Online Radio & Electronics Course

Reading 34

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FREQUENCY MODULATION

WHAT IS FREQUENCY MODULATION?

Before defining Frequency Modulation, let's just go over what we mean by 'modulation' as applied to radio communications. Modulation is the process of converting low frequency signals into high frequency signals, so that they can be radiated and recovered at a remote location. Usually the 'intelligence' we are trying to transmit is audio, but of course can be other things such as encoded video or data. With amplitude modulation, we combine the modulating audio with the radio frequency carrier in such a way that the <u>total wave power</u> of the transmitted wave is made to vary. Remember, the <u>total</u> wave power varies. The RF <u>carrier power</u> in an AM system remains constant.

Now besides using the modulating audio to change the total power of the wave being transmitted, we can make the modulating audio change the frequency of the RF carrier.

FM is a system by which the modulating audio is made to change the instantaneous frequency of the carrier.

Popular belief is that in an AM signal the carrier power varies, and in an FM signal the carrier power remains constant. This is incorrect. In an AM system the total wave power varies with modulation and the carrier power remains constant, and in an FM system the total power transmitted remains constant and the power in the carrier varies.

The simplest FM transmitter is where the modulating audio is fed directly to an oscillator, and the voltage level of the modulating audio causes the frequency of the oscillator to change.

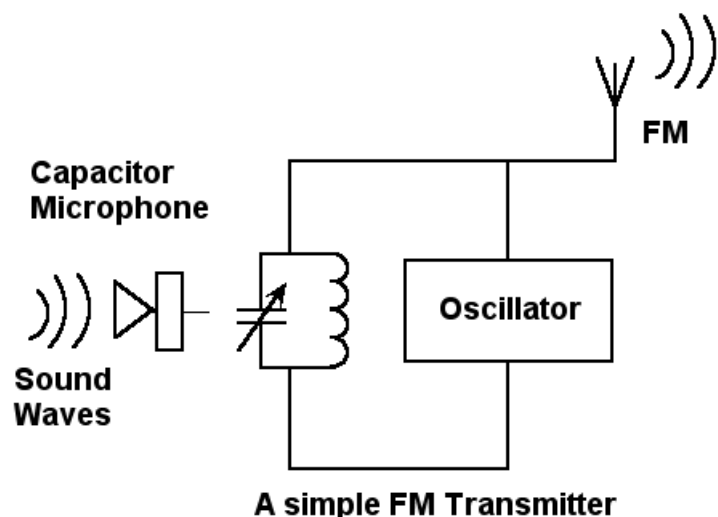This is how most FM wireless microphones work.

A simple FM Transmitter

Figure 1.

A SIMPLE FM TRANSMITTER

Modulating audio could be fed to a capacitor microphone that forms part of a LC circuit that in turn controls the operating frequency of an oscillator. Hence, the voltage of the modulating audio controls the frequency of the oscillator. This signal could then be

amplified and transmitted to the receiver. At the receiver we would need a method of converting RF frequency changes back into audio.

BACK TO THE VOLTAGE CONTROLLED OSCILLATOR (VCO)

In an earlier reading we discussed how voltage could be used to control the frequency of an oscillator. We did this by using a varactor diode. The reverse bias on the diode changed its capacitance and in turn that capacitance change was used to make small changes in the operating frequency of a quartz crystal. To jog your memory, have a look at the circuit of the VCO again, shown in figure 2.
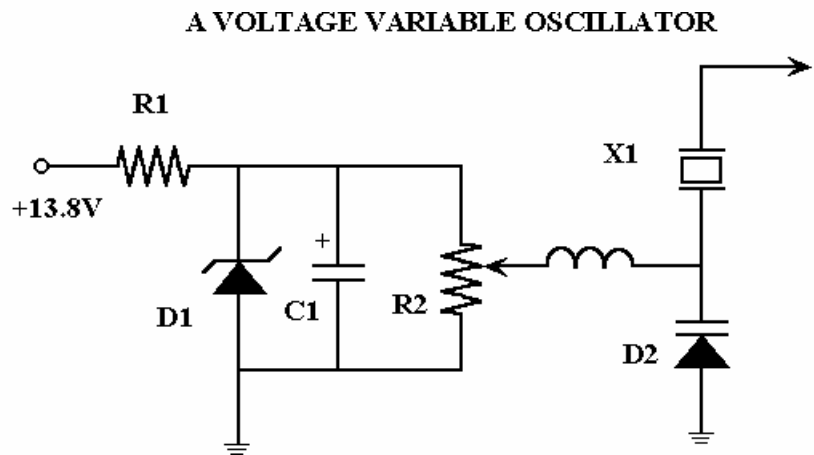
A VOLTAGE VARIABLE OSCILLATOR

Figure 2.

The potentiometer (variable resistor with a centre tap) can be adjusted to bring about small changes in frequency of the quartz crystal. Everything to the left of the radio frequency choke (RFC) in the circuit provides a regulated DC voltage using a zener diode.

Now what if we did away with the zener diode and applied (modulating) audio to the varactor diode. We would have a method of producing FM.
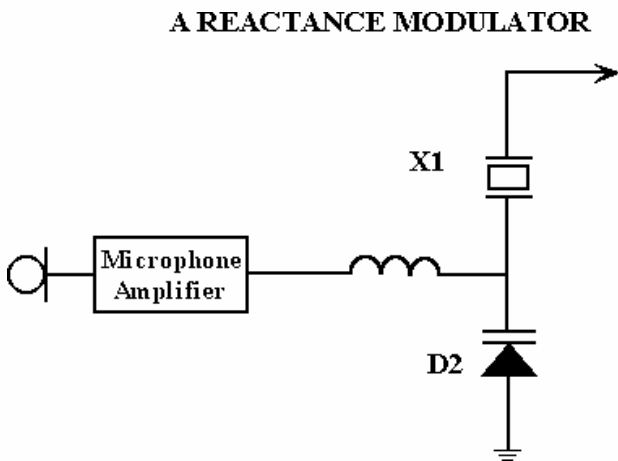
THE REACTANCE MODULATOR

A REACTANCE MODULATOR

It is that easy to produce Frequency Modulation. Simply feed the modulating audio to a varactor diode and the diode will bring about frequency deviations of the quartz crystal proportional to the modulating audio.

