression it is easy to see that the behavior of the system is determined by the factor of the AGC and the filter F(s).

F(s) is usually a low pass filter, since the bandwidth of the loop must be limited to avoid stability problems and to ensure that the AGC does not respond to any amplitude modulation that could be present in the input signal.

3.6 Single Side Band (SSB) demodulation with ACG

In electronic transmission design, a sideband is the portion (one side of spectra) of a modulated carrier wave that is either above or below the basic (baseband) signal.
The portion above the baseband signal is the upper sideband; the portion below is the lower sideband.

In regular amplitude modulation (AM) transmission, both sidebands are used to carry a message. In some forms of transmission, one sideband is removed (single-sideband transmission) or a portion of one sideband is removed.

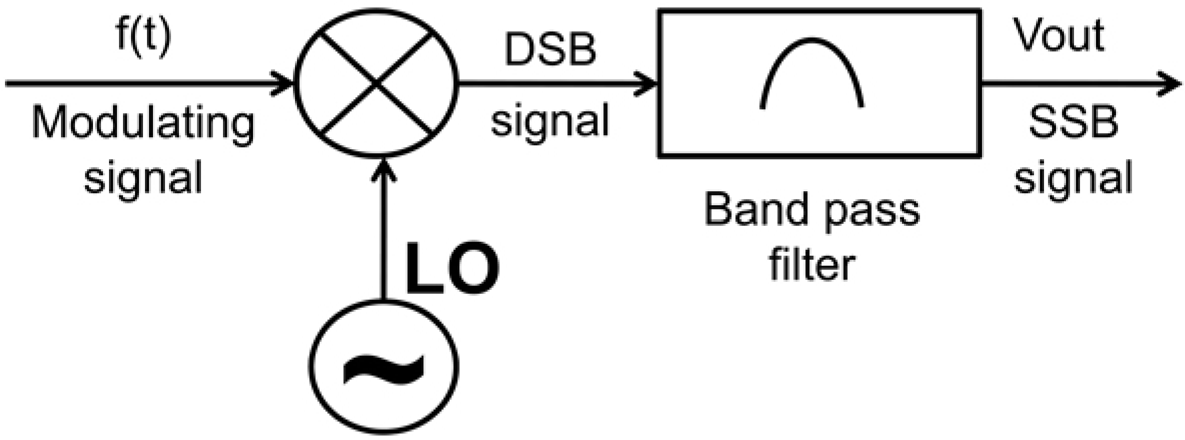

Figure3.13 - The simple block diagram structure of SSB

The SSB or Single Side Band is a type of AM without the carrier and one sideband. By opposition, DSB or Dual Side Band is AM with the carrier suppressed, but both upper and lower sidebands are used. By removing some of the components of the ordinary AM signal it is possible to significantly improve its efficiency.

It is possible to see how an AM signal can be improved by looking at the spectrum of the signal. When a carrier is modulated with an audio signal, for example a tone of 1 kHz, then two smaller signals is seen at frequencies 1 kHz above and below the main carrier.

If the tones are replaced with audio like that encountered with speech of music, these comprise many different frequencies and an audio spectrum with frequencies over a band of frequencies is seen. When modulated onto the carrier, these spectra are seen above and below the carrier.

The amplitude modulation is very inefficient from two points. The first is that it occupies twice the bandwidth of the maximum audio frequency, and the second is that it is inefficient in terms of the power used. The carrier is a signal and in itself carries no information, only providing a reference for the demodulation process.

The single side band modulation improves the efficiency of the transmission by removing some unnecessary elements. In the first instance, the carrier is removed - it can be re-introduced in the receiver, and secondly one sideband is removed - both sidebands are mirror images of one another and the carry the same information.

From the receiving point of view, DSB is compatible with SSB receivers; the receiver is only rejecting the unwanted sideband. When there is a use of both sidebands, they are used to carry two separate channels of information, and it is called ISB, or independent side band.

Today, the DSB modulation is seldom used today. It is still used in FM stereo transmission for the 38 kHz audio channel difference (L-R) subcarrier.
The easiest method for generating SSB is to modulate a carrier, and then filter the output to remove the unwanted sideband and carrier frequencies. This method is losing about two-thirds of the generated power into a filter. An alternative method for generating SSB is to use the phasing method.

In this case, two modulators (mixers) produce the desired sideband while suppressing the unwanted carrier and other sideband.

Today, most of SSB exciters first generate a DSB signal which is then processed into SSB. An AM signal with one sideband partially suppressed is called VSB or vestigial side band.
This is used in television transmission to reduce bandwidth while still allowing AM detection schemes to be used.

A SSB signal can be sent with a carrier to reduce occupied bandwidth, and this is called CSSB or compatible SSB. This approach has little advantage over AM other than the reduction in bandwidth and selective fading effects.

The spectrum of signals is shown next:

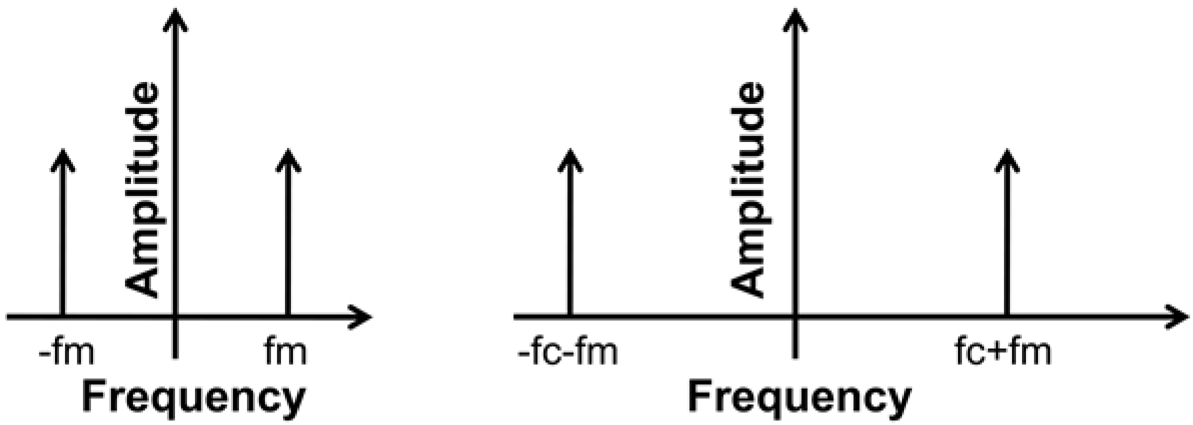

Figure3.14 - The spectrum of baseband and USSB signal

Lesson N.4: FM and PM - Angular modulation and demodulation

Objectives: **To know:**

- **the characteristics of the frequency modulation and demodulation**
- **the characteristics of the phase modulation and demodulation**
- **the characteristics and the parameters of the angular modulation and demodulation**

Requisites:

- **Lesson 1 of this theoretical guide**

Arguments:

- **Definition and theory of frequency modulation**
- **Difference between FM and PM**
- **Frequency Shift Keying**
- **Power and bandwidth of the signal**
- **Performance of PLL FM demodulator**

4.1 Definition of frequency modulation

The frequency modulation (FM) process is a form of modulation in which changes in the frequency of the carrier wave correspond directly with changes in the baseband signal.
This process is also considered an analog form of modulation, because the baseband signal is typically an analog waveform without discrete.

In order to understand FM, an important concept is that of frequency deviation.
The amount of frequency deviation a signal experiences is a measure of the change in transmitter output frequency from the rest frequency of the transmitter.

The rest frequency of a transmitter is defined as the output frequency with no modulating signal applied. For a transmitter with linear modulation characteristics, the frequency deviation of the carrier is directly proportional to the amplitude of the applied modulating signal.

It is defined as:

$$k_f = frequency\_deviation / V = [kHz / V]$$

The frequency modulation (FM) process is a form of modulation in which changes in the frequency of the carrier wave correspond directly with changes in the baseband signal.
This process is also considered an analog form of modulation, because the baseband signal is typically an analog waveform without discrete.

4.2 Theory of frequency modulation

From the mathematical point of view, we will represent this by describing the steps required to modulate the frequency of a sinusoidal carrier.

$$x_{FM}(t) = A(t)\cos(\psi t))$$

Where, ψ(t) is the phase of the signal xMF (t). We should make a parallel analyze with amplitude modulation. Simplifying AM, phase modulation could be written as:

$$\psi(t) = 2\pi f_p t + \varphi_p$$

In the case of amplitude modulation, the amplitude is time variant A(t). In the case of frequency modulation, A(t)= constant, and the signal transfer is described by ψ(t), as a phase deviation of the carrier. So, by integration the instantaneous pulsation of the transferred signal into carrier, the general mathematical model of the frequency modulation is:

$$x_{FM}(t) = A_C \cos\varphi_{FM}(t) = A_C \cos(\int_0^t 2\pi f_i(\tau)d\tau) = A_C \cos(2\pi f_C t + 2\pi k_F \int_0^t f_i(\tau)d\tau$$

by simplifying the equation,

$$x_{FM}(t) = A_C \{\cos(2\pi f_C t)\cos[2\pi k_F y(t)] - \sin(2\pi f_C t)\sin[2\pi k_F$$

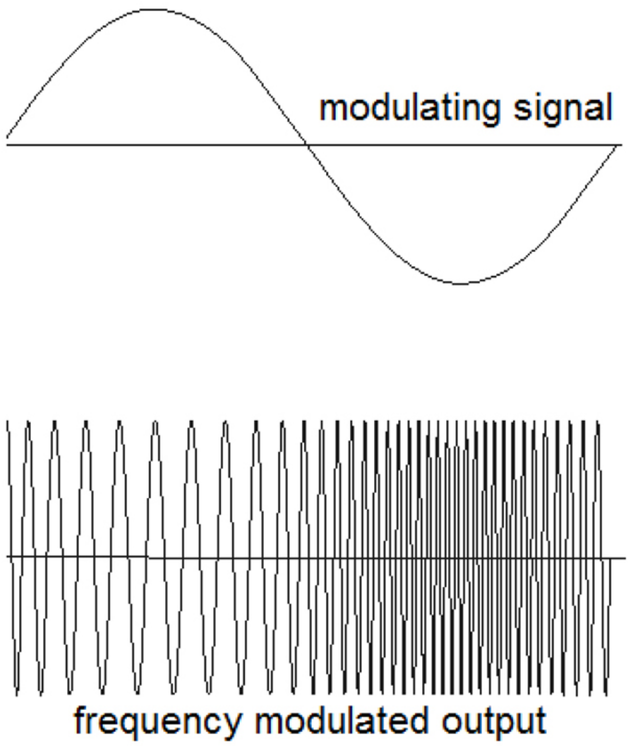

Figure 4.1 - This figure shows the a frequency modulate with a constant signal

4.3 FM versus PM

The question is what the difference between frequency modulation and phase modulation.
The general answer is that is not any difference, as either one will change the sine wave's frequency. Closer look reveals that there are some differences between the two forms of angle modulation. Now, we will not investigate deeper such differences. Practical speaking, it is possible to obtain FM from PM, but most today's FM systems will not generate FM by such method.

If we compare the two forms of modulation in the time domain, we will accept that the FM wave and PM wave appear quite similar; however, their timing appears to be out of synchronization. The FM wave has its maximum frequency deviation during the peaks of the input signal, while the PM wave has its maximum frequency deviation during zero crossings of the input signal. Without showing the phase relationship of the input wave to the modulated wave, it would be impossible to tell the difference between the two forms of angle modulation if one simply looked at the resulting modulated waveform.

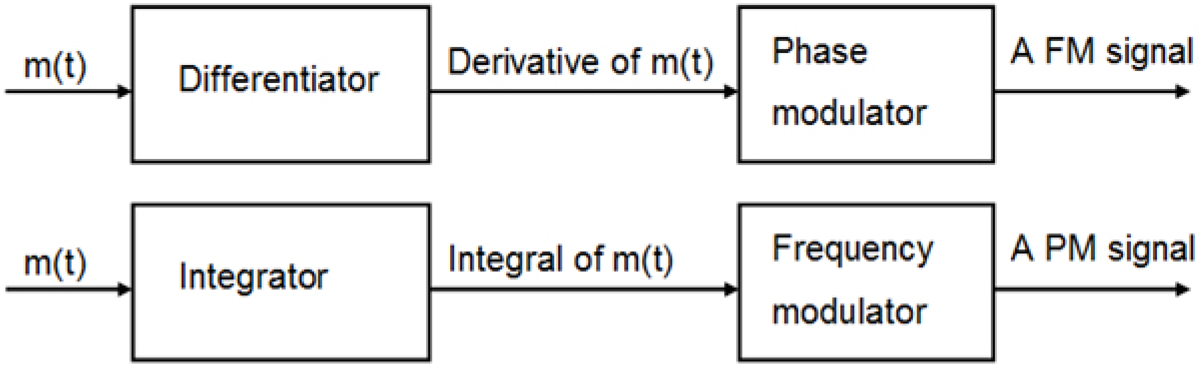

Figure 4.2 - The difference between FM and PM signals

Both FM and PM modulated signals are conceptually identical, the only difference is in fact that in the first case the phase is modulated directly by the message signal, and in case of FM signals, the message is first integrated. So, by using phase modulator we could create FM signal by integrating the message first.

4.4 Frequency Shift Keying

The binary version of FM is called Frequency Shift keying (FSK). The modulated signal could be fixed in a very simply way by establishing two frequencies for 1 logic and 0 logic.

$$x_{FM1}(t) = A_c \cos(2\pi f_1 t)$$, in response to 1 logic

$$x_{FM2}(t) = A_c \cos(2\pi f_2 t)$$, in response to 0 logic

Or

$$x_{FM1}(t) = A_c \cos[2\pi (f_c - \Delta f)t]$$, in response to 1 logic

$$x_{FM1}(t) = A_c \cos[2\pi (f_c + \Delta f)t]$$, in response to 0 logic

And Δf is called frequency deviation.

Index of modulation

The basic of model evaluation is related to the group kFy(t) compared with π/2:

- If max(kFy(t))<< π/2, the spectrum of modulated carrier is insignificant, and we call it tiny range frequency modulation;
- If max(kFy(t))>> π/2, the spectrum of modulated carrier is extremely large, and we call it large range frequency modulation;

As periodical functions $\cos[\beta \sin(2\pi f_c t)], \sin[\beta \sin(2\pi f_m t)]$, are periodical functions r to $2\pi / 2\pi f_m$ we will consider Fourier transformers terms:

$$x_{FM}(t) = A_C \sum_{-\inf}^{\inf} J_n(\beta) \cos(2\pi f_C t + n 2\pi k_m)t$$

Where, Jn(β) is p-order Bessel function.

Based on the value of modulation index (β), we can describe large or small range modulation. In most of the cases, there is accepted an approximation of MF signal:

$$x_{FM}(t) = A_C J_0(\beta)\cos(2\pi f_C t) + A_C J_1(\beta)\cos(2\pi f_C + 2\pi f_m)t - A_C J_1(\beta)\cos(2\pi f_C - 2\pi f_m)t$$

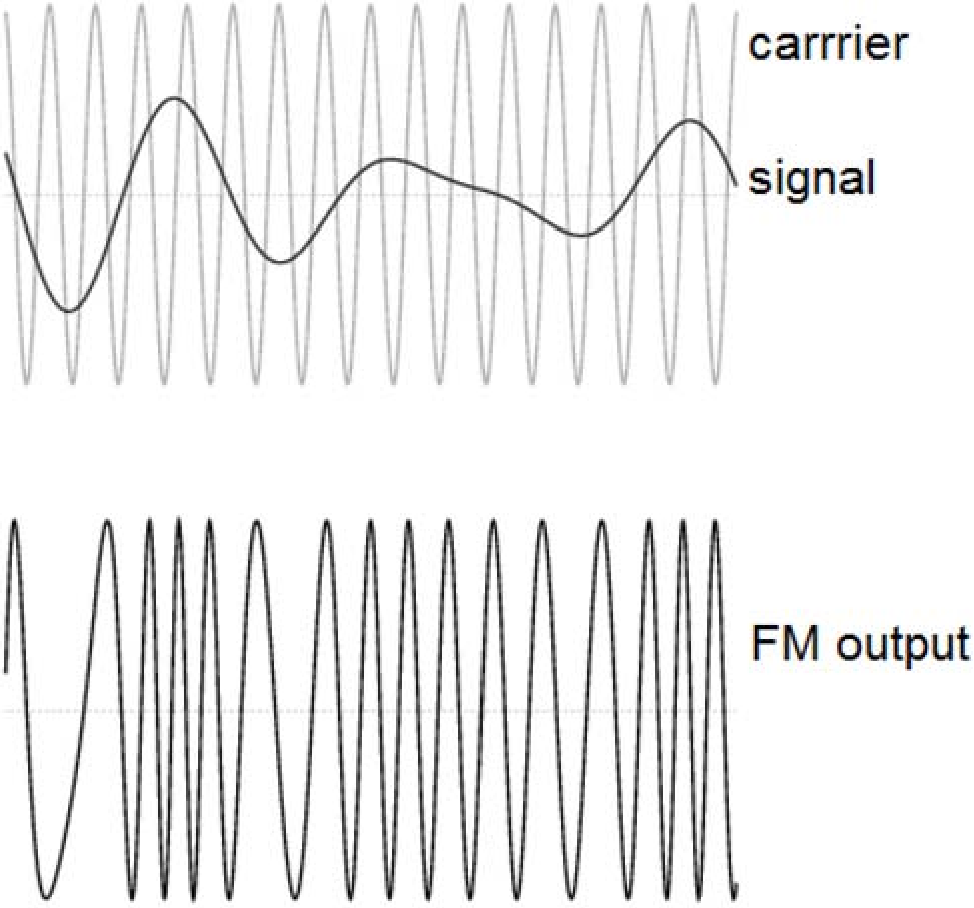

Figure 4.3 - This figure shows the a frequency modulate with a variable signal

For this case we can extract some global characteristics of frequency modulated signal.

4.5 Power of the signal and bandwidth

Average of power signal is calculated P_m

$$P_m = \frac{1}{T}\int_0^T x_{FM}{}^2(t)dt$$

Following the same procedures for simplification, with $\sum_{-\inf}^{\inf} J_K{}^2(\beta) = 1$, th

based only on carrier signal:

$$P_m = \frac{A_C^2}{2}$$

Signal bandwidth
Thinking that modulation index is in fact the ratio between frequency domain and modulation frequency, a particular case is when β>>1, then the bandwidth (B) is calculated as

$$B = 2\beta 2\pi f_m = 2\Delta\varpi_m$$

Back to the general case, it can be seen that the spectrum consists of the carrier plus an almost infinite number of sidebands spreading out on either side of the carrier at integral frequencies of the modulating frequency.

The relative levels of the sidebands can be read from a table of Bessel functions, or computer calculated.

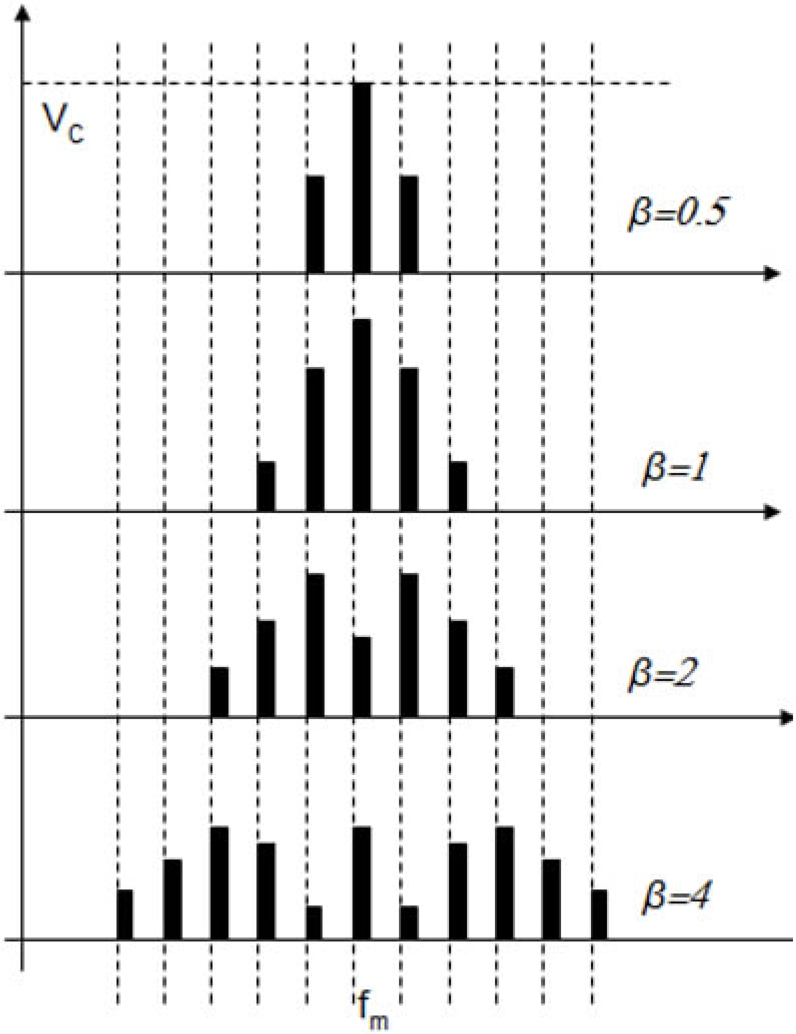

Figure4.4 - The spectra of FM with different modulation index

The literature presents the spectra of frequency-modulated signals with various values of modulation index for a constant modulation frequency.
It can be seen that for small values of the modulation index β (e.g. β = 0.5), the signal appears to consist of the carrier and two sidebands.
As the modulation index increases, the number of sidebands increases and the level of the carrier can be seen to decrease for these values.

4.6 Particular case- short exercise

There are imagined some parameters to be shown, to be evaluated.

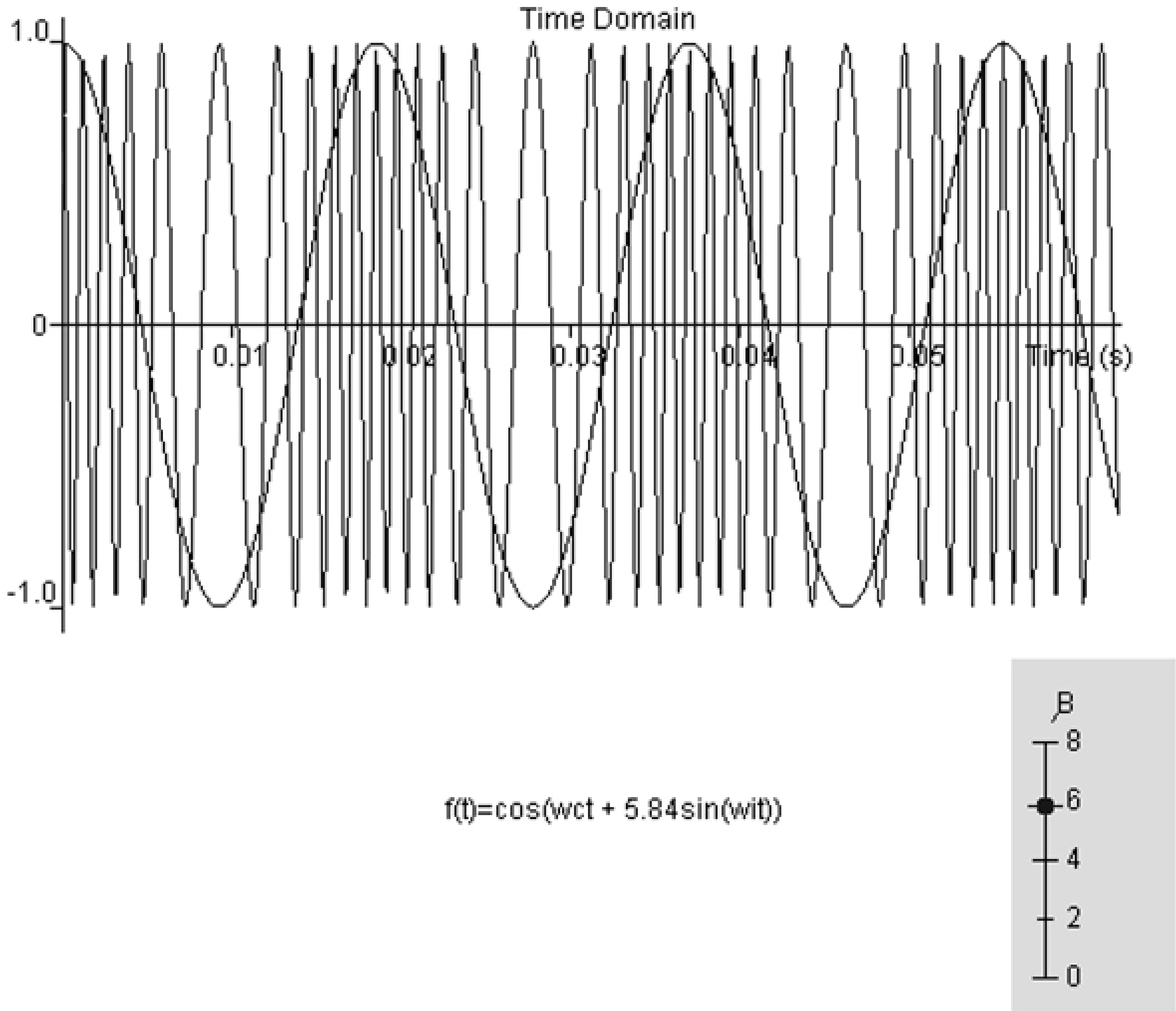

Figure 4.5 - Basic experiment panel

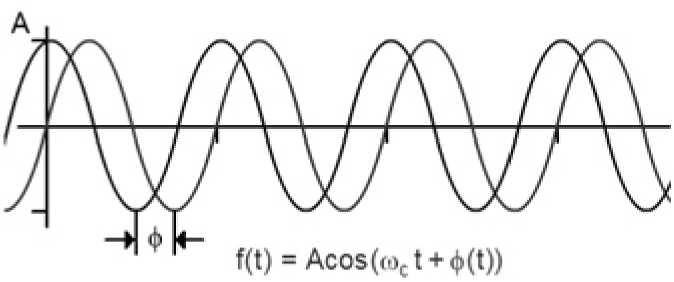

Figure 4.6 - Taking into account the phase of modulated signals

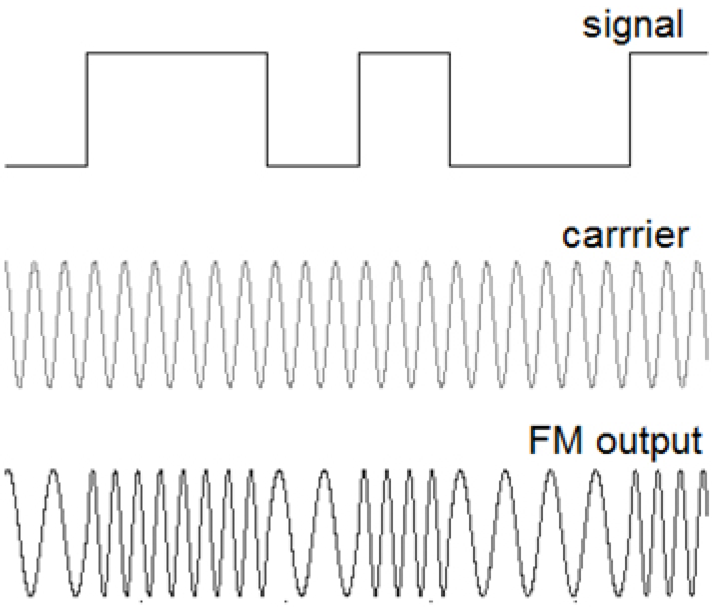

Figure 4.7 - This is a particular case of FM- signal shape is digital signal (only two values 0, 1)

4.7 PLL demodulator

FM modulated signal must be received. A FM receiver must be sensitive to the frequency variations of the incoming signals.

In order to convert the frequency variations into voltage variations, the demodulator must be frequency dependent. As we expect, the ideal response is a perfectly linear voltage to frequency characteristic. We expect that outside the bandwidth of the system, the response falls.

On the other hand, it is easy to see that the frequency variations of the signal are converted into voltage variations which can be amplified by an audio amplifier before being passed into headphones, a loudspeaker, or passed into other electronic circuitry for the appropriate processing.

The phase locked loop (PLL) is a non- linear system. When we put in into an FM demodulator, to analyze its performances is not an easy issue. A simple block diagram is illustrating its structure.

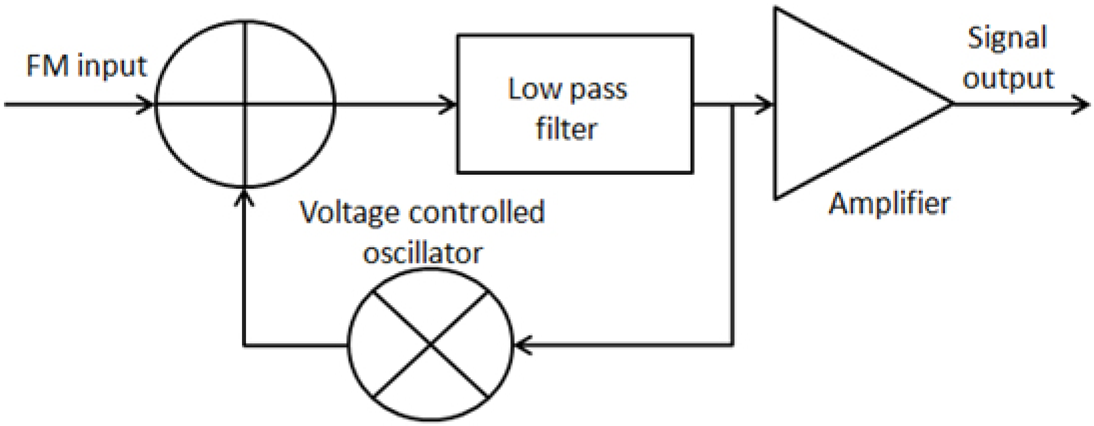

Figure 4.8 - Block diagram of PLL

As an FM demodulator, the output is taken from the LPF, as shown. Eventually, a signal amplifier should be required in order to use it. It is rather simple matter to describe the principle of operation of the PLL as a demodulator.

To analyze its performances, it is complicated by the fact that its performance is described by non-linear equations, and the solution to which is generally a matter of approximation. Let’s consider that in the block diagram we have an open loop form.

If we suppose there is an unmodulated carrier at the input. If the VCO was tuned precisely to the frequency of the incoming carrier, fc say, then the output would be a DC voltage, of magnitude depending on the phase difference between itself and the incoming carrier. If we consider two angles within the 360 range, the output would be precisely zero volts DC.
Then it is started to drift slowly off in frequency.

Depending upon which way it drifted, the output voltage would be a slowly varying AC, which if slow enough looks like a varying amplitude DC. The sign of this DC voltage would depend upon the direction of drift.

Let’s suppose now that the loop is closed. If the sign of the slowly varying DC voltage, now a VCO control voltage, is so arranged that it is in the direction to urge the VCO back to the incoming carrier frequency fc, then the VCO would be encouraged to ‘lock on’ to the incoming carrier.

4.8 PLL FM demodulator performance

The PLL FM demodulator is normally considered a relatively high performance form of FM demodulator or detector.
The key advantages of PLL FM demodulator are:

-	linearity range:	the linearity of the PLL FM demodulator is controlled by the VCO voltage to frequ within the PLL. As the frequency deviation of the incoming signal normally only s PLL bandwidth, and the characteristic of the VCO can be made relatively linear, th locked loop demodulators are normally very low.
-	manufacturing costs:	usually, the PLL FM demodulator is integrated into integrated circuit technology. Only a few external components are required, mostly related to resonant circuit of This situation makes the PLL FM as an attractive implementation for modern appl

Typical application
FM stereo broadcasting performs the multiplexing two signals and further combines them into a complex baseband signal that modulates the FM carrier.
The output of adder is sum of the two signals, or L+R signal, and the output of the other adder is difference between the two signals, L-R.

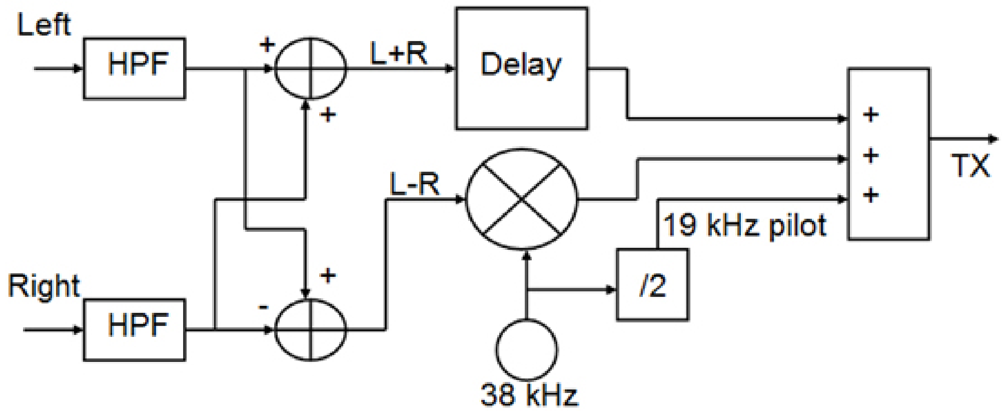

Figure 4.9 - This is a typical block diagram of FM stereo broadcasting.

For receiving FM stereo signals, typical block diagram is shown next

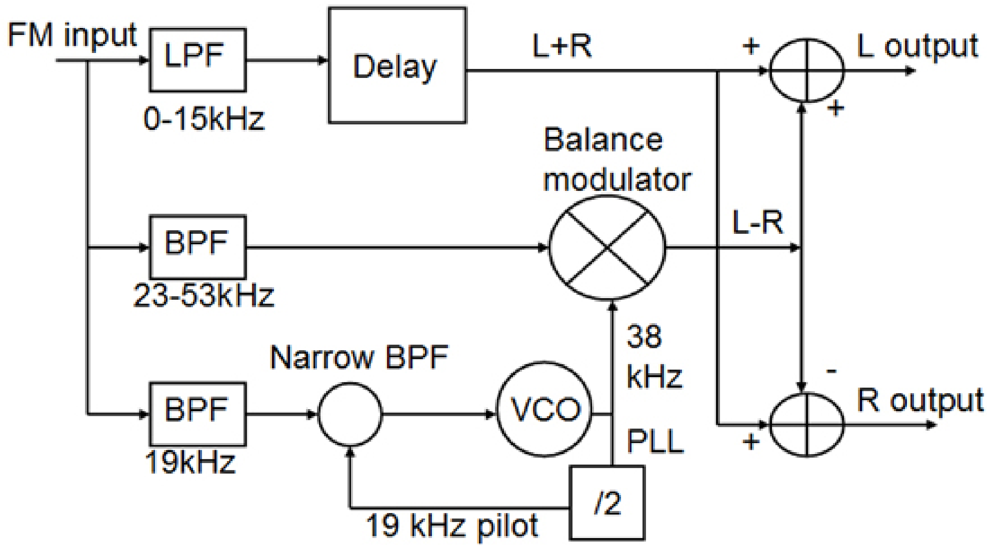

Figure 4.10 - Tipical diagram of stereo receiver

The FM input composite signal is separated by two BPFs and one LPF into three separate signals. The L + R signal, which occupies the 0- to 15-kHz range, does not undergo any additional signal processing, with the exception of the addition of a small time delay. The 19-kHz pilot tone is recovered by a narrow BPF centered at 19 kHz. This signal undergoes a frequency doubling to 38 kHz, and is then applied to a balanced modulator.

The 23-53 kHz filtered signal is recovered by another BPF centered at 38 kHz, and it is also applied to the previously mentioned balanced modulator. The balanced modulator output consists of the original left-minus-right (L – R) signal. At this point the L + R and L – R signals are applied to adder circuits (sometimes referred to as a matrix) that yield the separate left and right signals. The signals are then fed to identical audio amplifiers.

Lession N.5: Filters

Objectives:

To know:

- **the characteristics of the passive filters**
- **the characteristics of the active filters**
- **the different types filters and their order**

Requisites:

- **Lesson 1 of this Theoretical guide**

Arguments:

- **Passive filetrs**
- **Low Pass and High Pass filters**
- **band Pass filter**
- **Active filters**
- **High-order and N^{th} order filters**

5.1 Filters

An electrical filter is a circuit that can be designed to modify, reshape or reject all input frequencies of an electrical signal and accept or pass only those signals designed to be wanted by the circuits' designer.

