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CLASS C RF POWER AMPLIFIERS

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Class C RF Power Amplifiers

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Class C RF Power Amplifiers

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Introduction

A radio transmitter is a device for converting some form of intelligence into electrical impulses suitable for transmission through space. In its simplest form, a transmitter consists of a source of rf energy, called the master oscillator, and one or more stages of rf power amplification.

In practical transmitters, such as the one shown in block diagram form in Fig.1, there are a number of stages between the master oscillator and the antenna. Since each stage is a form of rf power amplifier, let's briefly discuss the particular role each one plays in the overall operation of the transmitter.

Immediately following the master oscillator is a stage called the buffer amplifier. Its purpose is to present a light, constant load to the oscillator, which helps maintain a stable oscillator frequency. The FCC requires that very close control of output frequency be maintained on all radio transmitters under its jurisdiction.

The next stage, called a frequency multiplier, produces an output whose frequency is some multiple of the input frequency. The presence of this stage permits the master oscillator to be operated at a frequency lower than the transmitted frequency. It is much easier to design highly stable oscillator circuits at the lower radio frequencies; therefore, one or more frequency multiplier stages are an essential part of most radio transmitters.

The driver and power output stages provide the remaining amplification necessary to supply power to the antenna. This output power may range from less than 100 watts to 1 million watts, depending on the transmitter type and the purpose for which it is to be used. In recent years low power circuits, which formerly used vacuum tubes, have been redesigned to use economical, efficient transistors. In many low to medium power mobile transmitters, transistors are used in all stages delivering as much as 75 watts of rf output power at 175 MHz. In other transmitters, all but the driver and final amplifier (power output) stages have been replaced by transistors. There is every reason to suppose that this trend will continue as the high-frequency power handling capability of transistors is improved.

The vacuum tube, however, is still used in high power stages of transmitters which are employed in the AM, TV, and FM broadcast fields. It will no doubt remain so for quite some time to come. Very large vacuum tubes are required to handle the enormous power outputs of these transmitters. There are three characteristics of all rf power amplifiers, transistors, or vacuum tubes which are of concern to us. They are linearity, power gain, and efficiency.

The linearity of an amplifier is a measure of how closely the amplified output follows the input; in other words, a measure of how much distortion is introduced into the output signal by the amplifier. Linear amplifiers, which introduce very little distortion into the signals they amplify, are a subject in themselves and will be considered in a later lesson.

The power gain of an amplifier, usually expressed in db, tells us how much the power level of the input signal is increased by the amplifier. Power gain depends on circuit design and the tube or transistor type used in the circuit. Beam power tetrodes have the highest power gain of any other conventional vacuum tube type. For this, as well as other reasons, the beam power tetrode is the most

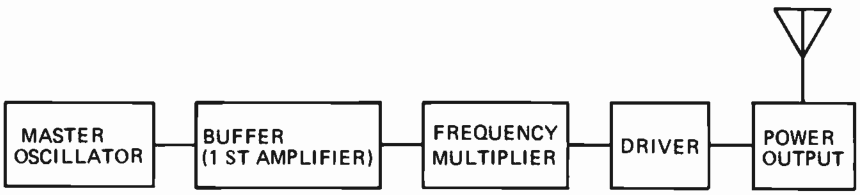


Figure 1. Block diagram of a basic transmitter. All the stages are operated class C.

commonly used tube in modern transmitters. The power gain of transistors does not compare favorably with that of vacuum tubes at higher radio frequencies. This limitation may be partially overcome by adding more stages or using more than one transistor in each stage.

The efficiency of an amplifier, expressed as a percentage, is the amount of dc input power to the stage actually converted to rf energy at the output. In a vacuum tube stage, the power input is the product of the plate supply voltage times the average current. For example, suppose the plate supply voltage is 3000 volts, the plate current 450 milliamps, and the power output of the stage 1000 watts. The dc input power to the stage is:

$$P = E \times I$$

$$= 3000 \times 0.45 = 1350 \text{ watts}$$

The efficiency of the stage can then be found by using the following formula:

$$\% \text{ Efficiency} = \frac{\text{Power Out}}{\text{Power In}} \times 100$$

$$= \frac{1000}{1350} \times 100 = 74\%$$

In a previous lesson, you learned that class C amplifiers give the highest practical efficiency, ranging up to 75%. This is compared to efficiencies of 35% to 50% for class B and as little as 30% to 35% for class A. However, the high efficiency of class C amplifiers is obtained at the expense of linearity. As you'll remember, output current flows for less than half the input cycle in a class C amplifier. This output current pulse bears little resemblance to the input signal which produced it and is therefore highly distorted.

If the output circuit of the class C amplifier is a resonant tank, this current pulse shock-excites the tank so that a complete sine wave is produced. Thus the nonlinearity of the class C amplifier is effectively eliminated when a single rf frequency (a sine wave) is to be amplified. This, along with the class C amplifier's high efficiency, makes it suitable for many rf power amplifier applications.

In the next section we'll discuss class C rf amplifier fundamentals. The information presented in this section applies to both vacuum tubes and transistors. Later, we'll discuss specific applications of vacuum tube and transistor amplifiers. In the final section, we'll talk about the various adjustments which may be made to both types of amplifier circuits.

RF Power Amplifier Fundamentals

In any amplifier, heat is generated by the current flow through the internal resistance of the stage. The power used to generate this heat represents wasted energy and subtracts from the power that could go to the output. The high efficiency of a class C amplifier is due largely to the fact that current flows for a relatively short portion of the input cycle. It is only during this short conducting period that power-wasting heat is generated within the amplifier. To begin our discussion of class C amplifiers, we'll consider the relationships between the current conducting time and the signal voltage waveforms in an operating class C amplifier.

CURRENT AND VOLTAGE RELATIONSHIPS

The graph in Fig.2 shows the output current pulse produced by an input signal at various dc bias levels. Look first at the signal at bias level 1. This signal is below the cutoff value of the amplifier for all of the negative half-cycle and nearly all of the positive half-cycle. Output current flows only for the time the input voltage goes above cutoff. Now look at the signal at bias level 2. It has the same amplitude as the first signal but, because we've increased the bias, this signal exceeds the cutoff level for a shorter period. As a result, the output current pulse produced also flows for a shorter period and is lower in amplitude. Increasing the amplitude of the input voltage has the same effect as decreasing the bias. That is, the output current flows for a longer time.

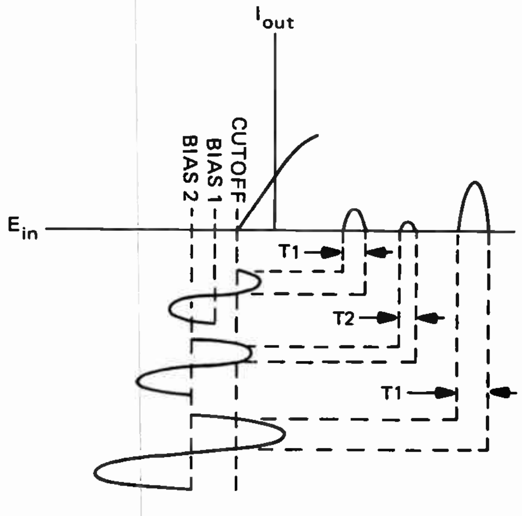


Figure 2. Output current pulse produced by input signals at various amplitudes and bias levels.

Another way of looking at these basic relationships is shown in Fig.3. Figure 3(A) shows two basic class C amplifiers; one uses a vacuum tube, the other a transistor. Figure 3(B) shows the voltage and current waveforms appearing at the inputs and outputs of the amplifiers. Again notice

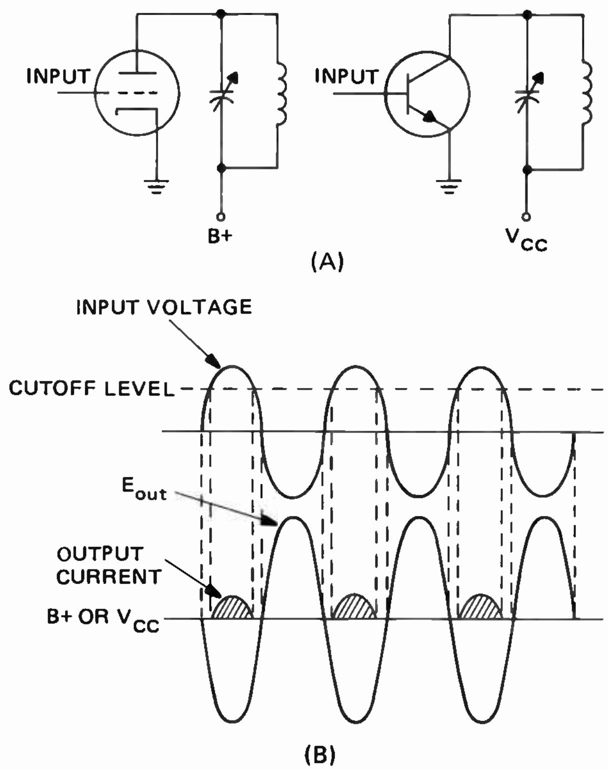


Figure 3. Basic relationships between current and signal voltages in a class C amplifier.

that output current flows only during the period when the input signal exceeds the cutoff level of the amplifier. The output voltage waveform E_{out} is produced by the flywheel action of the resonant output circuit.

Conduction Angle. There are 360 electrical degrees in one complete cycle of a sine wave. The number of electrical degrees the output current flows in a class C amplifier is called the conduction angle or operating angle of the stage. As you've seen, the operating angle depends on both the dc bias and the amplitude of the driving signal. Although amplifier efficiency is higher at the smaller operating angles, the power output is less because the output current pulse is reduced in amplitude and flows for a shorter period. Therefore, the operating angle must be a compromise between maximum efficiency and the highest power output. In making this compromise, the driving signal is maintained at a level sufficient to drive the stage into saturation while the bias is adjusted for the correct operating angle.

Driving Power. To drive a vacuum tube amplifier into saturation requires that the grid be driven positive. The positive grid draws current, causing power to be consumed in the grid circuit. Likewise, in a transistor amplifier, base current flows during the time the driving signal forward-biases the emitter-base junction. The result of this base current flow is that power is consumed in the base circuit.

The power consumed in the input circuit of a class C amplifier, called the driving power, must be supplied by the previous stage. Thus the input circuit of one class C amplifier represents the load on the stage which comes before it. Furthermore, this load presented by the input circuit varies over the period of an operating cycle, reaching a maximum when the input signal causes maximum current to be drawn. As we'll see later, we can use this current flow in the input circuit to develop bias for the stage.

TANK CIRCUITS

The resonant circuit in the output of a class C amplifier has several important jobs to do. We've already mentioned the most basic of these, that of changing the output current pulse into a complete sine wave. This resonant circuit is also required to present the proper load impedance to the stage, and to suppress the undesired harmonics generated within the stage. Let's discuss these last two in detail.