I have not drawn the complete circuit of the oscillator, as it could be any type. As you can see, the modulating audio is used to change the reverse bias on the varactor diode, hence changing its capacitance, and in turn the frequency of the oscillator.

Figure 3.

The RFC (radio frequency choke) is just to keep RF out of the AF circuit and vice-versa.

Just to make sure you have got exactly what we are on about here, let's go over it. Without any modulating audio the reactance modulator (which is just a fancy voltage controlled oscillator) will produce RF energy on one single frequency only. This is shown below in figure 4.2 - the frequency is irrelevant but it would be high, at least in the order of a few megahertz. This is called the **centre frequency**.

Then we have the audio frequency coming from the microphone, which in the case of radio voice communication would be somewhere between 100 Hz and 3 kHz maximum.
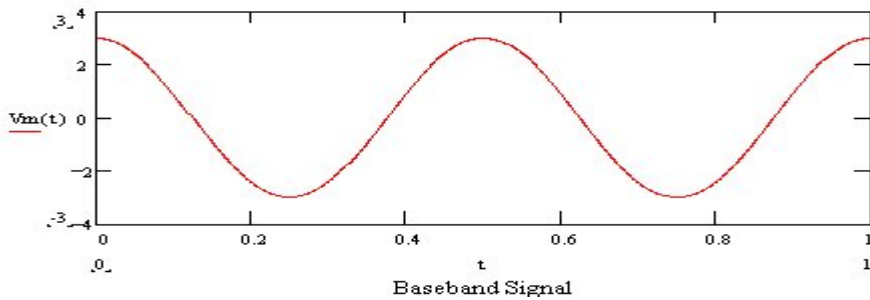
Figure 4.1 - The Modulating Audio:

Baseband Signal

Figure 4.2 - The Unmodulated RF Carrier:
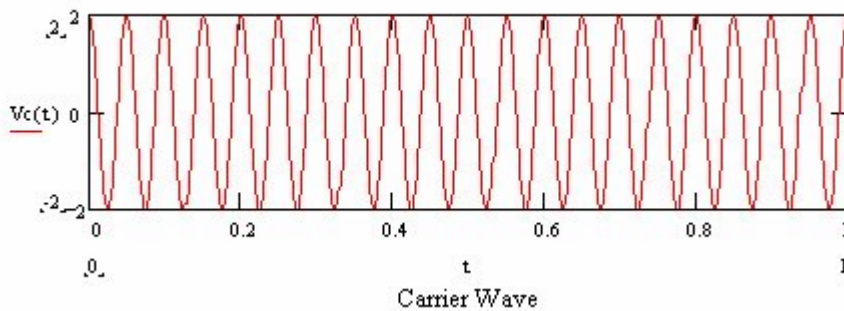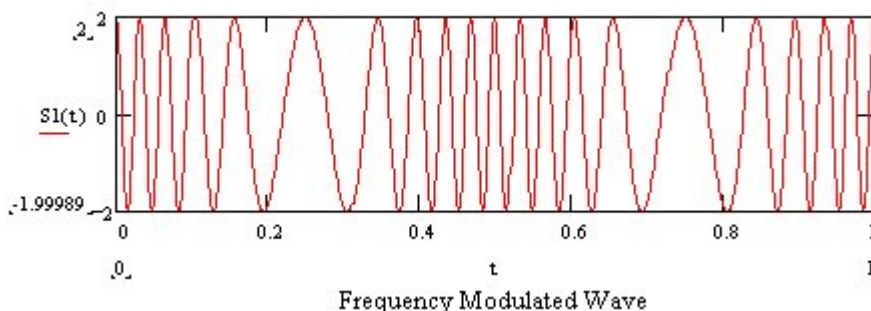

Carrier Wave

Figure 4.3 - The FM Modulated RF Signal:

Frequency Modulated Wave

Now, this modulating audio is applied to the varactor diode in the reactance modulator causing the frequency of the quartz crystal to change up and down from the centre frequency. So we end up with a frequency-modulated wave as depicted above in figure 4.3. The total wave power does not change, the frequency does.

These graphs could be shown on a device called an oscilloscope. The oscilloscope clearly shows the amplitude (power) of the FM wave does not change, but it is obvious that the average power of the centre frequency does change as it is not there all the time.

Frequency modulation is a method of modulating a carrier wave whereby the modulating audio causes the frequency of the carrier to change.

Major EH Armstrong developed the first working FM system in 1936, and in July 1939 began the first regular FM broadcasts from Alpine, New Jersey.

The capacitance of the diode in the reactance modulator has a direct effect on the oscillators frequency. The following important conclusions can be made concerning the output frequency:

1. The frequency of the sound waves arriving at the diaphragm of the microphone determines the <u>rate</u> of frequency change.
2. The amplitude of sound waves arriving at the diaphragm determines the <u>amount</u> of frequency change.

The previous diagrams illustrate the fundamental differences between an FM and AM wave. If the sound wave striking the diaphragm was to be doubled in frequency from 1 kHz to 2 kHz with the amplitude kept constant, the **rate** at which the FM wave swings above and below the carrier would change from 1 kHz to 2 kHz. However, since the intelligence of the sound wave has not changed, the **amount** of frequency swing above and below the carrier frequency will remain the same.

On the other hand, if the frequency of the modulating audio was constant but the amplitude was doubled, then the rate of frequency swing above and below the carrier would remain at 1 kHz, and the amount of **frequency deviation** (movement from the centre frequency) would double.

DEVIATION OF AN FM WAVE

Without modulation, an FM transmitter produces a single carrier frequency (fc). When modulation is applied, the output frequency will swing above and below the unmodulated carrier frequency (fc). The amount of decrease or increase around fc is called the frequency deviation.