For working in low frequency applications (up to 100kHz), passive filters are usually made from simple RC (Resistor-Capacitor), while higher frequency filters (above 100kHz) ask for RLC (Resistor-Inductor-Capacitor) components.

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5.2 Low pass filter principle

The simple First-order passive filters (1st order) can be built by connecting together a single resistor and a single capacitor in series across an input signal, (Vin) with the output signal, (Vout) taken from the ports of these two components.

Depending on which way around we connect the resistor and the capacitor with regards to the output signal determines the type of filter construction resulting in either a Low Pass Filter or a High Pass Filter.

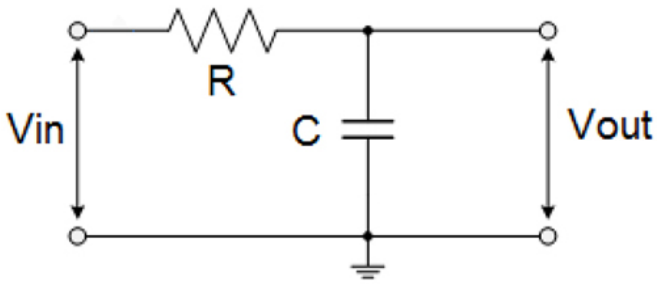

Figure 5.1 - Typical low pass RC filter

The main function of any filter is to allow signals of a given band of frequencies to pass unaltered while attenuating the others that are not wanted. In this case, there are two passive components within this type of filter design the output signal has a smaller amplitude than its corresponding input signal, therefore passive RC filters attenuate the signal and have a gain lower than one.

For the general calculus of the filters, the general formula is $V_{out} = V_{in}\frac{X_C}{Z}$, where, for this case, $Z = \sqrt{R^2 + X_C^2}$.

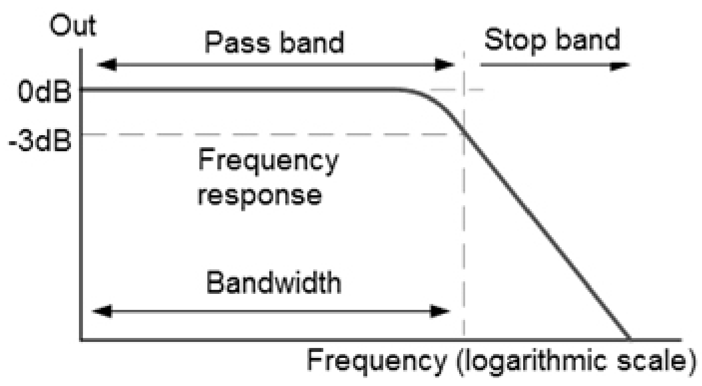

Figure 5.2 - This is frequency response of low pass filter (Bode diagram)

The low pass filter can transmit a DC component to the output. One important point is cut-off frequency. For each circuit it is calculated in accordance with circuit components.

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5.3 High pass filter principle

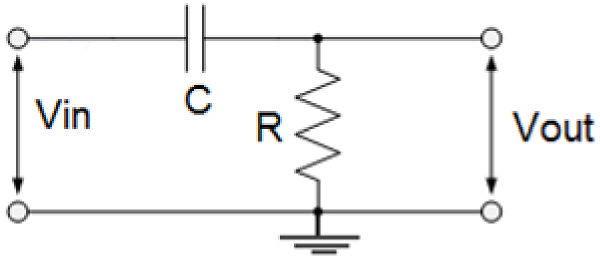

Figure 5.3 - Typical high pass RC filter

In this circuit, the reactance of the capacitor is very high at low frequencies so the capacitor acts like an open circuit until the cut-off frequency point of input signal (fc) is reached.
Above the cut-off frequency point the reactance of the capacitor has decreased sufficiently and acts more like a short circuit allowing the entire input signal to pass directly to the output.

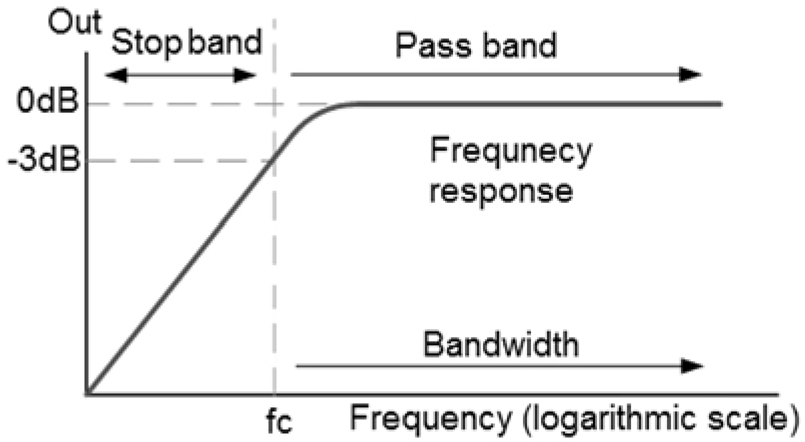

Figure 5.4 - This is frequency response of the high pass filter (Bode diagram)

The high pass filter cannot transmit DC component to output.

5.4 Band pass filter principle

The band pass filter is known generally as a Second Order Filter, (2nd-Order) because it has "two" reactive component within its circuit design. One capacitor in the low pass circuit and another capacitor in the high pass circuit.

Following the same logic, by connecting together a single low pass filter circuit with a high pass filter circuit, in series, it will produce another type of passive RC filter that passes a selected range or band of frequencies, or wide while attenuating all those outside of this range.

Band pass filters only pass signals within a certain "band" or "spread" of frequencies without altering the input signal or introducing extra noise.
This range of frequencies is commonly known as the Bandwidth and is defined as the frequency range between two specified frequency cut-off points (f_c), that are 3dB below the maximum centre or resonant peak.

So, we can simply define the term "bandwidth" as the difference between the lower cut-off frequency (f_{clower}) and the superior cut-off frequency (f_{cupper}) points.

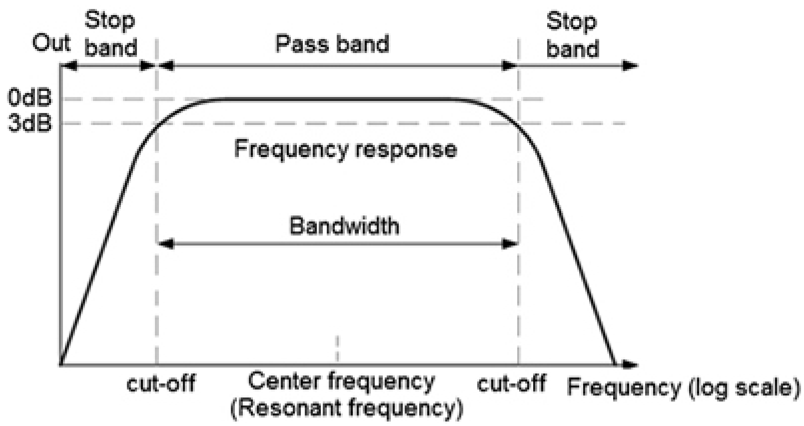

Figure 5.5 - This is frequency response of the band pass filter (Bode diagram)

The cut-off frequency points for a band pass filter (upper and lower points) can be found using the same formula as that of the low and high pass filters.

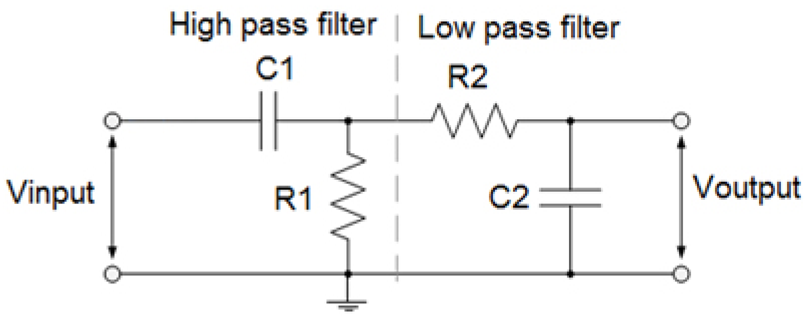

Figure 5.6 - Typical band pass RC filter

There is a simple method to calculate the "resonant" or "centre frequency" point of the bandpass filter were the output gain is at its maximum or peak value.

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5.5 Active filters

We have started designing passive filters, understanding that passive RC filters attenuate the signal and have a gain lower than one.
For this reason, let's imagine an amplifier of the output signals of RC filters.

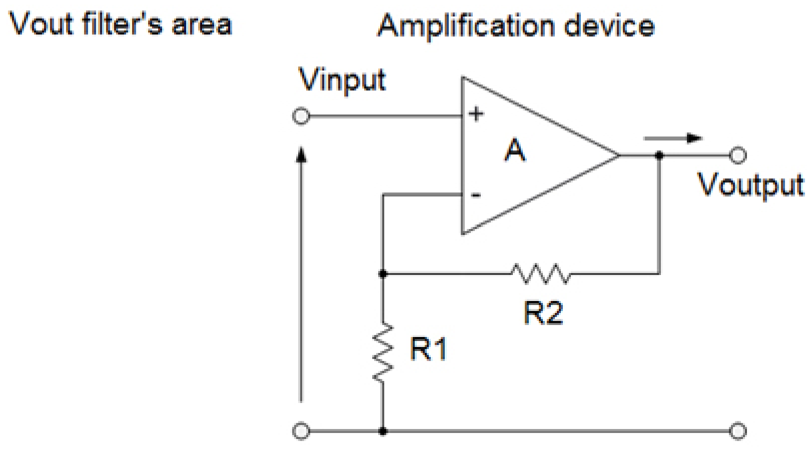

Figure 5.7 - Typical amplification module

The amplitude of the input signal (the output of passive filters) is increased by the gain of the amplifier and for a non-inverting amplifier the value of the pass band voltage gain is given as:

$$A = 1 + \frac{R_2}{R_1}$$

For a non-inverting amplifier circuit, the magnitude of the voltage gain for the filter is given as a function of the feedback resistor (R2) divided by its corresponding input resistor (R1) value.
Following this idea, an active band pass filter is built like:

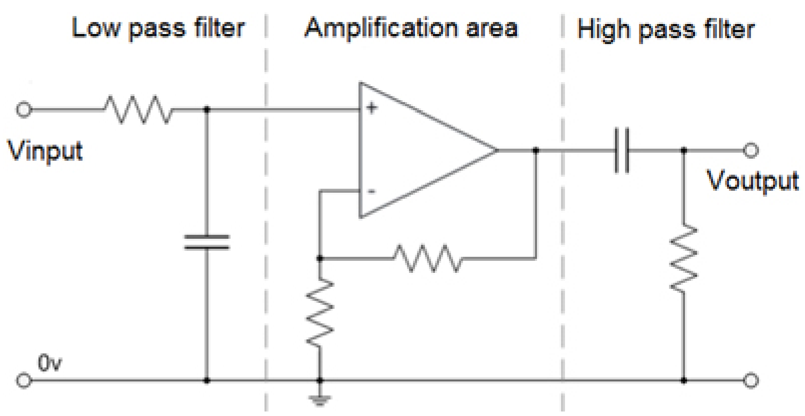

Figure 5.8 - Typical active band pass filter schematic diagram

The quality of filter is described by the quality factor Q,, that has the mathematical model:

$$Q = \frac{\mathrm{Re}\,sonant\_frequency}{Badnwidth}$$

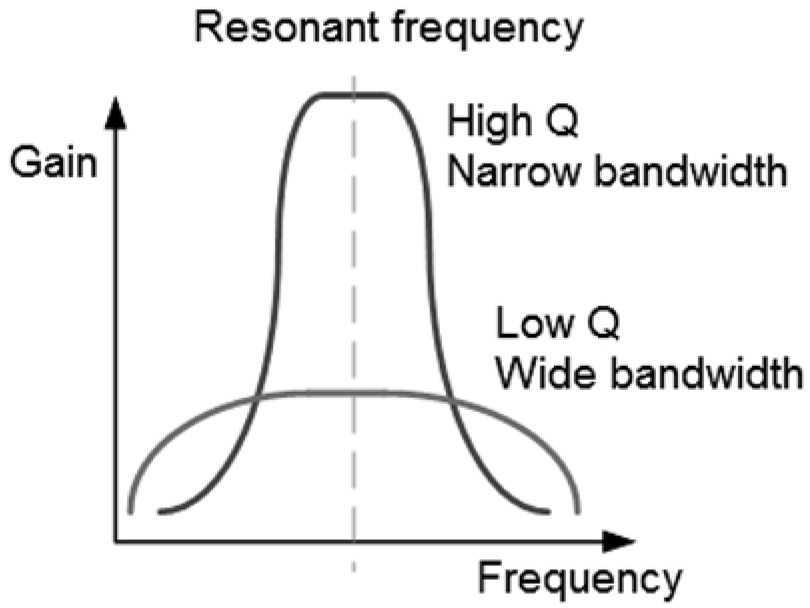

Figure 5.9 - Figuring the bandwidth in accordance with quality factor

5.6 High-order" or "Nth - Order" filters principle

In many applications that use filters to shape the frequency spectrum of a signal such as in communications or control systems, for a simple first-order filter may be too long or wide and so active filters designed with more than one "order" are required. These types of filters are commonly known as "High-order" or "Nth-Order" filters.

The level of them complexity, or filter type, is defined by the filters "order", and which is dependant upon the number of reactive components such as capacitors or inductors within its design.
We also know from literature that the rate of roll-off and therefore the width of the transition band depend upon the order number of the filter and that for a simple 1st-order filter it has a standard roll-off rate of 20dB/decade or 6dB/octave. Then, for a filter that has an "Nth" number order, it will have a subsequent roll-off rate of 20n dB/decade or 6n dB/octave.

So a first-order filter has a roll-off rate of 20dB/decade (6dB/octave), a second-order filter has a roll-off rate of 40dB/decade (12dB/octave), and a fourth-order filter has a roll-off rate of 80dB/decade (24dB/octave), just for exercising the roll-off rate. Usually, high-order filters, such as third, fourth, and fifth-order are formed by cascading single first-order and second-order filters. For example, two second-order low pass filters can be cascaded together to produce a fourth-order low pass filter, and so on.

Theoretically, there is no limit to the order of the filter that can be formed, as the order increases so does its size and cost, also its accuracy declines.
Practically, as with the first and second-order filters, the third and fourth-order high pass filters are formed by simply interchanging the positions of the frequency determining components (resistors and capacitors) in the equivalent low pass filter.

From the designing point of view, the frequency response of the high-order filter approximation function is also often referred to as "maximally flat" (not ripples!) because the pass band is as flat as possible at 0Hz (DC) with no ripples until the cut-off frequency at -3dB, and then rolls-off down to zero in the stop band.
This is because it has a Quality Factor, Q of 0.707.

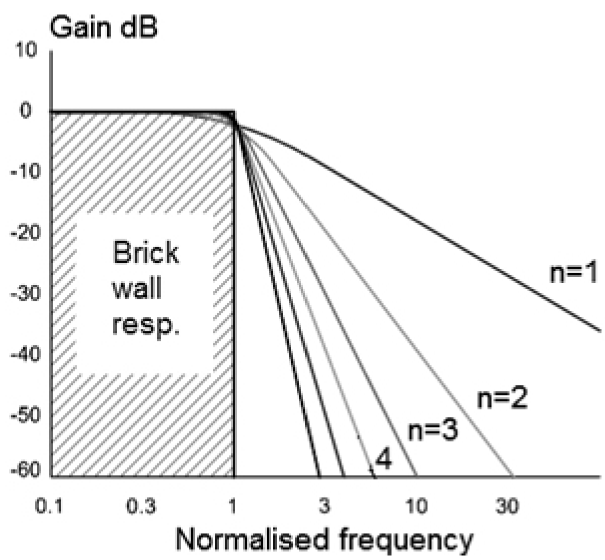

Figure 5.10 - This is frequency response of the high-order filter (Bode diagram)

One main disadvantage of the high-order filter is that it achieves this passband flatness at the expense of a wide transition band as the filter changes from the passband to the stopband.
It also has poor phase characteristics as well.
The ideal frequency response, referred to as a "brick wall" filter and the standard high-order approximations, for different filter orders are shown below.

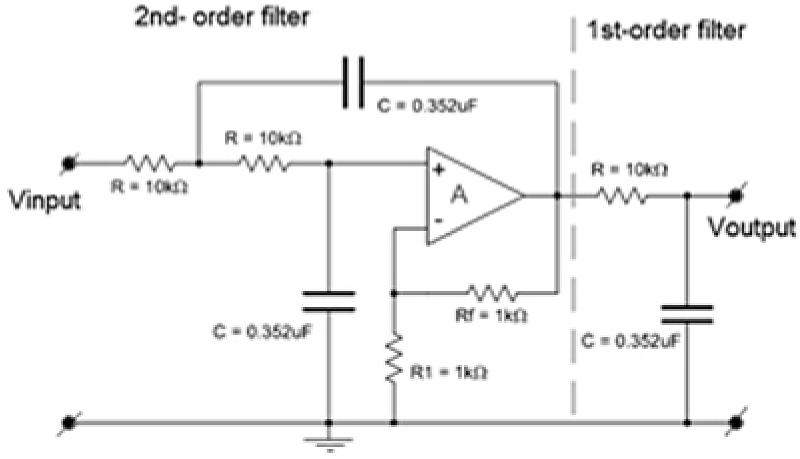

Figure 5.11 - This is typical schematic circuit for an active high-order filter

PRACTICAL GUIDE

Unit N.1: Introduction

Objectives:

- Familiarize the student with the architecture of the transmission educationl system and with the different stages composing it
- Familiarize the student with the amplitude modulated basic modulation
- Study the amplitude response operation of the output signal as to its input
- Study the operation limits of the Amplitude Modulation system by adding some random noise

Requisites:

- theoretical knowledges on the analog transmission

Operative instruments:

- DL 3155AL2 or stabilized supply source
- 20MHz double trace oscilloscope
- set of connection cables

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Introduction to experimental kit

For running these sets of experiments, dear reader, you may find the basic of knowledge in the theoretical manual.

The present practical manual is written as an applicative part of the theoretical part.
As soon as you will start thinking to information transmission, you are welcomed to read, to practice to DL 3155M60 implementation, and to imagine a lot of other transmission scenarios.

The structure of practical manual, the schematics, and the notations from the practical manual respects theoretical manual. Both are used for running these experiments, but not only.

We suggest, for any student, before running experiments, to read the theoretical part, in order to resume the required basic knowledge.

Experimental setup is design in a way that students have almost all needed devices, tools, and accessories for running experiments.
In our didactical vision, any experimental kit consists of:

- base station, with some important possibilities;
- experimental module which is dedicated to a set of experiments.

In order to run experiments, you might not be worry about technical characteristics of the kits components, because they are designed to be 100 % compatible.

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Signs descriptions

	Warning situation: please be carefully with running this experiments
	Electrocution danger: please be carefully with manipulating
	Advise and help board from technicians. Recommendation from tutors and theoretical manual authors
	Theory and technical explanation. Even some readings have been done before; it happens that some deeper explanation must be affected. So, please take care with these extra explanations.
	Tasks items. They must be passed step by step.
	This area where you can write your comments. They are welcomed feedbacks to the authors of the manuals, to the technicians, and to the experiments coordinators
	It is your time! By following recommendations, please complete the experimental work

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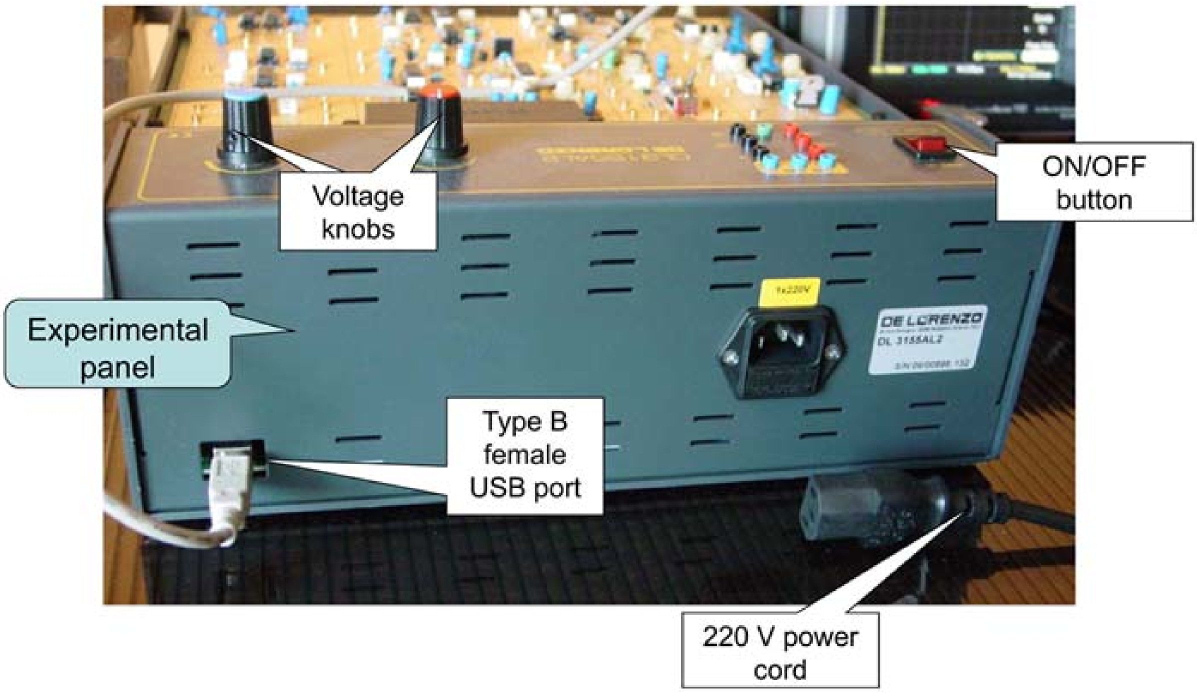
Experimental kit description

Kit connection



The experimental kit is supplied to the lab power network (220 V, 50Hz, +/- 5%).
The DL 3155AL2 is also equipped with USB port type B female for future experiments.

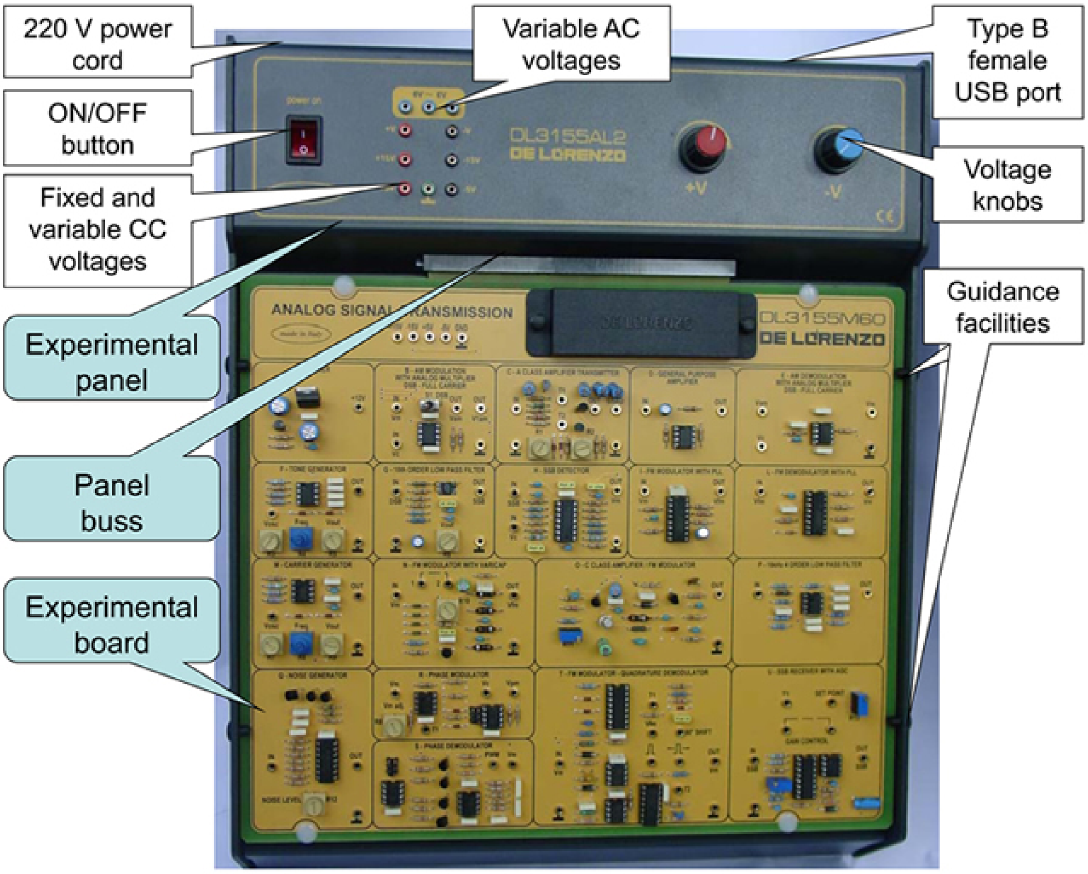
The power cord requires power supply grounding. Please check if grounding is done accordance with your local regulations.

Panel kit
The experimental kit consists of:
Base station (DL3155AL2), or experimental panel, with some important possibilities:

•	+/- 5 V, fixed voltage;
•	+/- 12V, fixed voltage;
•	+/- U, variable voltage;
•	 2 x ≈ 6 V, alternative voltage

Please pay attention to the front panel of DL3155AL2 and observe these facilities.

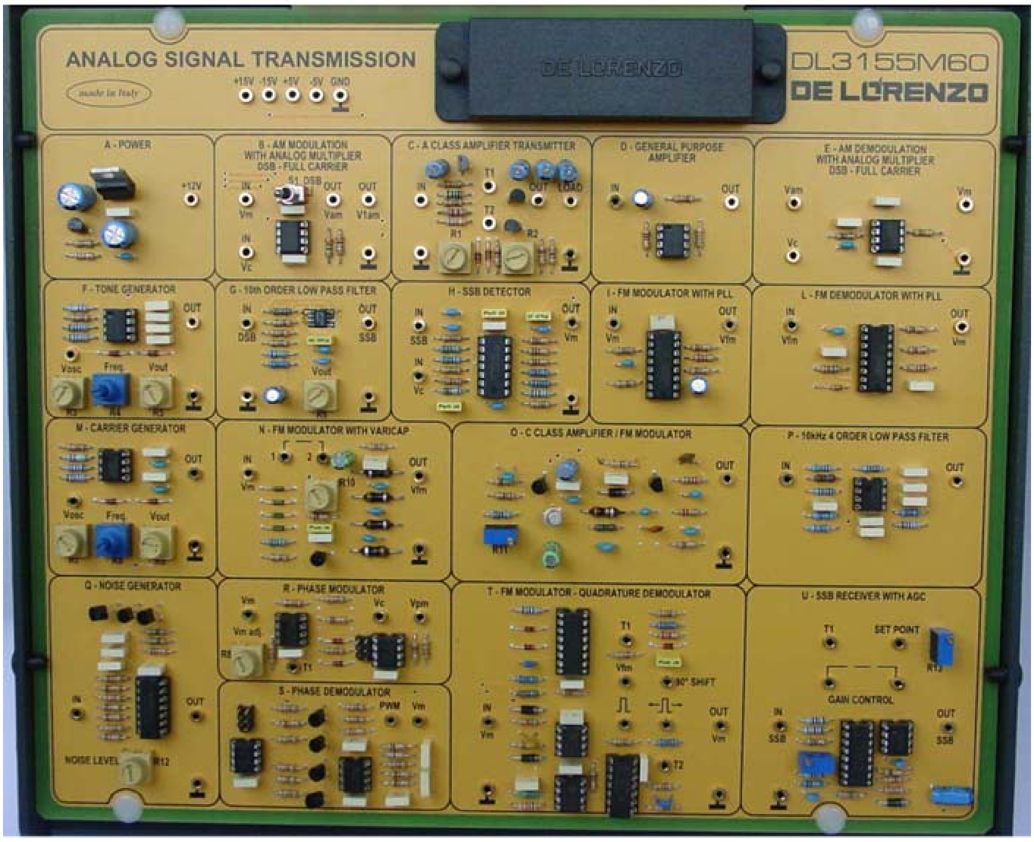
The full description of the front panel of experimental kit is figured next.



Experimental board (DL3155M60) is dedicated to a set of experiments.
The board consists of electronic circuits which are grouped by logical destination, and by handling requirements:

•	stabilized power supply -**A**;
•	oscillators and tone generators: • tone generator (audible frequency)- **F;** • general purpose amplifier- **D**; • carrier generator- **M**; • noise generator – **Q.**
•	AM modulation with analogue multiplier DSB full carrier-
•	A class amplifier transmitter- **C**
•	Demodulating facilities: • AM demodulation with analogue multiplier DSB- full carrier- **E**; • SSB detector- **H**; • 10^{th} order low pass filter- **G**; • 10 kHz 4 order low pass filter- **L**.
•	FM modulators: • FM modulator with varicap- **N**; • FM modulator with PLL- **I**; • C class amplifier/ FM modulator- **O**; • phase modulator- **P**.
•	FM demodulator: • FM demodulator with PLL- **L**; • phase demodulator- **S**; • FM modulator- quadrature demodulator- **T**.
•	SSB receiver with AGC- **U**;

	Please pay attention to the analogue signal transmission board (DL3155M60) and locate the electronic circuits.

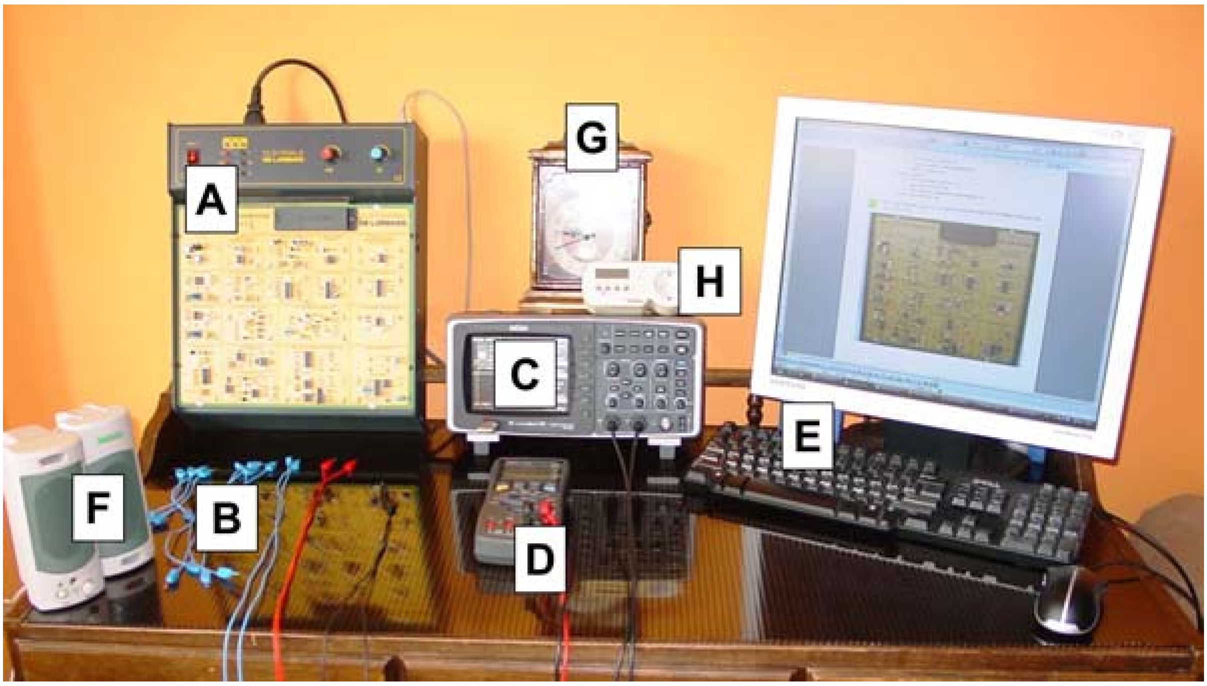


As I mentioned before, the technical staff designed the experimental board in a way of simplifying the work and for understanding easier the specific didactics. For any comments and related recommendations, please fell free to write to the technical staff your recommendations.

Additional accessories and instrumentation

Experimental setup structure

For running the transmission processes experimental we propose a minimal structure, which is figured next.



The meanings of marked elements are:

- ***transmission experimental kit* - A** (delivered by DeLorenzo, and included in package);
  - the main part of the experimental setup consist in control panel and transmission circuits board, which are correlated with the other structure elements;
  - the power of DeLorenzo experimental kits consists on the fact that they do not demand special requirements. They ask for normal labs endowment;
  - quite few extra measure should be taken, like grounding, screening, and using some time non magnetic screw drivers. All these measure will be prompted in time.
- ***wiring sets* - B** (delivered by DeLorenzo, and included in package);
  - colored wires are needed for connecting together electronic circuits, in order to fulfill the experimental tasks;
  - in labs, usually they are coloring regulations, and the experiments conductors, or supervisors follows them. They are required in order to make easy the process of experiments validation;
  - if they are not such regulations, the experiment conductor must establish such regulations. In the next paragraph, we will show such handling regulations;
- ***didactic oscilloscope* - C** (lab endowment);
  - because the experimental setup generates, uses, and process frequencies before 1 MHz, a general use oscilloscope, with minimal requirements will be used (two measuring channels, cutting frequency around 50 MHz);
  - there is a recommendation related to saving measured data from oscilloscope: for PC processing work, there is recommended an oscilloscope with serial communication with PC, or with flash memory drive.

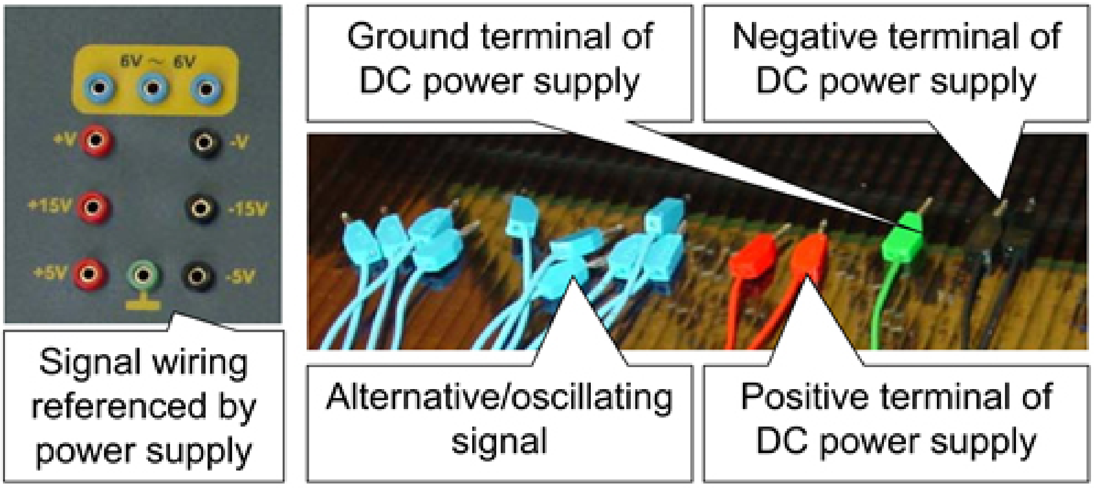
•	***multimeter* - D** (lab endowment); • for handling using voltages in transmission experiments a general use multimeter will be used.
•	***PC* - E** (lab endowment); • The recommendations for the PC are only related to the power of processing requirements; • USB 2 is found today in any PC endowment;
•	***audio "processor"* - F** (lab endowment); • when we start experimenting with transmission procedures, it is recommended also to have a well control to data that have to be transmitted. Please imagine that in any house there is &an audio system that reproduces in well condition any 1000 to 15000 Hz audio signal. So, for not very professional applications, any PC audio speakers are enough.
•	***time "controller"* - G** (lab endowment); • any elaborated experiments are scheduled, so there is demanded a practicing plan, they are issues to be passed. There is not bad to have a nice wall clock in your lab.
•	***energy meter* - H** (lab endowment); • running experiments is nice approach, but any additional requirements are related to energy consuming. Who did not heard about energetic certificate of electronic devices. For instance, the standard CEI 62XXX and EN 55XXX are good support for evaluating energetic class of equipments. A non expensive energy metering device you might find in any lab for evaluating how energy costly is your experiment. For instance, my experiment experimental panel consumes about 26 Watts hour. Not bad!

Experimental wiring regulations
When running any experiments it is very important to understand any possible dangers, any polluting sources, and any disturbances. The transmission processes is also the subject of interferences under Electromagnetic Compatibility regulations.