Load Impedance. In order to obtain the maximum power gain from a class C amplifier, or any other amplifier for that matter, the impedance of the load must match the internal impedance of the amplifier. However, do not confuse power gain with power output. It is quite possible that an amplifier operating with a matched load, for maximum power gain, is not delivering its maximum output power.

This is especially true for transistor rf power amplifiers. These amplifiers are very often designed to operate from automotive type battery supplies, thus limiting collector supply voltages to the 12 to 28 volt range. With the load matched to the internal impedance of the amplifier, there may be insufficient collector current flow to give the required power output. Using a value of load resistance lower than the input impedance of the stage results in a greater collector current flow and higher power output. Therefore, in some cases power gain must be sacrificed for power output.

Factors Affecting Impedance. Since the output tank circuit must offer the correct load impedance for the class C amplifier, let's look at some of the factors which affect this impedance. We know that to be resonant, the L and the C of the tank circuit must offer equal reactances at the operating frequency. If we increase the value of L, the value of X_L will increase. To maintain resonance, we must increase X_C by decreasing C. Having increased the value of X_L and X_C by equal amounts, we have increased the total impedance of the circuit without affecting the resonant frequency.

Any resistance present in the tank acts to decrease the total impedance of the circuit. The values of C, L, and R are related to total impedance by the following formula:

$$Z = \frac{L}{CR}$$

From the formula, you can see that increasing the ratio of L to C in the tank causes the impedance to increase. Increasing the resistance in the tank causes impedance to decrease. This leads us to a discussion of tank circuit Q.

Circuit Q. The Q of a coil, as you know, is the ratio of its reactance to its resistance or:

$$Q = \frac{X_L}{R}$$

A capacitor also has a Q, but its value is very large due to the capacitor's low internal resistance. The Q of a tank circuit, therefore, is equal to the Q of the coil.

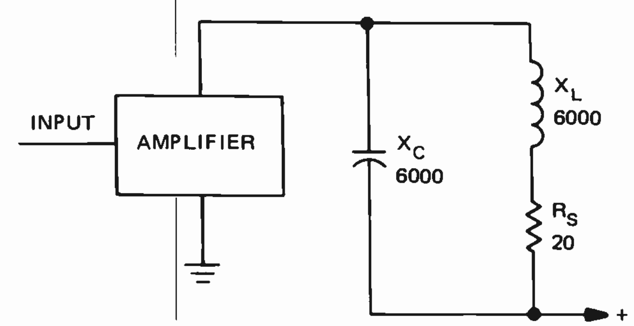


Figure 4. Amplifier with a parallel-tuned output tank showing impedances at resonance.

Figure 4 shows an amplifier with a parallel-tuned tank circuit in its output. At the operating frequency of this amplifier, let's assume the X_L of the coil is 6000 ohms and its resistance (R_S) is 20 ohms. The Q of this unloaded tank circuit then is:

$$Q = \frac{X_L}{R_S} = \frac{6000}{20} = 300$$

The Q of unloaded tank circuits in practical transmitters may range from 200 to 800. Figure 5(A) shows the same amplifier of Fig.4 inductively coupled to a load. This load might be a transmission line, an antenna, or another rf amplifier.

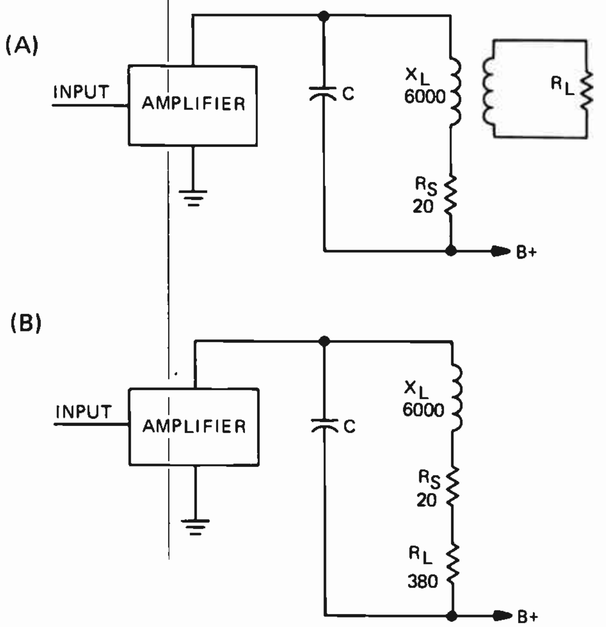


Figure 5. Tank circuit coupled to a load and its equivalent circuit.

The effect of this load is to reflect an additional resistance into the tank circuit. The equivalent circuit, shown in Fig.5(B), contains this reflected resistance (R_L') in series with the resistance of the coil. The exact value of R_L' depends on the value of the load resistance as well as the coupling to the load. We'll assume a value of 380 ohms for our discussion. The Q of the tank now becomes:

$$Q = \frac{X_L}{R_L' + R_S} = \frac{6000}{400} = 15$$

Thus, the Q of the tank circuit went from an unloaded value of 300 to the loaded value of 15 due to the resistance reflected into the tank circuit by the load. From the previous discussion of tank impedance, you know that this additional resistance in the tank also decreases the impedance of the tank. Tank Q and tank impedance are closely related quantities. Factors which change one will also change the other in the same direction.

Let's see now why this is important. You know that only the resistance in a circuit consumes power. Inductive and capacitive reactances, under conditions of resonance, merely transfer energy back and forth between themselves. Therefore, when the tank is loaded, all of the power in the circuit is consumed by the resistances R_S and R_L'. The power consumed in R_L' represents power consumed by the load, while that consumed by R_S is dissipated as heat in the tank circuit. From Ohm's law we derive that $P = I^2 R$, so the power consumed by the load far exceeds that lost as heat in the tank. This is because of a larger value of R_L'. We can actually calculate the efficiency of the tank circuit by the formula:

$$\%\ \text{Efficiency} = \left(1 - \frac{Q_L}{Q_U}\right) \times 100$$

In our example, the unloaded Q (Q_U) was 300, and the loaded Q (Q_L) was 15. Therefore:

$$\%\ \text{Efficiency} = \left(1 - \frac{15}{300}\right) \times 100$$

$$= (1 - 0.05) \times 100 = 95\%$$

Suppose we increased the coupling to the load and reflected a larger value of resistance into the tank. This would decrease Q_L without affecting Q_U, resulting in a higher tank circuit efficiency.

But remember, tank impedance is dependent on the resistance in the tank, so changing the coupling to the load changes the tank impedance. Since the stage is designed for best operation at a particular tank impedance, there is only one correct value of loading on the tank.

Reducing Harmonics. The output pulse of a class C amplifier contains, in addition to the fundamental, numerous harmonic frequency components. As you'll learn later in this lesson, this fact enables us to operate a class C stage as a frequency multiplier. The output tank circuit offers maximum impedance at the frequency to which it is tuned. Harmonic frequencies, seeing a relatively lower impedance, are not developed across the tank circuit in any great magnitude. The circuit which couples the tank to its load is usually designed with harmonic suppression in mind. Sometimes, special traps must be used in output coupling networks which either shunt the harmonics to ground or block their passage to the antenna.

An additional precaution against harmonic radiation is to use an electrostatic shield between two inductively coupled circuits. A shield of this type, called a Faraday screen, is shown in Fig.6.

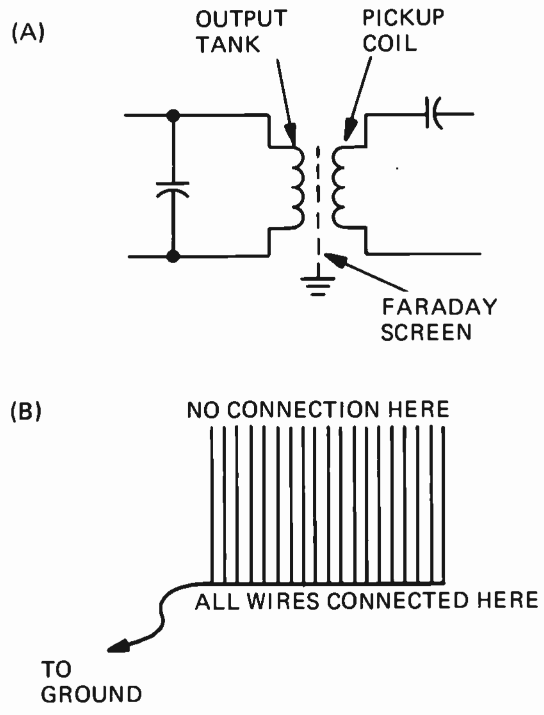


Figure 6. An electrostatic shield or Faraday screen between the output tank and the antenna pickup coil is used to prevent harmonic currents from flowing through the capacity between the coils.

The Faraday screen consists of a number of wires fastened together at one end and open at the other. The ends of the wires that are connected together are grounded. Capacitively coupled harmonic currents will flow to the screen wires rather than to the pickup coil. At the same time, because the wires do not form closed circuits, there can be no voltage induced in them by the magnetic field. Therefore, they do not interfere with the inductive coupling between the output tank and the link coil. This method is very effective in reducing the transmission of harmonics from an output tank circuit to an antenna or transmission line.

COUPLING METHODS

The resonant tank in the output of a class C amplifier forms the basis of the coupling circuit to the amplifier's load. We know that the impedance presented to the stage by the output tank circuit depends, to a large measure, on the equivalent resistance in the tank. We also know that the value of this equivalent resistance is primarily that reflected into the tank by the load. To obtain the correct tank impedance for the amplifier, the coupling must reflect a certain value of resistance into the tank. In most cases, the actual value of the load resistance connected directly across the tank would not reflect the correct resistance into the tank. Therefore, the coupling method must give an impedance transformation. The simplest way to accomplish this impedance transformation is to use a transformer as a method of inductive coupling.

Inductive Coupling. Figure 7 shows two amplifier stages inductively coupled together. In this circuit, the resistance reflected into the output tank for amplifier 1 is adjusted by varying the spacing between the coils. Varying this spacing also adjusts the drive to amplifier 2. The circuit is designed to reflect the correct resistance and provide the proper drive at the same setting.

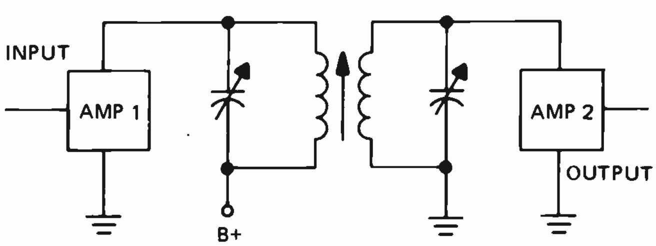


Figure 7. Inductively coupled amplifiers.