MODULATION INDEX OF AN FM WAVE

The modulation index is the ratio of the amount of frequency deviation to the audio modulating frequency. Suppose an FM broadcast transmitter has a deviation of 75 kHz when a modulating audio frequency of 15 kHz is applied to the microphone input, then:

modulation index (μ) = deviation / modulating frequency
modulation index (μ) = 75 / 15 = 5.

The modulation index has no unit. In this text, the modulation index will be denoted by the Greek letter (μ). Sometimes the modulation index is denoted by the letters (mf).

There is a **special case** concerning modulation index, and this is when the frequency of the modulating audio is at its **highest**. In such a case the modulation index is called the **deviation ratio**. For amateur FM transmissions, the deviation is restricted by regulation to 5 kHz. Since the highest modulating audio used for voice communications is 3 kHz, the deviation ratio for an amateur FM transmitter is then:

Deviation / highest modulating frequency = 5 / 3 = 1.66.
Remember the modulation index is known as the deviation ratio when the highest modulating audio frequency is used. The examiner will expect you to know this.

SIDE FREQUENCIES OF AN FM WAVE AND MODULATION INDEX

When a carrier (fc) is frequency modulated by a signal (fs), then theoretically an infinite number of side frequencies will be produced on each side of the carrier.

The **distance** between each of the side frequencies is equal to the frequency of the modulating signal (fs).

The **amplitude** of each side frequency does not follow any simple pattern and is dependent upon the modulation index (μ) or deviation ratio.

At a certain distance from the carrier (fc) the power in the side frequencies will decrease to a level where they can be considered insignificant. **A side frequency is considered insignificant if its voltage amplitude is less than 10% of the unmodulated carrier amplitude**. The number of significant side frequencies is directly proportional to the modulation index and can be found from: significant side frequencies = 2 (μ + 1) where μ = modulation index. To determine the relative amplitude of each of the side frequencies it is necessary to consult the table of Bessel functions of the first kind.

Table 1.
BESSEL FUNCTIONS OF THE FIRST KIND

Mf	J0	J1	J2	J3	J4	J5	J6	J7	J8	J9	J10	J11	J12	J13	J14
0.000	1.00														
0.250	0.98	0.12	0.01												
0.500	0.94	0.24	0.03												
1.000	0.77	0.44	0.11	0.02											
1.500	0.51	0.56	0.23	0.06	0.01										
2.000	0.22	0.58	0.35	0.13	0.03	0.01									
2.405	0.00	0.52	0.43	0.20	0.06	0.02									
3.000	-0.26	0.34	0.49	0.31	0.13	0.04	0.01								
4.000	-0.40	-0.07	0.36	0.43	0.28	0.13	0.05	0.02							
5.000	-0.18	-0.33	0.05	0.36	0.39	0.26	0.13	0.05	0.02	0.01					
7.000	0.30	0.00	-0.30	-0.17	0.16	0.35	0..34	0.23	0.13	0.06	0.02	0.01			
10.00	-0.25	0.04	0.25	0.06	-0.22	-0.23	-0.01	0.22	0.32	0.29	0.21	0.12	0.06	0.03	0.01

This is not in the exam, however it is worth your while knowing how Bessel Functions are used in FM. When you have the time, that is, passed your exam, you can come back to this or write to me for more material I have written on Bessel Functions.

Example 1. A commercial FM broadcast transmitter has a deviation ratio of 5 and the frequency of the modulating audio is 15 kHz. Determine (a) the amount of frequency deviation, and (b) the occupied bandwidth (BW) of the transmission.

Answer: the deviation is 75 kHz and the occupied bandwidth is 180 kHz.

Explanation:

We learnt that modulation index = deviation / frequency of the modulating audio. This is more often written as $\mu = \Delta f / fs$. This equation can be transposed to give:
Δf (deviation) = μ x fs, and for our problem this gives
Δf (deviation) = 5 x 15 kHz = 75 kHz.

The significant number of side frequencies present in the transmitted wave is given by:
significant side frequencies = 2 (μ + 1) = 2 (5 + 1) = 12.
Since each of these 12 significant side frequencies are spaced apart by an amount equal to the frequency of the modulating audio (15 kHz), then the bandwidth is:
BW = 12 x 15 kHz = 180 kHz.

An easier method (and the one you should remember):

A slightly easier method for determining the bandwidth of an FM wave can be derived. In the last example we found the bandwidth from:
    BW = 2 (μ + 1) x fs

By multiplying each term within the brackets by fs we get:
    BW = 2(μ x fs + 1 x fs).

Since μ x fs equals the deviation, the equation can be simplified to:
    BW = 2(Δf + fs).

The bandwidth (how much spectrum space it occupies) is equal to 2(Δf + fs). Where Δf is the deviation and fs is the frequency of the modulating audio.

Example 2.

Assume an amateur transmitter has a peak deviation of 3 kHz (which is typical), and that the frequency of the applied modulating audio is 3 kHz (typical as well). With the aid of Bessel functions, construct the spectrum of this wave as would be seen on a spectrum analyser.

The modulation index 'μ' of this wave is 1. There are 2(μ+1) or 4 significant side frequencies (two each side of the carrier) each spaced apart by (fs), or 3 kHz. By consulting a table of Bessel functions, we see that the voltage amplitude of the carrier for a modulation index of 1 is 0.77 and the first two pairs of side frequencies have amplitudes of 0.44 and 0.11. The third pair of side frequencies has a relative amplitude of 0.02 and since this is less than 10% of the unmodulated carrier value of 1, the third pair of side frequencies can be considered insignificant.

We can now construct the graph from the above information, as shown in figure 5.