For these reasons, we will kindly ask you to understand safety regulations. The experimenting box is built in respect of EU specific regulations, but you are not exempt from experimenting regulations. For our purpose, the only requirement is related to the fact that power supply must by grounded according to local regulations.

The experiments usually are conducted by technical personnel, but even you are experimenting as a hobby, please understand that the equipments are built up following electric regulations. So, you must be careful with short-circuits regimes. Firing safety regulations are also very important to be followed. We propose a minimal wiring checking actions.

We mentioned you above about wiring regulation. As a general rule, it is recommended to chose the signal/ power wires by following master's rules. In our case, the rules have to be correlated with experimental panel's rules.



On the left side, there is a capture of panel box with supply terminals. On the right side we have proposed meaning for cable's use.
You will be surprised to find how easy is to wire electronic modules, then to check for electric rules, if you are accustomed with respecting wiring regulations!

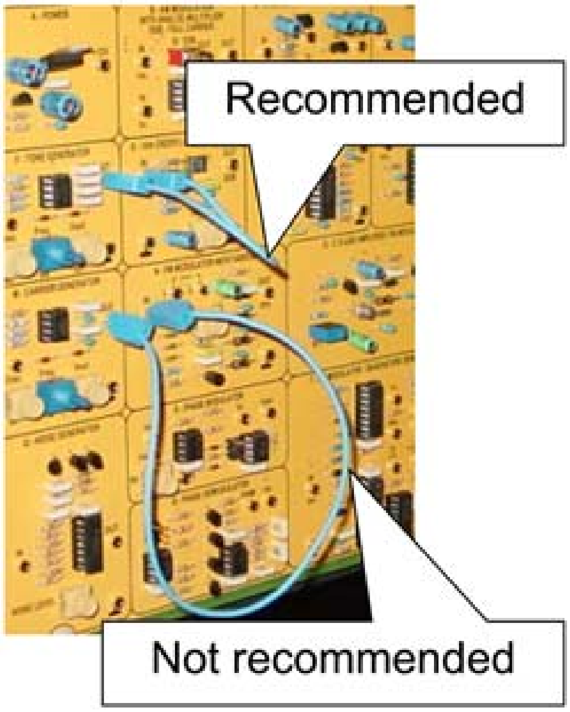
Handling high frequency signal
In any high frequency experiments we are worried about disturbances and pollution from external electromagnetic radiance sources. We are also worried about interferences between our signals with other electronic devices electromagnetic fields.

These are subjects of Electromagnetic Compatibility regulations. To prove also the presence of polluting frequencies please check a very simple experiment as figured next.

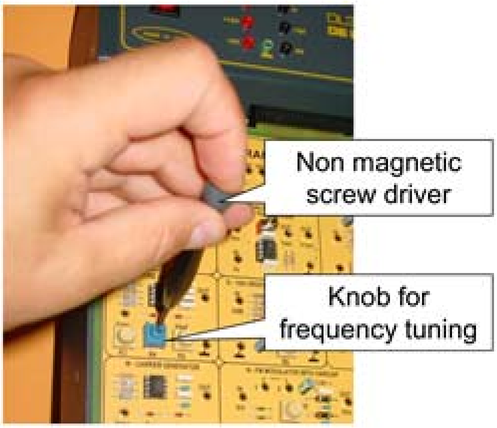
Simply, by touching the active wire of the oscilloscope measuring cable, on the display we figure out about 2 $V_{pk\text{-}pk}$ alternative (noise) signals. So what! Well, please switch on the available PC audio speakers, and then touch the signal terminal. Is it vexating you? I think so!

In the same manner, such disturbing signals will vexate the experiment results. Who want to modulate and transmit noise over the seas?
For acquiring, measuring, and presenting signals they are two situations when electromagnetic compatibility regulations have to be filled.
a. The oscilloscope uses screened cables, tested with IEC 61010 regulations, in order to avoid disturbances.

b. They are situations when some output ports have to be connected to the next input of electronic circuit. In this case the used cables have to be as short as possible, and the way of lied cables should be chosen as far as possible by the possible noise sources.

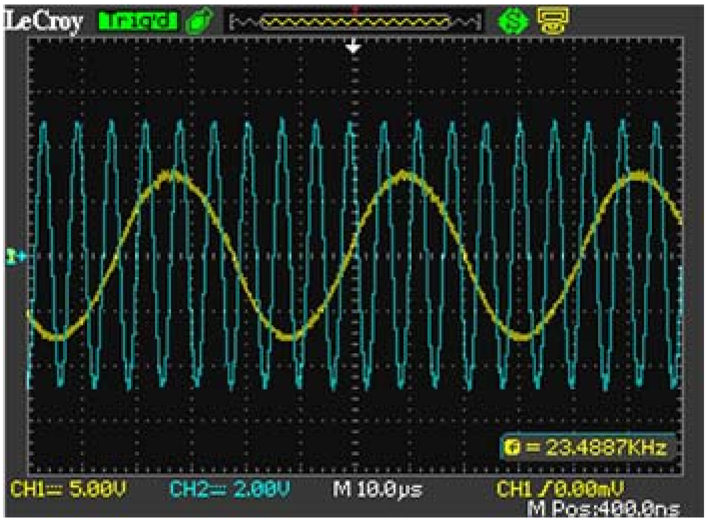


Handling with frequency tuning
In transmission experiments we work with oscillators. For passing experiments issues we need to tune, and to adjust central frequency. When we manipulate oscillator's parameters, the basic rule of frequency stability is to keep constant oscillators parameters (capacitance and inductance). For that reason, we must avoid any magnetic or capacitive disturbances.
On the transmission experiment board they are some facilities to change oscillator's frequency in a linear variation.



The handling knobs are accessible for fine tuning action by screw drivers. Any lab must have some proper non magnetic screw drivers. Please use them for successful results of experiments.

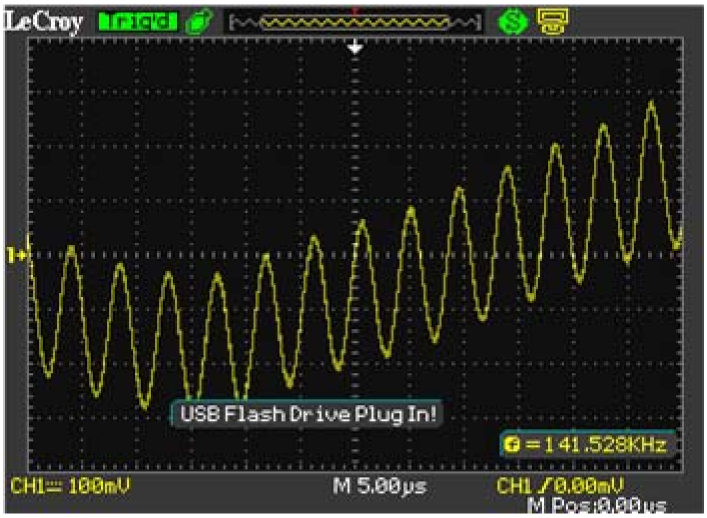
Managing with two signal measurements
In many experiments some minimal measurements ask for at list two synchronized signals measurements. For instance, there is necessary to evaluate the audio oscillator signal by comparing it with carrier signal, in the process of data transmission. Please exercise this task by manipulating scope probe in accordance with oscillator's knobs.



Managing experimenting errors

Additionally, there are a lot of strange situations for the conductor of the experiments, or for hobbyist. Many of them are cleaned up as soon as the electrical rules are checked.

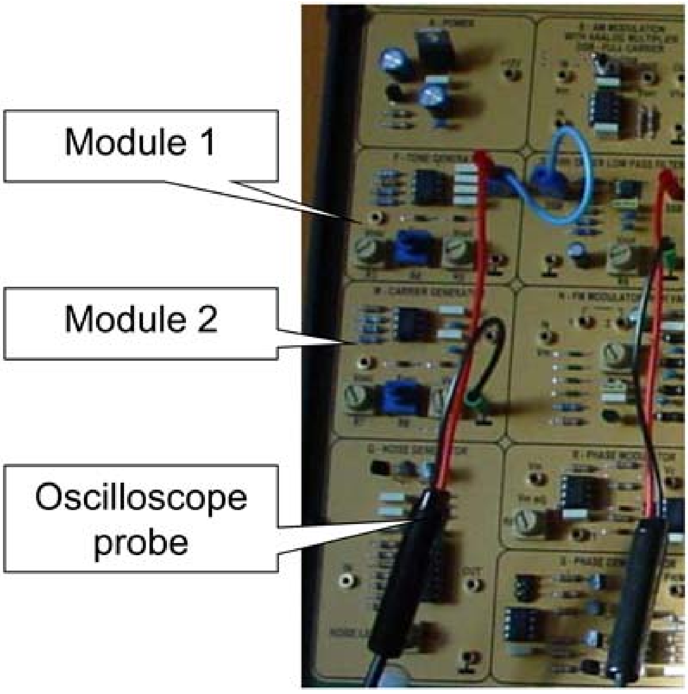
If in some cases, you get some strange figures on the oscilloscope's display.



In this case, if you did not adjust the outputs of the electronic modules in order to form such signal please check electric connections of the oscilloscope's probe. Based on my experience, such signal shape comes from the fact that the signal reference (signal ground) is not effective.

The same effect comes from the fact that one of the important electric rules is not fulfilled. In many electronic devices, there is working distinction between signal and supply ground.

On the other hand, in other electronic circuits some time there is insulation between signal grounds at different modules.



In the upper figure, please pay attention to the wrong connection of the oscilloscope's probe ("hot"- red wire is connected to module's 1 port, ground wire- black is connected to module's 2 ports).

Unit N.2: Understanding oscillators

Objectives:

- Understanding the characteristics of radio-frequency oscillators (RFO), and how to work with them
- Designing and operating with oscillators

Requisites:

- Minimum level of communication techniques understanding
- Medium level of electronics components and devices understanding
- High level of health and safety risks understanding
- Communication systems theoretical manual DL 3155M60

Operative instruments:

- DL 3155AL2 or stabilized supply source
- 20MHz double trace oscilloscope
- set of connection cables

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Where we start from

As we read about, an oscillator is a signal generator converting its dc supply voltage into a continuously repeating oscillating output signal without any input signal. The oscillator generates the carrier or local oscillation signal used in any communication system.

As we know, it includes an amplifier with a positive feedback network consisting by a resonant circuit.
Take care, when the power is first applied to the circuit, some signal noise will appear in the circuit and is amplified by the amplifier and then fed to the input through the feedback circuit through a resonant circuit with filter function.

The feedback circuit allows passing the signal frequency equalling the resonant frequency and rejects other frequencies.
The feedback signal will be amplified and fed back again. If the feedback signal is in phase with the signal at input and voltage gain is enough, the oscillator will be operation.
This situation is called stable oscillation.

Identifying the oscillators

For the experiment purpose, we have to identify the available devices, with all interactive ports.
Interactive ports (inputs, states, and outputs ports) are related to:
- oscillator's outputs,
- controlling voltages inputs,
- resonant circuits controls.

Usually, the controlling voltages inputs are terminals (one port referenced to the general ground, or to the zero supply), also called ***test points***.
The resonant circuits controls are variable resistors, capacitors or inductance coils.
Test points are called also the terminals designed to collect oscillator's signals.

Please be carefully with using ports: don't connect oscillator's inputs and outputs ports together, manipulate carefully power cables in order to make proper connections.

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M - Carrier generator

The most important module is ***Carrier generator***, with the code ***M*** on DL3155M60 board.
The quality of this circuit imposes the quality of the transmission system in any implementation.

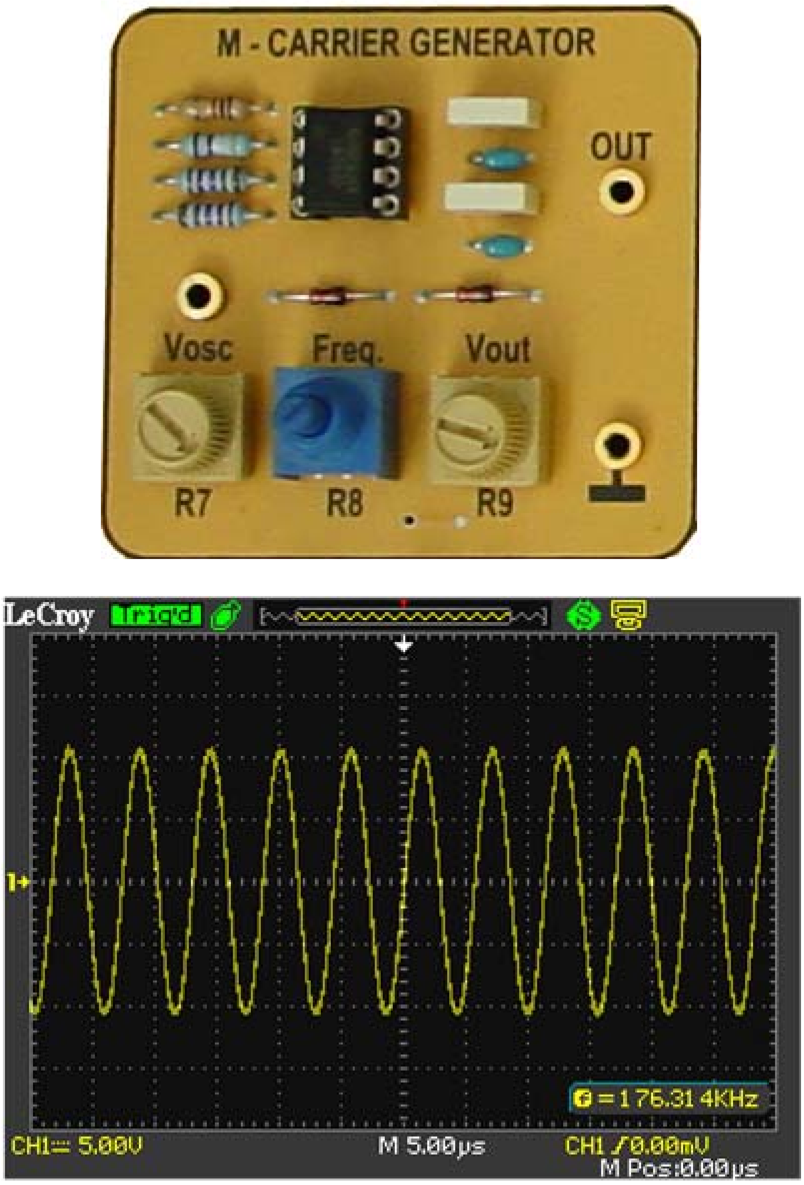


Figure 2.1 - The Module designed for exercising carrier frequency control

Tasks to study

	Locate the oscillator circuits on the module DL 3155M60. Locate the ports. Locate the controlling possibilities.

Please identify it, then, by consulting the circuit diagram understand the role of the main components (R7, R8, R9). Also please make the differences between the controlling manner of the oscillating voltage (R7), of the output voltage (R9), and of the oscillating frequency (R8).

- Note the extreme values of oscillating parameters.Make your comments

Set the input of oscilloscope to AC position and connect to output terminals (O/P). Observe the waveform and frequency.

- Note the extreme values of oscillating parameters. Make your comments

Use the resonant circuit controls for frequency variation. Please observe and note the oscillator's frequency bandwidth.

- Note the extreme values of oscillating parameters. Make your comments

Quiz. Test yourself and be ready to accept new questions

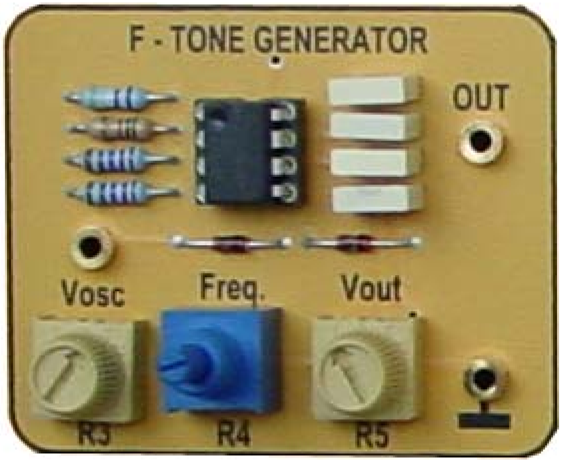
A. Can you calculate the oscillator frequency? There is a difference between calculated and measured values? Explain your answer please.

B. What is the function of each capacitor or inductor in analyzed oscillator circuit? Can you point differences between played roles of main inductors and capacitors in difference oscillator's circuits?

C. In your experiments did you detect frequencies that are useless? When the operating frequency is in radio-frequency range, why we must pay attention to the layout of circuit and the length of wire?

F- Tone generator

The same important module is also the ***Tone generator***, with the code ***F*** on DL 3155M60 board. It's quality also imposes the quality of the transmission system in any implementation.



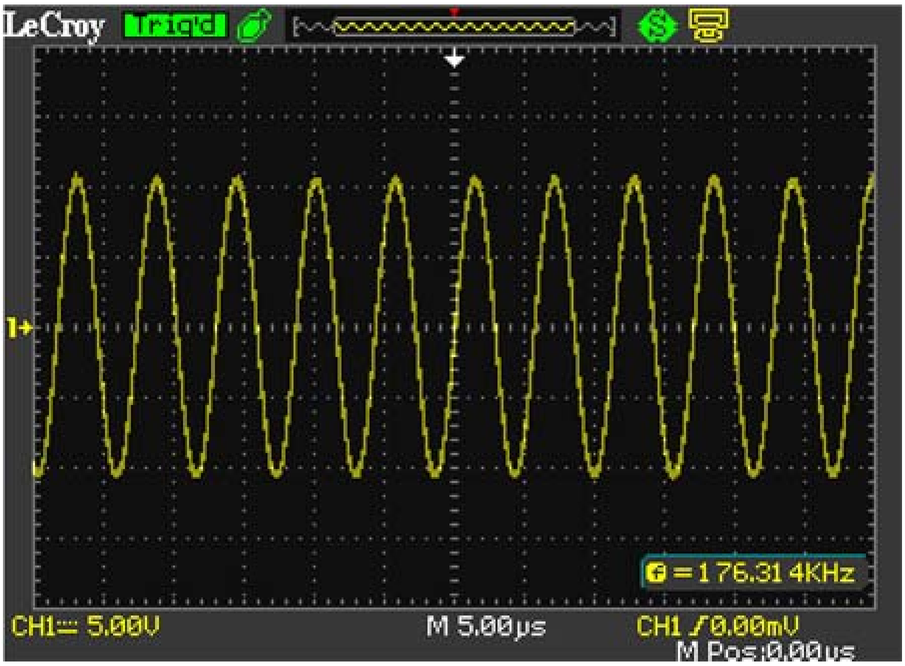


Figure 2.2 - The module is designed for exercising tone generator control, as information signal for carrier modulation

Tasks to study

	Locate the tone generator circuits on the module DL3155M60. Locate the ports. Locate the controlling possibilities.

Please identify it, then, by consulting the circuit diagram understand the role of the main components (R3, R4, R5). Also please make the differences between the controlling manner of the oscillating voltage (R3), of the output voltage (R5), and of the oscillating frequency (R4).

- Note the extreme values of oscillating parameters. Make your comments.

Set the input of oscilloscope to AC position and connect to output terminals (O/P). Observe the waveform and frequency.

- Note the extreme values of oscillating parameters. Make your comments

Quiz. Test yourself and be ready to accept new questions

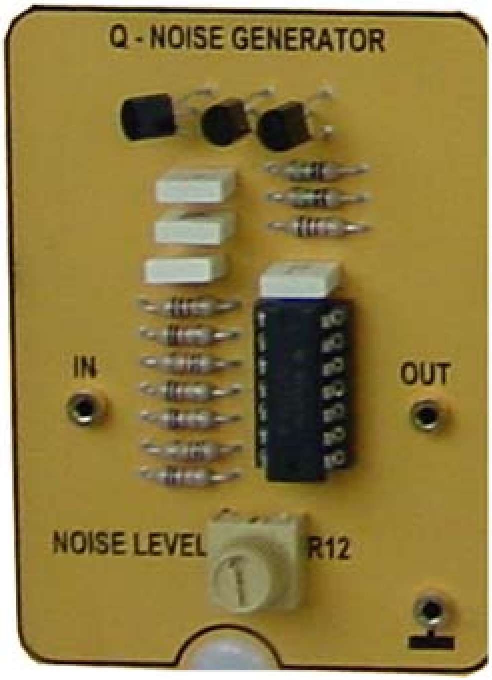
A. Can you calculate the oscillator frequency? There is a difference between calculated and measured values? Explain your answer please.

B. How the audio frequency could be detected? Use a proper device.

C. In your experiments did you detect frequencies that are useless? When the operating frequency is in audio-frequency range, what kind of precautions should be done?

Q - Noise generator

The most important module is **Noise generator**, with the code Q on DL3155M60 board.
The quality of this circuit imposes the quality of the transmission system in any implementation.



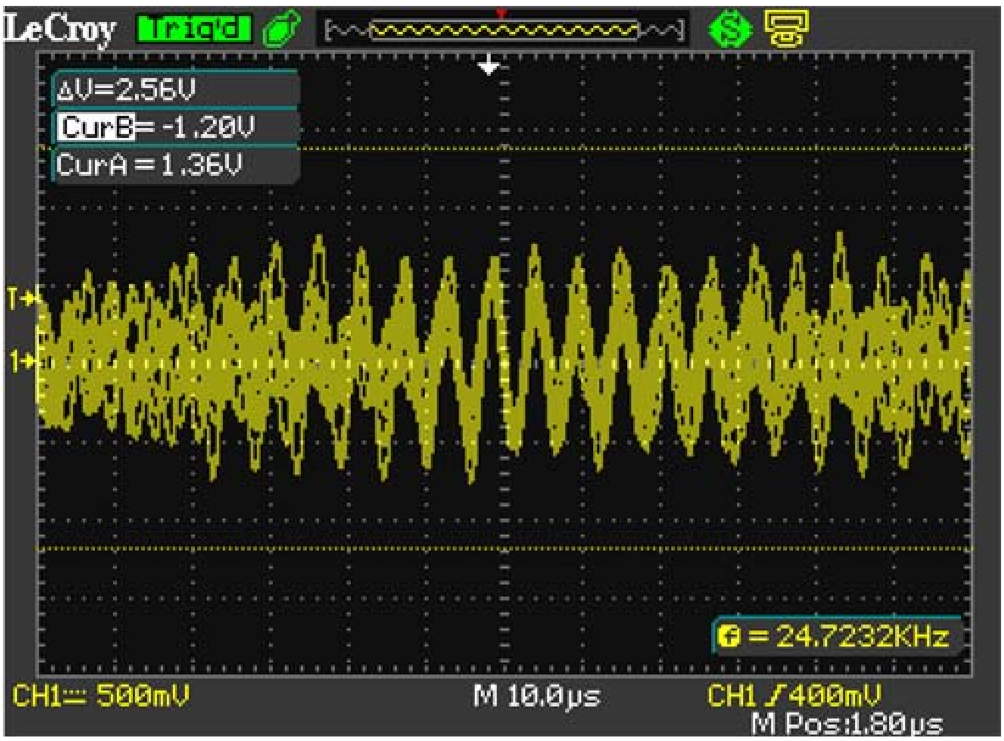


Figure 2.3 - The module is designed for practicing with noise generator, as distortions of information signal

Tasks to study

	Locate the noise generator circuits on the module DL 3155M60. Locate the ports. Locate the controlling possibilities.

Please identify it, then, by consulting the circuit diagram understand the role of the main controlling component (R12).

- Make your comments

Set the input of oscilloscope to AC position and connect to output terminals (O/P). Observe the waveforms and frequencies. Adjust the noise level controlling system (R12). Please note waveforms variations, and ratio of frequencies amplitudes.

- Make your comments

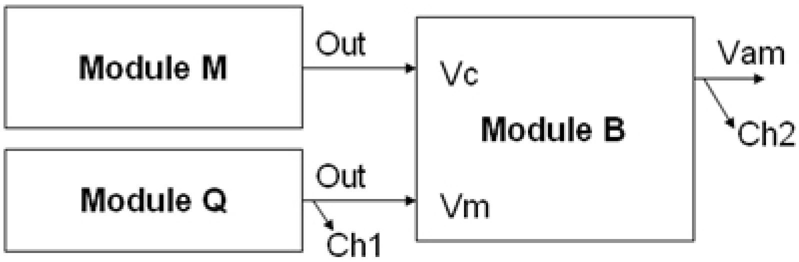
Quiz. Test yourself and be ready to accept new questions

A	What is white noise?

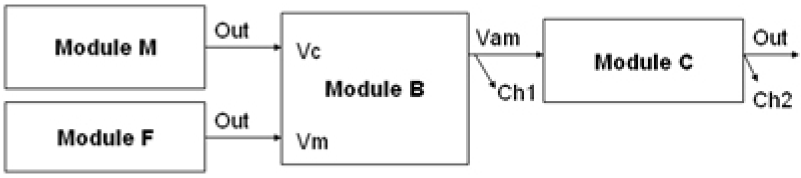
B	In your experiments did you detect frequencies that are useless? When the operating frequency is in audio-frequency range, what kind of precautions should be done?

Fault Simulation

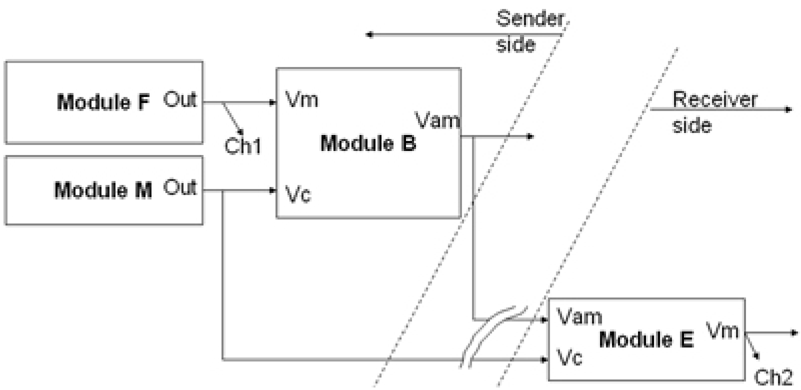
Let's imagine we are in the middle of one experiment:



Or:

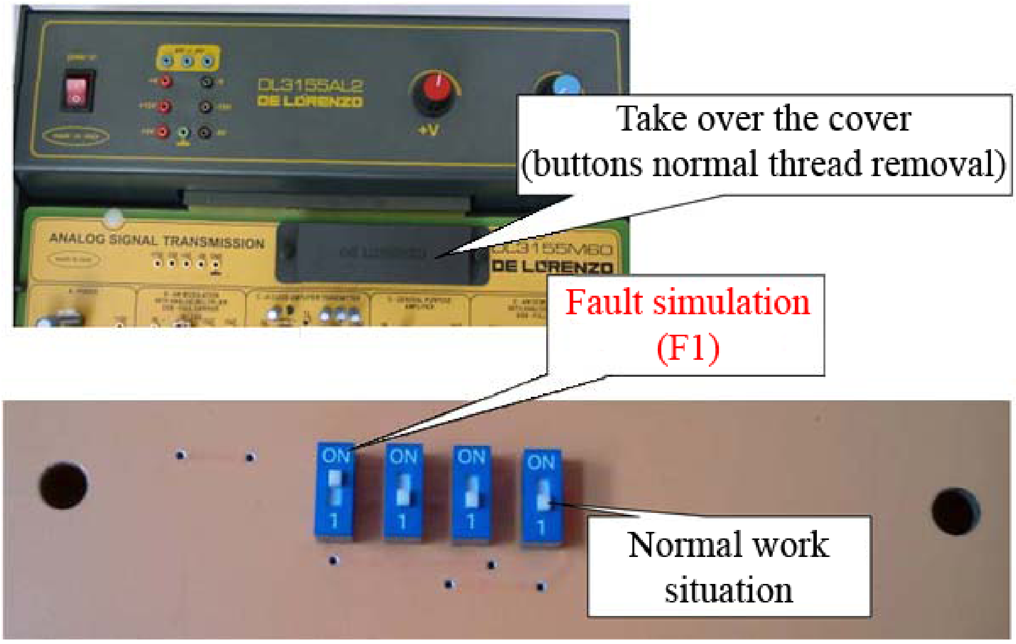


Or this:



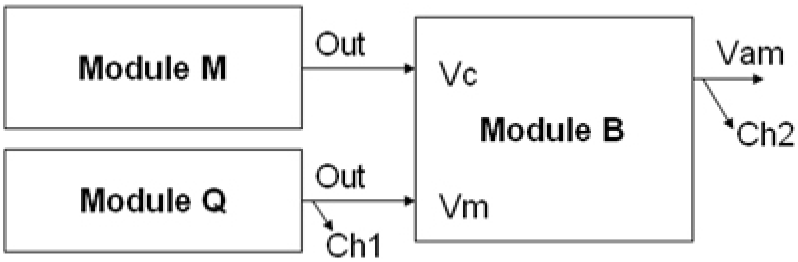
Now, let's rice up a question! What happened here in the experiment if there is not signal in Ch2? In order to start searching for the fault cause now we proved that we need to understand what is the role of each components/ modules for experiments.

The fault scenarios are simulated with dip- switches. The F1 is imagined in next figure.
The first switch is steady commuted in ON position.

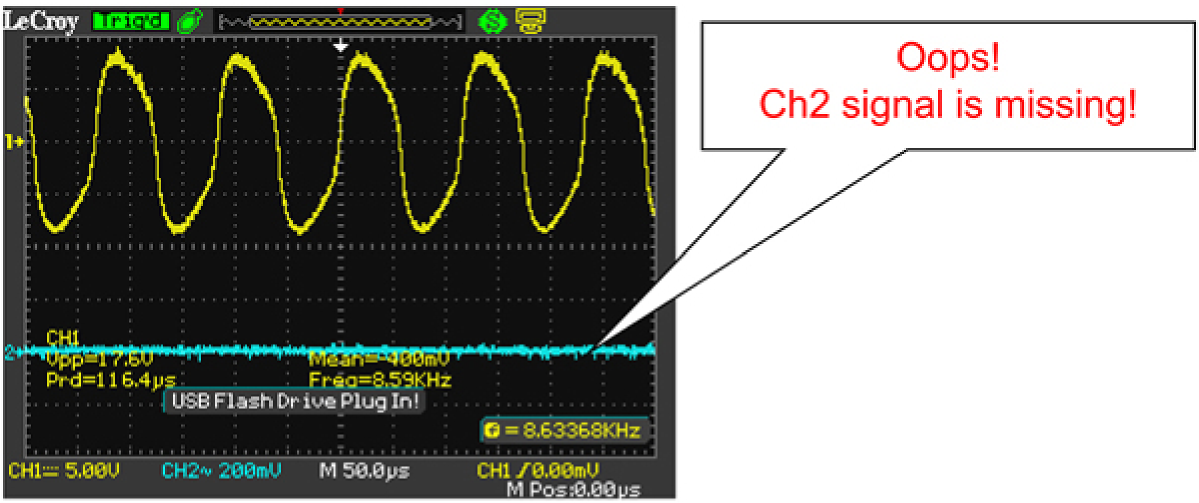


If the panel is connected directly to your PC via De Lorenzo TIME interface, then the fault can be inserted directly by clicking on INSERT shown below.

Let’s suppose that we are in the middle of the next experiment.
Then, you, as conductor of the experiment, switch the dip-switch for F1 scenario.



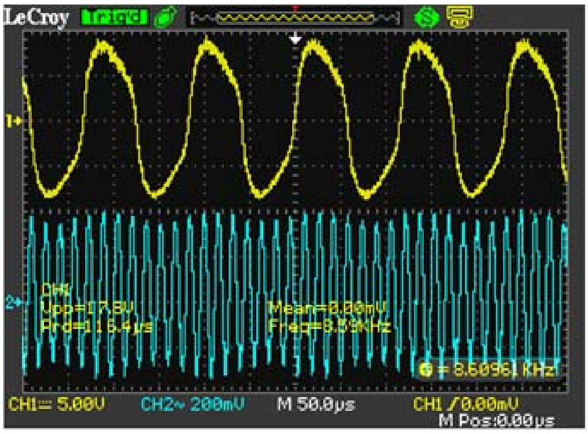
The signal at the module B (Ch2) will look like:

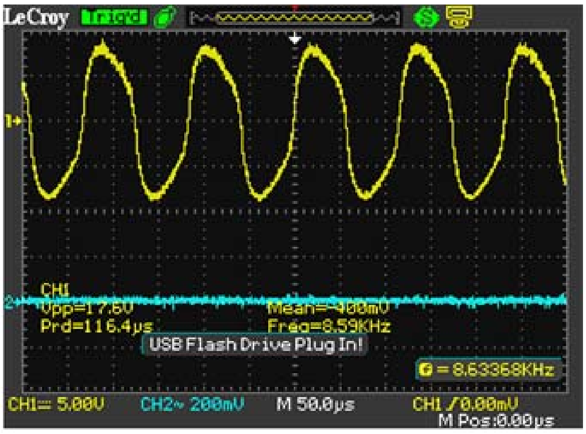


Let’s translate it in English (or in your born language!): the output of the modulator is missing!
Some remarks: the fault experiment does not cover board, or components damages; the power supply might not be imagined as a cause of faults

If we succeeded with well translation, the half of problem is solved. This is because it means that we understand the processes they are running in our experiment. So, this is the fault detection start point.

In order to start designing faults detection algorithms, let’s compare the normal (Exp3) and fault situations (F1), at the output of module B.





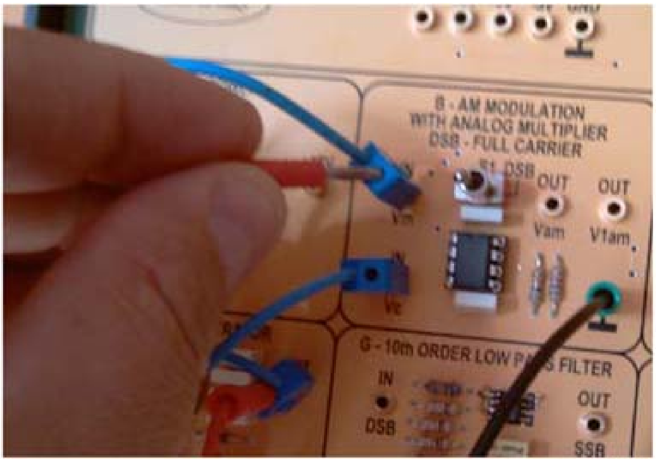
In the normal situation (left figure) we see tone generator signal, we see the envelope of modulating signal, and we see the high frequency. What about right image? Nothing at the output of module B. There is not carrier, there is not modulating signal!

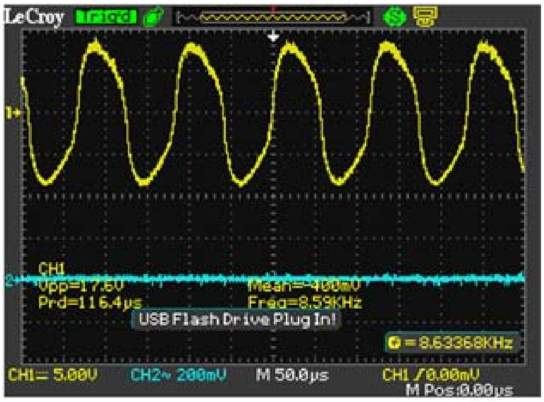
Is the oscilloscope cable right connected?

Please check it! If YES, let’s go next.
Because we understand the role of connected modules, we can explore deeper the situation by putting here a new question.

Supposing that the modulator module (B) works well, what input signal is missing, carrier or tone signal?

By understanding how the modulator works, we could have an answer.
If not, let’s use the scope to diagnose the situation.