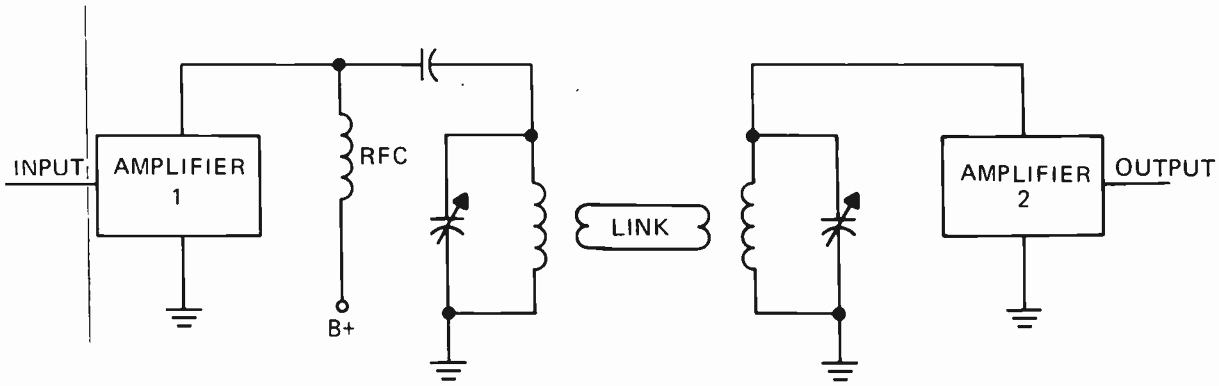


Figure 8. Link coupled amplifiers.

A variation of inductive coupling is shown in Fig.8. This method is called link coupling. It consists of a coil with only a few turns of wire inductively coupled to an output tank. A similar coil is inductively coupled to the load. The connection between the two coils is usually by means of shielded coaxial cable, so it may run some distance with very little loss. Link coupling may also be used between the final power amplifier in a transmitter and a low impedance transmission line. As in the conventional inductive coupling already discussed, the coupling is adjusted by varying the spacing between one of the link coils and the tank. Sometimes the link itself is tuned by a variable reactance. When this is done, the tuned link provides additional suppression of harmonics generated in the previous stages.

Notice that the method of applying B+ to the stage in Fig.8 differs from that of Fig.7. In Fig.8, this voltage is applied through a radio frequency choke (rfc). The rfc offers a very high impedance at the operating frequency, so it keeps the signal voltage out of the power supply. When the power supply, the tank circuit, and the stage are connected in series, as in Fig.7, the amplifier is said to be series-fed. When the power supply, tank circuit, and stage are in parallel (or shunt), as in Fig.8, the amplifier is said to be shunt-fed.

Tapped Tank Circuits. Another method of obtaining an impedance transformation is to connect the load across only a portion of the tank coil. Such a method is shown in Fig.9(A). The value of resistance reflected into the tank is dependent on the position of the tap. In Fig.9(B), the tank capacitor is split to provide the impedance transformation. The values of C1 and C2 determine the value of load resistance seen by the tank (reflected resistance).

The methods shown in Fig.9(A) and 9(B) may be combined as shown in Fig.9(C). With this circuit arrangement, the internal impedance of the stage, as seen by the tank, is transformed to a higher value by the tapped coil. Using this method the required value of loaded Q in the tank may be maintained in spite of low values of internal impedance (such as found in transistor stages). The values of C1 and C2, as before, determine the value of load resistance seen by the tank.

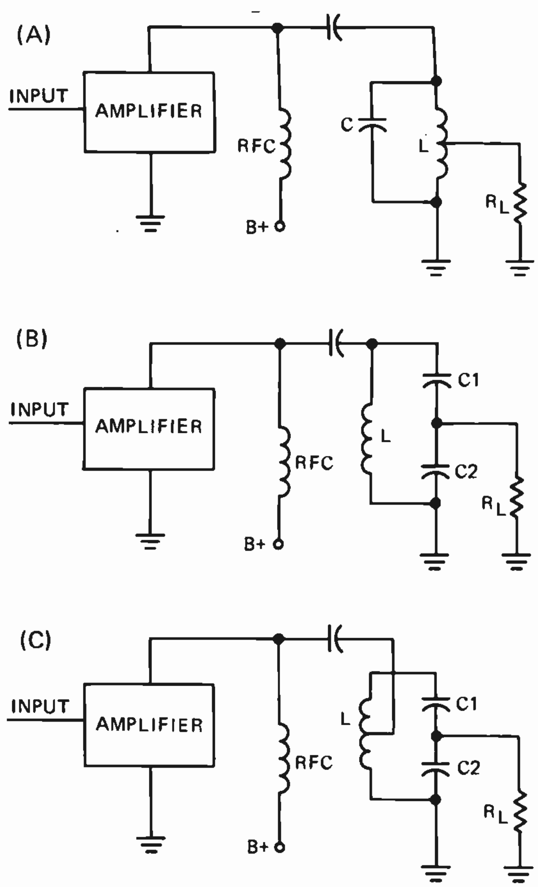


Figure 9. Tapped tank circuits used for impedance transformation.

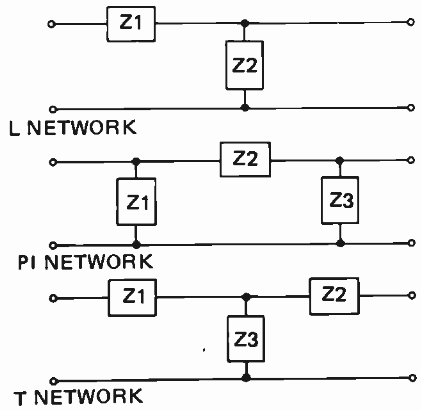


Figure 10. Networks used to couple an amplifier to its load.

Network Coupling. Figure 10 shows three types of networks frequently used to couple class C amplifiers to their loads. The various arms of each are shown as impedances in the figure. In practical networks of this type these impedances will be combinations of L and C components. Later on in your course you'll learn to calculate the reactance values for the arms of these networks to give a required impedance transformation. For now, it is enough for you to know that they fulfill all the requirements of a tank circuit and can provide impedance transformations over a very wide range of values. In addition, these circuits can be easily designed to attenuate undesired harmonic frequencies.

Figure 11 shows an example of how each of these networks (L, π, T) is used to couple an amplifier to its load. The network used in any particular application depends on the magnitude of the impedance levels to be transformed. The networks themselves may be coupled together in a variety of combinations to provide the proper load to the amplifier and greater harmonic attenuation. It is important to remember in the examples of this section that R_L may represent the input of another amplifier, a transmission line, or an antenna. The only difference between these three types of loads, as far as the amplifier is concerned, is the impedance level. An antenna or transmission line usually offers a lower impedance, and therefore a greater load, to an amplifier than the input circuit of another amplifier.

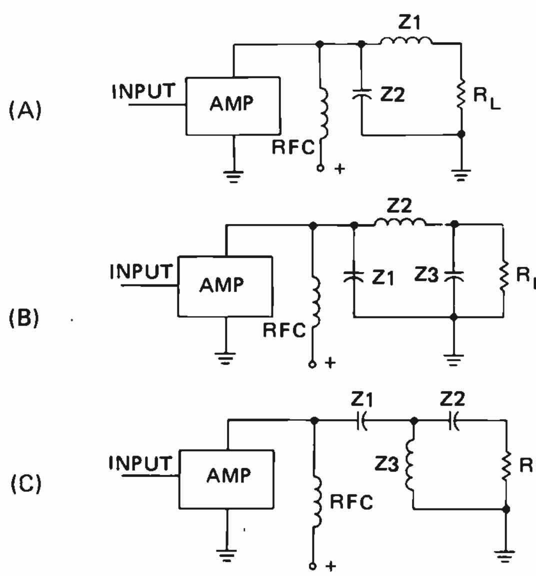


Figure 11. Amplifiers using L, π, and T network coupling to load.

SELF-TEST QUESTIONS

Please check your answers on page 30.

1 What is the primary reason class C amplifiers operate at higher efficiencies than class A or class B amplifiers?

2 What two factors affect the operating angle of a class C amplifier?

3 Is impedance matching between an amplifier and its load always desirable? Why?

4 Suppose we wanted to increase the impedance of a parallel-tuned tank without changing the coupling or the resonant frequency. What components in the tank should be changed and in what direction?

5 If the coupling to a tank is adjusted to increase the load on the tank, what would happen to the loaded Q? The unloaded Q? The efficiency of the tank?

6 If the loaded Q of a tank decreases, what would happen to the tank impedance?

7 How does the output tank circuit reduce the harmonics present in the output of a class C amplifier?

8 How would you increase the coupling between two inductively coupled amplifiers?

9 Is amplifier 1 (shown in Fig.6) series-fed or shunt-fed?

10 In a transistor rf power amplifier, the internal resistance of the stage is found to load the output tank so heavily that a high enough loaded Q cannot be obtained. If the amplifier is connected to the tank as shown in Fig.6, what change could be made in the circuit to increase the loaded Q of the tank?

11 Normally, which would cause a heavier load on a class C amplifier: an antenna or another class C amplifier?

Vacuum Tube RF Power Amplifiers

Now that you have a basic understanding of rf power amplifiers, let's take a detailed look at some applications which use vacuum tubes. In the first section we'll discuss methods of obtaining the class C bias necessary to get the proper conduction angle from the amplifier. Then we'll look at some of the methods employed to ensure stable amplifier operation. Finally, we'll take a look at some practical rf amplifier circuits, including frequency multipliers.

BIAS METHODS

Some typical class C bias methods are shown in Fig.12. The three most common biasing methods are shown at (A), (B), and (C).

External Bias. In Fig.12(A), the bias is obtained from an external bias supply and is coupled through an isolating rf choke to the grid of the stage. The rf choke acts as a high impedance and prevents the power supply circuit from acting as a shunt for the radio-frequency energy.

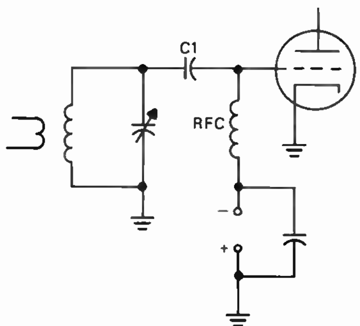


Figure 12(A). External bias.

Grid-Leak Bias. In Fig.12(B), grid-leak bias is used. With this method of biasing, the grid current that flows during the crest of the positive half of the cycle of the incoming signal charges capacitor C1 to a high negative value. During the interval between grid current pulses, the capacitor discharges through grid-leak resistor R_g. The discharge current develops a steady negative voltage across R_g. The value of this voltage depends upon the value of R_g and on the current through it. One advantage of this circuit is that the bias adjusts itself when the driving power is changed. Increasing

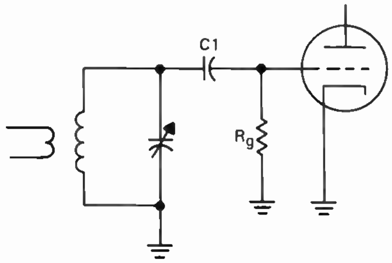


Figure 12(B). Grid-leak bias.

the driving power increases the grid current and therefore increases the voltage drop across R_g. Thus, with grid-leak bias, changing the driving power does not appreciably change the operating angle.