Figure 5.
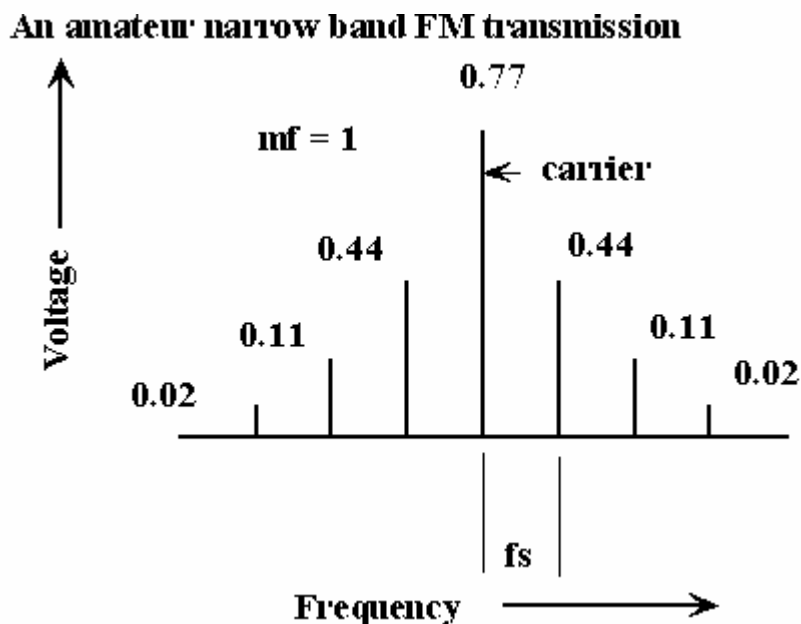
Show - using a table of Bessel functions that the total power of an FM wave remains constant.

A table of Bessel functions gives the **relative voltage amplitude** for the carrier and each pair of side frequencies. Since the **power is directly proportional to the square of the voltage, the relative power amplitude of the carrier and its side frequencies can be found by squaring each value listed in the table**. In addition, since the total power in an FM wave remains constant**, the sum of the squares of the values in the Bessel table should equal 1 for any particular modulation index.**

The figures given on tables of Bessel functions are for practical purposes rounded, usually to two decimal places. So the sum of the squares will be almost, but not quite, 1. If we used Bessel functions taken to say six significant places, we would find the result is extremely close to 1.

Modulation Index = 1

Relative	E	E^2	Multiply by	Power
J(0)	0.765198	0.585528	x1	0.585528
J(1)-	0.440051	0.193645	x2	0.387290
J(2)-	0.114903	0.013203	x2	0.026406
J(3)-	0.019563	0.000383	x2	0.000766
J(4)-	0.002477	0.000006	x2	0.000012
			Total Power:	1.000002

Table 2.

Here we have demonstrated that **the total power in an FM wave remains constant**. If we did the same for any other modulation index we would find the same result. The reason it may seem I have carried on about this a bit is because there is a popular false belief that the power of the carrier of an AM wave varies with modulation and the power of an FM carrier remains constant - when precisely the opposite is the case. I will not be asking any assignment questions about Bessel functions. However, please do remember the structure of an FM wave and how to calculate the bandwidth.

SUMMARY SO FAR

AM - a system of modulation whereby the modulating audio is combined with the carrier in such a way as to create new frequencies (sidebands) and where the total wave power transmitted varies with modulation.

FM - a system of modulation whereby the modulating audio cause the frequency of the carrier to vary in proportion to the modulating voltage. The total power transmitted does not vary. The power in the carrier does vary and can fall to zero (for example, look at mf = 2.405 in table 1).

THE ADVANTAGES OF FM WHEN COMPARED TO AM

An FM system provides a better signal-to-noise ratio than an AM system. Put simply, this means it has less noise.

During its transmission (propagation), a frequency modulated wave will be subject to noise and interference voltages. The effect of these unwanted voltages is to vary the **amplitude**

and phase of the FM wave. The noise amplitude variations have no effect on the performance of the system. Intelligence, information or modulation if you like, are not carried in the amplitude of an FM wave. **Amplitude variations are removed in the FM receiver stage called the limiter**. The phase deviation caused by noise means the carrier is effectively frequency modulated and some noise will appear at the output of the receiver.

The improvement in the signal-to-noise ratio that FM has over AM is primarily dependent upon the system deviation ratio (**D**). This is why FM broadcast stations sound so good. The degree of improvement can be determined by the equation:

Signal-to-noise ratio improvement = 20Log(**D** sqrt 3) dB (not a must know).

The deviation ratio used in amateur communications (narrow band FM) is 5/3 or 1.66. The signal-to-noise ratio advantage over AM, by using the previous equation, is found to be 9.21 dB. Nothing to get excited about, but it is none-the-less a worthwhile improvement. Great advantages in signal to noise ratio are only realised by commercial broadcasters. FM broadcast stations have a system deviation ratio of 5, giving a signal-to-noise ratio advantage of 18.75 decibels compared to AM.

THE DISADVANTAGES OF FM WHEN COMPARED TO AM

FM has a greater bandwidth requirement than AM. Narrow band FM occupies 2(5+3) = 16 kHz of spectrum space compared to AM's 6 kHz (or in the case of single sideband 3 kHz).

FM systems generally have a much wider bandwidth than AM systems. This makes FM (or any wider bandwidth signal) more prone to selective fading. Selective fading applies to HF propagation – see the appropriate reading for more information if needed.

CAPTURE EFFECT

Another disadvantage of FM is that of **capture effect**. Capture effect is the ability of an FM receiver to lock onto and capture one transmitting station only. This can be a good thing but it can lead to weaker signals not being heard at all. Typically, if a signal is about 3dB weaker than the desired signal then it will not produce an output from the receiver. The receiver's limiter stage has "captured" the stronger signal. This explains the reluctance of airport authorities to switch to FM. Using an AM system allows an aircraft to break into an occupied channel should an emergency situation develop.

PRE-EMPHASIS AND DE-EMPHASIS

Pre-emphasis is used at the transmitter to boost the modulating audio frequencies above 1 kHz. De-emphasis is employed in the receiver to restore the modulated audio back to its original power distribution.