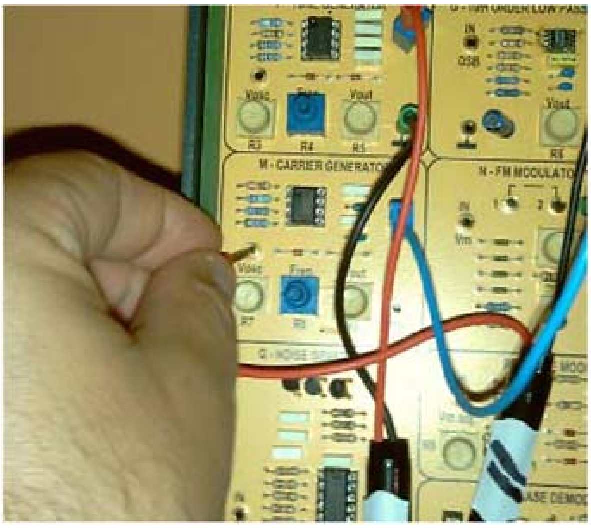


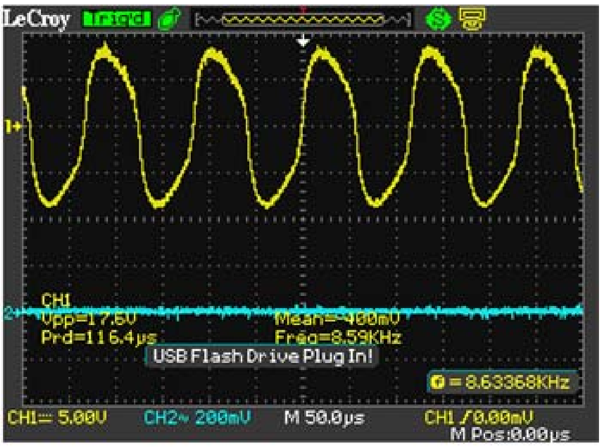


Signal is missing at carrier input

Is the connecting cable right connected?
Please check it! If YES, let's go next.

Let's go to the module M- carrier generator.
This module, as you remember has two outputs (one is test point; the other is output port for experiments).
Next oscilloscope diagnose is related to check if we have signal in one of outputs.





Signal is missing at carrier outputs

The final diagnose is: the carrier generator is not working properly. We are happy that is only simulation of the fault!

⚠	We are kindly advice you here, stop searching deeper for the fault. If you want to know how the simulation of the fault is done, we could advice you.

Unit N.3: Understanding filters

Objectives:

- Understanding the characteristics of filters
- Understanding the use and manipulation of active filters
- Designing and operating with filters
- Controlling N-order filters with integrator circuit

Requisites:

- Minimum level of communication techniques understanding
- Medium level of electronics components and devices understanding
- High level of health and safety risks understanding
- Communication systems theoretical manual DL 3155M60

Operative instruments:

- DL 3155AL2 or stabilized supply source
- 20MHz double trace oscilloscope
- set of connection cables

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Where we start from

As we read about, filters, which in every communication systems, are designed to pass a specified band of frequencies while attenuating all signals outside this band.

They are classified according to filtering range, frequency response in pass band, and circuit component. Classified by filtering range, there are four types of filters: low-pass, high-pass, band-pas s, and band-reject filters.

Because the frequency response is the main reason of use those, according to frequency response in pass band, there are two types of filters: Butterworth and Chebyshev filters. One the other hand, according to circuit component, they are active and passive filters.
The passive filters are the circuits that contain only passive components (resistors, inductors and capacitors) connected in such a way that they will pass certain frequencies while rejecting others.

The active filters employ active components (transistors or operational amplifiers) plus resistors, inductors and capacitors.
Active filters are widely used in modern communication systems, because they have the following advantages:

- the transfer function with inductive characteristic can be achieved by particular circuit design, resistors can be used instead of inductors;
- the high input impedance and low output impedance of the operational amplifier means that the filter circuit is excellent in isolation characteristic and suitable for cascade;
- active components provide amplification, therefore active filters have gain;

In the next experiment activity, we will focus on the characteristics of the N-order low-pass and high-pass active filters (including second-order filters).

As we already know, a low-pass filter is an electronic circuit that has a constant output voltage from dc up to a cutoff frequency. As the frequency increases above the cutoff frequency, the output voltage is attenuated.

The cutoff frequency, also called the 0.707 frequency, the 3dB frequency, or the corner frequency, is the frequency where the output voltage is reduced to 0.707 times its pass band value.

Identifying the panel filters

For the experiment purpose, we have to identify the available devices, with all interactive ports.
Interactive ports (inputs and outputs ports) are related to:
- filter's outputs (voltage level, cutoff frequency);
- controlling the filtering DSB into SSB;

Usually, the controlling voltages output are terminals (one port referenced to the general ground, or to the zero supply), also called ***test points***.
For controlling the outputs, there is a variable resistor.
Test points are called also the terminals designed to collect oscillator's signals.

Please be carefully with using ports: don't connect filter's inputs and outputs ports together, manipulate carefully power cables in order to make proper connections.

G - 10th order low pass filter

For running transmission system an important module is ***10th order low pass filter***, with the code ***G*** on DL 3155M60 board.

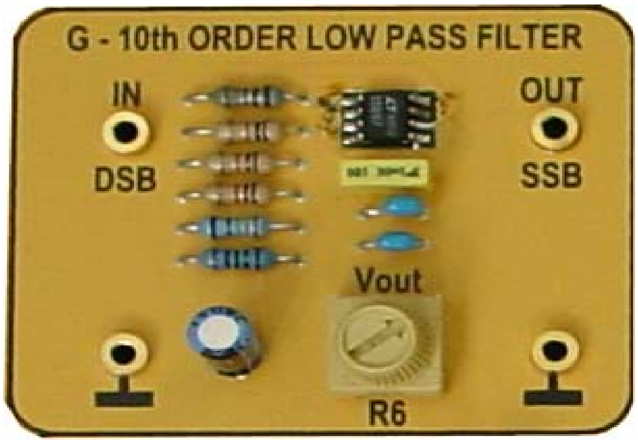

Figure 3.1 - The module is designed to exercise low pass filter characteristics

Tasks to study

Locate the 10th order low pass filter circuits on the module DL 3155M60.

Locate the ports. Locate the controlling possibilities.
Understand ports meaning.

Please identify it, then, by consulting the circuit diagram, try to understand the role of the main components (R6).

- Make your comments

Set the input of oscilloscope to AC position and connect to board input terminals (O/P).
Observe the waveform and frequency for DSB.
Then, set the input of oscilloscope to AC position and connect to output terminals (O/P).
Observe the waveform and frequency for SSB.

- Make your comments

Use the variable resistor controls for controlling the output voltage level variation.

- Make your comments

Quiz. Test yourself and be ready to accept new questions

A. Why there is a need to have access to control the output voltage level on Nth order filter?

B. What are the advantages of the active filters?

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P - 10kHz 4th order low pass filter

This module ***(10 kHz 4th order low pass filter)***, with the code ***P*** on DL 3155M60 board, is used for controlling the audible signals- as information signal into the transmission process.
The quality of this circuit defines the distortions into the process of modulation/demodulation.

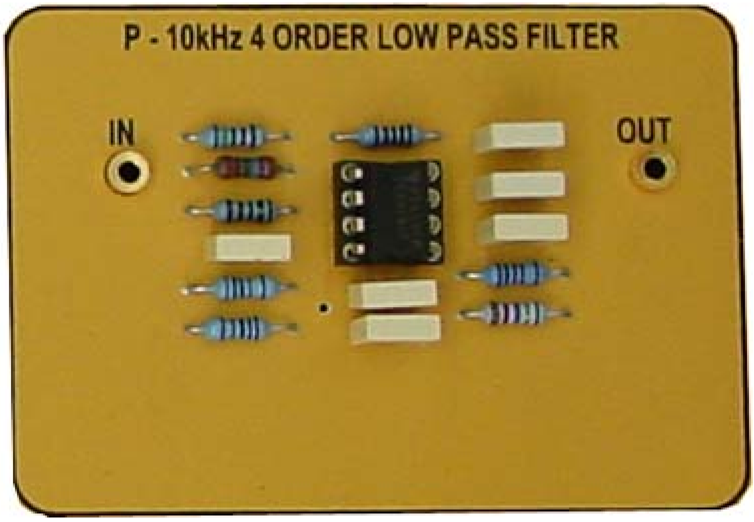


Figure 3.2 - 4th order low pass filter with 10 kHz cut-off frequency

Tasks to study

Locate the 10 kHz 4th order low pass filter circuits on the module DL 3155M60. Locate the ports. Locate the controlling possibilities.
For this experiment you must use a frequency generator with the central frequency around 10 kHz, and an oscilloscope with measurement range focused on 10 kHz frequency. Connect the output of the generator to IN port of the filter. Connect the oscilloscope inputs to the OUT port of the filter

With the frequency control button from the generator, adjust the frequency, and note the signal behavior to the oscilloscope. Pay attention to the level of voltage.

- Make your comments

Locate the tone generator circuits on the module DL 3155M60.

Locate the ports.
Locate the controlling possibilities.

Connect the output of the tone generator to the input of the 10 kHz 4th order low pass filter.
Connect the oscilloscope inputs to the OUT port of the filter.
Adjust tone generator frequency and note the signal behavior on the oscilloscope display.

- Make your comments

	Locate the noise generator circuits on the module DL3155M60. Locate the ports. Locate the controlling possibilities.

Connect the output of the noise generator to the input of the 10 kHz 4th order low pass filter.
Connect the oscilloscope inputs to the OUT port of the filter.
Adjust noise generator voltage level and note the signal behavior on the oscilloscope display.

- Make your comments

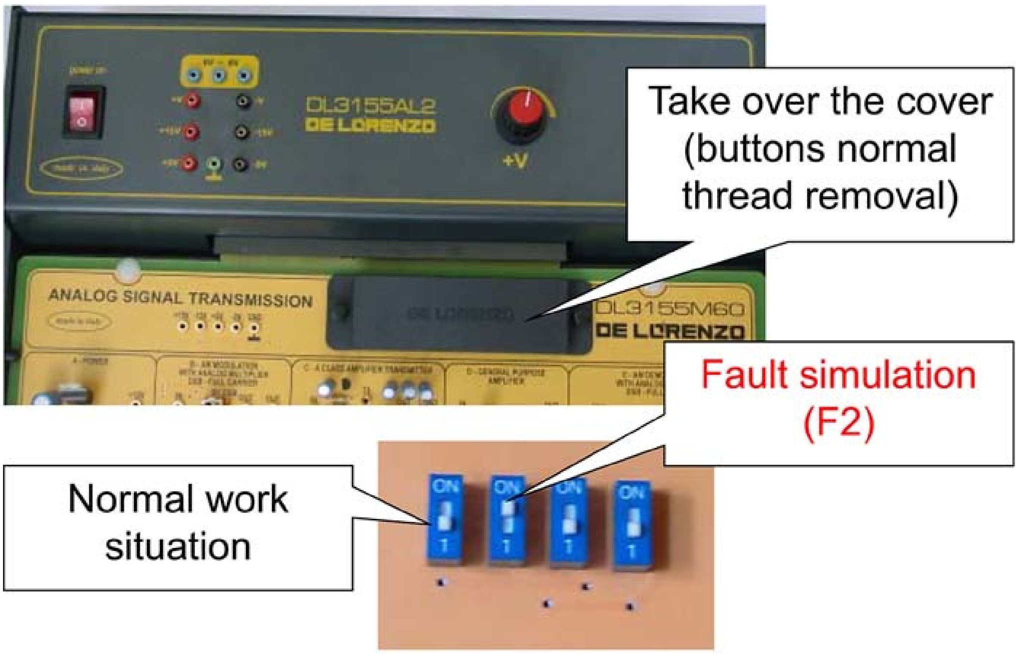
Quiz. Test yourself and be ready to accept new questions

A	Please imagine another method for evaluating 10 kHz 4th order low pass filter behaviour.

B	If we want to change the bandwidth of this kind of filter how can we proceed?

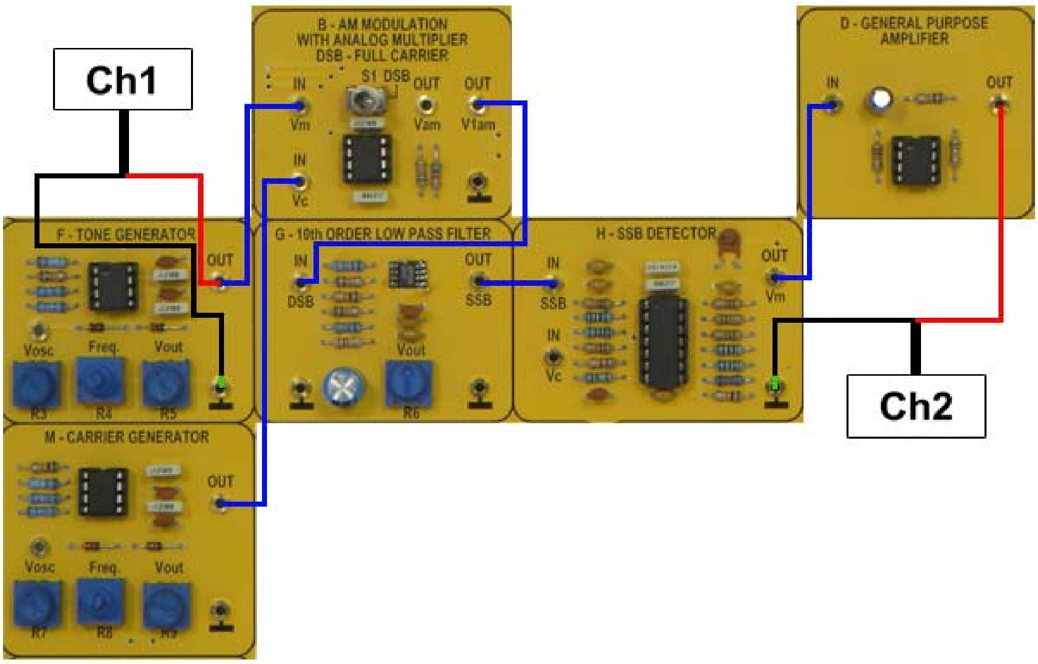
Fault Simulation

Now, the situation is a bit more complicated. The fault scenarios are simulated with dip- switches. The F2 is made by adjusting the switches like in next figure.



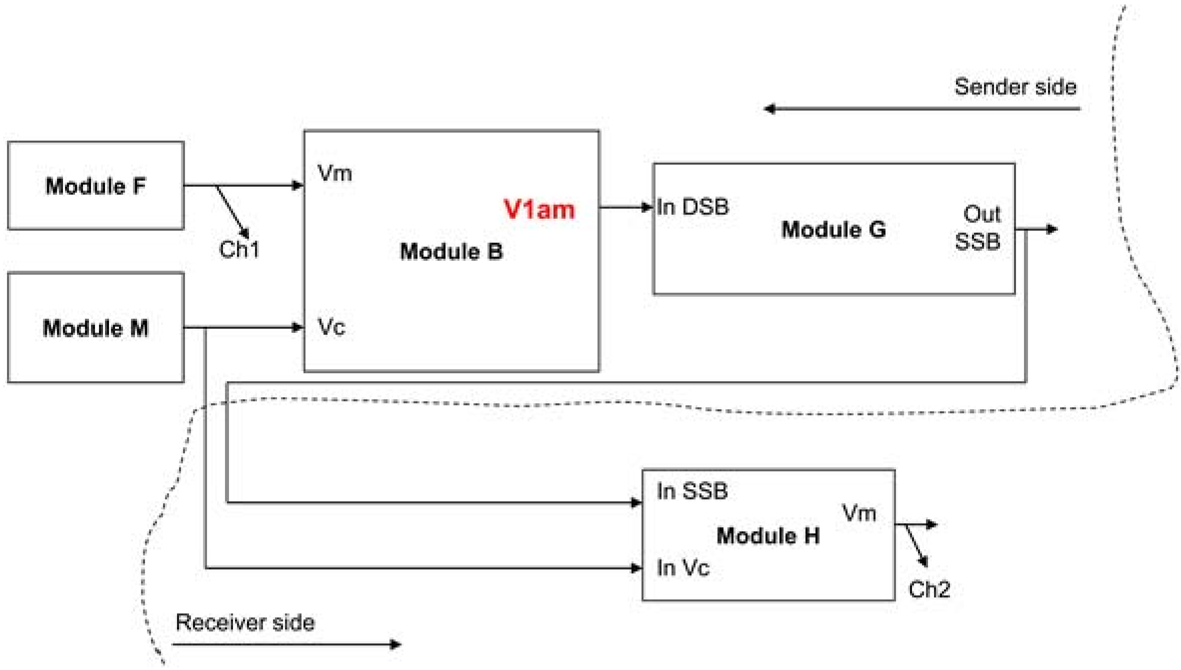
If the panel is connected directly to your PC via De Lorenzo TIME interface, then the fault can be inserted directly by clicking on INSERT shown below.

The experiment under investigation is shown in next figure, and probably you remember it very well.

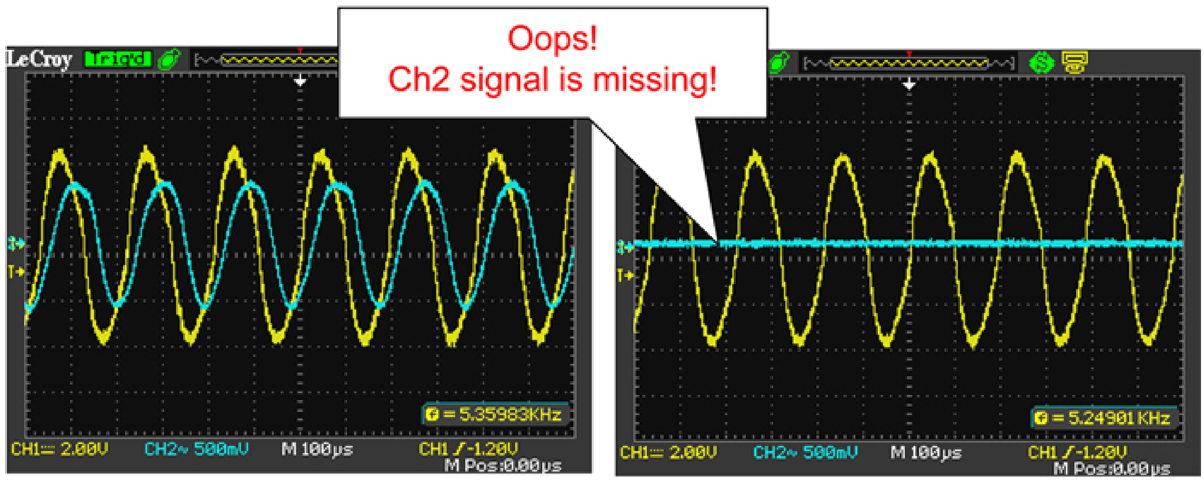


Remember please, we were experimenting SSB receiving signal. Because of low level of the output of SSB detector, we were using also the general purpose amplifier.

Let's suppose the fault is happening when we are excited by the nice results we get with SSB detection.
We show that in next figure- you remember it!



In our simulation, for the fault situation we get no signal at the output of module D.
The fault situation F2 make a difference on Ch2 like next figure.



Normal working situation

Fault situation

Let's try to translate this situation: the output signal of the amplifier is not detectable- it is extremely low.

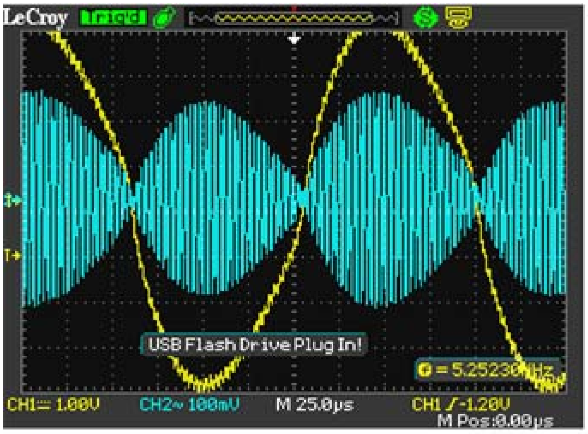
What are the possible causes?
Looking on the block diagram, we might say that they are six modules with six possible causes. Because the modules are serial connected from functional point of view, we might have the possibility to investigate modules from the left to right or in opposite direction.

What is the best "direction" to investigate?
We should start thinking to answer, by evaluating the levels of signals we are manipulating during experiments.

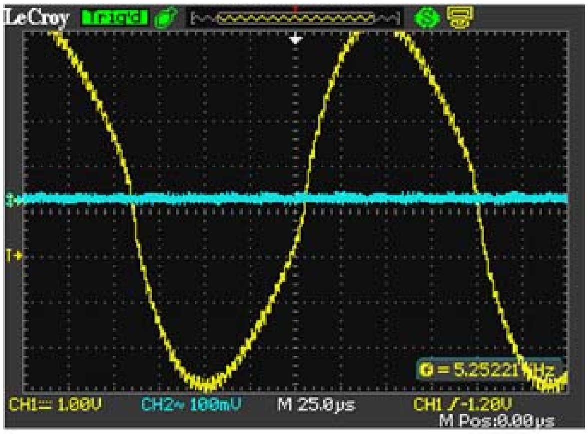
What are the easiest signals to be detected?
The tone generator and the carrier generator have high level of voltage (volts!). The modulator and the filter have also strong signals. More difficult situation is to check the output of the SSB detector (around 50 mV!), where also the ratio signal/ noise is low. Well, after this review, signal, we know the answer: from the left side to right side (from generators).

As in previous scenario (F1), we need to be sure about any other possible outside causes of faults. Then, we use the scope and we check the output signals, module after module.

In our case, after few actions, we will get a situation similar with next figure.



The input of module G



The output of module G

The final diagnose is: the 10th order low pass filter is not working properly. Again, we say that we are happy that is only simulation of the fault!

Beside the actual fault explanations, we would like to recommend you to be careful with manipulation of the cables during measurements- it is very easy to make a short circuit because they are a lot of metallic parts of electronic circuits!
So, during measurements, please handle with one cable a time, use both ends of one cable in one hand, and do not touch any other conductive parts then the connection points.

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Unit N.4: Understanding amplitude modulation (AM)

Objectives:

- Understanding the principle of amplitude modulation (AM)
- Understanding the waveform and frequency spectrum of AM signal and calculating the percent of modulation
- Designing an amplitude modulator using dedicated circuits (AD 633)
- Measuring and adjusting an amplitude modulator circuit

Requisites:

- Minimum level of communication techniques understanding
- Medium level of electronics components and devices understanding
- High level of health and safety risks understanding
- Communication systems theoretical manual DL 3155M60

Operative instruments:

- DL 3155AL2 or stabilized supply source
- 20MHz double trace oscilloscope
- set of connection cables

Where we start from

As we know already, the modulation is the process of impressing a low-frequency intelligence signal onto a high-frequency carrier signal.
The amplitude modulation (AM) is a process that a high-frequency carrier signal is modulated by a low frequency (information signal) modulating signal (usually an audio).
For the amplitude modulation, the carrier amplitude is changing its parameter with the modulating amplitude.

The main parameters of this process are:

- A_{DC} - dc level
- A_m - audio amplitude
- A_c - carrier amplitude
- f_m - audio frequency
- f_c - carrier frequency
- m - modulation Index or depth of modulation

As we read about, the audio signal is contained in the side bands so that the greater the sideband signals the better the transmitting efficiency.
From the theoretical part we know that the greater the modulation index, the greater the sideband signals and the better the transmitting efficiency.

In practical applications, the modulation index is usually less or equal to 1; if m > 1, it is called over modulation. An useful comparison between various balanced modulator outputs under various input frequency conditions should be done in order to understand:

- the types of AM;
- the modulation index;
- the spectra of different types of AM;
- the dissipation power for understanding the efficiency of the AM transmission

For the practical evaluation, we would like to remember you the fastest formula for modulation index calculus.

$$m[\%] = \frac{(A_c + A_m) - (A_c - A_m)}{(A_c + A_m) + (A_c - A_m)} 100$$

Remember, the notations that contain “c” are related to carrier signals, and the notations that contain “m” are related to modulating signal (information signal). Now, in upper equation we speak about manipulating with amplitudes of signals.

Identifying the panel filters

As you understood, we are in the process to experiment few modulating techniques. Let's indentify the available devices, with all interactive ports.

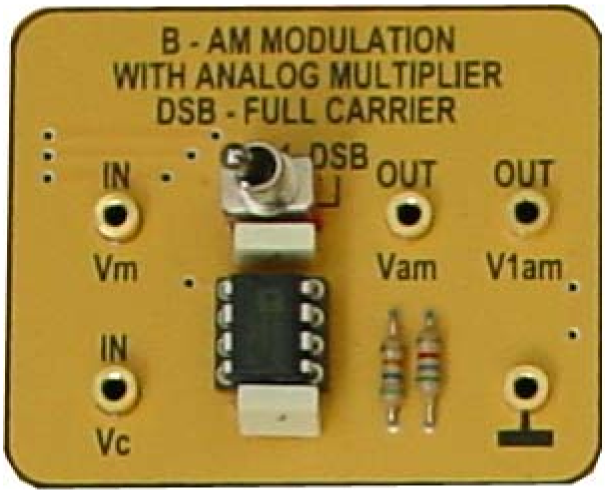


Figure 4.1 - The available module will help to experiment the AM modulation

On the theoretical part, that supports this experimental kit- amplitude modulation section, there is a full description of possibilities of running with amplitude modulation techniques.
On the transmission experimental board (DL 3155M60, module B), the implementation is around AD 633, analog multiplier (used for modulation/demodulation, phase detection, as voltage-controlled amplifier/attenuator/filter).

Interactive ports (inputs and outputs ports) are related to:
- information (modulating) signal - V_m;
- carrier signal- V_c;
- output signal at different levels (V_{am}, V_{am1})
- controlling the type of modulation (DSB modulation/ stop modulation);

	Please be carefully with input signal, and check with the parameters: • V_m= 3 V_{pp}, f_m= 5 kHz; • V_c= 10 V_{pp}, f_c= 100 kHz; where, V_{pp} is peak-to-peak voltage.

	Please be carefully with using ports: don't connect module's inputs and outputs ports together, manipulate carefully power cables in order to make proper connections.

	The ground terminal (the signal reference) is the same with the power ground terminal. The only precautions should be related to check for ground terminal continuity.

B - AM modulation with analogue multiplier DSB - full carrier

This is the first important experiment in data transmission process.
Of course, many other are important, or helpful, but this module shows us how to manipulate carrier in order to obtain transmission of data through modulating process.

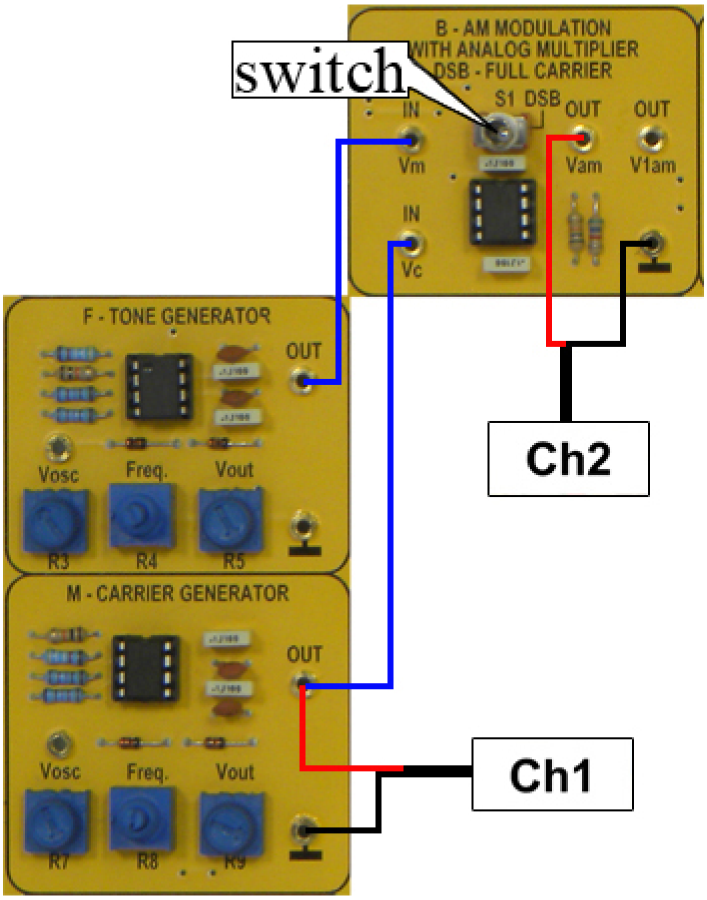


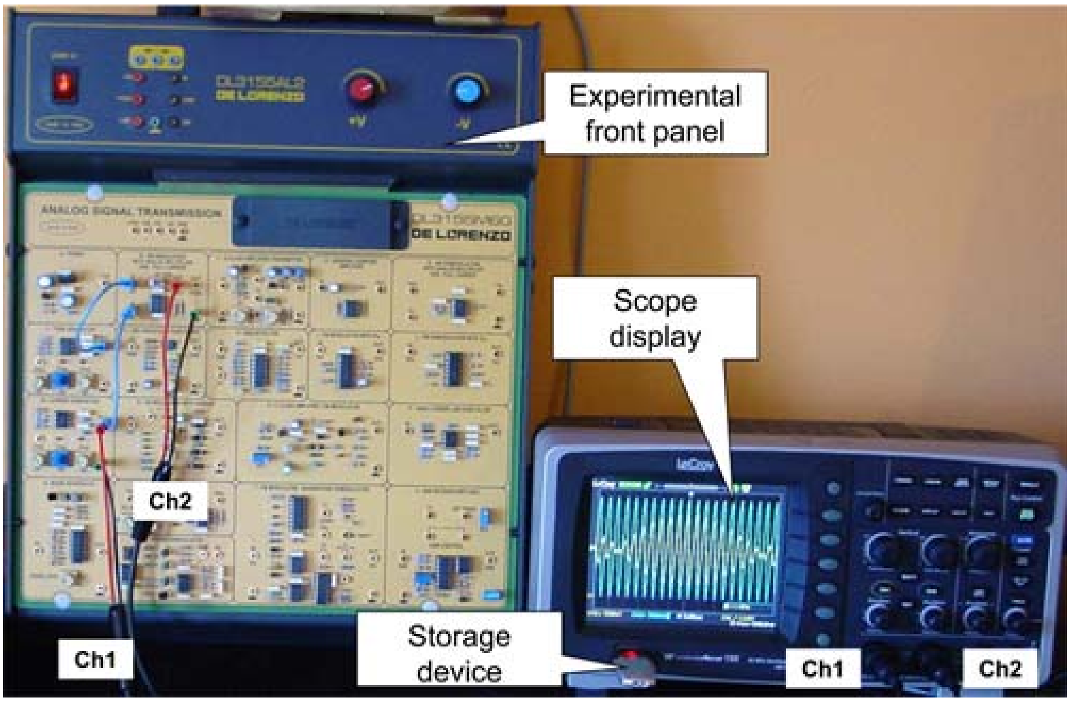
Figure 4.2 - The module is designed to exercise low pass filter characteristics

For this experiment, there are necessary three modules, as you perhaps expect.

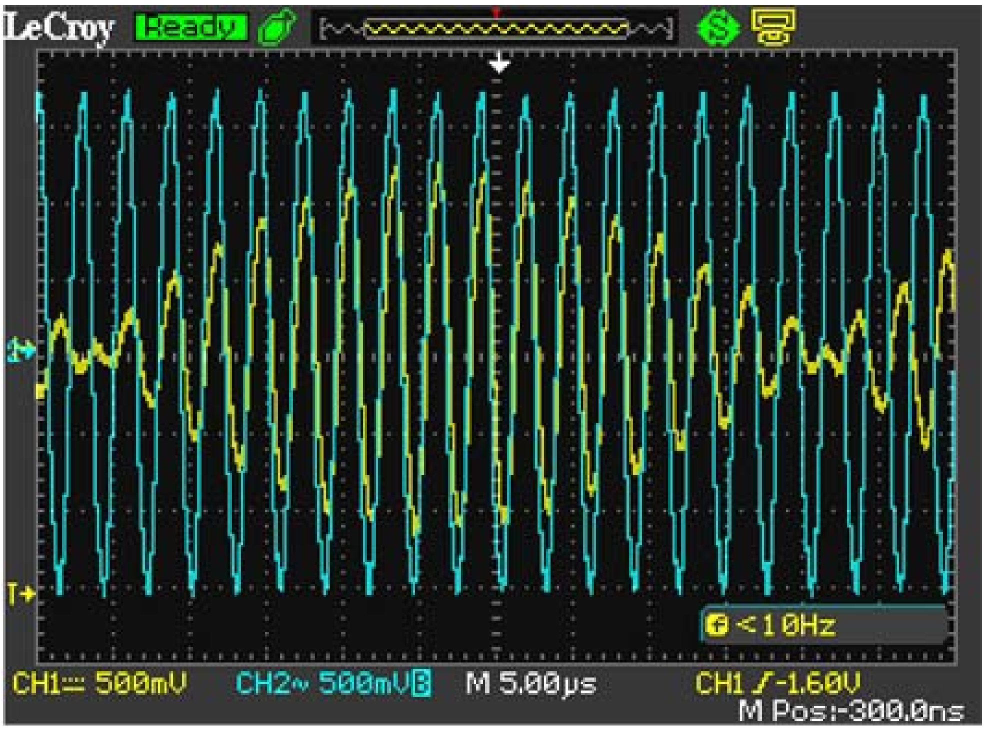
The carrier generator (module M) is injecting signal to the input (marked B) of AM modulation with analogue multiplier DSB full carrier (module B).
The audio generator (module F) is injecting signal modulating circuit (module B) through the specific input (marked A).

By following established local regulations and by taking proper length cables we connect step by step modules as upper figure shows.
For running this experiment, the main controlling features are *Freq*. knobs, and S1 switch (module B).

In order to figure out the frequencies shapes, and effects, we use two oscilloscope measuring channels (Ch1, Ch2). One channel is used for acquiring, and displaying carrier frequency, the other one is used for shown the modulated result.



Eventually, some extra facilities could be used into the process of measurements and present- an external storage.
In this case, the image from the oscilloscope will be nice digitally printed.



Experiment objectives

	Understanding the principle of amplitude modulation (AM). Understanding the waveform and frequency spectrum of AM signal and calculating the percent of modulation.

Understanding the principles of designing an amplitude modulator.

Measuring and adjusting an amplitude modulator circuit.

To provide a familiarization with the use of a multiplier-circuit unit to generate an amplitude-modulated (AM) signal, with an adjustable modulation factor (m).

To examine both time & frequency displays of an AM signal.

To measure the percentage modulation (m %), and the percentage of total power in both sidebands and in the carrier versus the modulation index (m).

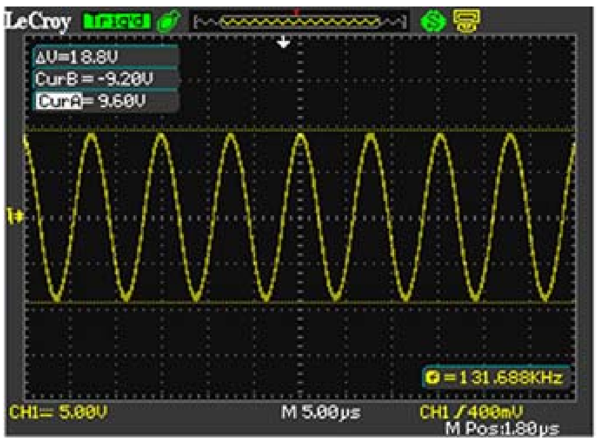
To investigate the use (& limitation) of envelope detection in demodulating AM signals.

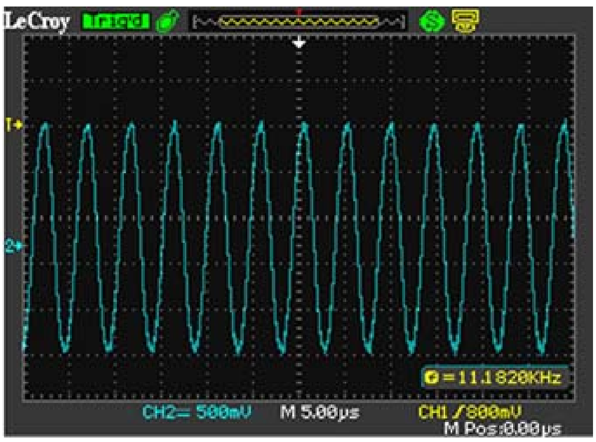
Additionally, we are focused on:
- Learning how to generate double-sideband suppressed carrier and single-sideband modulated signals.
- Learning how to test and adjust double-sideband suppressed carrier and single-sideband balanced modulators.