A disadvantage of using grid-leak bias alone is that if there is a failure in the preceding stage, so that no excitation is supplied to the grid of the amplifier, there will be no bias developed. Excessive plate current will then flow, and if the circuit is not suitably protected, the tube and its associated components will be damaged.

Combination External and Grid-Leak Bias. To protect the tube against loss of bias, a combination of external and grid-leak bias, shown in Fig.12(C), is often used. This circuit has the self-adjusting features of the circuit in Fig.12(B), and at the same time provides enough bias to protect the tube if there is a failure in the preceding stage.

Cathode Bias. The tube can also be protected against the loss of excitation by using the cathode bias combination, shown in Fig.12(D). The amount

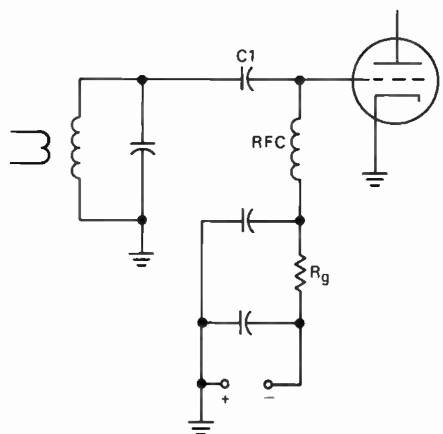


Figure 12(C). Combination external and grid-leak bias.

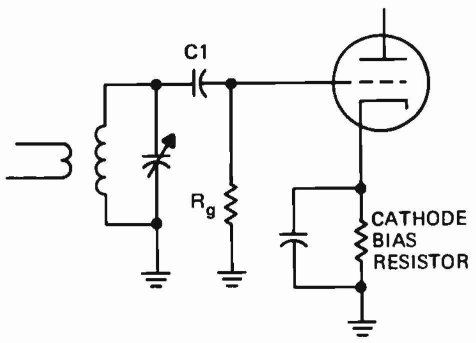


Figure 12(D). Cathode bias.

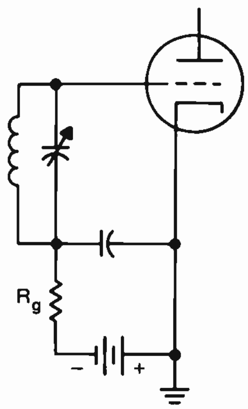


Figure 12(F). Variation of circuit shown in Fig.12(C).

of protective bias, in either Fig.12(C) or Fig.12(D), is chosen so that the plate current through the tube multiplied by the plate voltage is equal to or less than the maximum safe plate dissipation of the tube. The disadvantage of cathode bias is that part of the power supplied to the plate circuit of the tube is wasted in the cathode resistor. In large high-power stages this might be a substantial amount.

Variations. The circuits in Fig.12(A) through 12(D) show the basic class C bias methods. There are also some minor variations of these circuits.

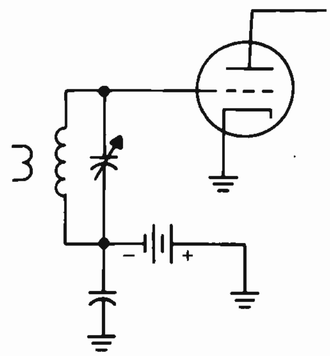


Figure 12(E). Variation of circuit shown in Fig.12(A).

For example, the circuit shown in Fig.12(A) may be rearranged as in Fig.12(E), eliminating the rf choke and coupling capacitor. The circuit in Fig.12(C) may be rearranged as in Fig.12(F). Perhaps we should remind you that you will find minor variations in many circuits.

Do not conclude that a circuit is necessarily different from the basic circuit you have studied just because it has been changed slightly. Study the circuit carefully and you will probably find that the method of operation is basically the same.

AMPLIFIER STABILITY

Class C amplifiers using triode tubes will go into self-oscillation easily because of feedback between the input and output circuits. As you will learn in a later lesson, a tuned-grid, tuned-plate oscillator is simply an unstable class C amplifier.

The feedback path in a triode is through the grid-to-plate capacity. Since this capacity is quite large in a triode tube, enough energy from the plate circuit can be fed back to the grid to overcome the grid-circuit losses and cause the stage to oscillate at a frequency near the resonant frequency of the tuned circuits.

Oscillation will not take place if the plate tank circuit is tuned precisely to resonance; the tank circuit must be slightly detuned to sustain oscillation. Precise adjustment, however, is very difficult. Even if you were able to make such an adjustment, the amplifier would be unstable. Slight changes in supply voltages and load would cause it to go into oscillation.

The feedback signal in a triode amplifier must be neutralized to prevent oscillation. The stage is neutralized by feeding a second signal back into the grid circuit. This second signal must be of opposite polarity and of the same amplitude as the signal fed into the grid circuit through the grid-plate capacity of the tube in order to cancel the feedback.

The most basic method of neutralization is shown in Fig.13. In this circuit, a coil is inserted between the grid and the plate. In series with the coil is a blocking capacitor, which keeps the dc plate voltage off the grid of the tube. It has no other effect on the neutralizing circuit. The value of the coil is chosen to resonate with the grid-to-plate capacity at the frequency to which the amplifier is tuned.

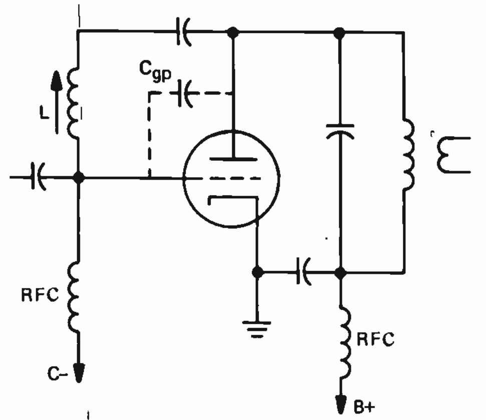


Figure 13. Neutralization for a triode tube stage.

The current through the coil lags the voltage 90°; the current through the capacity leads the voltage 90°. Therefore, the currents through the coil and the capacity are 180° out of phase and cancel each other. The disadvantage of this simple and basic method of neutralization is that it must be retuned when the operating frequency is changed. Let's look at other methods of neutralization.

Plate Neutralization. Figure 14 shows plate or "Hazeltine" neutralization. In this circuit arrangement, the coil used in the plate tank circuit is tapped, and the tap on the coil is operated at rf ground potential by grounding it through the capacitor C.

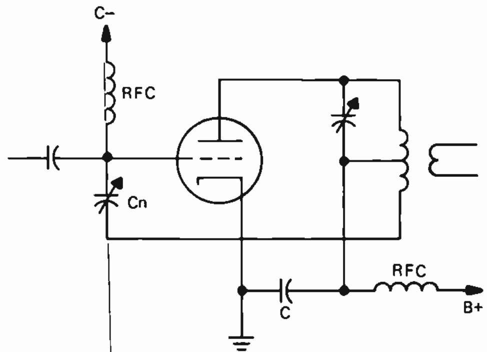


Figure 14. Plate or "Hazeltine" neutralization.

A signal voltage is developed between ground and the bottom end of the coil that is out of phase with the voltage at the plate end of the coil. By connecting the bottom of the tank circuit to the grid of the amplifier through the capacitor Cn, which is called the neutralizing capacitor, we can get a signal at the grid that will cancel the feedback from plate to grid through the tube capacity. Capacitor Cn is adjustable so that we can apply the exact amount of signal needed to cancel out the signal fed back through the tube.

The plate neutralization system can be considered as a balanced bridge circuit. A bridge circuit is shown in Fig.15(A). The input voltage is applied between terminals A and B and the output voltage is taken off between C and D. If the ratio of impedance Z1 to impedance Z2 is equal to the ratio of Z3 to Z4, the voltage between terminals C and D will be zero, and we say the bridge is balanced.

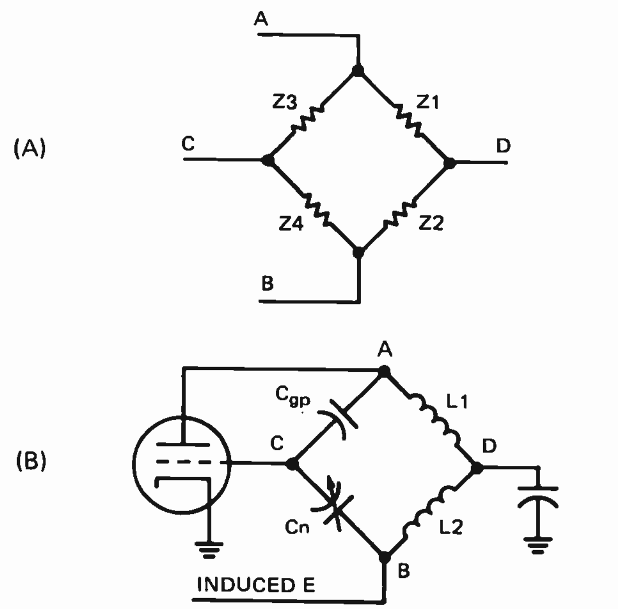


Figure 15. Equivalent bridge arrangement of plate neutralization circuit.

The plate neutralization system in Fig.14 can be redrawn as a bridge circuit as shown in Fig.15(B). The voltage is applied to terminal A from the plate of the tube. The voltage applied to terminal B is the voltage induced in the lower half of the coil in the plate circuit of the tube. We have labeled this half of the coil L2, and the upper half L1. Terminal C of the bridge is connected to the grid of the tube and terminal D is grounded. L1 is made equal to L2 by center-tapping the coil. When Cn is adjusted to equal C_{gp}, the ratio of L1 to L2 will be equal to the ratio of C_{gp} to Cn. Then the bridge will be balanced, so there will be no voltage fed back to the grid circuit from the output circuit.

With this type of circuit, once the stage is neutralized it will remain neutralized over a reasonably wide frequency range, if the coil is exactly center-tapped, so that L1 is exactly equal to L2. If L1 is not exactly equal to L2, the stage can still be neutralized simply be making the ratio of the impedance of C_{gp} to the impedance of Cn equal to the ratio of the impedance of L1 to the impedance of L2. If there is an appreciable difference between the values of L1 and L2, the frequency range over which the stage will remain neutralized becomes limited.