Pre-emphasis and de-emphasis have the effect of improving the signal-to-noise ratio at the receiver for higher audio frequencies.
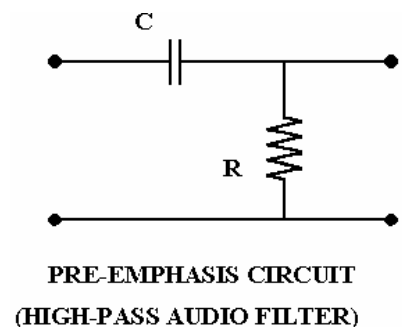

PRE-EMPHASIS CIRCUIT
(HIGH-PASS AUDIO FILTER)

Figure 6.

Pre-emphasis - boosting of audio frequencies above 1kHz in the FM transmitter.
De-emphasis - attenuation of frequencies above 1kHz in the FM receiver.

Frequencies contained in human speech mostly occupy the region from 100 to 10,000 Hz, but most of the power is contained in the region of 500 Hz for men and 800 Hz for women. The problem is that in an FM system the noise output of the receiver increases linearly with frequency, which means that the signal-to-noise ratio is poorer at the high audio frequencies.
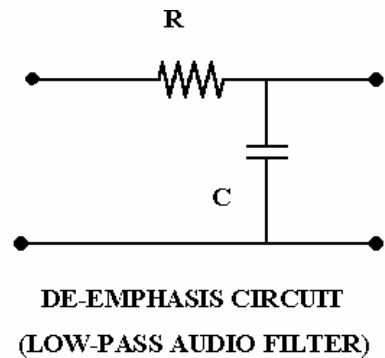

**DE-EMPHASIS CIRCUIT
(LOW-PASS AUDIO FILTER)**

Figure 7.

Also, noise can make radio reception less readable and unpleasant. This noise is greatest in frequencies above 3 kHz. The high frequency noise causes interference to the already weak high frequency voice. To reduce the effect of this noise and ensure an even power spread of audio frequencies, pre-emphasis is used in the speech amplifier of the transmitter.

A pre-emphasis network in the transmitter accentuates (boosts) the audio frequencies above 1 kHz, **so providing a higher average deviation across the voice spectrum**, thus improving the signal-to-noise ratio at the receiver. Without corresponding and converse de-emphasis at the receiver, the signal would sound unnatural.

Pre-emphasis and de-emphasis are obtained by using simple audio filters. Normally we use LC filters for high and low pass filtering, however for simple low power audio, where we are not concerned about some resistive loss, simply RC filters can be used. You may be required to identify one of the circuits in an AOCP exam. The pre-emphasis circuit produces higher output at higher frequencies because the capacitive reactance of 'C' is decreased as frequency increases. As the resistor and the capacitor form a voltage divider and the reactance of the capacitor has decreased, more voltage must appear at the output across the resistor.

A similar reasoning, but reverse in effect, can be applied to the de-emphasis circuit. The standard time constant ($T=CR$) for these circuits in amateur transceivers is 75 microseconds (uS).

IN SUMMARY

Pre-emphasis is used in the transmitter to boost (emphasise) the higher audio frequencies in speech (above 1000 Hz).

Pre-emphasis increases the average deviation of a transmitter resulting in better signal-to-noise ratio. In addition, the effects of higher frequency noise is less noticeable.

De-emphasis in the receiver attenuates the high audio frequencies in the receiver to restore the original audio as closely as possible.

THE BLOCK DIAGRAM OF AN FM RECEIVER

Figure 8 below shows the partial block diagram of an FM receiver. All the stages preceding the last IF amplifier would be the same as that of any superheterodyne receiver.

The bandwidth of the IF amplifier would be approximately 16 kHz in an amateur FM receiver. Compared to an AM receiver, the IF amplifiers would have more gain to allow for full limiting action to occur at the limiter. We will discuss the limiter stage shortly. By the time the FM signal reaches the limiter it has been amplified sufficiently for the limiter to be able to remove any amplitude modulation. **Any amplitude modulation would be noise in an FM system**. The stage following the limiter is called the **discriminator**. The purpose of a discriminator is that of an FM demodulator. An FM demodulator's job is to convert frequency variations in the signal back to the original modulating audio (the audio that went into the microphone at the transmitter). The discriminator can work on one of several principles. You will not have to know how these work but you may have to know what they are called. The basic type of discriminator are the Foster-Seeley, Ratio Detector, and the PLL type. The PLL type being the most commonly used FM demodulator today.

All these discriminators perform the same function in that they convert frequency variations into audio output. The greater the frequency deviation the higher (louder) the audio output voltage.

Portion of an FM receiver

IF AMP — LIMITER — DISCRIMI NATOR — AF AMP

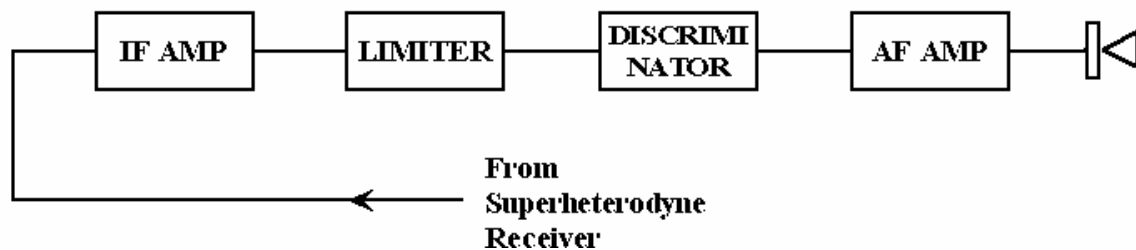From Superheterodyne Receiver

Figure 8.

The de-emphasis circuit restores the audio to its original power distribution by attenuating frequencies above 1 kHz. All FM receivers contain a "**mute**" or "**squelch**" circuit to cut off the input signal to the audio amplifier in the absence of an FM input signal.

OPERATION OF THE LIMITER STAGE

Figure 9.