Tasks to study

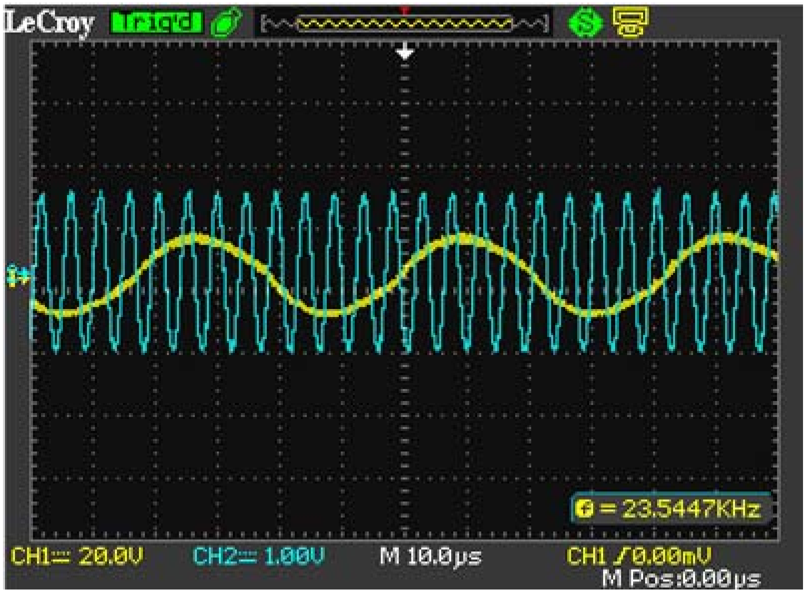
First of all, start running experimental panel.
Be sure to adjust the two frequencies (audio) and carrier, at the desired values. Eventually, control the quality of audio device with the PC speakers.

All the controls are well done with the oscilloscope. Be having displayed the data on the display, with minimum actions you know exactly the main parameters.





or,

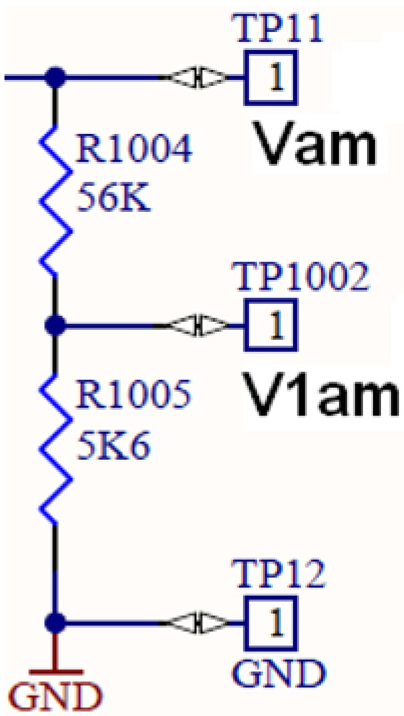


If you are satisfied with the nice pictures, note the main parameters of the oscillating signals. Additionally, please consult the theoretical part of the manual. Based on theoretical formulas, please make the calculations, for evaluating what kind of shapes you expect to get.

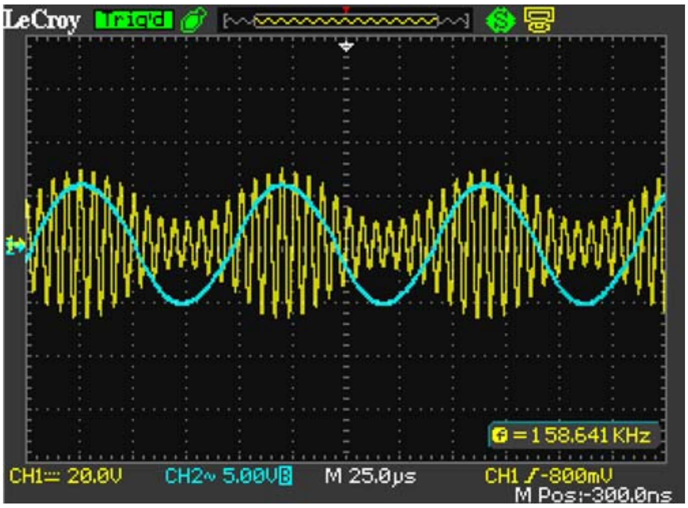
Now, please remember how many types of amplitude modulations we might have. Let's start with first type. Put the switch S1 (from module B) on the left, and let's see the shape of resulted signal, on the output V_{am}.

One of the biggest requirements for amplitude modulation is the amplitude control, which is hardly influencing the quality of the modulation- you know one unpleasant situation: over modulation.

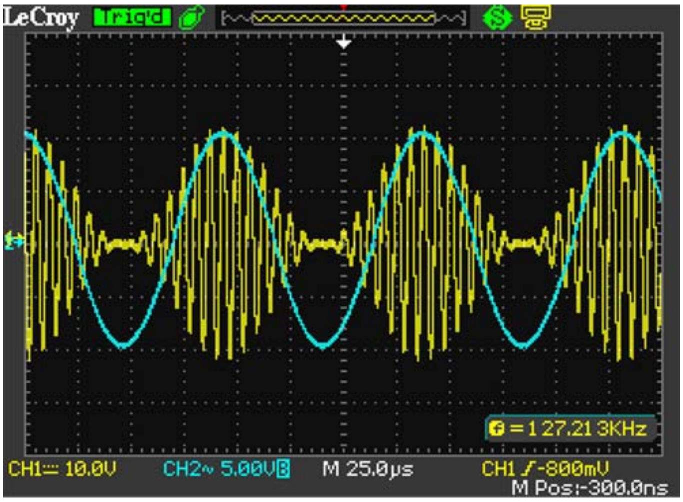
In order to have well control with the amplitude signals, there are some knobs for such adjustment, but also there is a voltage divider at the output of some modules. Here there is such implementation.



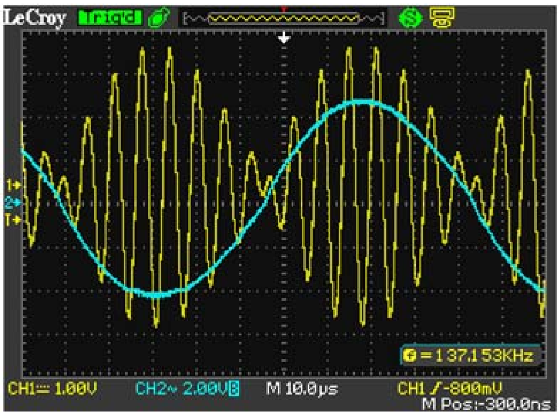
Let’s visualise the acquired data on the oscilloscope. The very low frequency is the audio signal, and high frequency (158,641 KHz) is modulated signal.

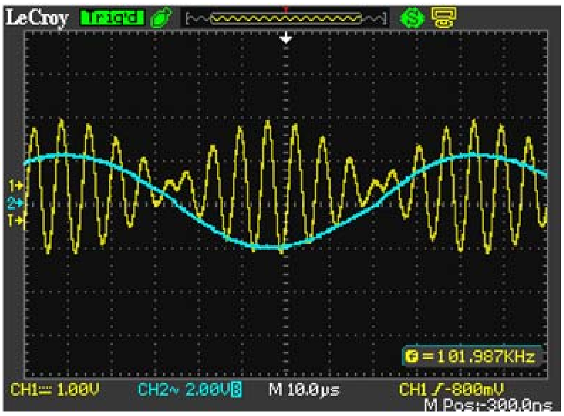


Please remember the big disadvantage of this type of modulation.
Now, play with amplitude knobs of the oscillators (audio and carrier). Did you get such image?



Please note, how, and when you got this situation, because, in latter experiment, probably you will get bad results, and one of the reason might by the actual situation.
Let’s go with the second type. Put the switch S1 (from module B) on the right side, and let’s see the shape of resulted signal, on the output V_{am}.



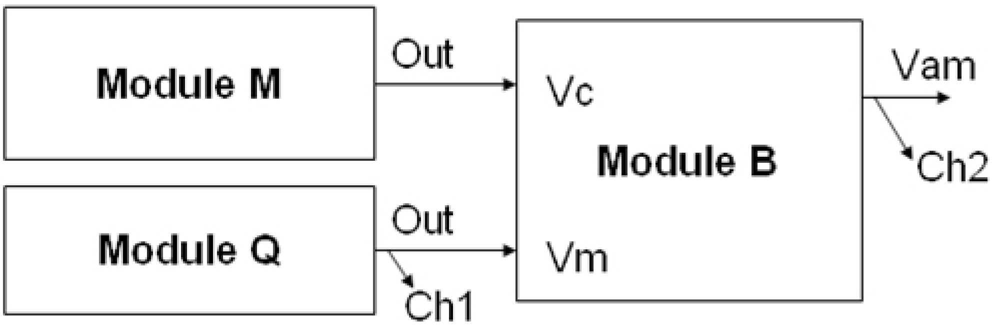


Do you get differences when you change the oscillating parameters? I hope not because we didn't. No any distortions! Again, the very low frequency is audio signal. Please remember another big advantage of this type of modulation (DSB modulation).
Now, let's imagine that we are using not an audio generator, but we are using a football speaker signal.

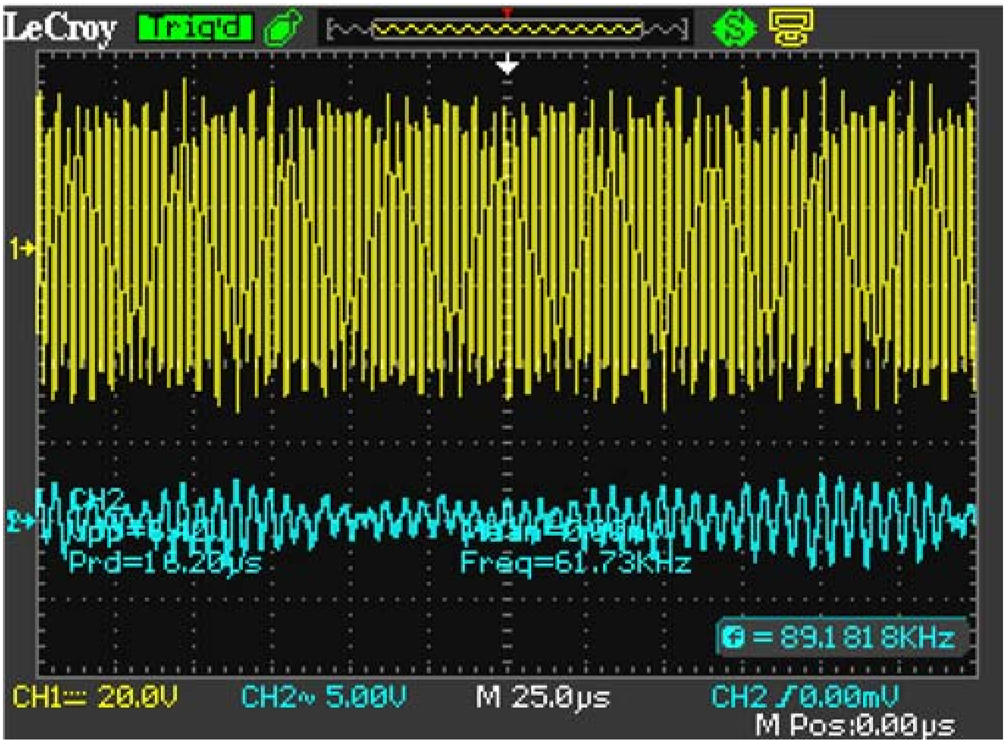
⚠	For such experiment extension, please take care with the level of the signal. The input of the AM modulator is limited to 3 V_{pk-pk}.

In order to simulate such situation we are using the noise generator module (module Q). The connections diagram between modules is shown next.
For connecting the noise generator we have to use a bit longer cable.

Don't forget to use the same colour for the cable.

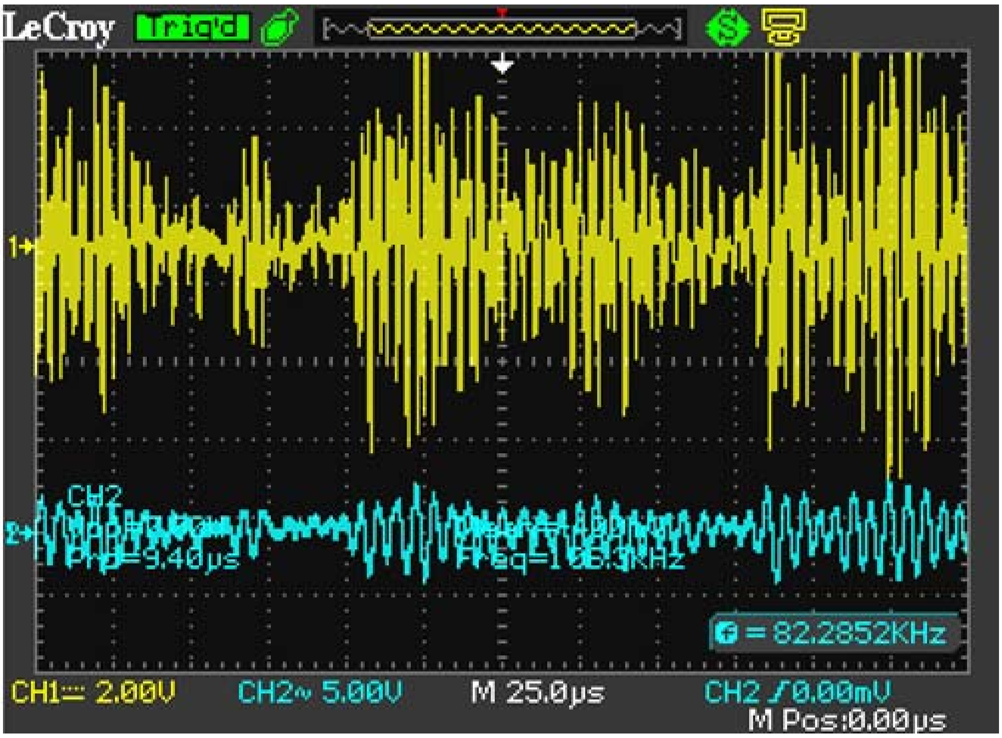


The oscilloscope measuring cables are connected as upper figure mention: Ch1, Ch2.
Let's see what it happens.



When want full modulation without cleaning DC part, the upper figure shows modulated signal. The lower signal is collected from noise generator, and the upper signal is modulated signal.

When we use DSB modulation, the signals shapes (noise and modulated signal) look like next figure.



So, in second part of the modulating experiment we have used noise generator.
This electronic module is very useful because, in first instance we simulate an audio signal (a speech), and also we can use it as disturbance for any data signal.
By the way, did you check the noise output with PC speaker? Please do that, you will be accustomed with controlling the quality of audio devices.

Quiz. Test yourself and be ready to accept new questions

Please make records for fixed values of Vc, Vm, fc, Fm. Then organize your record in a way you might get well conclusions. We recommend you a table with a header like next one:

Fm	Output waveform	Output signal spectrum	M [%]	Notes
8 kHz				
10kHz				
……..				

In your experiment, which method might be used to measure the modulation index of an AM signal with a very low percentage modulation? Explain why.

As far as you know the AM broadcast station technical standards specify that the % of modulation must be maintained @ 85 - 95%; comment on any possible disadvantages (based on your experimental results) that will occur by ignoring this specification.

D - General purpose amplifier

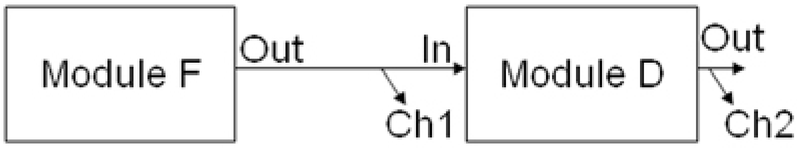
Speaking about extensions of the experiments, some times is happening that the data signal is two small.
A general purpose microphone generates to the output port few mV (for instance 5 mV).

Remember, the input of the modulator from previous experiment accepts data signal up to 3 V. How can we manage such small signal (almost 1000 times smaller)!?
For that reason, we need an amplifier simple, without many restrictions or extra requirements.

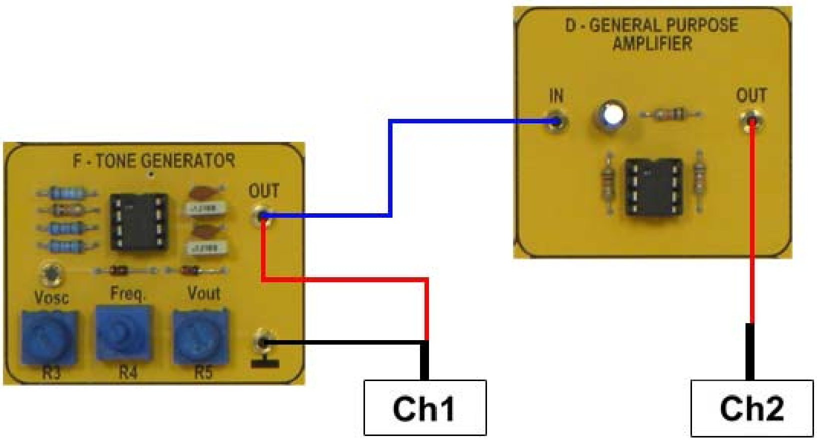
The module D- general purpose amplifier comes to fulfil such simple requirements

Amplifying audio signals

Let's connect electronic modules like in next figure:

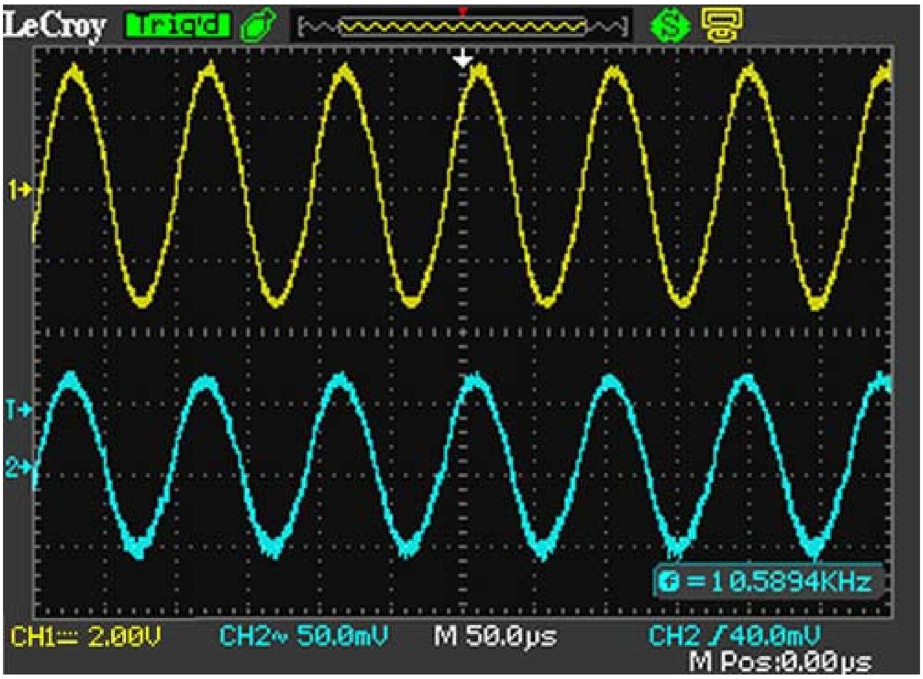


Then check if you make cables' connections like we did.



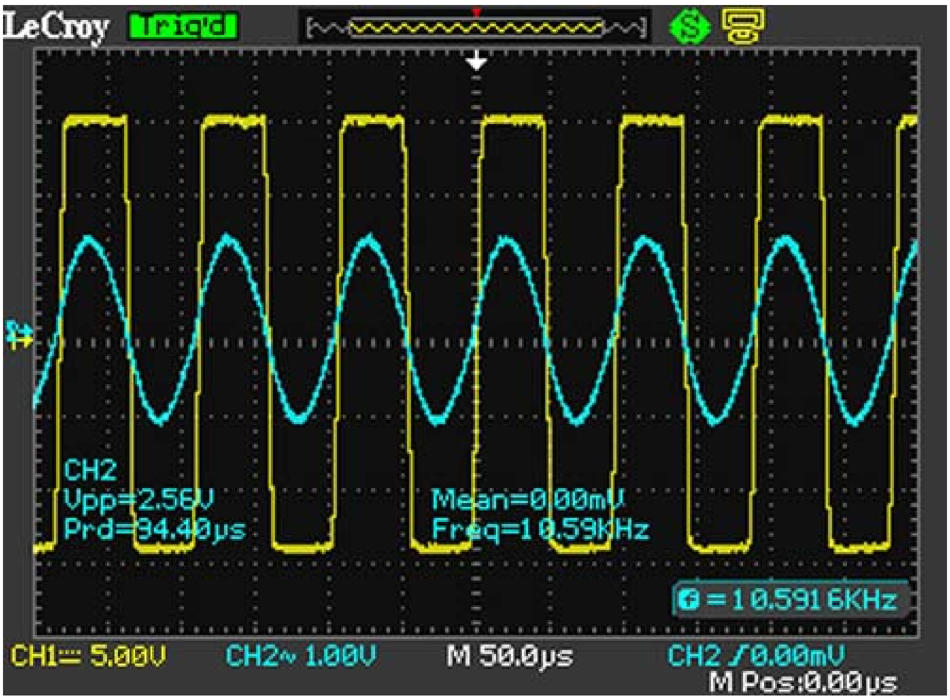
Now, we have well exercise in running experiment, in using oscilloscope. Let's experiment with measuring how the audio signal is amplified.
For better figuring the measured shapes, we measure in the same time in channels 1, and 2.

Let’s see if you get the same frequency shapes.



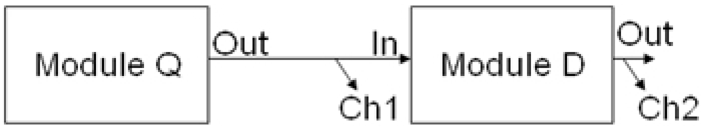
Please pay attention to the displayed values on the figure.
We will see that de lower signal is original audio signal (the oscilloscope scale is 50 mV/div), and the upper signal is amplified signal (2 V/div).
What is the ratio?

Now let’s manipulate with new output value of audio signal.
The result is next



Can you explain what this is? What is happened? Make your comments please!

Following the same procedure let’s connect other modules.
Next figure shows that.



Please exercise the next part of experiment in the same manner as previous experiment.
Make your comments please!

We know that you are very ingenious.
Excellent! But please don’t think to connect the output of the general purpose amplifier a loudspeaker.
I also wouldn’t recommend you to test this with available PC speakers.

Quiz. Test yourself and be ready to accept new questions

A	What is de meaning of amplification factor? Calculate amplification factor of the module

B	Please find other definitions of amplification factor. Where they are used for?

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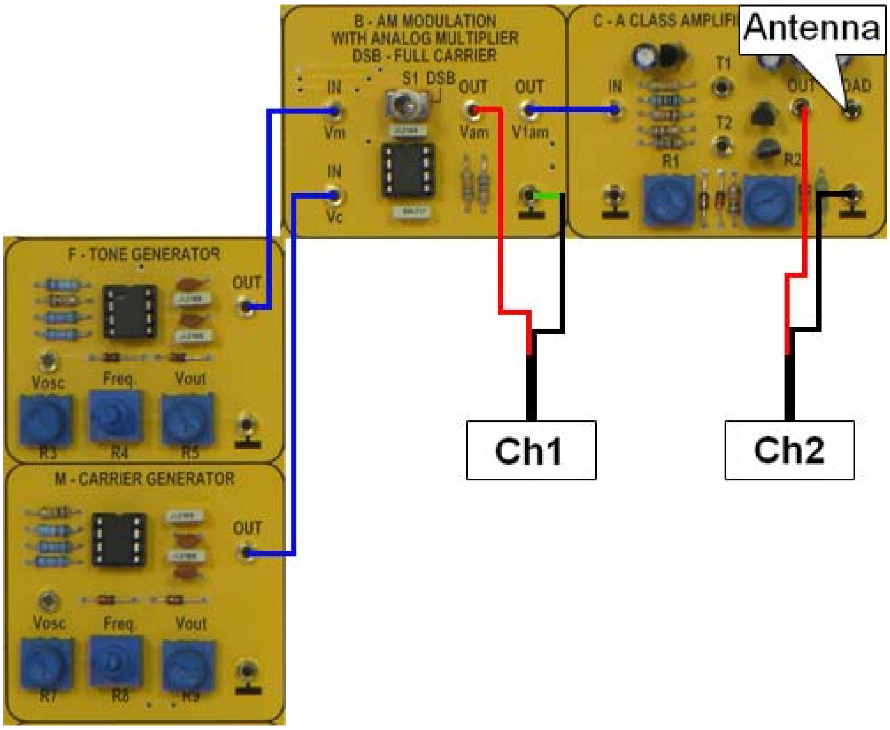
C - A class amplifier transmitter

We want to invite you to imagine a radio receiver. It receives information sent over the air. But who send this information? As you know, a sender broadcasts information.

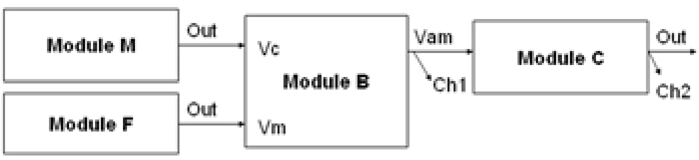
For that matter, in one of previous experiments we have used a carrier together with an audio generator as inputs of an AM modulator.
The output of the modulator has to be connected to an electronic device capable to send information over the air. Such device is a specific amplifier that has the load an aerial antenna.

A typical amplifier module for transmission experiments is designed as A class amplifier.
Let’s go back to experimental module.

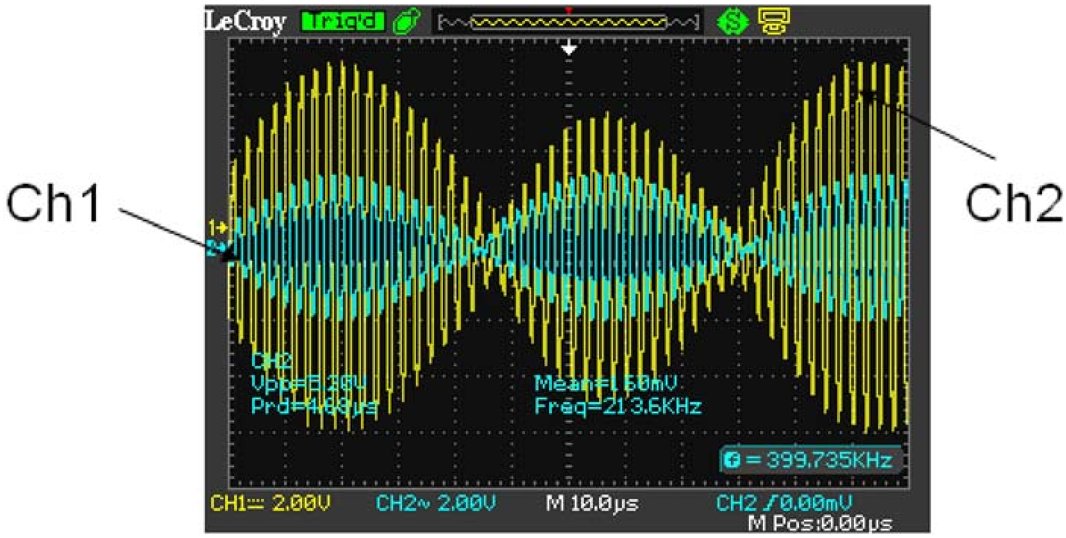
The experiment includes one more module - C (an A class amplifier).



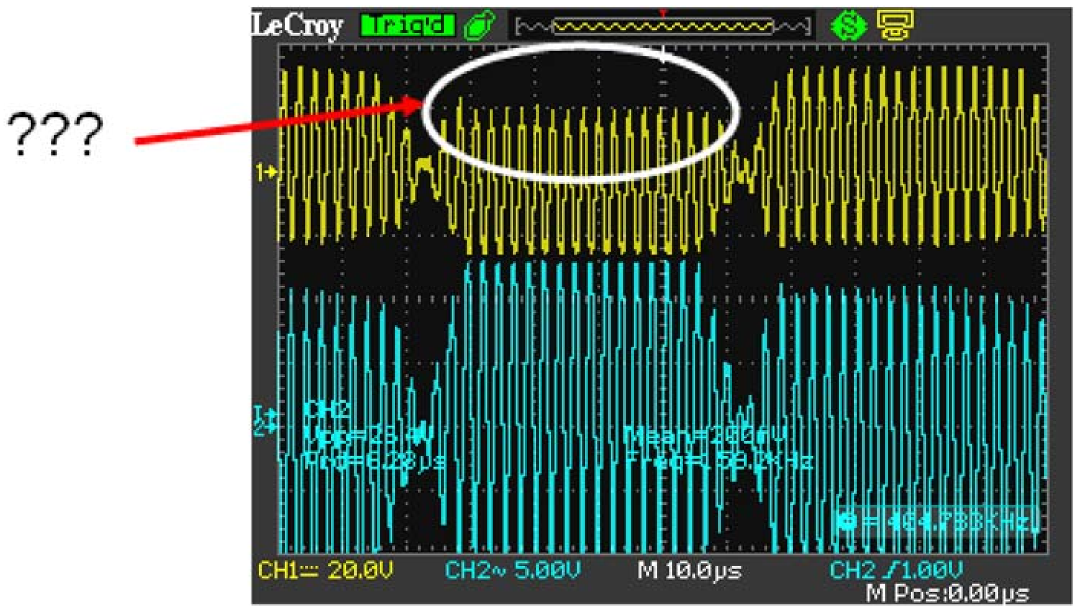
The block diagram of the connections is shown next.



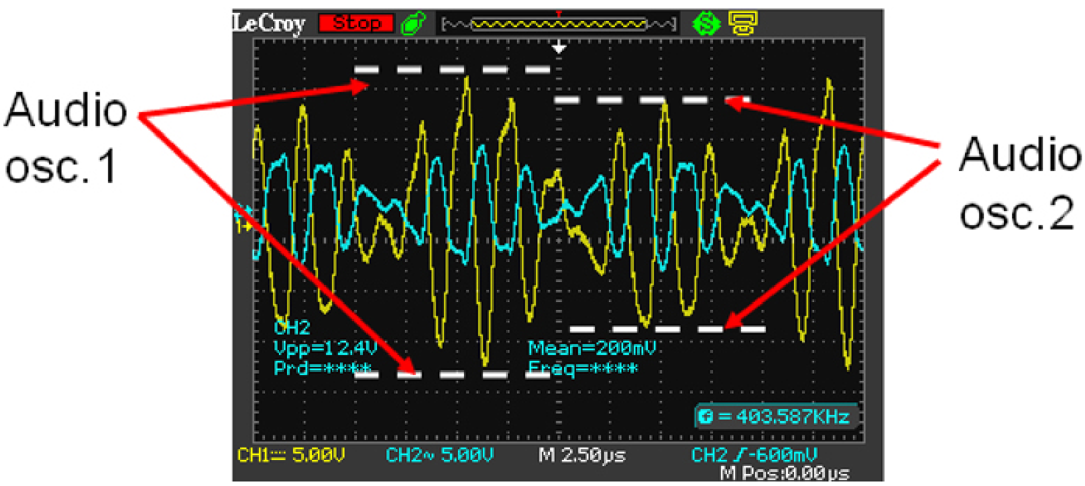
The next scope window figure shows the shape of the output signal. The Ch1 channel records the output of the module B (Vam), and Ch2 channel records the output of the module C. By using the superposition method we could analyze the work accuracy of the electronic modules.



By the way, did you get shapes of the signals like next figure? We did.
Can you analyze them, bring comment related, and make recommendations for avoiding such situations?



One more, did you observed differences between two successive oscillating envelopes of the A class amplifier?

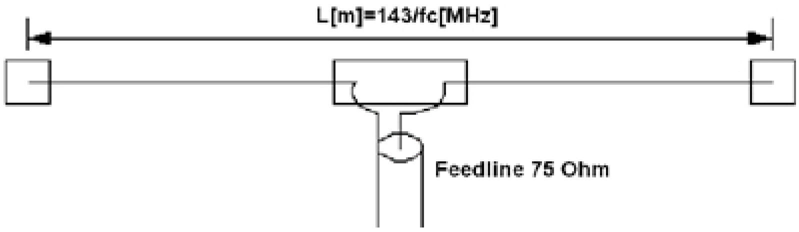


Can you analyze them, bring comment related, and make recommendations for avoiding such situations?

On the other hand, we want to review the classified frequency coverage, used in radio transmission:

- FM 87.5 - 108.0 MHz
- ***LW 153 - 513 kHz***
- MW 520 - 1710 kHz
- SW 1711 - 29999 kHz

Today’s our carrier is settled to 399.45 kHz. So we are inside of a Long Wave radio system.
The principle of designing an antenna for this frequency is shown in figure.
In other words, the length of the antenna dipole is 357, 99 m!!!! Not a problem!



Based on these explanations, you are invited to design and build such antenna.
Then extend it, but without any accidental electrical connections with other conductive parts.
With proper terminal, connect the antenna to the LOAD terminal of the module C.
Now you are ready to run new experiment with broadcasting audio signal on the air.

For more information, please read other references related to building, adjusting radio senders.
By the way, are you familiar with frequencies licensing?
Can you analyze the work of all modules, bring comment related, and make recommendations for work with them?

Quiz. Test yourself and be ready to accept new questions

Make a list of cautious actions for running experiments with amplifiers. There are particularities related to high frequencies?

Can you define the concept of load? Please find the differences between audio loads and high frequencies load.

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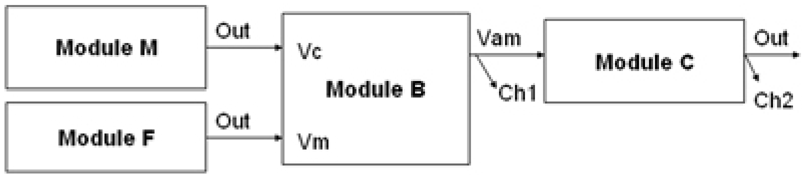
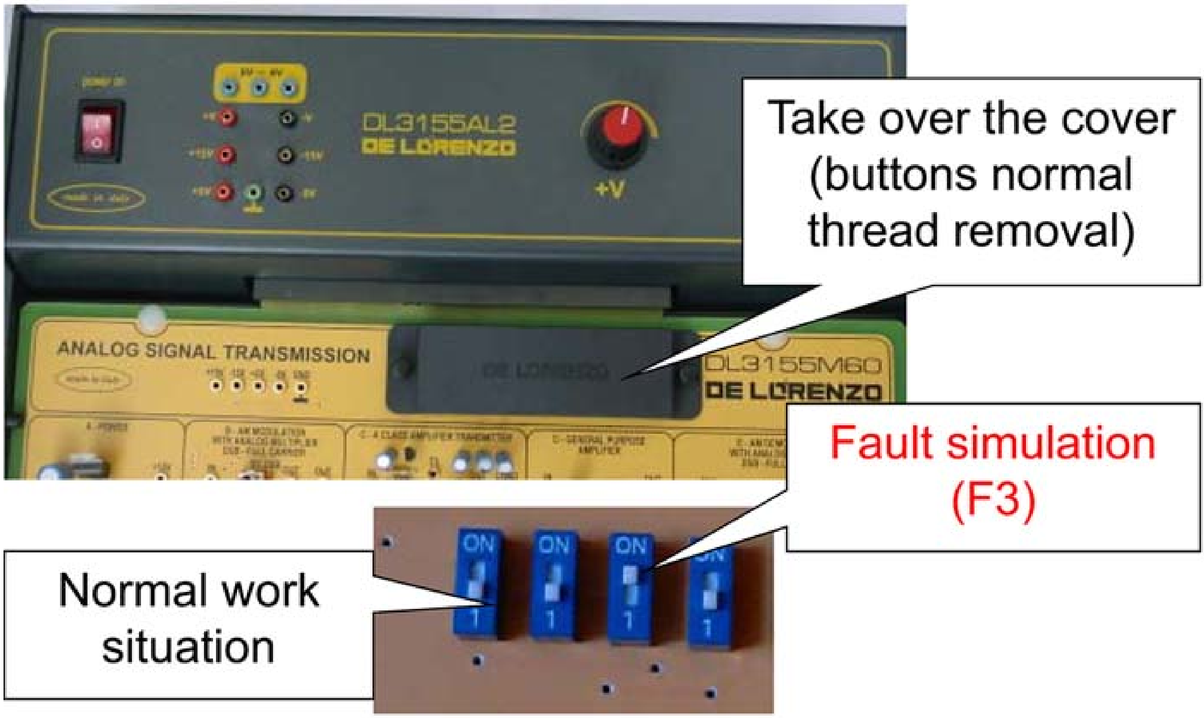
Fault Simulation

In the same manner, as previous situations we are simulating new situation.