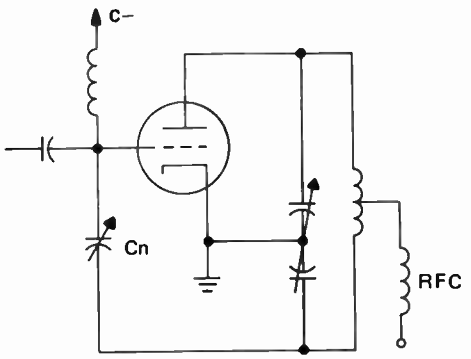


Figure 16. Another plate neutralization circuit.

Another circuit for plate neutralization is shown in Fig.16. In this circuit the center tap on the coil is not operated at rf ground potential; it is connected to B+ through an rf choke. The ground point is taken at the rotors of a split-stator tuning capacitor. A split-stator capacitor is a variable capacitor with one set of rotor plates and two sets of stator plates that are insulated from each other. The voltages at the two ends of the coil are equal and of opposite phase. The circuit in Fig.16 can be shown as a balanced bridge by substituting the sections of the split-stator tuning capacitor for coils L1 and L2 in Fig.15(B). The rf ground in both circuits is made through a capacitor (marked C in Fig.16).

Grid Neutralization. The tapped-grid circuit arrangement shown in Fig.17 can also be used for neutralization. This is referred to as grid neutralization or Rice neutralization. The neutralizing signal is taken from the plate and applied to one end of a tapped coil in the grid-tuned circuit. The rf ground connection is made to the center tap on the grid coil. The polarity of the feedback signal, introduced through the neutralizing capacitor Cn to one

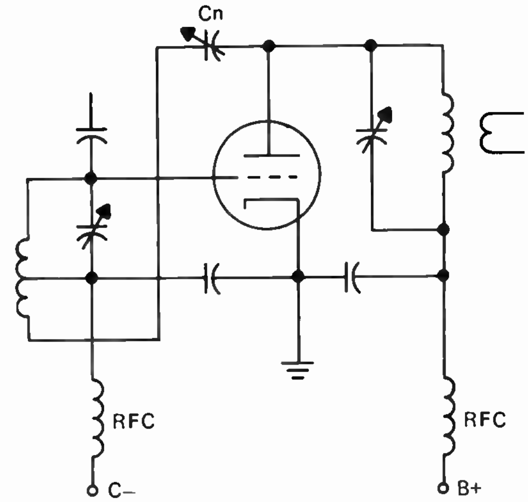


Figure 17. Grid or "Rice" neutralization.

end of the grid coil, is in phase with the signal that is fed directly to the other end of the grid coil through the plate-grid capacity.

By properly adjusting the neutralizing capacitor Cn, voltage fed through it can be made equal to the voltage fed through the tube capacity. These two voltages will cause equal currents to flow through the grid coil in opposite directions. These currents will induce new voltages in the grid coil which will tend to cancel the voltage fed through Cn and C_{gp}. A split-stator version of grid neutralization is shown in Fig.18. Its operation is essentially the same as that of Fig.16.

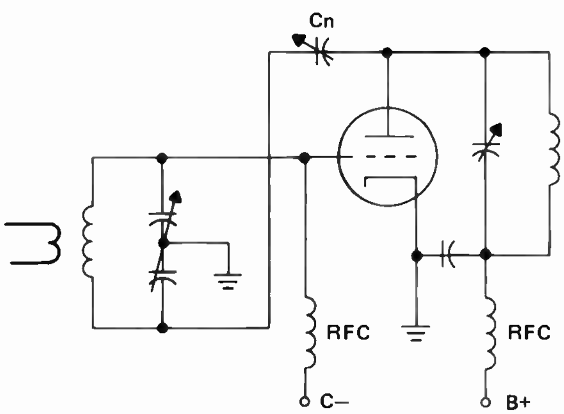


Figure 18. Split-stator grid neutralization.

Inductive Neutralization. Still another method of neutralization is shown in Fig.19. This is referred to as inductive neutralization, because the neutralizing signal is obtained by inductive coupling between the plate and grid-tuned circuits. The signal induced in the grid circuit by the inductive

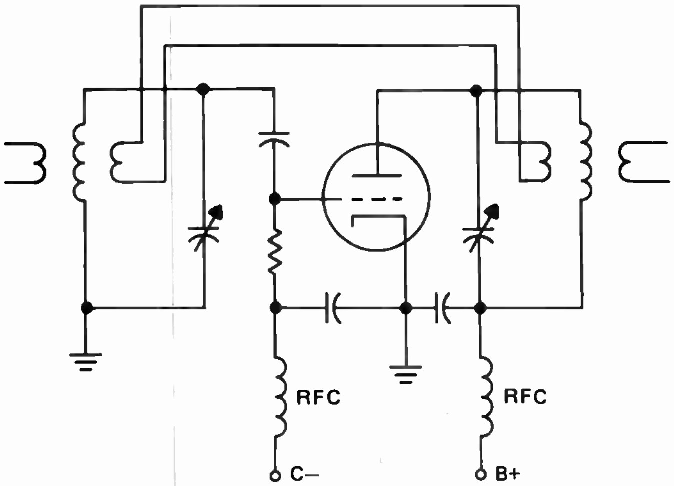


Figure 19. Inductive neutralization.

link is opposite in polarity to the feedback signal, and gives feedback cancellation.

Parasitics. Neutralization of an amplifier is used to prevent oscillation at the frequency to which the grid and plate circuits are tuned, in other words, at the signal frequency. Some amplifiers go into oscillation at frequencies far removed from the desired signal frequency. Oscillations of this type are called "parasitic oscillations" or "parasitics."

The neutralization circuits we have just studied can do nothing to prevent this type of oscillation. Long leads, tube interelectrode capacities, rf chokes, and bypass capacitors are the major inductive and capacitive elements that cause parasitic oscillations.

Parasitics may exist at low or high frequencies, or at both low and high frequencies at once. They cause low operating efficiency and instability in the stage, erratic meter readings, radiation of improper carriers and sidebands, distortion, overheating of the amplifier tube, and premature breakdowns in the circuit parts. If grid-leak bias is used in the stage, parasitics will also cause changes in the grid bias.

Figure 20(A) shows a typical class C amplifier stage. At the operating frequency, the grid and plate circuits are tuned by the coil and capacitor combinations L1-C1 and L2-C4. The stage is prevented from oscillating at the operating frequency by the signal fed back through neutralizing capacitor Cn.

Figure 20(B) shows what the effective circuit would be if this stage were producing low-frequency parasitic oscillations. The grid circuit is now tuned by the parallel combination of the grid choke, RFC1, and the grid bypass capacitor, C2.

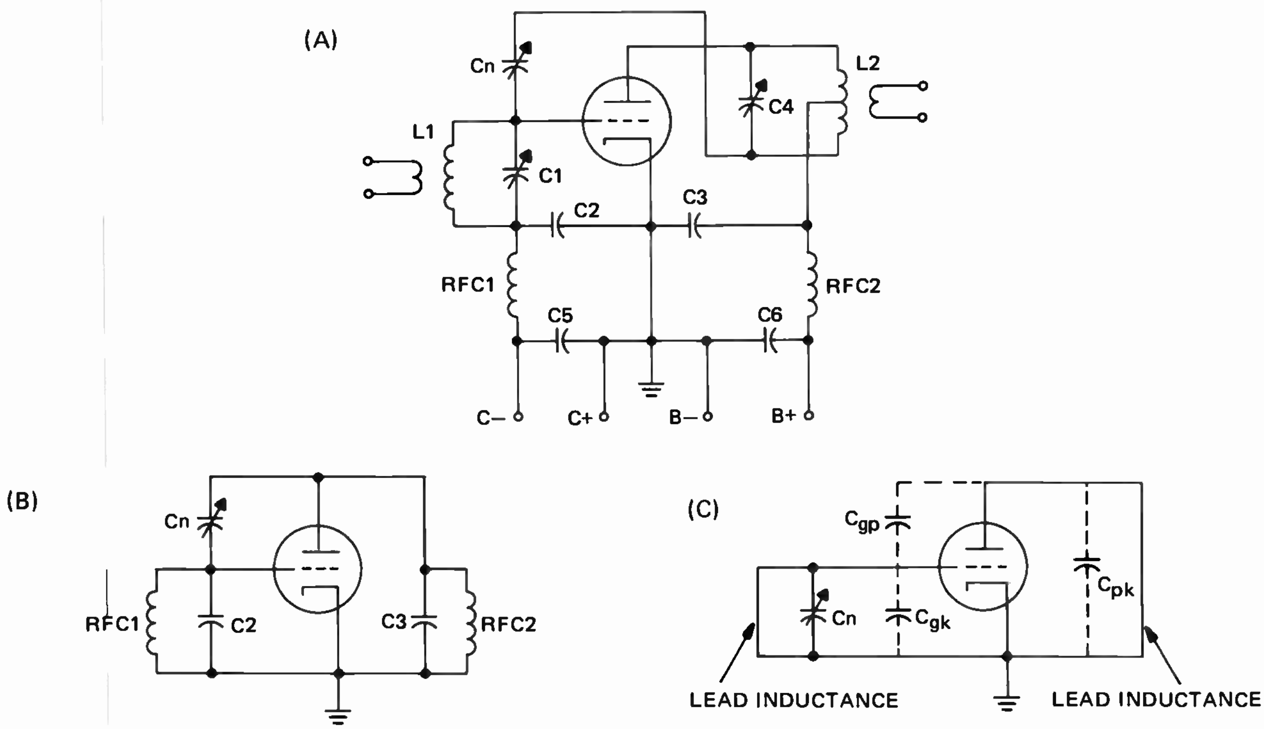


Figure 20. A typical class C stage is shown at (A); the effective circuit that produces low-frequency parasitics is shown at (B); the effective circuit that produces high-frequency parasitics is shown at (C).

Since these oscillations usually take place at frequencies below 200 kHz, coil L1 has very little reactance and serves merely as a connecting lead from the grid to the junction of C2 and RFC1. This places the rf choke and grid bypass capacitor in parallel between grid and ground.

The tuned circuit in the plate at the low frequencies is the plate bypass C3 and the plate rf choke. Here, too, the regular tank coil L2 has practically no reactance at the oscillation frequency, and serves simply as a connecting lead. The neutralizing capacitor Cn is now effectively in parallel with the tube grid-plate capacity and increases rather than reduces feedback. Coils L1 and L2 do have a slight reactance at the parasitic frequency, so tuning capacitors C1 and C4 can make slight changes in the parasitic oscillation frequency.

The effective circuit for the stage, if it were producing high-frequency parasitic oscillations, is shown in Fig.20(C). In this case, the grid and plate circuits are tuned by the inductance of the leads between the tube elements and the tank circuits and the grid-to-cathode capacities. At the high frequencies, the capacities of C1 and C4 are so high that they act only as connecting leads in the inductive circuit. Capacitors C2 and C3, which are even larger in size, have practically no reactance at this frequency. The neutralizing capacitor Cn now appears between grid and ground and is effective in determining the frequency of the grid circuit. Feedback is through the capacity between grid and plate.