HOW A LIMITER WORKS

JFET LIMITER — Saturation — Cutoff

Ouput to discriminator with noise removed

INPUT CONSISTS OF SIGNAL PLUS NOISE NOISE IS AMPLITUDE

INPUT (GATE) TO LIMITER

Figure 9 shows what happens to the input signal after being fed to a limiter. Positive and negative amplitude peaks drive the FET (though other active devices could be used) into **cutoff or saturation**, therefore **slicing off the amplitude variations which represent noise**. **The limiter is actually an over-driven RF amplifier**. If the signal entering the limiter was not of sufficient amplitude to drive the FET between cutoff and saturation then limiting action would not occur. This happens on weak signals and considerable noise is heard at the receiver output. In an attempt to ensure full limiter action, most FM receivers have more IF amplification than AM types.
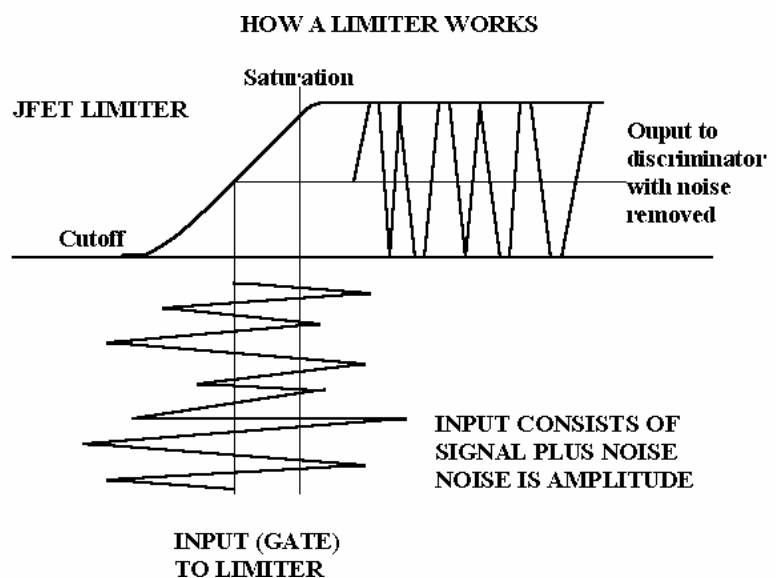
The limiter shown in figure 9 consists of a sharp cutoff FET. You can see that the input to the gate has amplitude variations (noise). The output from the drain has all of the amplitude variations removed. The intelligence is in the frequency that is left unchanged by the limiter.

THE FOSTER-SEELEY DISCRIMINATOR

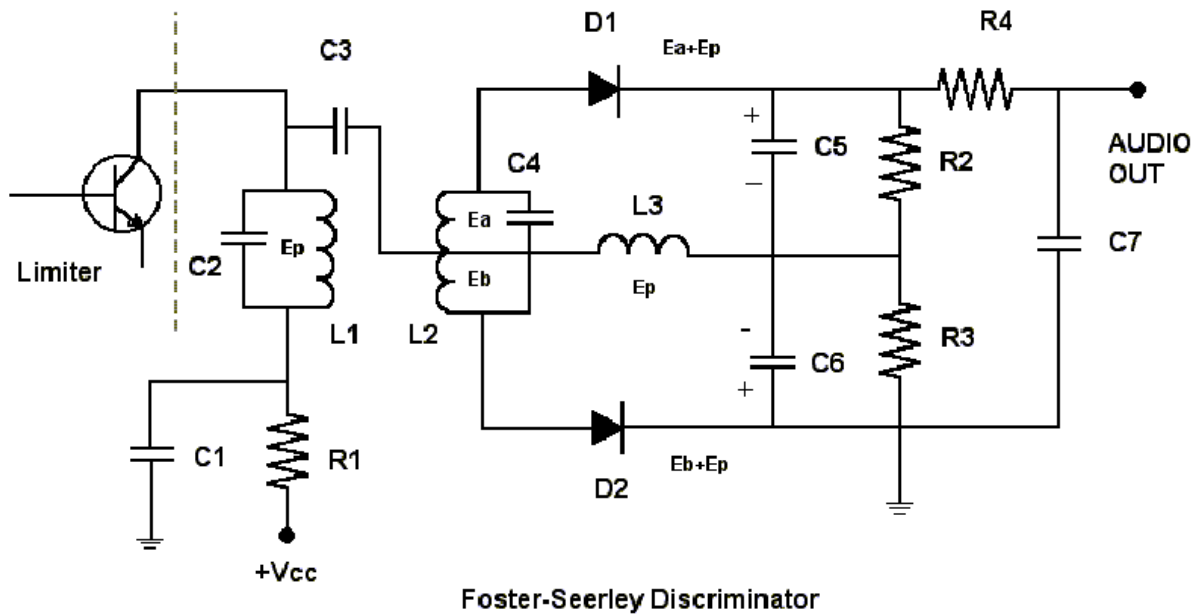Foster-Seerley Discriminator

Figure 10.

The circuit and its operation are explained in detail. You do not need to explain the operation. You may need to know the schematic's name and to be able to identify it.

Operation:

Remember first - the purpose of any FM demodulator is to convert radio frequency variations back to audio frequency variations.

In order to understand the operation of the circuit it is important to realise that the capacitances C1, C3, and C6 have very low reactances at the operating frequency. The left side of L3 is connected to the top of L1 via C3, and because of the low reactance of C3 there is no appreciable voltage across C3, therefore the left end of L3 and the top of L1 are at the same potential. Similarly, the right end of L3 is connected by C6 and C1 to the bottom end of L1, and is the same potential as the bottom of L1.

L3 is electrically in parallel with L1. The FM signal from the last IF stage produces a voltage Ep across L1, and since L1 is in parallel with L3 then the voltage Ep will also appear across L3. There will be a voltage induced into L2 from L1. Because of the centre tap, half of the induced voltage will appear as Ea on the top half of L2, and as Eb on the bottom half. The total voltage applied to D1 is then Ep + Ea, and the voltage applied to D2 is Ep + Eb.