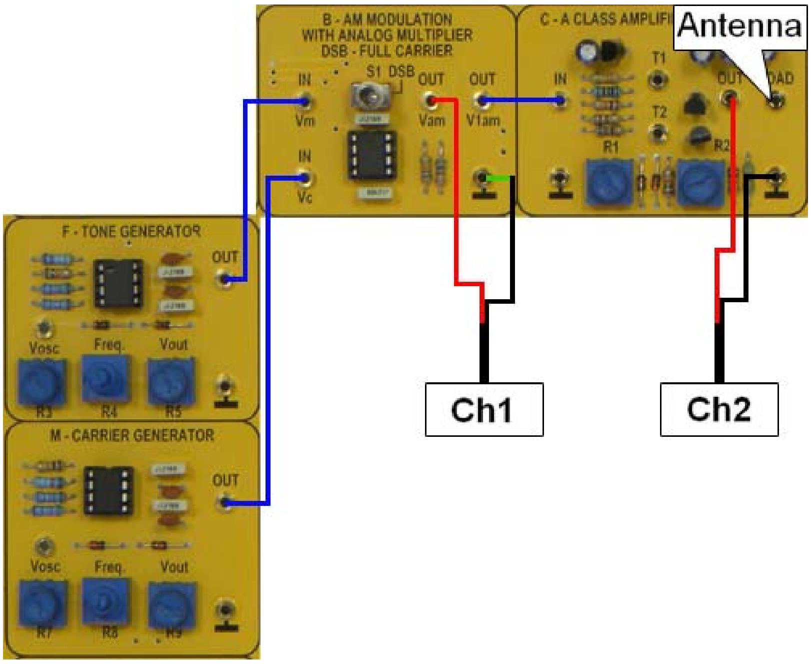
Now, we say that is routine to detect the fault in such few modules interconnected.
The fault scenarios are simulated with dip- switches.

The F3 is made by adjusting the switches like in next figure.

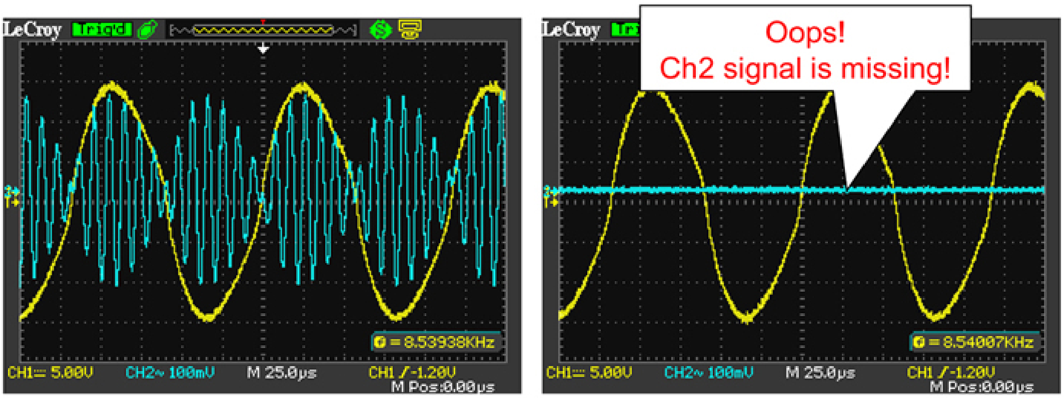


If the panel is connected directly to your PC via De Lorenzo TIME interface, then the fault can be inserted directly by clicking on INSERT shown below.

This simulation is affecting again the amplitude modulation- the A class amplifier transmitter.



In this case, in the middle of the experiment we will get signal on the channel 2 like in next figure.



Normal working situation

Fault situation

As in previous experiments, and fault simulations, we are in the position to start searching in different ways.

Where we start from?
By exception, in this case, first of all we suspect that the last module is malfunctioning. Because there are some test points in this module, a skilled user will try to check the presence of signals also in test points.

What we do next?
Another recommended action is to disconnect the C module from the module B., in order to be sure that there is not any influence between malfunctioning module and the other modules.

What about other modules? How they work?
In our simulation case, all other modules are working well. That's again very well! No other damages!

⚠	Beside the actual fault explanations, as we mentioned before, we would like to recommend you when you get some malfunctioning situations, as soon as you insulate the module with faults, please disconnect it from the other modules, and from the power supply if this action is possible.

Unit N.5: Understanding amplitude demodulation (AM demodulation)

Objectives:

- Understanding the principle of amplitude demodulation (AM demodulation)
- Detecting the AM demodulation
- Evaluating the quality of detected signal in AM demodulation
- Measuring and adjusting an demodulated signals

Requisites:

- Minimum level of communication techniques understanding
- Medium level of electronics components and devices understanding
- Understanding the amplitude modulation mechanism
- High level of health and safety risks understanding
- Communication systems theoretical manual DL 3155M60

Operative instruments:

- DL 3155AL2 or stabilized supply source
- 20MHz double trace oscilloscope
- set of connection cables

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Where we start from

	In previous experiment (Exp3) we where dealing with modulating a carrier with data signal (audio) through AM method. So, we put the foundation of radio sender.

On the other side, at receiver side, the basic idea is to receive such modulated signal, and to remove the carrier for efficient use of audio signal.
Both parts represent the dream of all radio amateurs around the world.

The main parameters that describe the demodulation process are:

- A_m - audio amplitude
- A_c - carrier amplitude
- f_m - audio frequency
- f_c - carrier frequency
- the spectrum of the audio signal at the output of demodulating device

For the practical evaluation, we would like to remember you the fastest formula for modulation index calculus.

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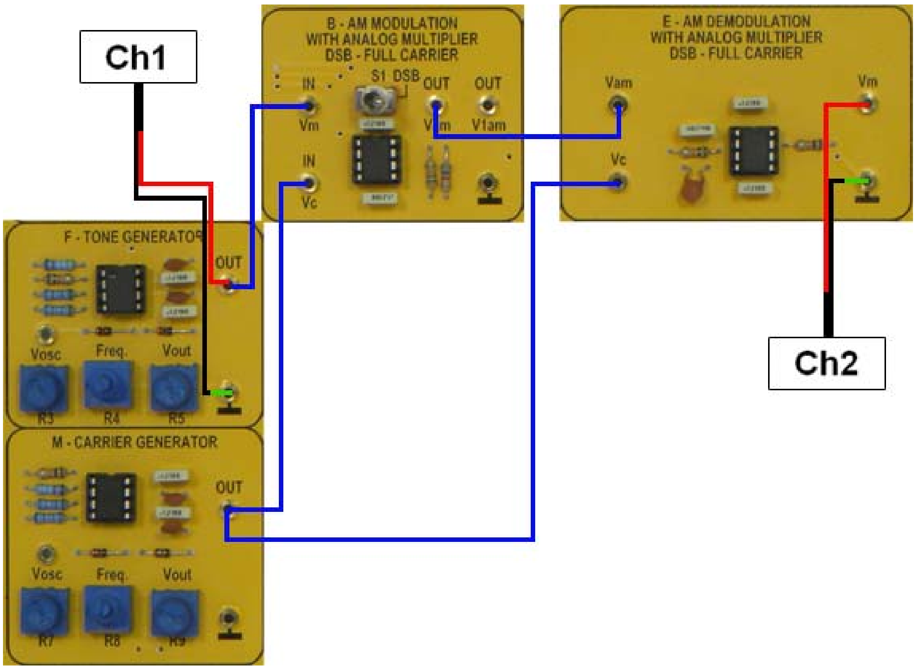
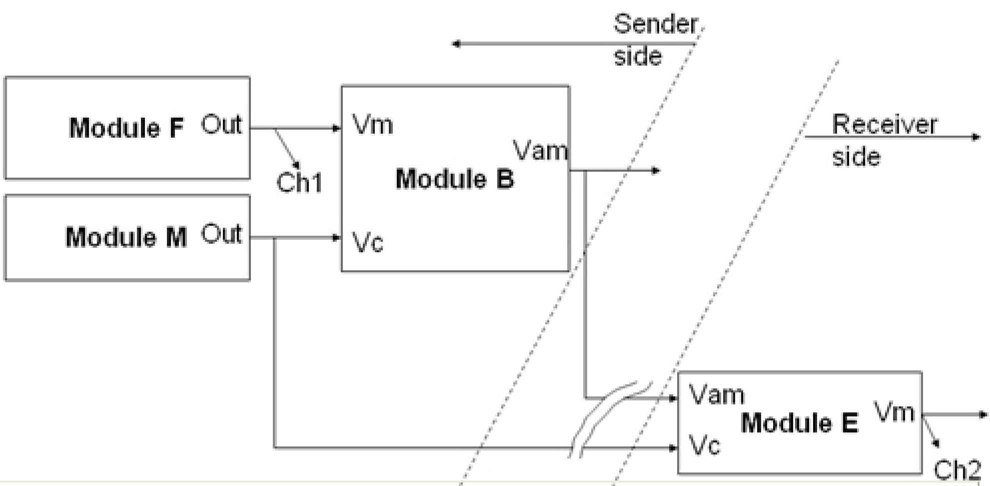
E - AM demodulation with analogue multiplier DSB - full carrier

Now, we are familiar with experimenting with De Lorenzo boards.
Let's build up the demodulating process. So, in order to experiment with demodulation, we need modulated signal, so previous experiments are welcomed and required.
We are also familiar with connecting electronic modules.
The experimental circuits are shown next.



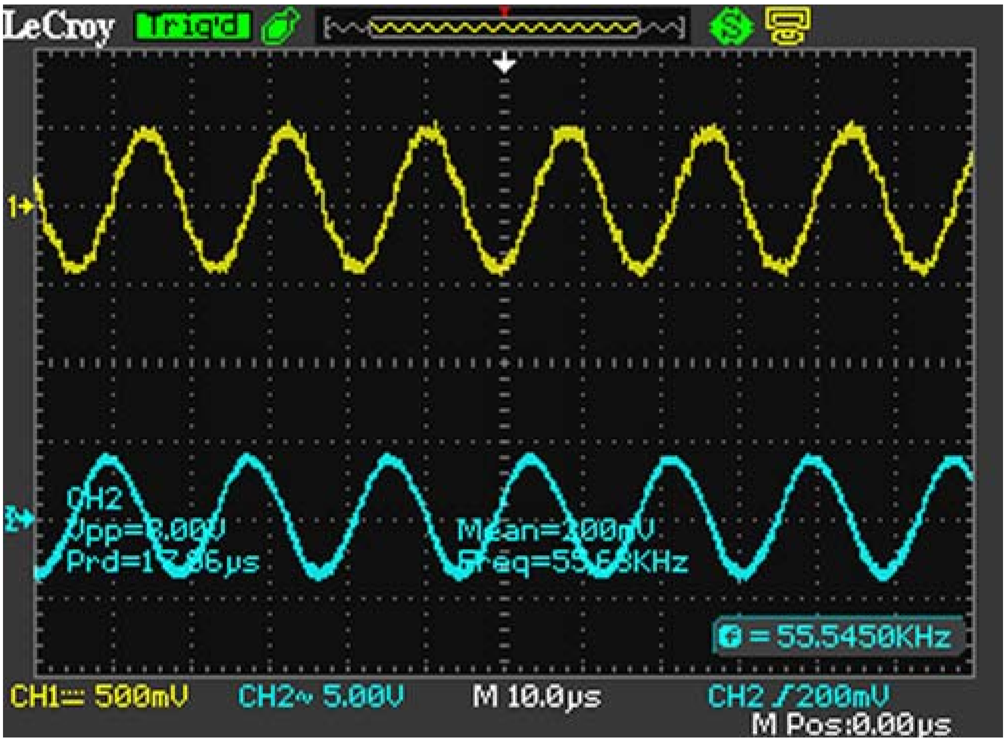
The block diagram connection is shown next. You will pay attention to the sender's, and receiver's side.
There is not place and time to talk about the area between dashed lines.



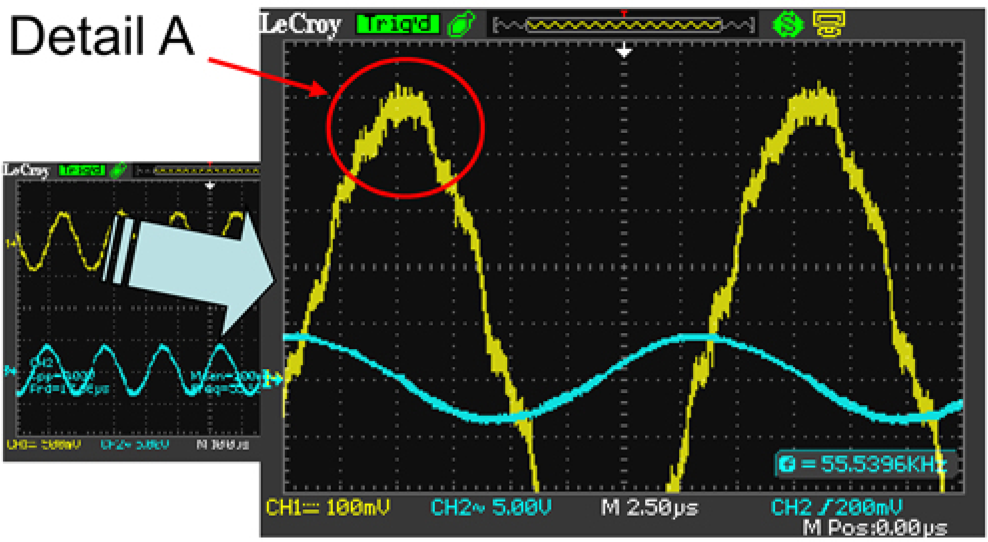
As you understand, it is the simplest experiment for demodulating AM signal.

Let's run it and see what we get on the oscilloscope.
There is the place for a little indication: most of oscilloscopes have a facility to adjust displaying parameters, let's say- an auto mode of measuring.

In order to save time during acquiring data, try to use such facility. We did it, and we got next signal shapes.
T ake a look to measuring scales, and to shapes of the signals. If you have mood for that, test the result with your PC speakers.

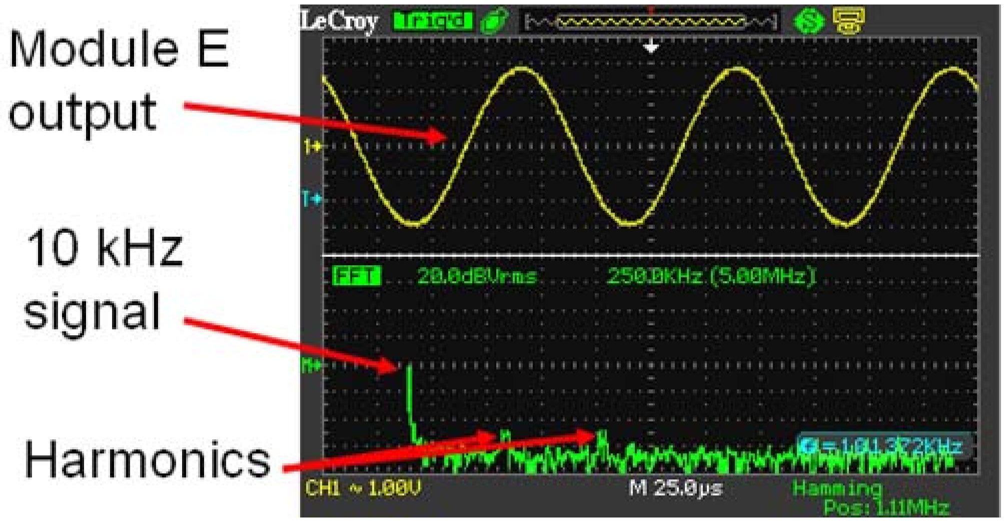


Please measure signals, analyze the work of demodulator, bring comment related, and make any observations related to the obtained audio signal.
Well, now let’s go dipper into received signal.
Probably, when you tested the audio signal with PC speakers, you found some unpleasant sounds. You have right! If we make a zoom of the received signal (the output of module E), we will see a picture like next one.
See the detail A!



It is the time to analyze what is here. There are some methods to do these.
The easiest way is to use a spectral analyzer that will analyze all signals that composes such signal.

Most of modern oscilloscopes have at list a FFT math function. Find it on your scope, run it and analyze the result.
If you are not very familiar with scope functions, please take out the measuring cable for original audio signal, in order to be sure you will analyze the E module output.



My eyes tell me that the audio signal result is OK (the module E output).
The FFT function tells me that the signal is composed by a 10 kHz signal (20 dB Vrms), and some harmonics.
Now, depending by next processing signals, some time this simple processing stage is enough, or not.

Quiz. Test yourself and be ready to accept new questions

A. Please review the advantages of using DSB techniques of modulation/demodulation.

B. Who influences the demodulated signal parameters?

C. What kind of measure do you propose in order to ensure the quality of demodulated signal?

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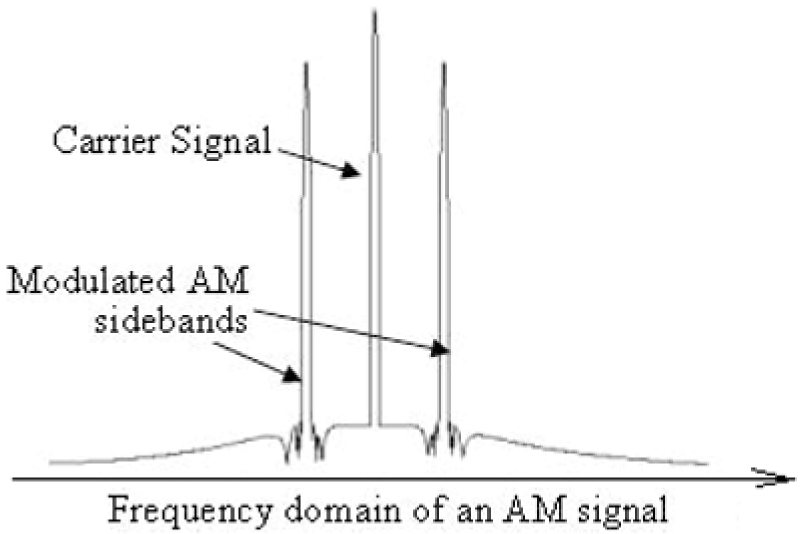
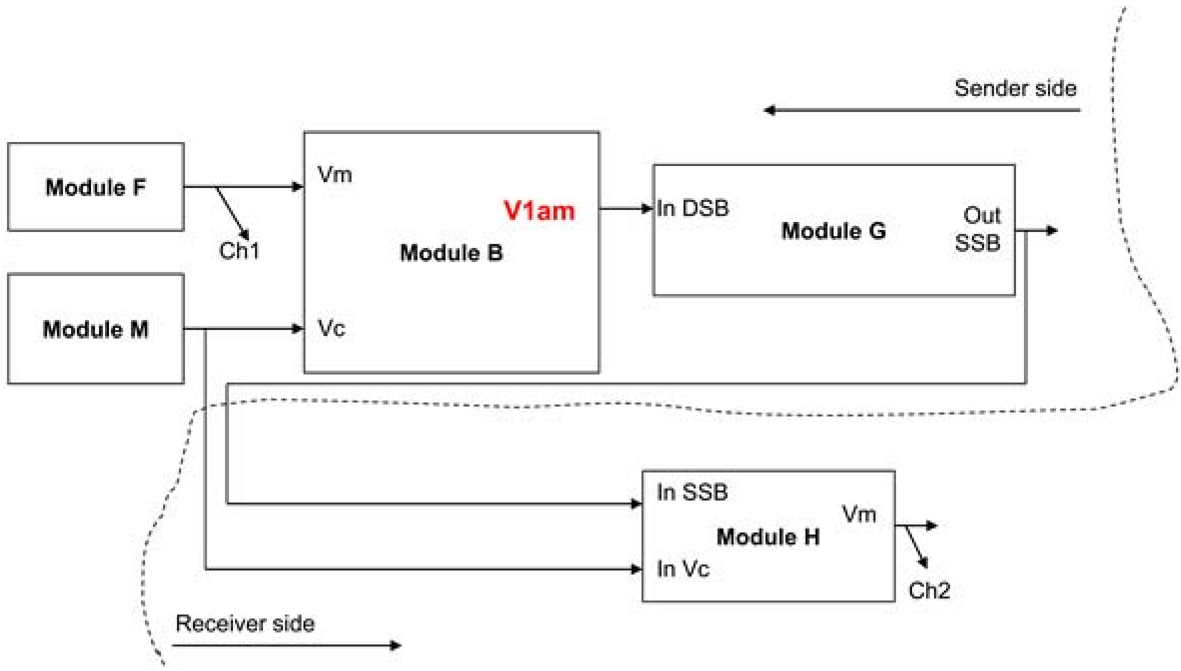
E - SSB - detector

Now, we know what does mean AM, DSB, SSB.
Now, it is the time to understand how to manipulate signals for energy saving in transmission process.
One method is based on adapting the modulation type in accordance with transmitted data. For instance, there are a lot of applications that uses SSB modulation.
So, now we are trying to understand how to use successfully the SSB modulation.

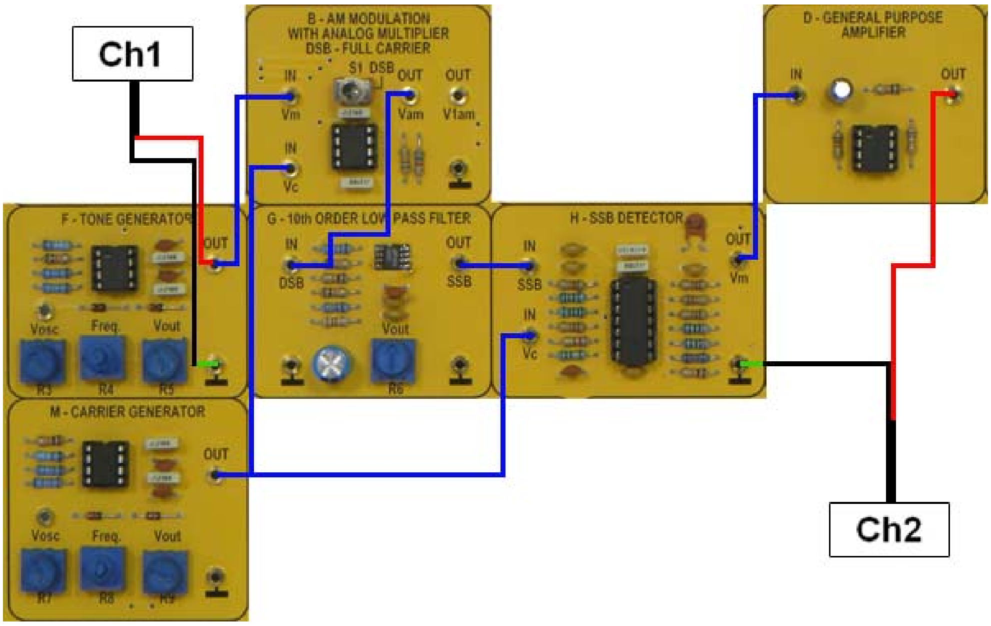
By using De Lorenzo experimental board DL 3155M60 we have modulated signal with DSB- full carrier.
In order to experiment using SSB detector, we have two sidebands, and it is necessary to remove on side of the modulated signal.
Next figure shows the "positions" of signal for better understanding how to remove one AM sideband.



For our experiment we use a 10th order low pass filter- module G to remove one side of AM signal. After that we can use a SSB detector.

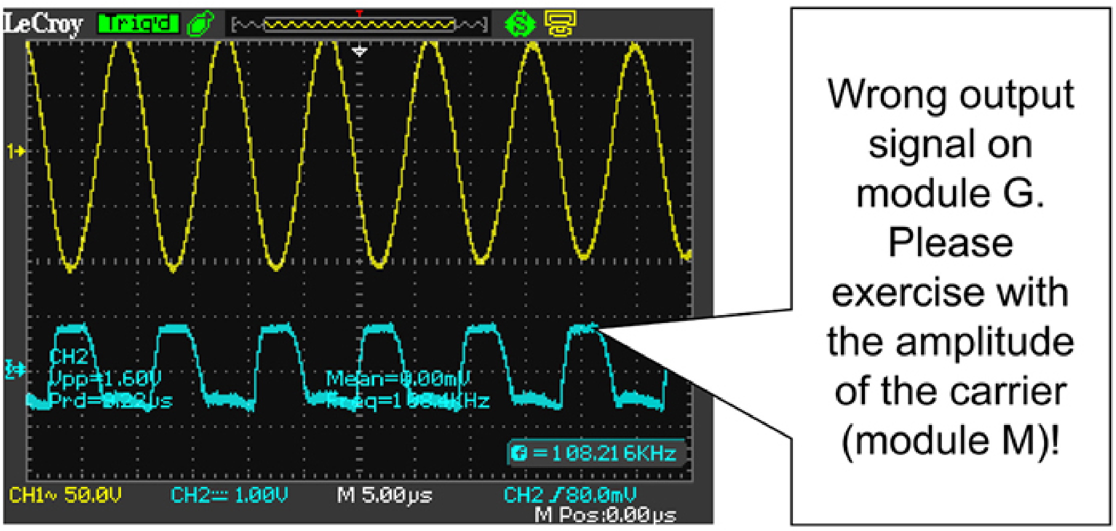


Well, remember, we want to recuperate the audio signal that we have used in modulation process at the highest quality.
That's why we have to compare extracted audio signal with the original one.
The connection diagram of modules who are involved in this experiment is shown in upper figure.
The proposed experimental panel looks like next figure:



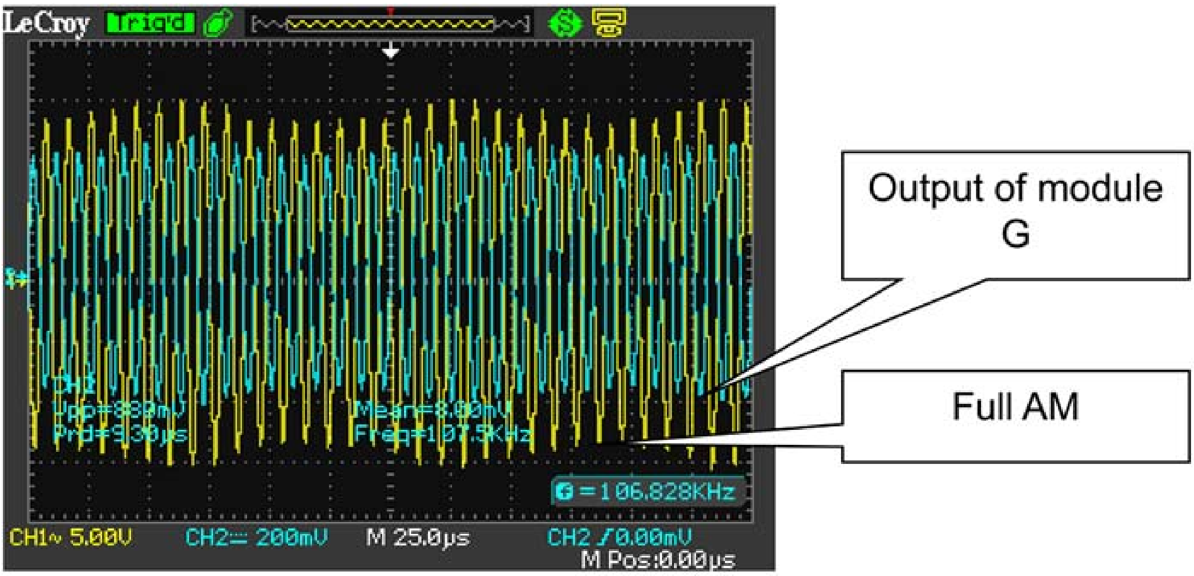
	Let's run the experiment. From the beginning, I would like to prevent you that we need to test results almost at borders frequencies of audio generator- 4 kHz, and 50 kHz. The reason of this is to test the adjustment work of 10th order low pass filter.

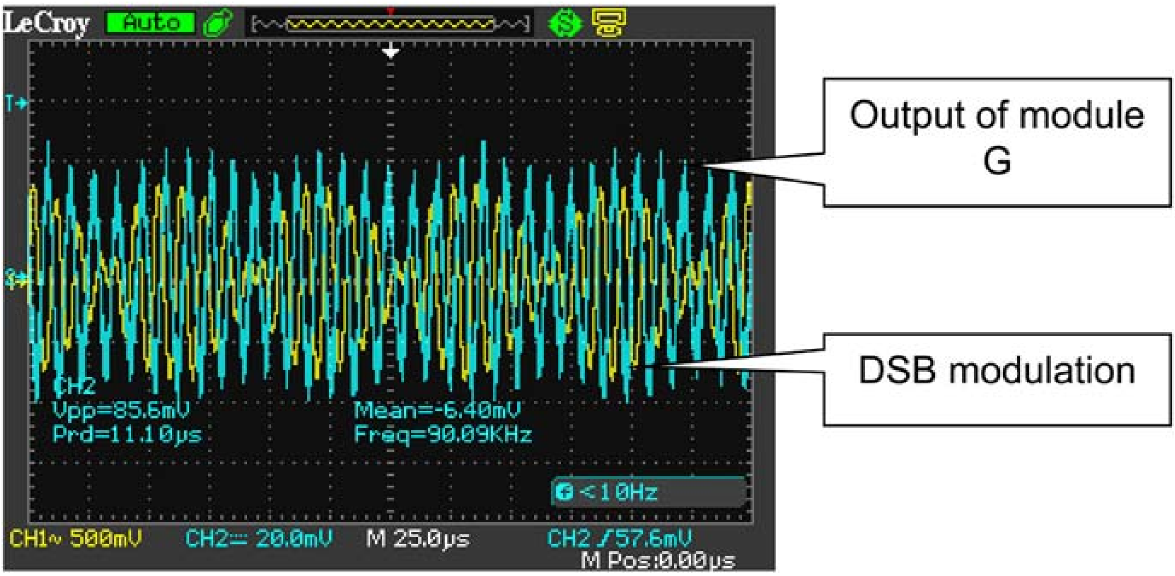
	First of all we exercise removing sidebands. By switching between S1 and DSB from module B and testing the results on the output of module G, understand how to use 10th order low pass filter. Please exercise this, by adjusting audio generator frequency. Adjust also the level of carrier in order to work with fine oscillations through the electronic module.



After taking some precautions, we can play with audio generator frequency, and module B switch.

By adjusting also the output voltage of the filter, we can analyze its effect over the modulated signal.





Now, we are sure that we have well SSB modulated signal.

It is the time to run the SSB detection experiment.

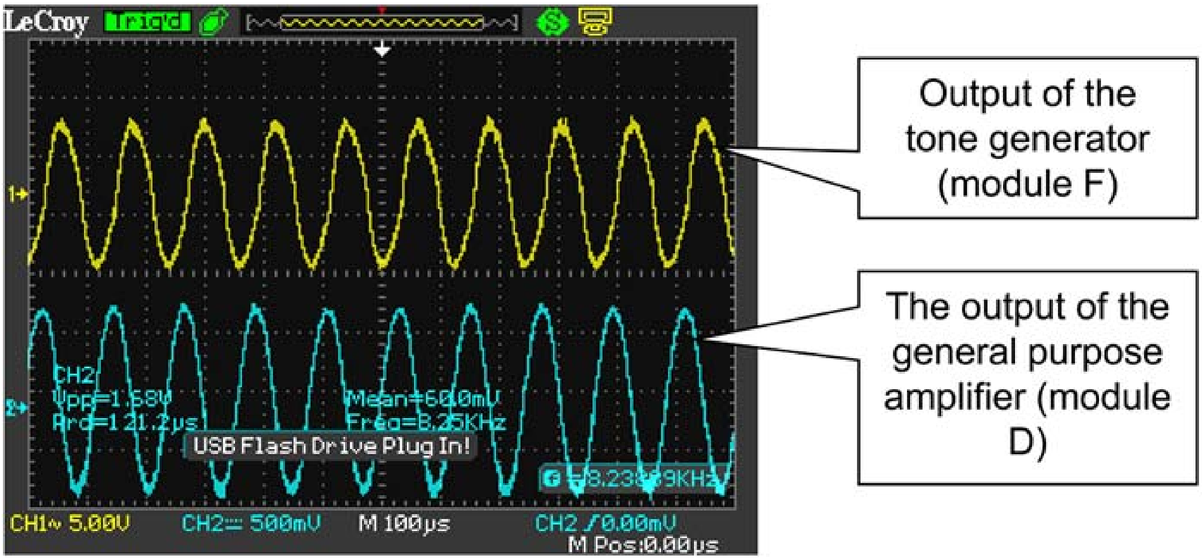
A particular detail: on this picture you will see that the output of module H is connected to the input of module D; the second measurement channel is connected to the output of general purpose amplifier.
I am kindly asking you to justify this action.

Please remember that we need to check some experimental situations:

- the level of audio signal;
- the level of the carrier;

- the border frequencies of the audio generator;
- the border frequencies of the carrier;
- the output level of the filter

The list of the experimental situation is not close, but it is recommended to check at list these, in order to be sure about SSB transmission process.
When you acquire the signals (the output of module B, and the output module D), after proper adjustments of the knobs, you will get some concluding figures.



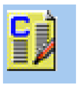
Are you satisfied? I do!
The audio signal, at both borders of tone generator is well recuperated.

Please exercise with your experiment, in order to get nice shapes of the signal.
Please also do some FFT processing of the output signal of the SSB detector in order to be sure that the signal is accurate

Quiz. Test yourself and be ready to accept new questions

Please review the advantages of using SSB techniques of modulation/demodulation.

Who influences the demodulated signal parameters?

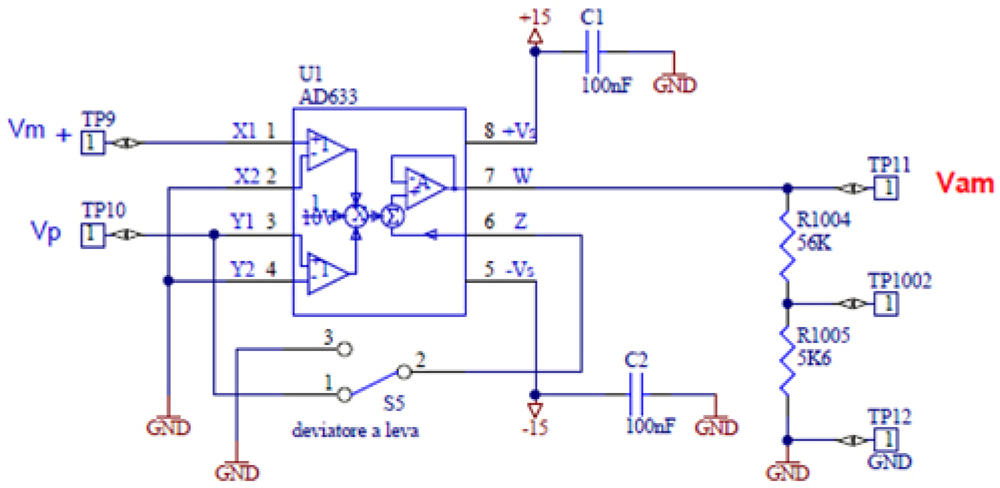


What kind of extra measure do you propose in order to ensure the quality of demodulated signal?

U - SSB receiver with AGC

We are now familiar with AM, DSB, SSB.
We should remember that, the gain control is a method for adjusting the receiver sensitivity for the best reception of signals of widely varying amplitudes.

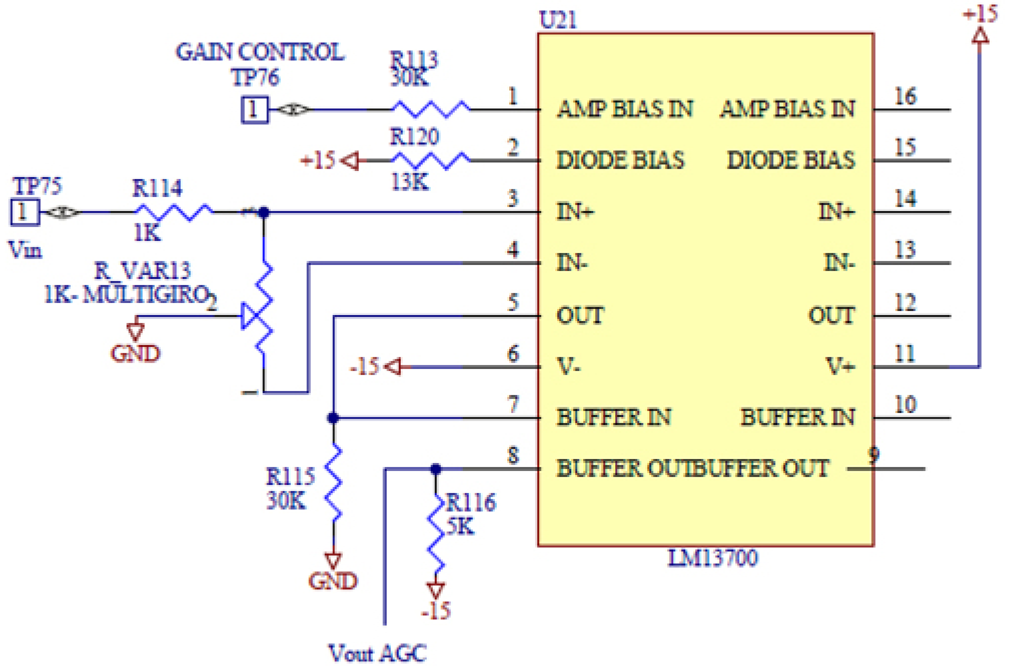
The complex form of automatic gain control (AGC), or instantaneous automatic gain control (IAGC) is used during normal operation.