Preventing Parasitics. In the effective circuit of either Fig.20(B) or 20(C), parasitic oscillation can be prevented by making the resonant frequency of the grid circuit higher than that of the plate circuit. This may be done in Fig.20(B) by making capacitor C2 smaller than C3 or by making RFC1 smaller than RFC2.

The most satisfactory method for suppressing very high-frequency parasitic oscillations in a class C amplifier stage is by using parasitic suppressors. The purpose of these parasitic suppressors is to increase the circuit losses at the parasitic frequency. Examples of these suppressors are shown in Fig.21.

The suppressors are low resistances, usually around 100 ohms, in parallel with small rf chokes. At the normal operating frequency, these small coils, L1 and L2, have very low inductive reactances, and the signal frequency can pass through them with no loss. At the frequency of the parasitic oscillation, however, these coils have very high reactance and force the parasitic signal to flow through resistors R1 and R2. The loss of parasitic signal in the two resistors is great enough to prevent the tube from going into oscillation at these high frequencies.

Although a commercially manufactured transmitter should be free of parasitic oscillations, occasionally a new transmitter being tuned up for the first time will have them. Parasitics can also occur if parts are replaced by those of a different make. Whenever modifications are made in an amplifier stage, that stage should be checked for both high- and low-frequency parasitics. Low-frequency parasitics will sometimes be evident as sidebands of the carrier frequency. The most common indication of high-frequency parasitics is an unusually high plate current and low output.

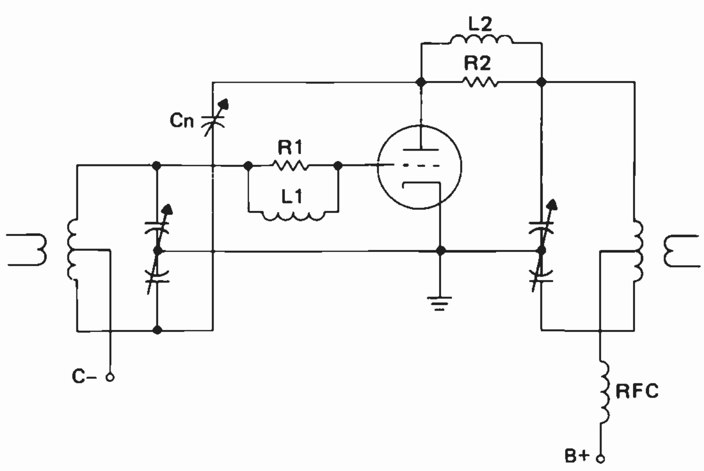


Figure 21. Parasitic suppression methods.

MULTIELEMENT TUBE STAGES

In a previous lesson you learned about the characteristics of screen-grid, pentode, and beam-power tubes. Let us review briefly their characteristics with respect to their use as class C amplifiers.

If a tetrode or pentode tube is used in the stage, the screen grid of the tube acts as an electrostatic shield between the grid and plate, which reduces the grid-to-plate capacity. Therefore, tetrode and pentode tubes are less susceptible to feedback and self-oscillation, and usually do not require neutralizing.

Multielement tubes have a higher power gain than triodes. In other words, for the same amount of grid driving power, you can get a higher power output from a stage using a multigrid tube than from one using a triode. This means that fewer stages are needed to get the desired power output. It also means that better shielding must be used between input and output circuits to prevent external feedback. Even a very small amount of feedback from the output of a stage back to the input can cause oscillation. Also, because of the higher power gain, parasitic oscillations are more common in tetrode and pentode stages than in triode stages.

In screen-grid (tetrode) class C amplifiers, the minimum plate voltage must not be allowed to swing lower than the screen voltage because the screen then offers a greater attraction to the electrons than the plate. The secondary emission effect, due to electrons bouncing off the plate and being pulled to the screen instead of falling back onto the plate, determines the minimum to which the plate voltage can swing. The grid excitation is adjusted so that the grid swings far enough positive so that the tube draws maximum permissible peak plate current without exceeding the dissipation rating of the plate and grid electrodes.

A pentode tube permits a greater plate voltage swing and, therefore, an even higher power gain. It does so by using a suppressor grid at cathode potential between the plate and screen grid to prevent secondary electrons from moving to the screen grid. Thus, the plate current remains independent of plate voltage to a much lower value of plate voltage. The suppressor grid forces the secondary electrons coming off the plate to return to the plate.

A beam-power tube has characteristics similar to those of a pentode. The tube elements are shaped in such a way that they control the electrons flowing between the cathode and the plate of the tube. Proper shaping of the electrodes sets up a potential barrier between screen and plate to suppress secondary emission.

You will find many multielement tubes used in transmitter equipment because of their higher power gain and simplicity of neutralization. Beam-power tetrodes are the most common.

In many circuits using multielement tubes, no neutralization is used. However, the power gain of these tubes is so great that only a small amount of feedback will set up instability and oscillations. Keeping feedback below the level that will cause instability or oscillation is a real problem. Even if a stage does not oscillate when it is first manufactured, there is no guarantee it will not be unstable when the tube in the stage is replaced. To overcome these problems, manufacturers often neutralize stages using multielement tubes.

In a class C stage using a multielement tube, the screen voltage has as much control, or more, on the plate current and power output as the actual value of the plate supply voltage. The plate supply voltage, however, must be high enough to obtain the necessary plate voltage swing across the plate-tuned circuit. Because the screen grid has so much control, the power output in some transmitters is controlled by varying the screen-grid voltage.

The correct voltage must be applied to the plate of a multigrid tube at all times. If the plate voltage drops to zero or is lower than normal, the screen grid may be damaged. Under these conditions the screen current may be so high that it exceeds the safe dissipation factor.

Screen voltage and current also vary with the grid excitation, particularly if the screen voltage is obtained through a dropping resistor. An increase in grid excitation will cause the screen current to rise and the screen voltage to fall. A decrease in excitation will have the opposite effect.

When a tetrode or pentode stage is being tuned and loaded, the plate and screen voltages should be reduced. Most transmitters using tetrode or pentode power stages have provisions for reducing these voltages during tuning. A nonresonant or unloaded plate tank causes the minimum plate voltage to drop below the screen voltage. Under these conditions, the screen draws excessive current. This may destroy high-power tetrodes in a matter of a few seconds. After the tuning and loading are roughly adjusted, full voltage can be applied to the tube and the adjustments carefully peaked.

MULTITUBE STAGES

To get a higher output from a class C stage, two tubes can be connected in parallel, or in push-pull. For very high power, tubes are operated in push-pull-parallel; that is, two sets of parallel-connected tubes are operated in push-pull.

Parallel Operation. Figure 22 shows a stage with two tubes connected in parallel. In parallel operation, the output power is approximately twice that from a single tube, if the correct driving power is applied and the circuit components and electrode voltages have the correct values.

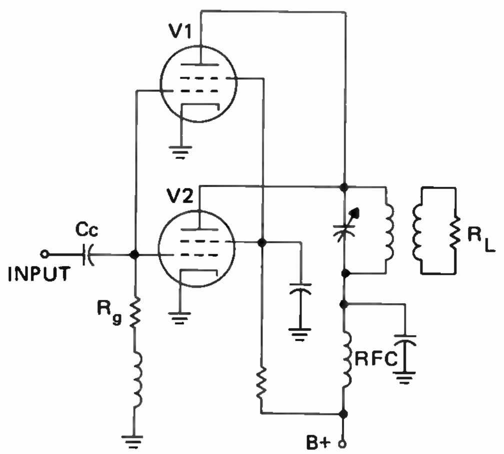


Figure 22. Class C amplifier with two tetrodes connected in parallel.

The grid current is doubled, because with two tubes the grid impedance is approximately halved. The driving power needed for the parallel amplifier is twice that needed for a single tube.

When grid-leak bias is used with the class C stage, the value of the grid resistor must be cut in half to get the same grid bias at twice the grid current.

The internal or plate resistance is also halved because of the parallel connection and doubling of the peak plate current. Thus, the same tuned circuit voltage is developed with twice the plate current. It is the higher amplitude plate current pulses exciting the tuned circuit that develop the added power delivered to the load in parallel operation.

Push-Pull Operation. A push-pull amplifier is shown in Fig.23. The input excitation is applied with equal amplitude but opposite polarity to the grids of the push-pull stage. The ground point of the circuit is at the center of a split-stator variable capacitor.

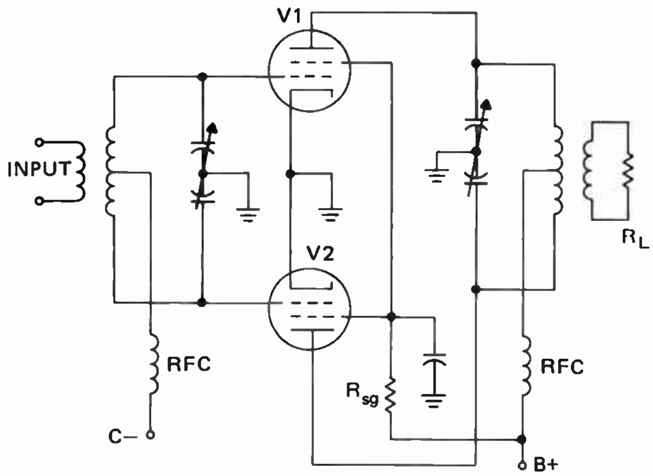


Figure 23. Push-pull class C amplifier using tetrodes.

As in the case of parallel tube operation, the grid and plate currents drawn are twice as great as those drawn by a single tube. To retain balanced operation of the push-pull stage where each tube performs an equal share of the work, the supply voltages are applied at the midpoints of the coils so as not to imbalance the stages.

Balanced operation is necessary to prevent possible overloading of one of the tubes because of any uneven dissipation of power by the grid or plate. Imbalance can be caused by tubes that are not exactly matched or by a mismatch between the grid or plate circuit components. Thus, both the circuits and the tubes must be matched and balanced to get proper operation of the stage.

FREQUENCY MULTIPLIERS

A frequency multiplier stage is a class C amplifier that is used to generate an output signal whose frequency is some multiple of the applied signal frequency. For example, the frequency of the output of a doubler stage is twice the frequency of the input. The frequency of the output of a tripler is three times the frequency of the input. A doubler with an input frequency of 10 MHz would have an output signal of 20 MHz. A tripler with an input frequency of 10 MHz would have an output signal of 30 MHz. Multipliers can be used to generate signals of even higher multiples of their input signal frequency, but the higher the multiplication the lower the output. Thus, you can expect less output from a tripler than from a doubler using the same tube type. A multiplier generating a signal four times the frequency of the input signal would have an even lower output than a tripler using the same tube.