At the centre frequency (no deviation) Ep + Ea = Ep + Eb. Also, the currents I2 and I3 through R2 and R3 will be equal and opposite. Likewise, the voltages appearing across R2 and R3 will be equal, and there will be no net output voltage. The output voltage is the net

or resultant voltage across R2 and R3. This is as it should be with no deviation. At the centre frequency, the tuned circuit consisting of L2 and C4 is resonant, and as such the current and voltage are in phase.

At any frequency other than the centre frequency (when modulation is present) there is a change of phase between Ea, Eb and Ep. This means that Ep + Ea will not equal Ep + Eb. How much difference exists will depend upon the deviation from the carrier frequency. Whether Ep + Ea is greater or less than Ep + Eb will depend on which way the frequency swings. The overall result is that for any frequency other than the centre frequency, currents I2 and I3 will not be equal. The voltages R2 and R3 will not be equal and opposite, and a net voltage will appear at the output with an amplitude directly proportional to the deviation. By converting the deviation into amplitude variations we have recovered the original modulating audio.

A de-emphasis circuit is formed by R4 and C7. R4 and C7 form a simple low pass filter.

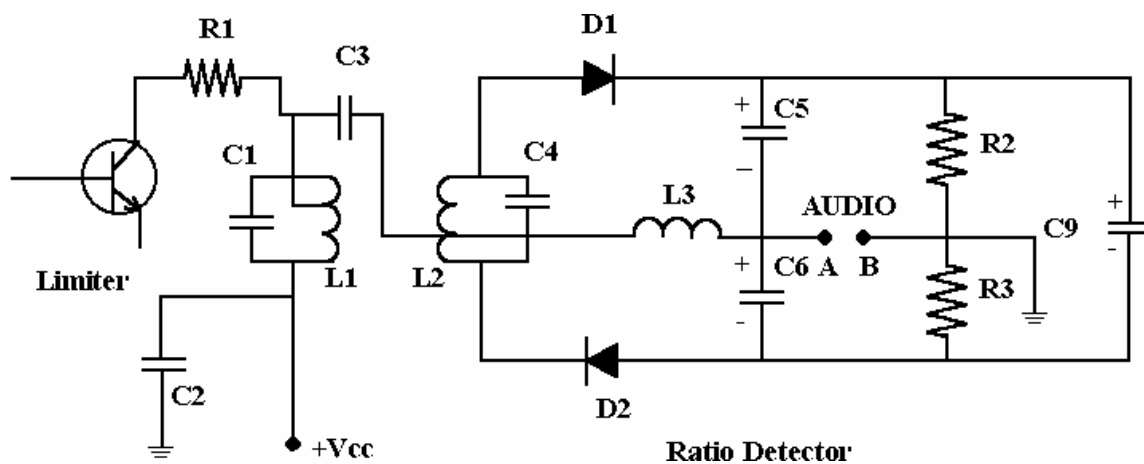## THE RATIO DETECTOR (another FM demodulator)

Figure 11.

In a ratio detector, one of the diodes is reversed so that current can flow in the overall circuit. The circuit operation is similar to the Foster-Seeley discriminator but the output voltage is now taken from the points (A) and (B) as shown in the schematic diagram. Because C9 is connected in parallel with R2 and R3, and also C5 and C6, the voltages C5 + C6 and R2 + R3 are held at a constant value by the charge on C9.

At the centre frequency there will be no potential difference between points (A) and (B). When deviation is present, one of the diodes will conduct harder and the voltage across its associated capacitor will increase; the voltage on the other capacitor will decrease (the total voltage C5 + C6 remaining constant); the circuit is unbalanced and potential difference must now exist between points (A) and (B). This net voltage is the de-modulated audio output. Because amplitude variations do not unbalanc the circuit, a ratio detector does not *theoretically* require a limiter stage. In practice however, most manufacturers provide a limiter stage, as large amplitude variations will produce some noise output.

PHASED LOCKED LOOP (PLL) FM DE-MODULATORS

Without doubt, this is the most common method of FM demodulation today. **For a full description of how a PLL system works, download the optional reading for this subject (supplementary for Reading 27)**. It is not necessary to have a broad knowledge of PLL systems for exam purposes. The bock diagram of a PLL FM detector is shown in Figure 12.
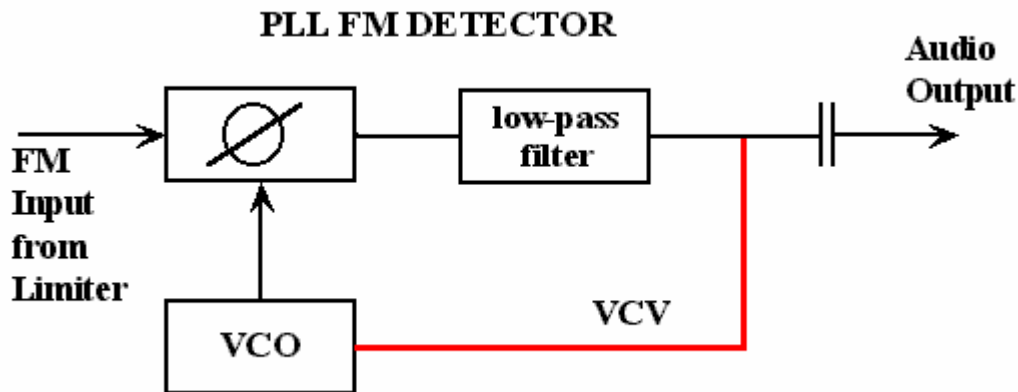

Figure 12.