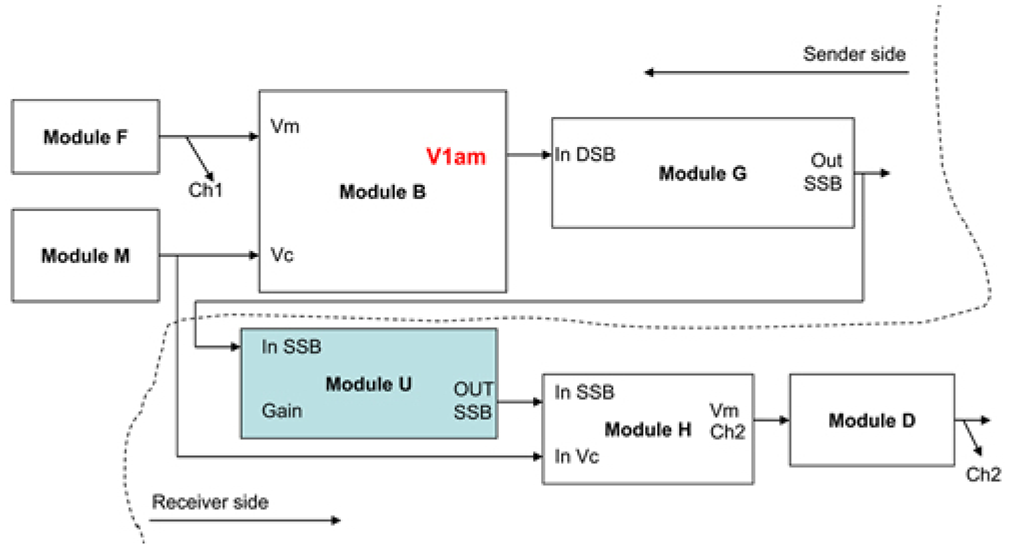
By analyzing the electronic circuit of AM modulator with analogue multiplier, you will observe that the output has two sockets Vam, V1am.
They belong to a voltage divider (1/10), which is used in our experiment for simulating low level of RF signal.

The experimenting board offer the possibility to experiment also AGC for low level of RF signal.
We are continuing the experimentation from previous experiment, by adding a ACG facility from module U.
The electronic module is built around integrate circuit LM13700 Dual Operational Transconductance Amplifiers with Linearizing Diodes and Buffers.

For our purpose is used only one operational amplifier. The diagram is extremely simple, with few external components.
The pin input 1 is used for controlling the gain for BIAS amplifier.
For the purpose of the experiment, we are using the variable DC voltage. Remember please- use the proper colour for the cable!



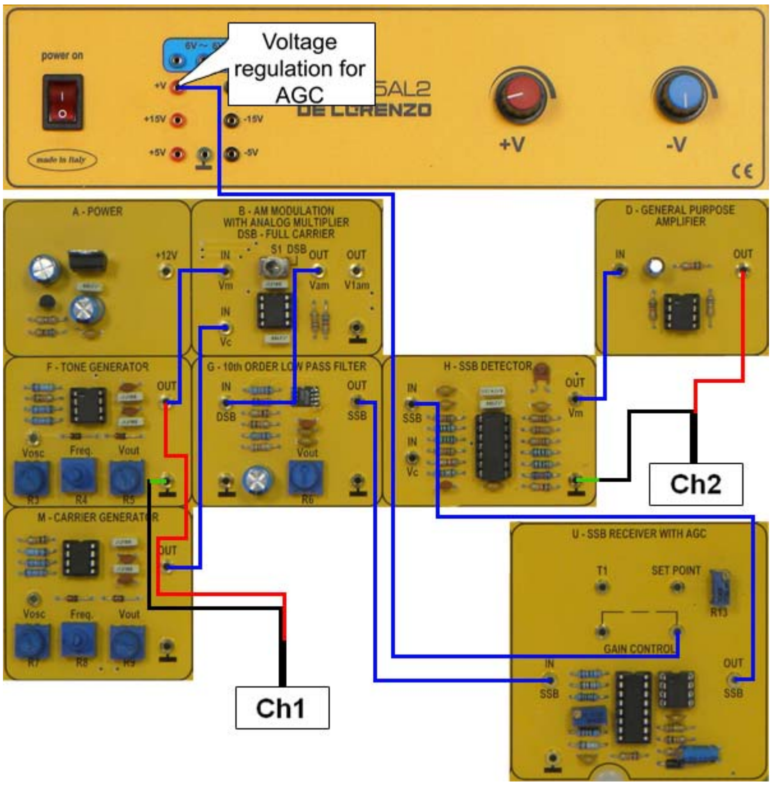
The block diagram of the improvement of experimental SSB receiver is shown in next figure.



The module is integrated in previous experiment diagram, with only minor changes (the AM modulated signal is collected from V1am, instead of Vam).
Again, the issue is to recuperate the audio signal that we have used in modulation process at the highest quality.

That's why we have to compare extracted audio signal with the original one.
The next figure shows the proposed experimental panel. As in previous experiment, we exercise removing sidebands.

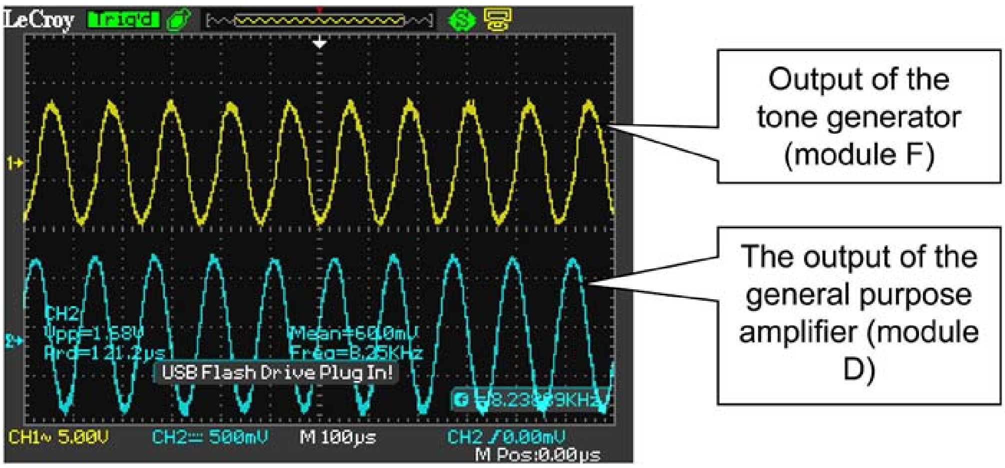
By switching between S1 and DSB from module B and testing the results on the output of module G, understand how to use 10th order low pass filter.
Please exercise this, by adjusting audio generator frequency.
Adjust also the level of carrier in order to work with fine oscillations through the electronic module.



Again, we should remember that we need to check some experimental situations:

- the level of audio signal;
- the level of the carrier;
- the border frequencies of the audio generator;
- the border frequencies of the carrier;
- the output level of the filter

Also in this case, the list of the experimental situation is not close, but it is recommended to check at list these, in order to be sure about SSB transmission process. When you acquire the signals (the output of module B, and the output module D), after proper adjustments of the knobs, you will get some concluding figures.



Again, we are satisfied with results.
The audio signal, at both borders of tone generator is well recuperated.

Please exercise with your experiment, in order to get nice shapes of the signal.
Please also do some FFT processing of the output signal of the SSB detector in order to be sure that the signal is accurate.

Unit N.6: Understanding frequency modulation/demodulation

Objectives:

- Understanding the principle of frequency modulation (FM)
- Understanding the waveform and frequency spectrum of FM signal
- Designing a frequency modulator using dedicated circuits
- Designing an frequency demodulator using dedicated circuits
- Measuring and adjusting modulator/demodulator circuits

Requisites:

- Minimum level of communication techniques understanding
- Medium level of electronics components and devices understanding
- High level of health and safety risks understanding
- Communication systems theoretical manual DL 3155M60

Operative instruments:

- DL 3155AL2 or stabilized supply source
- 20MHz double trace oscilloscope
- set of connection cables

Where we start from

The modulation is the process of impressing a low-frequency intelligence signal onto a high-frequency carrier signal.
In our case, the frequency modulation (FM) is the process that a high-frequency carrier signal is modulated by a low frequency (information signal) modulating signal (usually an audio).

For the frequency modulation, the carrier frequency is changing its parameter with the modulating amplitude.

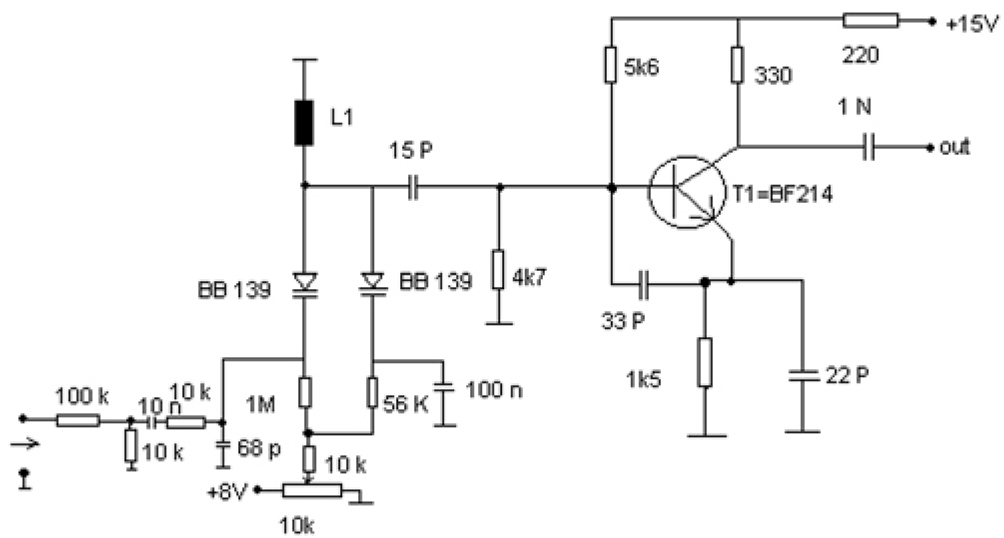
The main parameters of this process are:

- A_{DC} - dc level
- A_m - audio amplitude
- A_c - carrier amplitude
- f_m - audio frequency
- f_c - carrier frequency
- Pm- power of signal
- B- bandwidth of signal

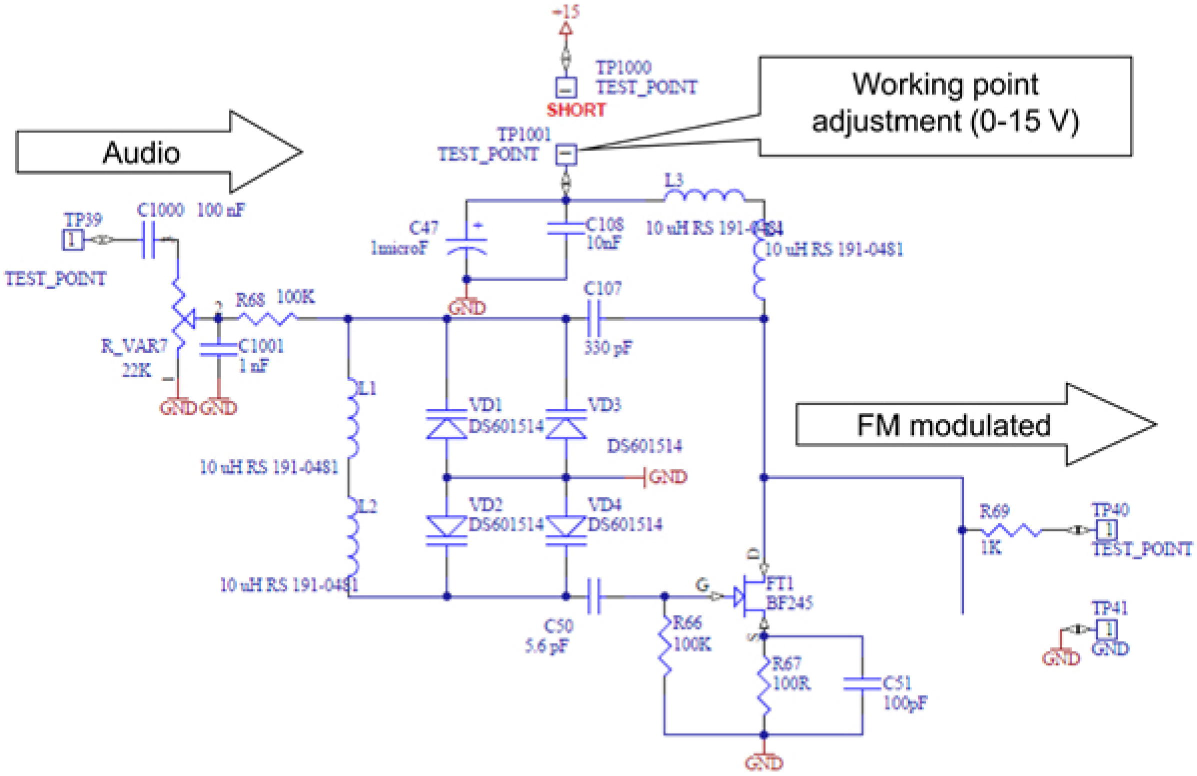
The easiest implementation, as most of the technical literature describe, is around a varicap diode, as the main part of high frequency oscillator.
The typical electronic scheme is built around one/ two varicap diodes, as part of resonant circuits.

The active component (a transistor in this case) is oscillating around working point.
This point is based on the DC polarization of base. For that reason it is necessary to setup up individually this point.

Next figure shows such typical electronic diagram, where a 10 kΩ variable resistor is used to establish the working point, reference to the power supply ground.



The audio signal is injected through a passive network of components to the varicap system for changing the frequency in accordance with the audio signal.
More accurate FM modulator is implemented by De Lorenzo, and shown in next electronic diagram.

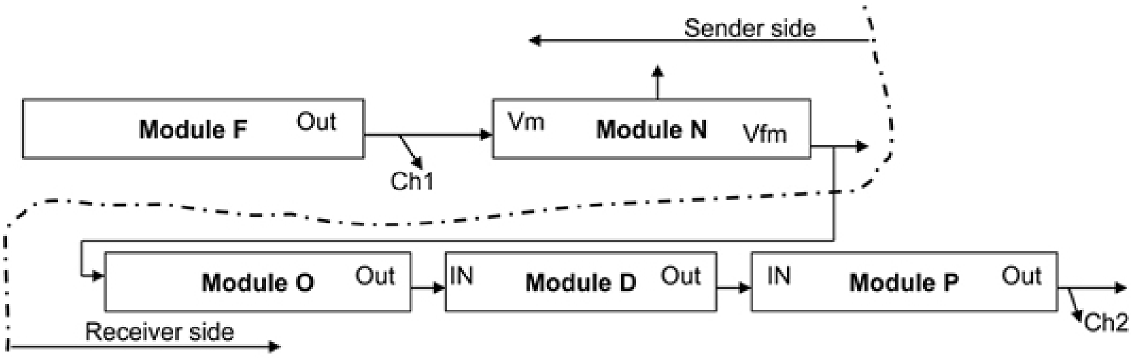


In order to start experiments with this circuit, we need a stabilized variable voltage power supply (0-15 V).

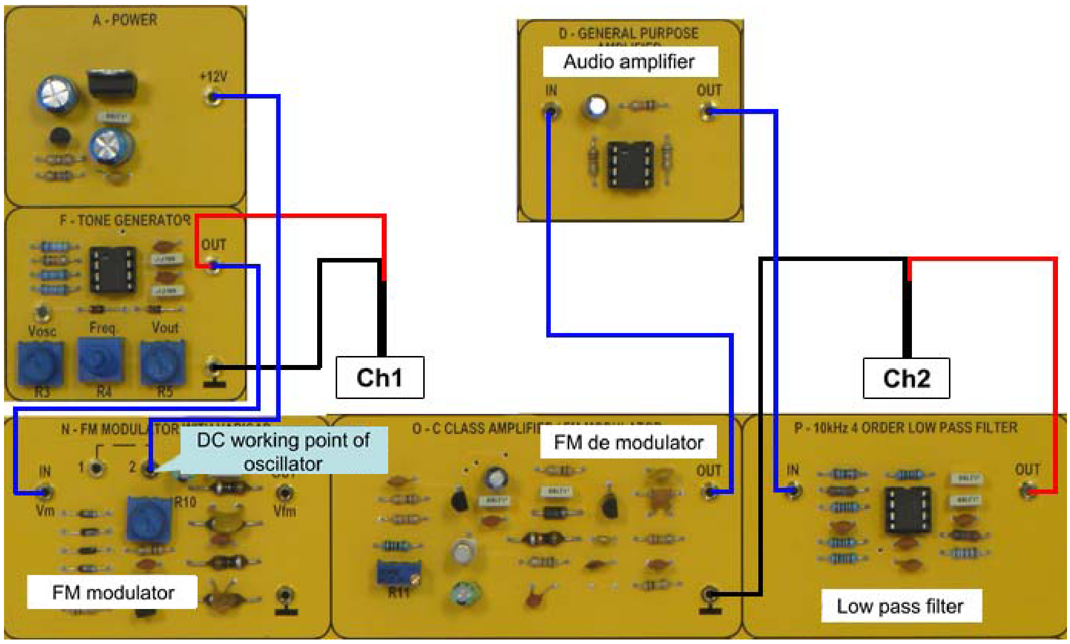
N - FM modulator with varicap; O - C class amplifier / FM demodulator

The experimental board offer the possibility to study this principle and technique of modulation. N module is a varicap based FM modulator.
In order to see how such modulator works, it is used together with module O- C class amplifier/ FM demodulator.

For this experiment, the block diagram of modules connection is shown next.

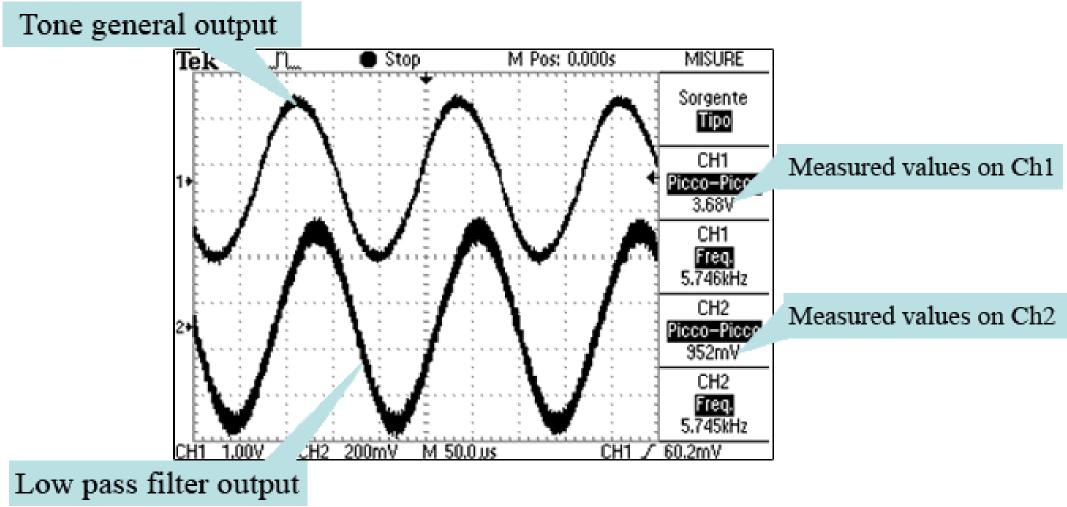


And, the picture of experimental panel is shown next.



Here there is figured also the measurement of the working point adjustment.

So, let's start working with frequency modulation module. With upper the figured upper connection we will proceed experimenting.
Remember, all the time we will observe the two oscillating signals (the output of the tone generator, and the output of the 4 order low pass filter



First of all we need to ensure the working point of the oscillator. For this purpose we setup the tone generator at 5 kHz.

Please use a screw driver for fine adjustment of the knob Second, we will adjust the working point of the FM modulator oscillator taking into account to keep clean the oscillating waves. Because of some passive electronic components, it happens that output of tone generator is bad influenced.
Please adjust the working point taking into account the quality of the tone generator output.
In the same time, adjust the working point for maximum output signal (we have got around 1 V at the output of 4 order low pass filter).

Next, we are adjusting the bandwidth of the FM signal. As you know, large range of tone voltage means bigger bandwidth of FM signal.
On the module N there is a knob that changes the level of the audio signal on the varicap diodes.
The bigger input signal means the bigger output signal.
Normally, there not need to touch the variable resistor R11 from the module O.
It is also used for establishing the working point of the C amplifier. Again, for eventually correction through this variable resistor, please also keep seeing the shape of the output voltage.

Quiz. Test yourself and be ready to accept new questions

What are the advantages of using FM modulator by using varicap diodes?

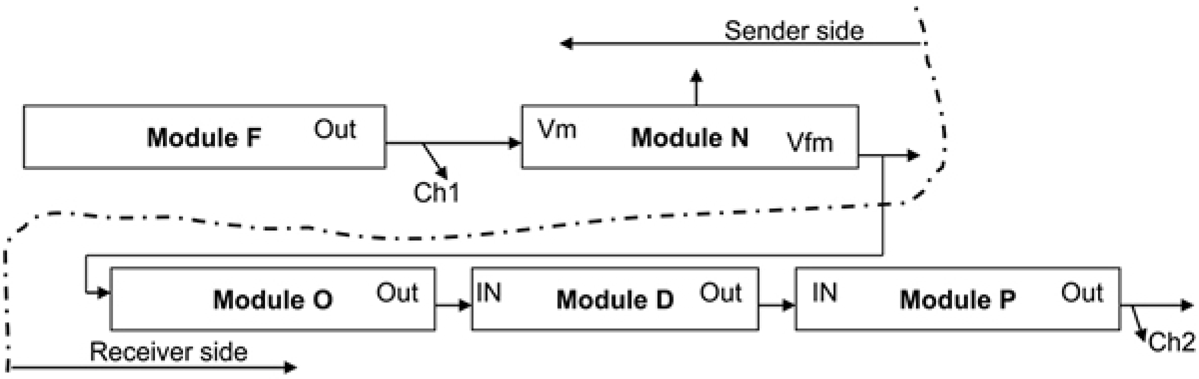
Please review the list of main influences over the demodulation process.

What are the main disadvantages of varicap based FM modulators

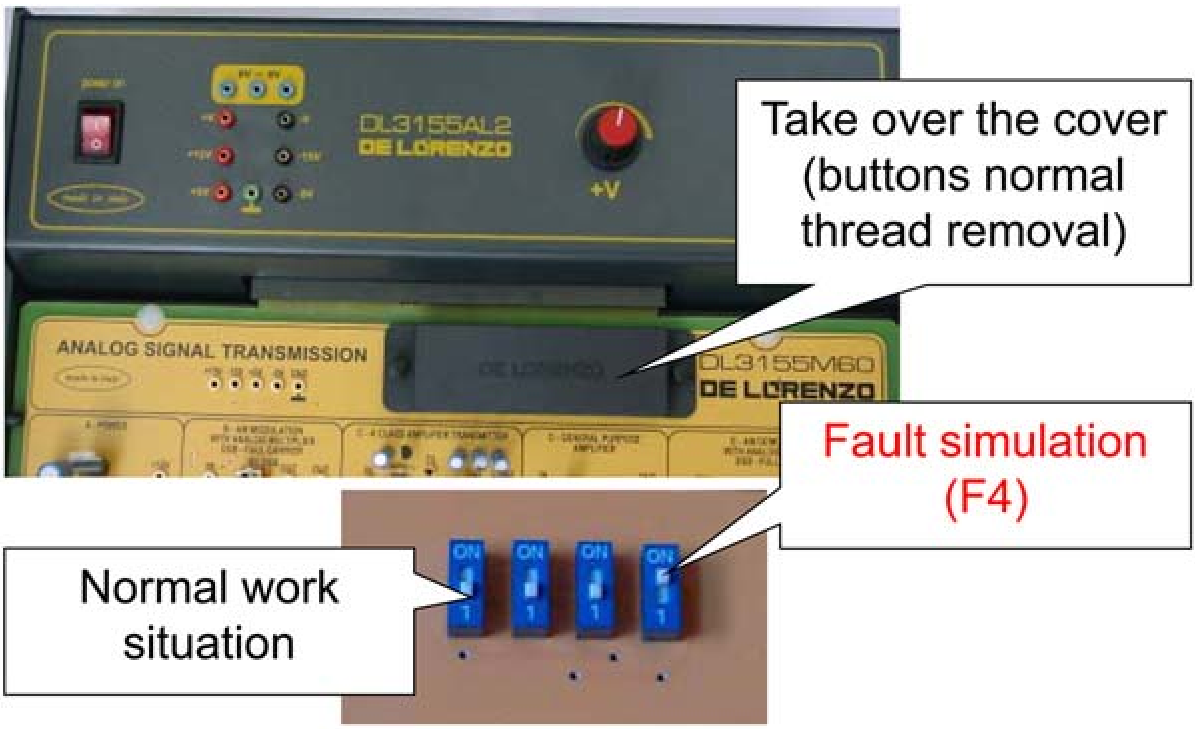
Please justify the use of electronic modules from this experiment.

Fault Simulation

In the same manner, as previous situations, we are simulating another case- a malfunction in C class amplifier/FM modulator.

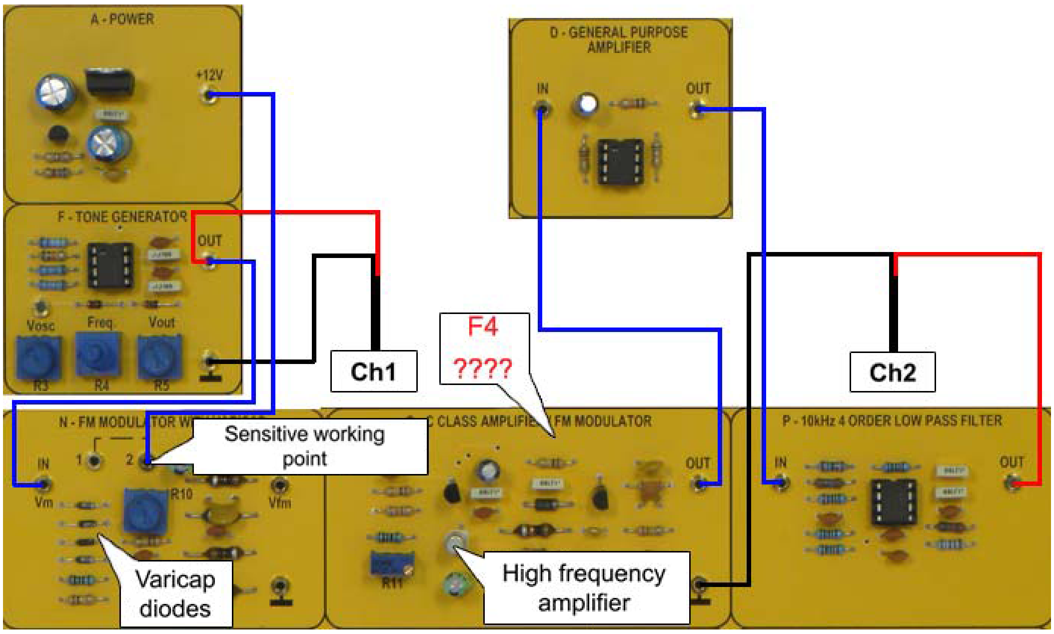


For this situation, the fault scenario (F4) is made by adjusting the switches like in next figure.



If the panel is connected directly to your PC via De Lorenzo TIME interface, then the fault can be inserted directly by clicking on INSERT shown below.

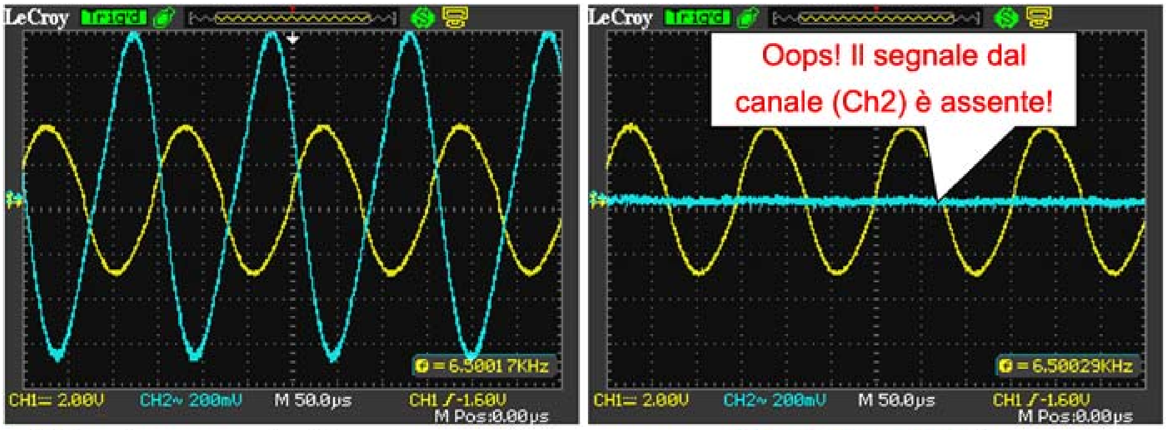
Now, we remember, we have exercised with this experimental board.



The experiment where we simulate this fault (F4) is one of the most complex, because it consists of electronic components with high sensitivity to external electric field (varicap diodes), a high frequency amplifier with low immunity to the disturbances.

In one of the previous fault simulation (F2), we have underlined the modules with high level of voltages, as modules where tests are done easier.
In present simulation, all the modules have low level of voltages (tens of mV). The algorithm of testing will be designed starting from different approach.

Let's see how the fault simulation looks like. Next figure shows the difference between normal and fault situation.



Situazione normale di funzionamento

Simulazione Guasto

Where we see the fault?
The fault is detected at the output of 10 kHz 4 order low pass filter (module P).

In normal situation, what is the purpose of the low pass filter?
The signal levels at input and output of module P is almost the same. It "cleans" the high frequency. If the output of this module is low, it might happen that, the general purpose amplifier has not any signal to amplify. So, for the purpose of our simulation, please check the input of the module P.

What about the input of general purpose amplifier?
The signal is detected, but it is noise "drowned". The detection is easy by fine adjusting of the tone generator's knob. In the fault situation there is not any application signal. In the process of fault diagnose, it is important to be sure what modules are not damaged.

What next? What should do next?
Because the high frequency signals have low amplitude, with normal measurement equipments, we must check all equipments which are easy detected if they work normally or not. We have two more modules. The varicap- based modulator can be diagnosed about malfunctioning.

Does the oscillator works well?
Now, with this preliminary probation, we will see at the output of the module N if it works or not. I suppose you got modulated signal, as in experiment 5. Well, we understood together that the "damaged" module is C class amplifier. We are happy that it is only a simulation of the fault. Please switch back the dip-switch in the normal position and make the nice experiment- Exp5.

	We want to worry any hobbyist that the work with high frequency oscillators and amplifiers. For instance, the module O also consists in a variable resistor (R11) that is accessible, but it has been used by the producer for adjusting the base potential of the first amplifier transistor. So, please do not touch it for normal experiments.

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Unit N.7: Understanding PLL frequency modulation (PLL FM)

Objectives:

- Understanding the principle of frequency modulation/ demodulation using PLL circuits
- Understanding the waveform and frequency spectrum of FM signal
- Designing an frequency modulator using dedicated circuits
- Measuring and adjusting modulator circuit

Requisites:

- Minimum level of communication techniques understanding
- Medium level of electronics components and devices understanding
- High level of health and safety risks understanding
- Communication systems theoretical manual DL 3155M60

Operative instruments:

- DL 3155AL2 or stabilized supply source
- 20MHz double trace oscilloscope
- set of connection cables

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Where we start from

As we know already, the modulation is the process of impressing a low-frequency intelligence signal onto a high-frequency carrier signal.
The frequency modulation (FM) is a process that a high-frequency carrier signal is modulated by a low frequency (information signal) modulating signal (usually an audio).

For the frequency modulation, the carrier frequency is changing its parameter with the modulating amplitude.

The main parameters of this process are:
- A_{DC} - dc level
- A_m - audio amplitude
- A_c - carrier amplitude
- f_m - audio frequency
- f_c - carrier frequency
- Pm- power of signal
- B- bandwidth of signal

Identifying the panel filters

On the theoretical part, that supports the experimental kit- frequency modulation section, there is a full description of possibilities of running with frequency modulation techniques.
On the experimental board there are some modules (electronic circuit) that are used to conduct well FM experiments.

First we will take into consideration to use again, the carrier generator (module M), the tone generator (module F), and some other modules depending by the level of experimentation.
Ones again, it is important to underline all blocks that are involved in FM modulation and demodulation.

I - FM modulator with PLL; L - FM demodulator with PLL

As you know from all domestic applications, the most popular ‘electronic’ method for FM/PM modulation and demodulation is the Phase Locked Loop (PLL).
The heart of PLL is the Double Balanced Mixer, where in a PLL, the DBM is used in combination with a low pass filter (or time constant) as a Phase Detector.

Remember please the block diagram of the PLL. Another important component of it is the voltage controlled oscillator.
A more general explanation of the PLL's behaviour is that it continuously adjusts the VCO output to maintain phase quadrature.
That means that any change in the input FM wave's frequency or phase tends to produce a DBM output which causes the VCO to ‘track’ the input.

In order to maintain a constant phase relationship they must keep ‘in step’ — i.e. their frequencies must always be the same.
Since the VCO's frequency depends upon the control voltage, it follows that this voltage varies in proportion with the FM wave's frequency. Remember, this one of biggest advantage of PLL FM modulator/demodulator.

In our implementation, there is an integrated device in MOS technology - CD 4046, that make our experiment very simple.