When a tank circuit is shock-excited into oscillation by a single current pulse, the circuit will continue to oscillate for a number of cycles. The number of cycles will depend on the losses in the circuit. Each cycle will be lower in amplitude than the preceding one because of these losses. With a high Q circuit, when the losses are low, there will be many cycles before the oscillation drops to zero.

In a frequency multiplier we take advantage of the fact that oscillation, once started, will continue for many cycles in a tank circuit. By using a tank circuit in the plate circuit of the tube that is resonant at some multiple of the input frequency, we can start the oscillation by feeding an rf signal to the grid. This will produce a plate current pulse that starts the tank circuit oscillating at its resonant frequency which may be two or three times the frequency of the input signal. This oscillation would soon die out, except on the second cycle in the case of a doubler, or the third cycle in the case of a tripler, where the grid of the tube will be driven positive again by the rf signal. This produces another plate current pulse which adds to the energy in the tank circuit and supplies the power needed to make up the circuit losses, so the oscillation in the plate circuit continues. Now let's look at some typical frequency multiplier circuits.

Single-Tube Multipliers. The circuit of a frequency multiplier is very simple; one is shown in Fig.24. It is even simpler than a regular class C amplifier. No neutralization is needed, even when the tube used is a triode, because self-oscillation occurs only if the input and output circuits are tuned close to the same frequency. In a doubler the output tank circuit is tuned to twice the frequency of the input circuit, in a tripler it is tuned to three times the frequency, etc.

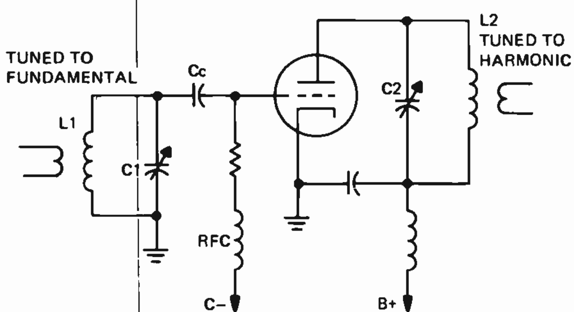


Figure 24. Basic frequency-multiplier stage.

The current waveforms in the tank circuit of a class C amplifier are compared with those in a frequency multiplier in Fig.25. Figure 25(A) shows the waveforms for a fundamental class C amplifier. The plate current pulse flows during part of each cycle of the incoming signal. The flywheel action of the tank circuit develops the fundamental sine wave output, shown by the dashed curve.

Figure 25(B) shows the waveforms for a single-tube doubler circuit. The tube is operated with a higher bias, so the plate current pulse flows

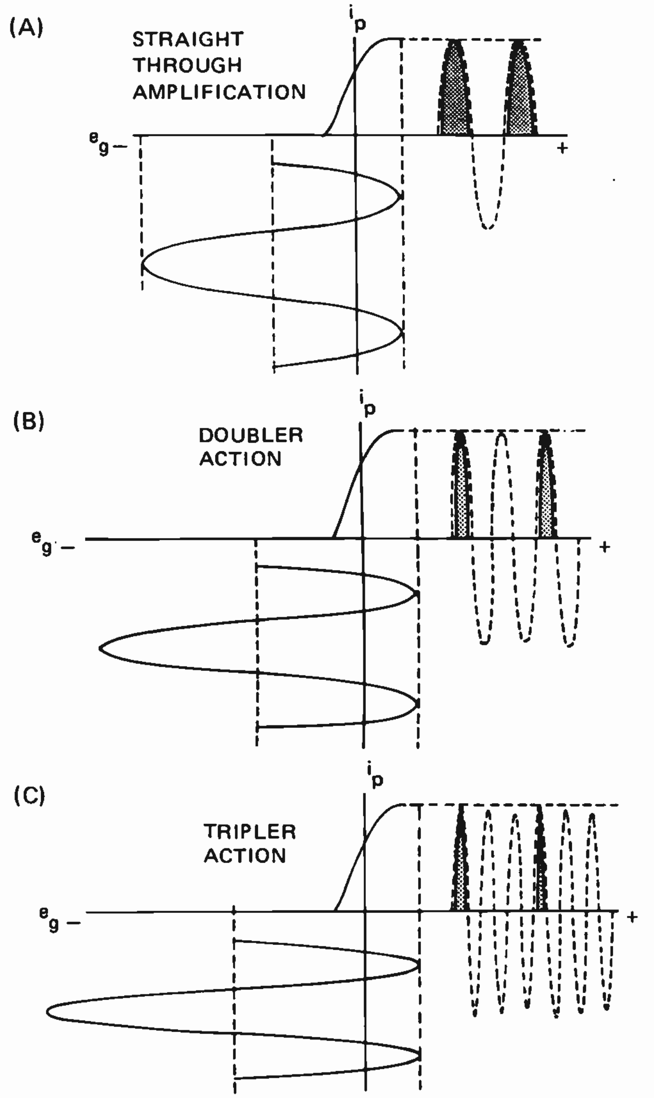


Figure 25. Waveshapes of frequency multipliers compared to the "straight-through" amplifiers.

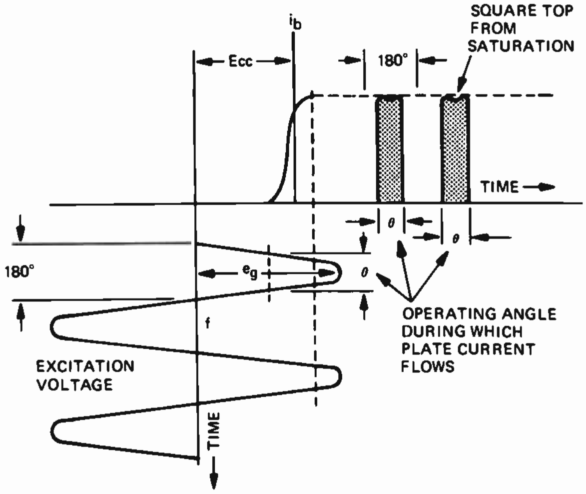
during a smaller part of each cycle of the incoming signal, and the flywheel action of the resonant circuit carries through that cycle and another cycle before the next plate current pulse arrives. Since the plate current flows only on alternate cycles of the output, the power output and efficiency of the stage are lower than for fundamental operation. The efficiency is usually less than 50%.

The higher the harmonic signal to which the tank circuit is tuned, the lower the obtainable power output and efficiency of the class C stage. Figure 25(C) shows the waveforms for a tripler. The tube is biased so that the plate current flows for a still smaller part of the cycle of the incoming signal, and the resonant circuit carries through three cycles before the next pulse arrives. Losses in the circuit cause the amplitude of each succeeding cycle to decrease. The efficiency of a tripler stage is even less than that of a doubler. The efficiency of a multiplier is kept as high as possible by using the proper values of L and C in the tank circuit and correctly shaping the plate current pulse.

The best pulse shape is a square top pulse like the ones shown in Fig. 26. This pulse shape can be obtained by operating the stage with a high bias and then driving the stage to plate-current saturation. For best doubler operation, the angle of current flow, indicated by the Greek letter θ (theta), should be somewhere between 90 and 120 degrees. With this angle of flow, the plate current pulse has a suitable and effective second harmonic content. For tripler operation, the angle of current flow is reduced to less than 90 degrees, and the third harmonic component is emphasized.

The plate tank circuit of frequency multipliers can usually be tuned over a wide enough range to resonate at more than one harmonic of the signal at the grid. Therefore it is important that you check the output frequency to be sure you have the correct harmonic. You can do this with an absorption wavemeter. You'll learn more about this instrument later in the lesson.

Two-Tube Multipliers. A special higher powered and somewhat more efficient doubler can be obtained by using the push-push arrangement shown in Fig. 27(A). In this circuit, the grids are supplied with signals in push-pull and the plates are connected in parallel.



OUTPUT FREQ	OPTIMUM OPERATING ANGLE (θ)
2f	90°-120°
3f	80°-100°
4f	70°-90°
5f	60°-70°

Figure 26. Multiplier operating characteristics.

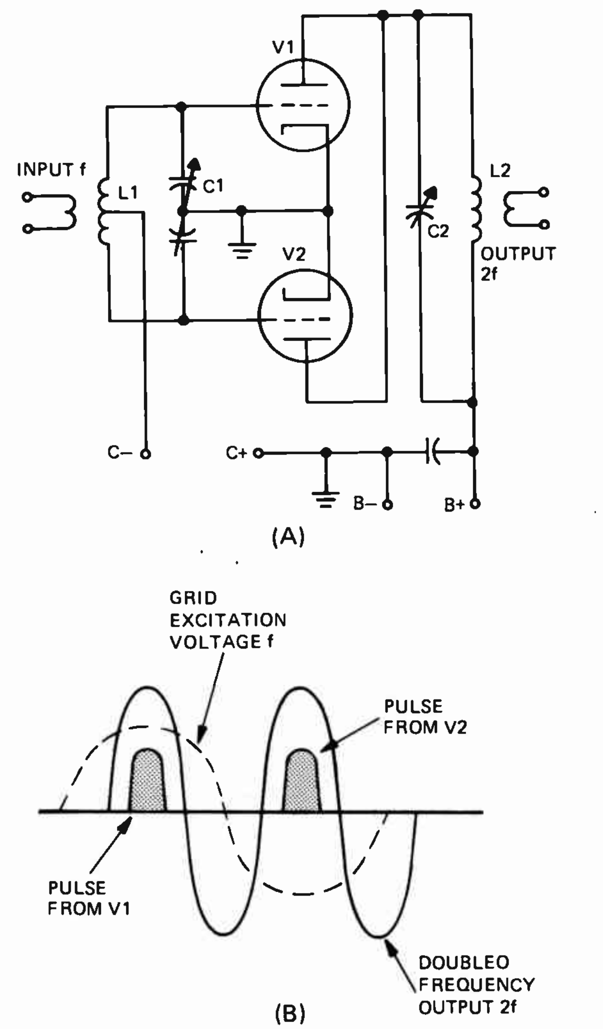


Figure 27. Push-push doubler for even harmonics.

The tubes are connected so that one will conduct on the positive alternation of the incoming signal, shown in dashed lines in Fig.27(B), and the other tube will conduct on the negative alternation. Thus, plate current pulses are fed to the output circuit once during each alternation of the doubled frequency. The efficiency and power output are higher than for a single-tube doubler.

The push-push frequency multiplier stage in Fig.27(A) operates well on even harmonics, but not on the fundamental or odd harmonic frequencies. Frequency doublers are used more often in transmitters, especially low-frequency transmitters, than the higher harmonic generators because of the higher output and efficiency. The grids are connected in push-pull, so they must be fed with balanced signals to get the proper output signal.

Since the doubler is the most frequently used type of frequency multiplier stage, let us list some of its characteristics:

1. The plate tank circuit is tuned to twice the grid circuit frequency.
2. It does not have to be neutralized.
3. The operating angle of the plate current pulse is approximately 90°.
4. The dc bias is about 10 times the plate current cutoff value.
5. The plate current pulse has a greater harmonic content.
6. It requires a very large grid-driving signal.
7. It has a low plate efficiency compared to a fundamental class C amplifier.