The FM signal from the limiter is fed to the phase detector. At the centre frequency only the carrier will be present and the PLL circuit will lock onto the carrier and is held there by the correction voltage produced by the phase detector (VCV). If the IF was 10.7 MHz then the VCO would now be oscillating at 10.7 MHz. When deviation is present, the carrier swings above and below the centre frequency. **The PLL circuit will attempt to stay in lock by producing a correction voltage (VCV)** which makes the VCO track the carrier frequency swing. **As the correction voltage is proportional to the amount of deviation from the centre frequency, the <u>correction voltage is in fact the de-modulated audio!</u>**

Let's look at the block diagram of the FM transmitters shown in figures 13 and 14. The transmitter in Figure 14 is the most common type. The audio voltage from the microphone is increased in level by the microphone amplifier. The amplified audio is then applied to the reactance modulator which, in most cases, consists of little more than a reverse biased varactor diode. The audio voltage will cause the effective capacitance of the diode to vary. The varactor diode is connected across the crystal in the 10.695 MHz oscillator; thus the applied modulating audio makes the crystal oscillator vary above and below its centre frequency. The microphone amplifier will have an internal gain control that can be adjusted to obtain the required amount of deviation from the crystal. The **output of the crystal oscillator has the required deviation for transmission**. The signal is then fed to the mixer stage along with a very high frequency carrier from the heterodyne oscillator. For stability, the heterodyne oscillator would consist of a complete PLL circuit.

At the output of the mixer will be the required transmit signal which is passed through a band pass filter in order to remove other unwanted mixing products. The FM signal is then amplified by the power amplifiers that typically consist of 3 to 4 active stages. RF amplification can be Class C for maximum efficiency.

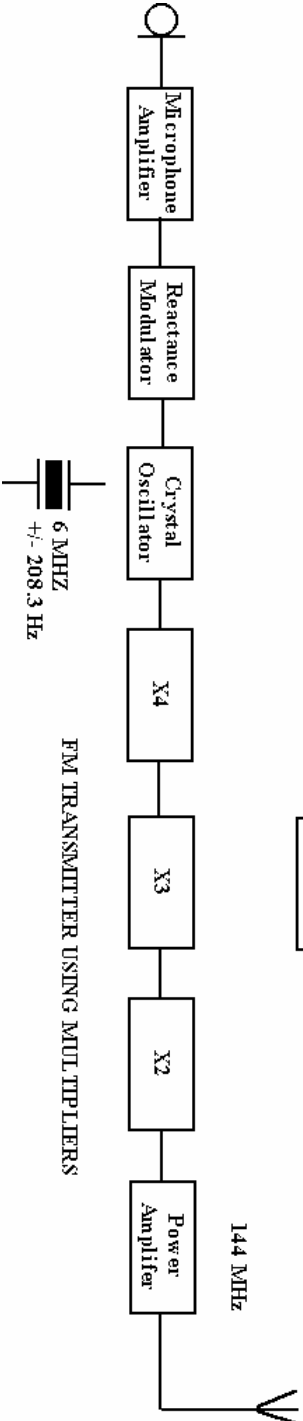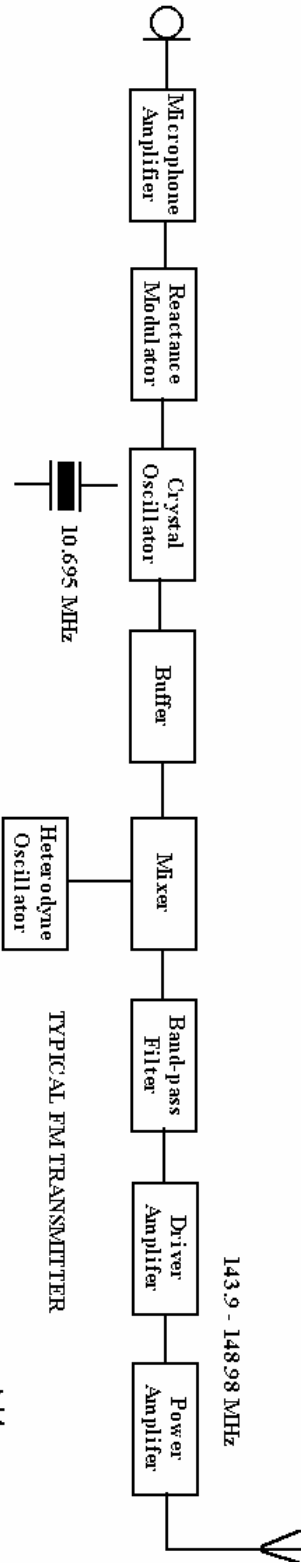# The complete block diagram of two types of FM transmitter:

Figure 13.   **Figure 14.**

Figure 13. FM TRANSMITTER USING MULTIPLIERS

Microphone Amplifier → Reactance Modulator → Crystal Oscillator (6 MHZ +/- 208.3 Hz) → X4 → X3 → X2 → Power Amplifier → 144 MHz

Figure 14. TYPICAL FM TRANSMITTER

Microphone Amplifier → Reactance Modulator → Crystal Oscillator (10.695 MHz) → Buffer → Mixer (Heterodyne Oscillator) → Band-pass Filter → Driver Amplifier → Power Amplifier → 143.9 - 148.98 MHz

The earlier amateur FM transmitters more often relied upon **frequency multiplication** to obtain the final transmit frequency. In these transmitters the frequency deviation produced by the reactance modulator was much less than the required +/- 5 kHz, **because frequency multiplication has the effect of multiplying the deviation as well**. A block diagram of an FM transmitter that primarily users frequency multiplication is shown in figure 13. With integrated PLL circuitry now commonplace, the heterodyne technique described is now universally used to obtain the required transmit frequency. **For examination purposes it is important to remember that frequency multiplication multiplies not only the carrier frequency but also the deviation of an FM signal**. Frequency heterodyning (frequency conversion by mixing), however, leaves the deviation unchanged. In FM broadcast transmitters where a deviation of 75 kHz is required, it is usual to use frequency multiplication to obtain the required deviation, then frequency heterodyning to place the FM signal on the correct transmit frequency.

Frequency heterodyning is just another name for a mixer and oscillator combination that forms a frequency converter.

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