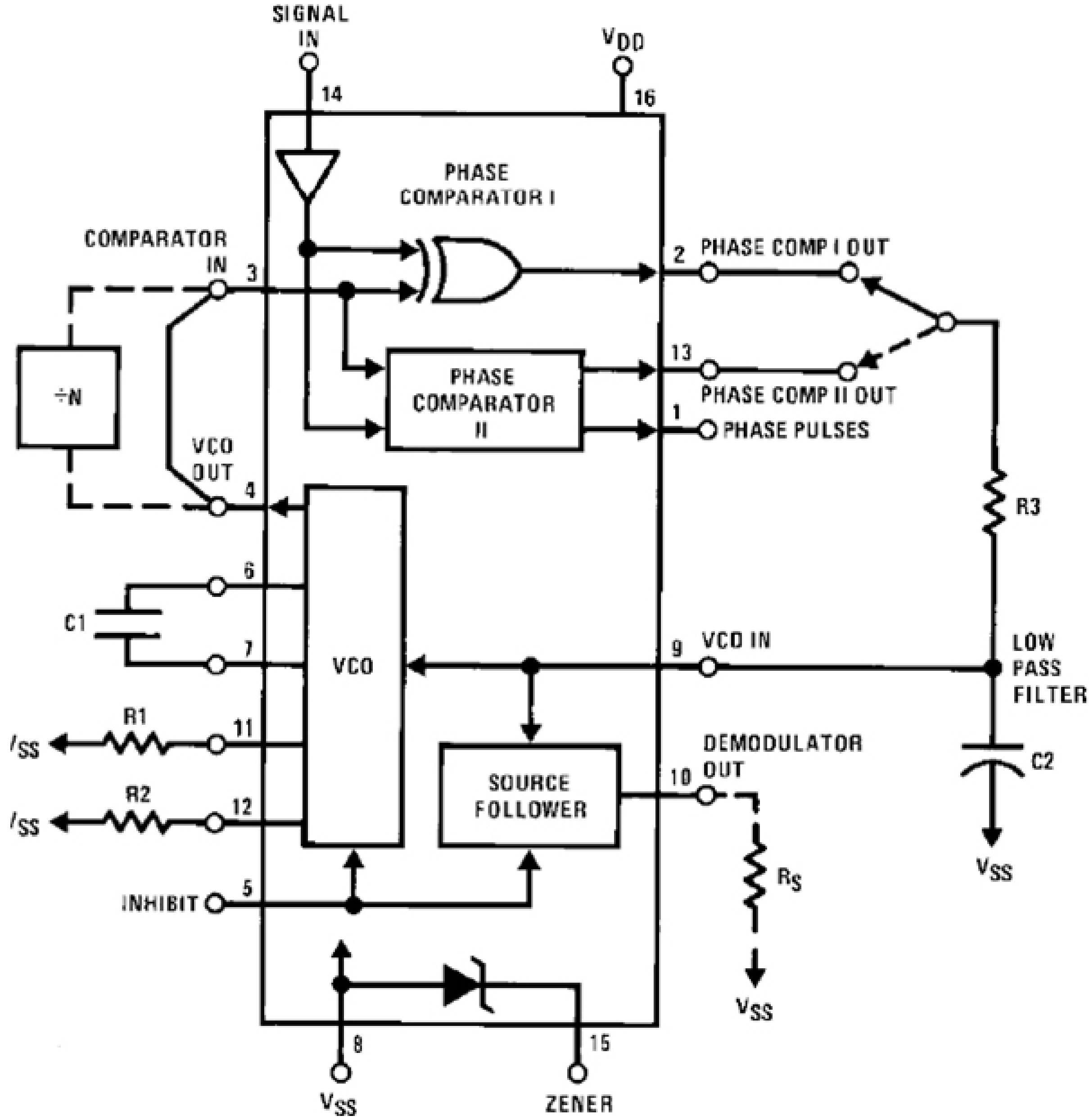
The PLL is a very interesting and useful building block available as single integrated circuits from several well known manufacturers.
It contains a phase detector, amplifier, and VCO, see Fig. 1 and represents a blend of digital and analog techniques all in one package. One of many applications and features is tone-decoding.

There has been traditionally some reluctance to use PLL's, partly because of the complexity of discrete PLL circuits and partly because of a feeling that they cannot be counted on to work reliably. With inexpensive and easy-to-use PLL's now widely available everywhere, that first barrier of acceptance has vanished.
And with proper design and conservative application, the PLL is as reliable a circuit element as an op-amp or flip-flop.

From the catalogue datasheet of 4046, we understand that the phase comparator II is an edge-controlled digital memory network.
It provides a digital error signal and lock-in signal (phase pulses) to indicate a locked condition and maintains a 00 phase shift between signal input and comparator input.

The linear voltage-controlled oscillator (VCO) produces an output signal (VCO Out) whose frequency is determined by the voltage at the VCOIN input, and the capacitor and resistors connected to pins C1, R1 and R2.

The source follower output of the VCOIN (demodulator out) is used with an external resistor of 10 kΩ or more.

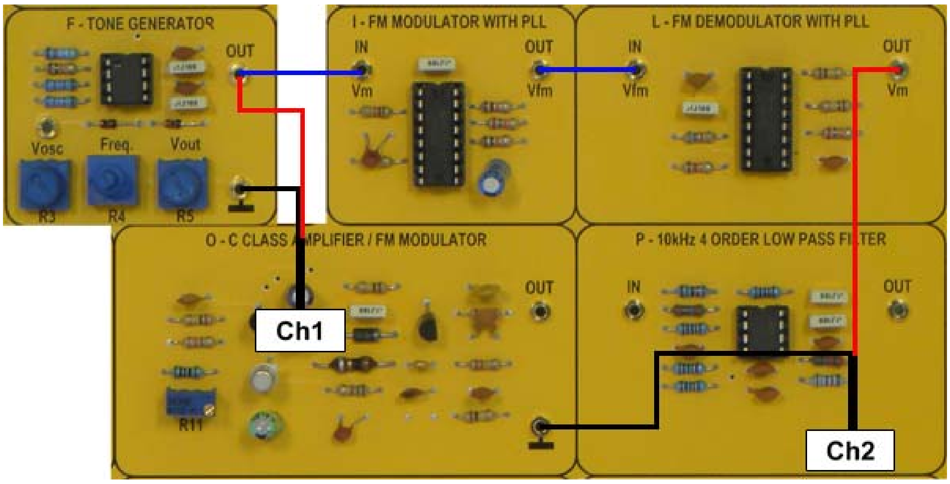


In upper figure there is the block diagram of internal structure of CD 4046. The VCO is externally controlled by C1, R1, and R2.
For running experiments with De Lorenzo product board (DL3155M60) we will use FM modulator with PLL (module I), FM demodulator with PLL, tone generator (module F), and 10 kHz 4 order low pass filter (module L)- we will explain why latter.

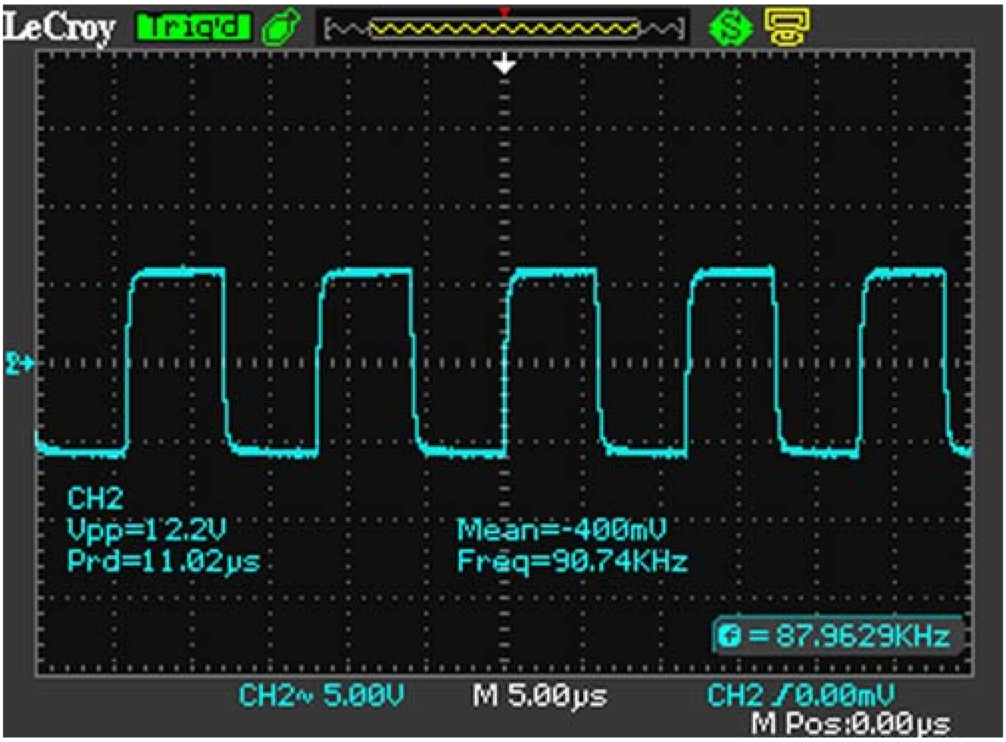
Here, there the time to fix up a technical detail: in any transmission, there is a specific parameter for the data that have to be sent- the bandwidth of the data.
Let's suppose that we want to broadcast audio tone in a bandwidth of 10 kHz. It means that we need electronic modules that process 10 kHz of data- like a low pass filter.

It is important to mention this, because this bandwidth is influencing the bandwidth of high frequency (please read the FM paragraph from theoretical manual).
Next figure shows the connections between modules.

In our experiment, the FM modulator, built around 4046, with the external element R1, C1, develop 88 kHz in VCO, with 12 $V_{pk\text{-}pk}$.



Again, our target is to send an audio signal through a transmission environment (like FM modulated radio wave), then, at the receiver side, to recuperate entirely the audio signal.
For that reason, main measurements are done at output of tone generator (Ch1), and at output of FM demodulator with PLL.
But they are also some intermediate measurements, like VCO parameters. Next figure shows that.

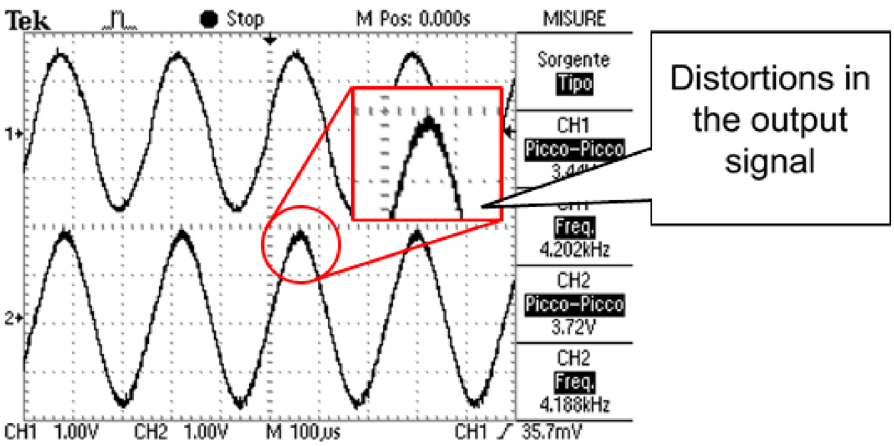


For our experiment, we are establishing the output of the tone generator at frequency below 5 kHz, with an amplitude maximum 4 $V_{pk\text{-}pk}$.

Tasks to study

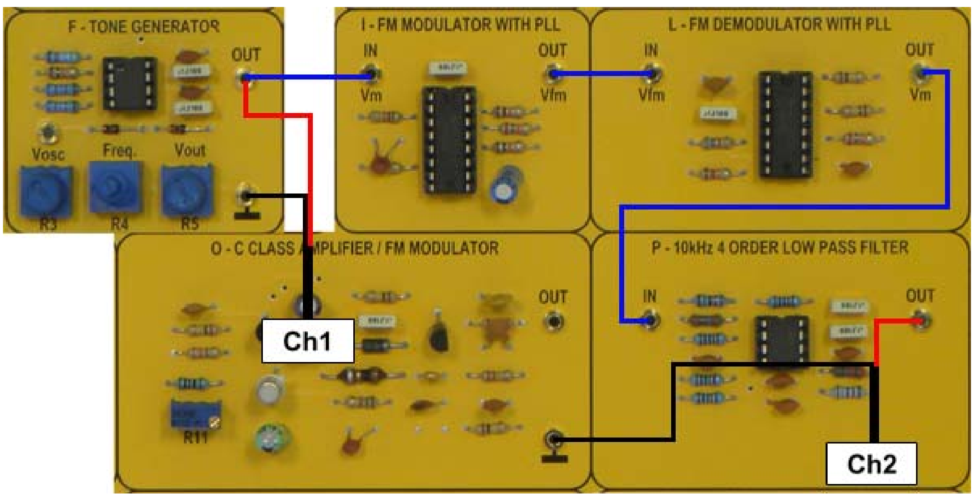
First, please make the proper setup, as previous presentation.

Then, make fine adjustments in tone generator’s frequency. Check the output of FM demodulator (module L).



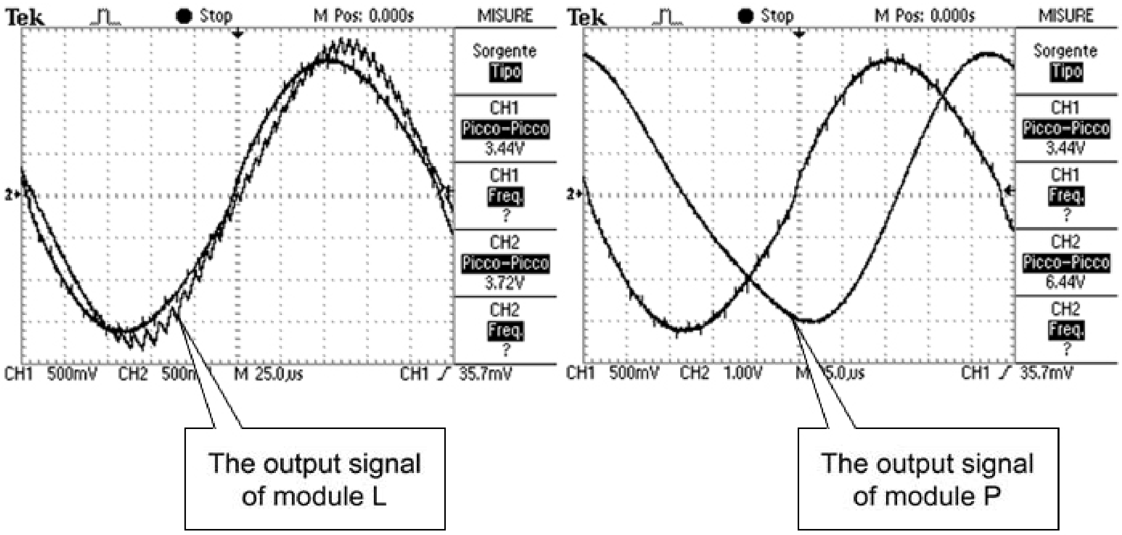
Please make your comments related. What about output phase signal? What about output level of signal?
Well, the signal is not very accurate.

What can we do next?
Please connect the output of FM demodulator to the input of 10 kHz 4 order low pass filter, like next figure.
Let’s see what is happening.



By measuring the output signal of module P, we will be much satisfied about results.
Next figure shows for comparison, the measured signal at the outputs of module L, and P.

Please make your comments related, about the quality of recuperated audio signal, and other related parameters.



As far as you followed the recommendations for experiments, and you got almost similar results, now we can go father with quality analysis.
An important parameter of modulation/ demodulation process is the linearity of the process.

First, based on catalogue data of the 4046, please calculate the VCO frequency, based on values of the electronic circuit: C1= 680 pF, R1= 27 kΩ, R2= 100 kΩ.
Check for the calculated results comparing with measured results.
Check for the linearity of the chosen values for external VCO components.

Second, into the imposed limits of tone generators parameters, with fixed amplitude of it, please make an analysis of the output signal in accordance with tone generators frequency.
Please complete the next table, and then build up a graph related.

F tone [kHz]	**U out filter [V]**	**Delay [%]**	**Obs.**
F1	**U = ?**		
F2	**U = ?**		
....	**.....**		

Please make your comments related.

Quiz. Test yourself and be ready to accept new questions

A Please review the advantages of using PLL technology in implementing FM transmission system.

B Who influences the modulation/demodulation process?

C Who are FM modulating/ demodulating parameters?

D Please prove the use of module P.

Unit N.8: Understanding FM modulator- quadrature demodulator

Objectives:

- **Understanding the principle of frequency modulation (FM demodulation)**
- **Understanding quadrature demodulation**
- **Evaluating the quality of detected signal in FM quadrature demodulation**
- **Measuring and adjusting an demodulated signals**

Requisites:

- **Minimum level of communication techniques understanding**
- **Medium level of electronics components and devices understanding**
- **Understanding the frequency modulation mechanism**
- **High level of health and safety risks understanding**
- **Communication systems theoretical manual DL 3155M60**

Operative instruments:

- **DL 3155AL2 or stabilized supply source**
- **20MHz double trace oscilloscope**
- **set of connection cables**

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Where we start from

As you red in theoretical manual, in the FM signal, the modulation is the deviation of a carrier from its nominal frequency.
The conventional method to demodulate this signal is to convert frequency deviation to phase and after that, to detect the change of phase.
In the quadrature demodulator, the modulated carrier is passed through a resonant circuit that shifts the signal by 90° at the centre frequency.

This phase shift is either greater or less than 90° depending on the direction of deviation. So, a phase detector compares the phase-shifted signal to the original in order to give the demodulated baseband signal.

We are using quadrature demodulators not only for frequency modulation, but also with digital modulation schemes such as FSK (frequency shift keying) and GFSK (Gaussian frequency shift keying).

Again, De Lorenzo, has used a simple implementation for frequency modulation- a Voltage Controlled Oscillator, implemented with LM137000 (Dual Operational Transconductance Amplifiers with Linearizing Diodes and Buffers).

As you know, with the component values shown, this oscillator provides signals at 87,245 kHz. The circuit oscillates at the frequency at which the loop phase-shift is 360° or 180° for the inverter and 60° per filter stage.

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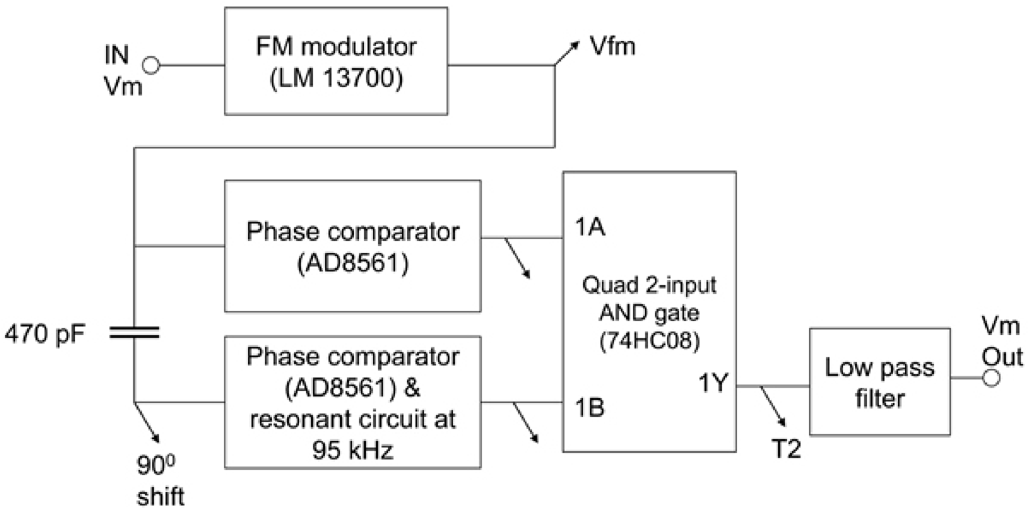
T - FM modulator - quadrature demodulator

For studying the principle of quadrature measurement, De Lorenzo offer an embedded electronic circuit (module T), where, there is possibility to study FM modulation, and also principles of quadrature demodulation.

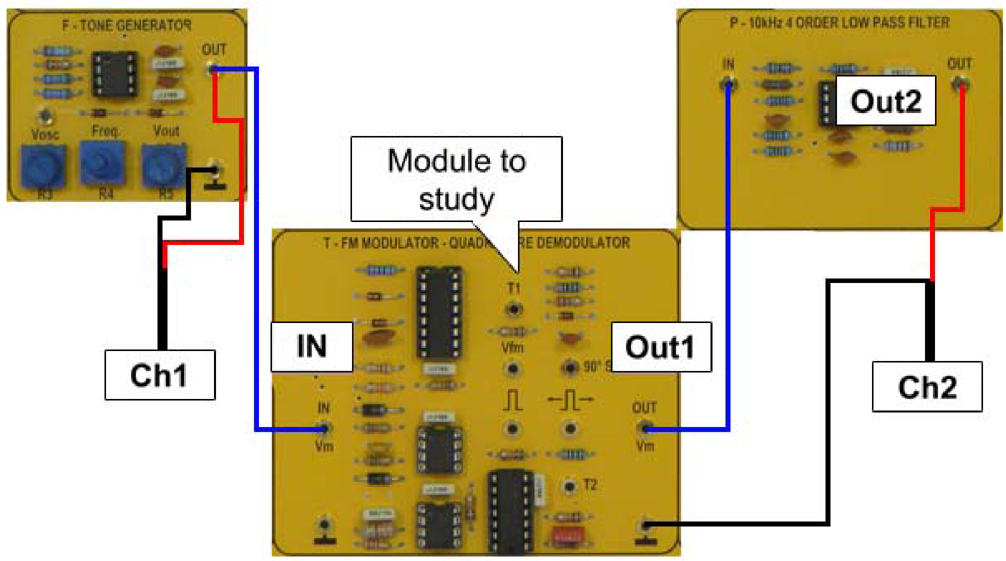
The block diagram of this electronic module is shown next:



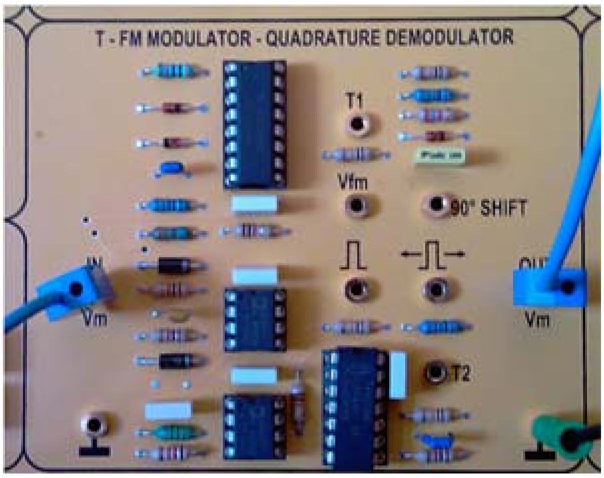
Vm is audio signal (3- 5 kHz, 3-4 V_{pk-pk}), Vfm represents frequency modulated signal, Vm(Out) represents to extracted audio signal by quadrature demodulator.

Tasks to study

As we are accustomed, let's make the proper setup.

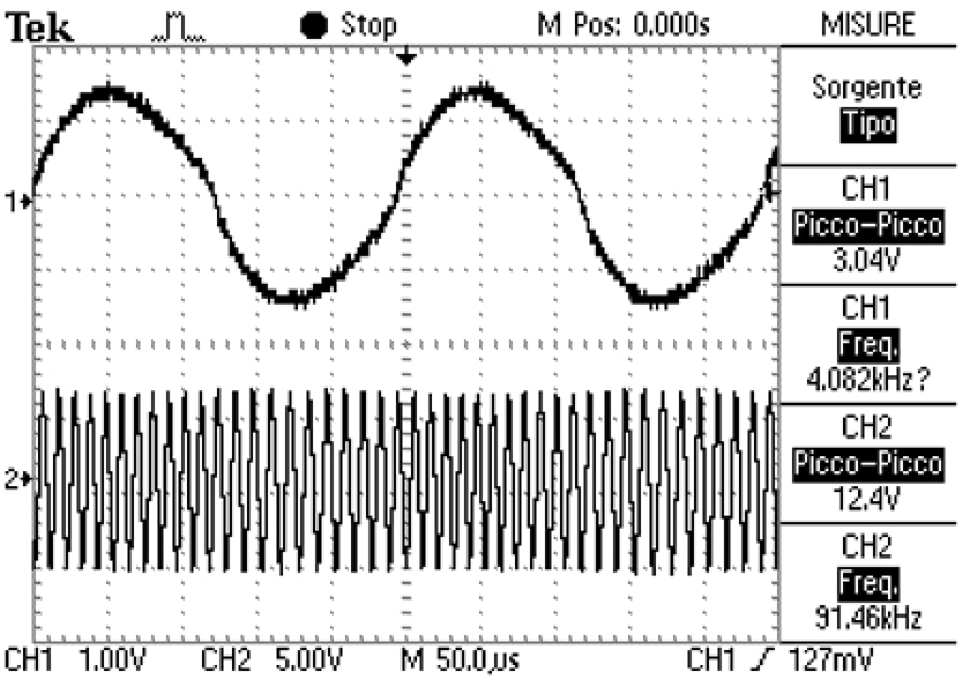


In order to understand the role of each electronic component from quadrature demodulator, we would suggest you to check each signal test point as mentioned on the board (please locate on the module T the test points: T1, T2, Vfm, 900 shift, Vm OUT).



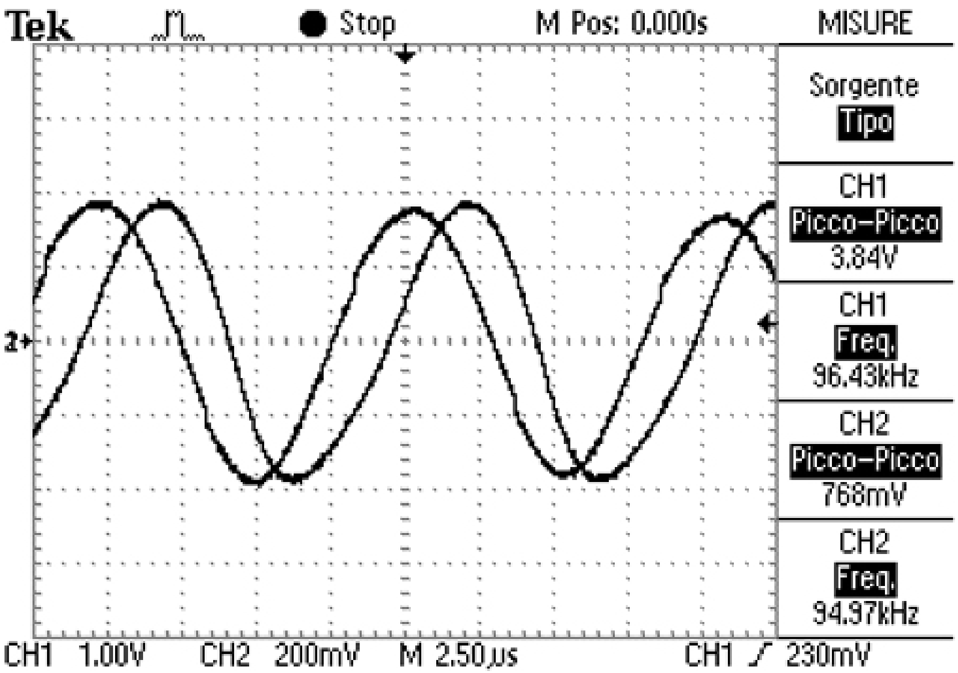
	First of all let's check the frequency modulated signal (Vfm). With input signal, by respecting the values limit, we are collecting the output of FM modulator.

The signal looks like next figure.

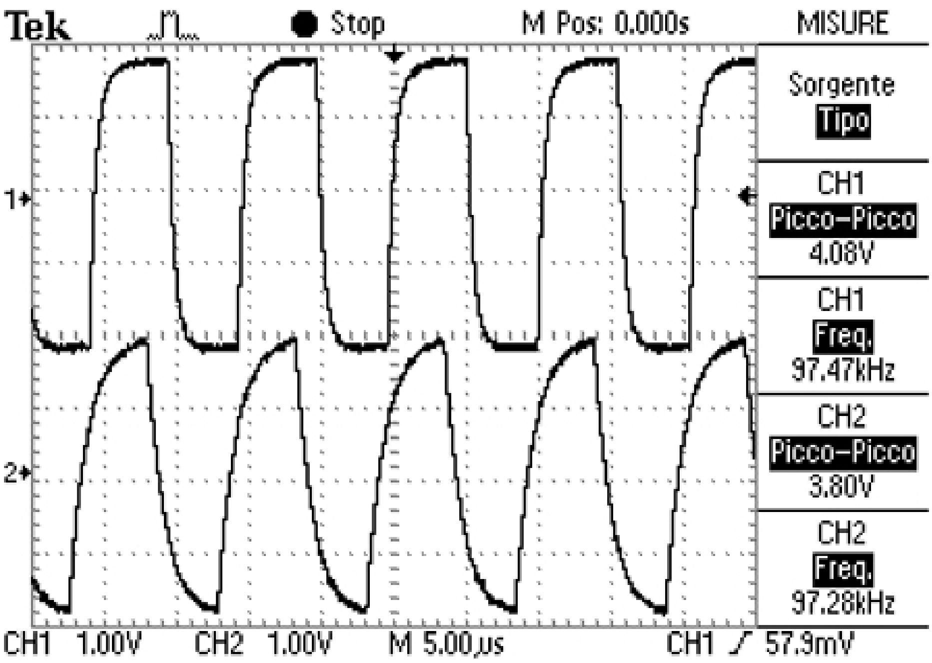


Then, we are checking the test points signals. First, let's check the Vfm, and 90° shift signal (90° shift is done by using 470 pF capacitor- see upper block diagram).

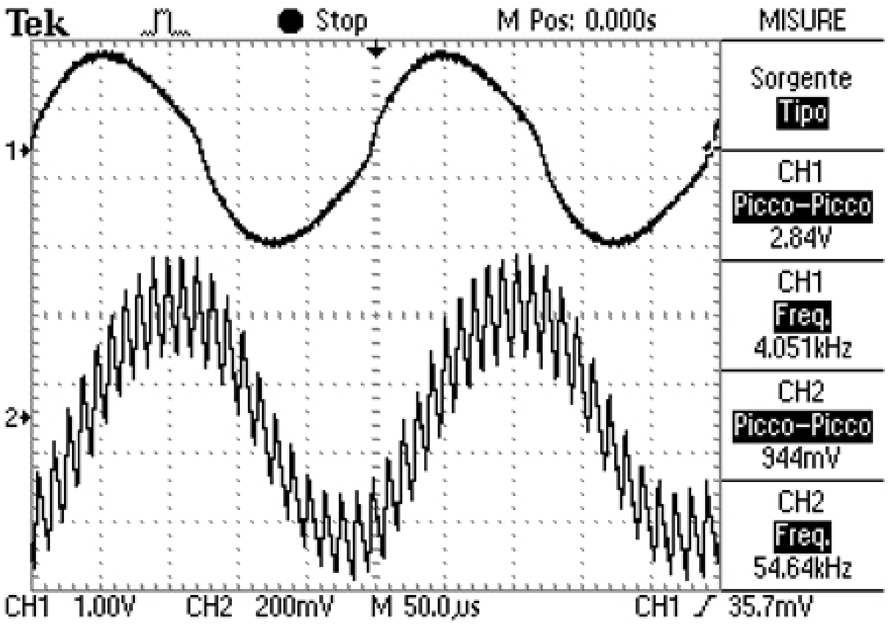
The shifted signal is also attenuated.



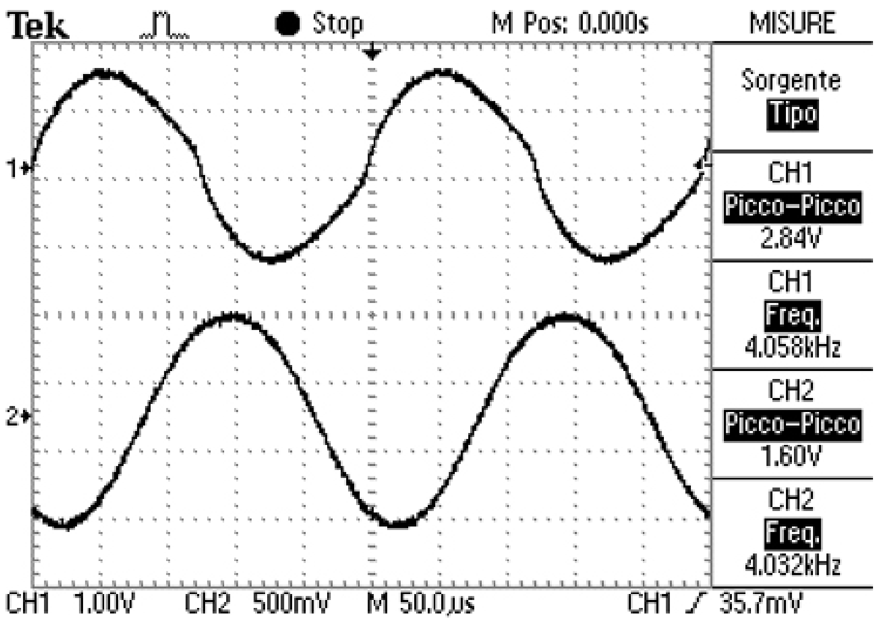
Let's see how the phase comparators process the signals.
Next figure shows the results



The outputs of this phase comparators are injecting the inputs of AND function for extracting the phase differences developed proportionally by the audio signal.
The output of this AND gate is shown next:



It consists on audio signal and also a high frequency signal. What we do next? We apply this complex signal to the input of a high order low pass filter.
The results are presented next.



It is a nice audio signal! Next stage of the experiment is related to qualitative analysis.
In the same manner with previous experiment, we have to evaluate the linearity of the FM modulator/ demodulator.

Into the imposed limits of tone generators parameters, with fixed amplitude of it, please make an analysis of the output signal in accordance with tone generators frequency.
Please complete the next table, and then build up a graph related.

F tone [kHz]	U out filter [V]	Delay [%]	Obs.
F1	U = ?		
F2	U = ?		
....	.....		

Please make your comments related.

Quiz. Test yourself and be ready to accept new questions

Why we use phase modulation? What is its advantage?

What is the essence of quadrature demodulation? What are the check points of quadrature demodulation?

Who influences the modulation/demodulation process?

Who are FM modulating/   demodulating parameters?

Please prove the use of module P.

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Unit N.9: Understanding PM - phase modulator/demodulator. Individual work

Objectives:

- Understanding the principle of phase modulation (PM) and demodulation
- Understanding PWM and its consequences
- Evaluating the quality of detected signal in PM demodulation circuits
- Measuring and adjusting an demodulated signals

Requisites:

- Minimum level of communication techniques understanding
- Medium level of electronics components and devices understanding
- Understanding the frequency modulation mechanism
- High level of health and safety risks understanding
- Communication systems theoretical manual DL 3155M60

Operative instruments:

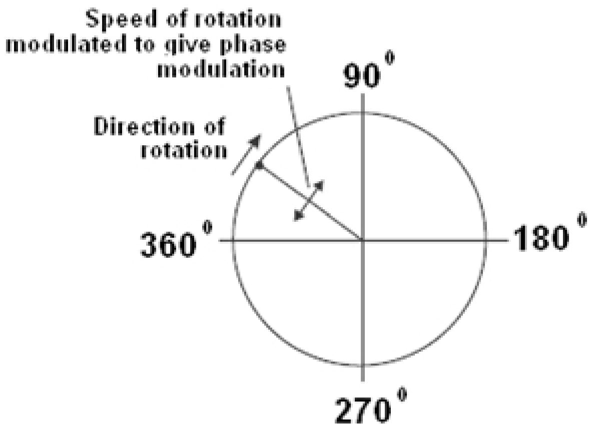
- DL 3155AL2 or stabilized supply source
- 20MHz double trace oscilloscope
- set of connection cables

Where we start from

The easiest way to start understanding how phase modulation works, it is first necessary to give a vector explanation of the phase.

As we imagine, a radio signal consists of an oscillating carrier in the form of a sine wave. The amplitude follows this curve, moving positive and then negative, and returning to the start point after one complete cycle.

This complicate phrase can also be represented by the movement of a point around a circle, the phase at any given point being the angle between the start point and the point on the waveform as shown in next figure.



So, modulating the phase of the signal means changes the phase from what it would have been if no modulation were applied.
The speed of rotation around the circle is modulated about the mean value. To achieve this it is necessary to change the frequency of the signal for a short time.

In other words, when phase modulation is applied to a signal there are frequency changes and vice versa. Phase and frequency are inseparably linked, as phase is the integral of frequency.
By symmetry of the experiment, the conventional method to demodulate this signal is to convert frequency deviation to phase and after that, to detect the change of phase.

Additionally, for an easy practical implementation, TL082 has been used.
Please consult technical sheet of this general purpose J-FET dual high speed operational amplifier.
Please take into account the cut-off frequency of this integrate circuit, it works well around 100 kHz.

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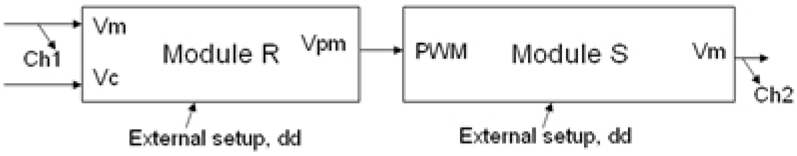
R - phase modulator; S - phase demodulator

For studying the principle of phase modulator/ demodulator (particular case- pulse width modulation), De Lorenzo offers the specific electronic circuits (modules R and S), where, there is possibility to study PM modulation, and also principles of phase demodulation, and to understand the main differences between FM and PM.

The proposed block diagram of the phase modulation/ demodulation is shown next:



Please imagine your own lab experiments master plan, with respecting next issues:

1. Develop your own setup in order to get change in the carrier phase according to the amplitude of the modulating signal.
2. Make your own observations change in width of clock pulses according to the amplitude of the modulating signal at the different clock frequencies.
3. Based on your measurements, please calculate modulation index of the modulated signal.
4. Study of PWM & demodulation using different sampling frequencies.
5. Please imagine your own setup in order to recovery of modulating signal from demodulator circuit.
6. Please speak about spectral efficiency in the case of working frequency and phase modulation.

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