As you can see, some of these characteristics vary widely from those of a class C amplifier operating as a fundamental frequency amplifier.

SELF-TEST QUESTIONS

12 What is the disadvantage of using only grid-leak bias in a class C amplifier?

13 What is the feedback path for oscillations near the operating frequency in a triode class C amplifier?

14 What is the main advantage of plate neutralization over the method of connecting a coil and blocking capacitor from plate to grid?

15 In Fig.16, what is the phase relationship between the signal fed through Cn and the signal fed back through grid-plate capacitance?

16 What inductive components in the grid circuit form part of the low-frequency parasitic resonant circuit?

17 What might be the cause of unusually high plate current and low rf output from a transmitter?

18 What two characteristics of tetrodes make them more useful as class C amplifiers than triodes?

19 What characteristic of tetrodes makes them more susceptible to parasitic oscillation than triodes?

20 How would the values of L and C in a tank used in a parallel-connected stage compare with those used in a similar single tube circuit operating at the same frequency?

21 How is balanced operation obtained in a push-pull stage?

22 In general, which multiplier would have a higher output, a doubler or a tripler?

23 Why are neutralization circuits unnecessary in a triode operated as a frequency multiplier?

Transistor Power Amplifiers

In their present state of development, transistors cannot amplify high-frequency signals to high power levels as well as vacuum tubes. However, power outputs greater than 100 watts or frequencies much above 900 MHz are seldom required in many communications applications. Chief among these is commercial mobile radio. In this application, the transistor's small size, low operating voltage, extreme ruggedness, and high overall efficiency make it ideally suited for use in mobile radio equipment.

The common emitter circuit is almost universally used for transistor rf power amplifiers because of its greater stability at radio frequencies. This circuit arrangement is often compared with the grounded cathode triode. As you might expect, it has much in common with the triode circuits you previously studied. With transistor amplifiers we are concerned with the biasing, efficiency, and stability, just as we were with the triode. In this section, we'll look at some typical circuits which illustrate the important features of transistor rf power amplifiers.

BIASING METHODS

You have learned that class C operation of a power amplifier results in the highest percentage of input power being converted to useful rf energy at the output. In the case of a transistor, where high-frequency power handling is a limitation, we are especially interested in getting the highest efficiency obtainable from the stage. It is not surprising then that most transistor rf amplifiers are operated class C.

Two practical methods of obtaining bias for class C operation are shown in Fig.28. The circuits illustrated employ npn transistors; pnp devices could just as well have been used (with all polarities reversed, of course).

In Fig.28(A), reverse bias across the emitter-base junction is developed by the R1-C1 network in the emitter circuit. When the input signal to the stage goes sufficiently positive, base and collector currents flow over the paths indicated by the solid lines. Both these currents flow through the emitter

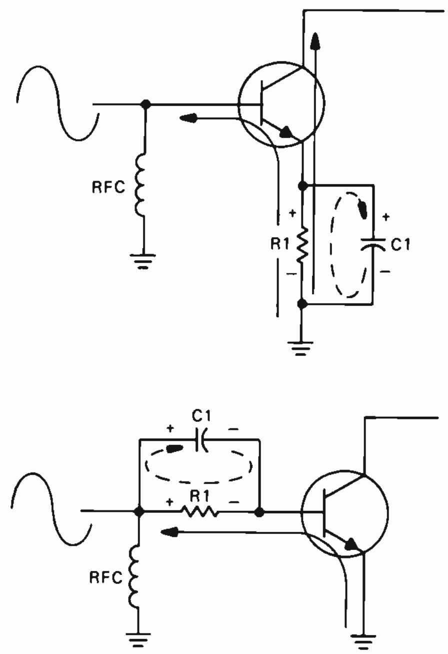


Figure 28. Methods of obtaining emitter-base reverse bias.

resistor, dropping a voltage of the polarity indicated. Capacitor C1 charges to the peak value of this voltage drop. During the time between positive-going portions of the input signal, C1 slowly discharges through R1 as indicated by the dashed line. The values of R1 and C1 are such that C1 does not discharge appreciably before the input signal again swings positive, thereby recharging C1. The emitter is thus maintained slightly positive with respect to the base by the charge across C1. Collector current does not flow until the input signal drives the base more positive than the emitter.

Reverse bias for the circuit of Fig.28(B) is developed in the base circuit, again by a parallel combination of R1 and C1. The base current drawn during the positive-going portion of the input signal (indicated by the solid line) drops a voltage across R1 as shown. C1 charges to the peak value of this voltage drop and essentially maintains its full charge during the time between positive-going portions of the input signal. This is possible because the discharge path for C1 (shown by the dashed line) is through R1, the value of which is chosen to permit only a very small discharge current to flow. Notice that the polarity of the charge on C1 is such that it subtracts from the positive-going portion of the input signal. This means that the input signal voltage must exceed the voltage across C1 before the emitter-base junction becomes forward-biased, allowing collector current to flow.

The two circuits shown in Fig.28 depend on the presence of an input signal to develop bias. With no input signal present, zero bias is developed. Unlike vacuum tubes, however, transistors do not conduct under zero bias conditions and are therefore self-protecting.

Not only are transistors nonconducting under zero bias conditions, but also a small forward-biasing voltage must exist across the emitter-base junction before collector current begins to flow. Figure 29 shows collector current plotted against emitter-to-base voltage for a typical rf power transistor.

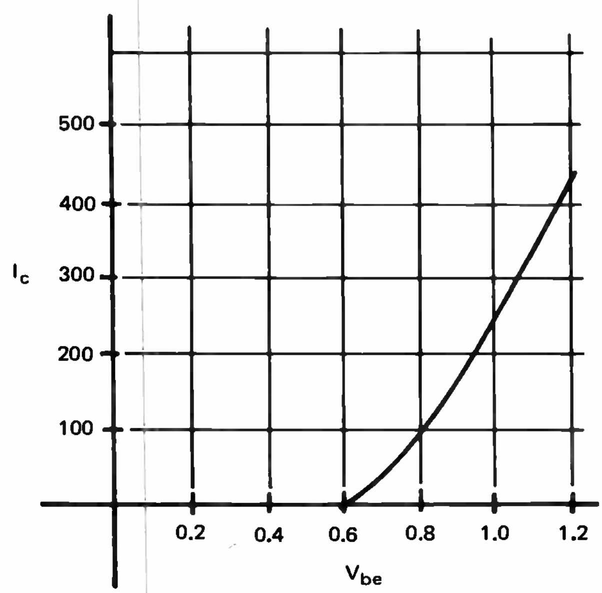


Figure 29. Collector current plotted against emitter-to-base voltage of a typical rf power transistor.

As you can see from the graph, collector current does not begin to flow until the emitter-base voltage reaches approximately 0.6 volt. When we operate a power transistor with zero bias, then, we are actually biased about 0.6 volt below collector current cutoff.

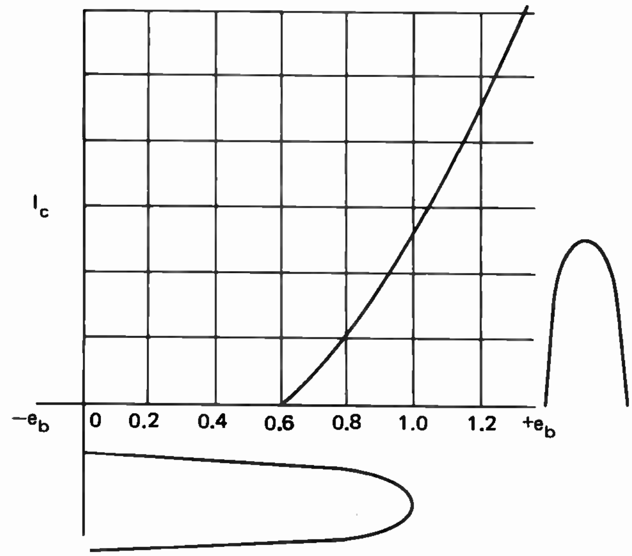


Figure 30. Curve of Fig.29 with signal applied showing collector current pulse.

Figure 30 shows the graph of Fig.29 with an input signal of 1 volt peak amplitude applied. Collector current flows only during the time the input signal is above 0.6 volt. The conduction angle here would be about 120° of the input cycle, well within the class C operating range. Input signal levels of such low amplitude are not unusual in power transistors because of the transistor's low input impedance. Input impedances actually range from several ohms to less than 1 ohm. With such a low input impedance, a relatively large input current is permitted to flow when the base-emitter junction of the transistor is driven positive.

Recalling that P = EI, you can readily see that the driving power to the stage is accounted for primarily by the high current flow which develops only a small voltage across the low input impedance. It follows, then, that a reverse-biased emitter-base junction is not always necessary for class C operation.

MULTITRANSISTOR AMPLIFIERS

As mentioned earlier, the power obtainable from a transistor used as an rf amplifier is rather limited as compared to vacuum tubes used in similar circuits. When more power is required than can be obtained from a single transistor, several transistors can be arranged in push-pull or parallel. In a push-pull arrangement, an input transformer is required to feed out-of-phase signals to the transistors. This transformer is also required to match the relatively high output impedance of the driver to the very low input impedance of the push-pull stage. Such a transformer capable of operation at high frequencies is very expensive to build. For this reason, multiple transistor stages are nearly always parallel-connected.

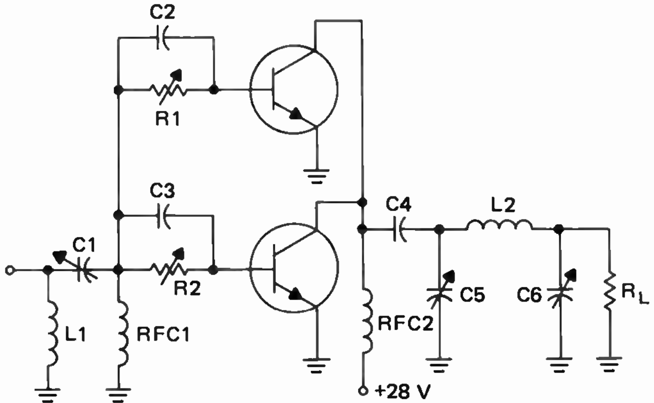


Figure 31. Two-transistor parallel-connected rf amplifier.

Figure 31 shows a two-transistor parallel-connected rf amplifier stage. C1 and L1 form an L network which provides the proper impedance match between the source and the input to the stage. The input signal is developed across RFC1, amplified by the transistors, and applied to the load through the output coupling network. Notice that we can vary the operating bias of the two transistors by adjusting R1 and R2. These adjustments are necessary so that the two transistors will share the load equally. In practice we would insert a milliammeter in the collector or emitter circuit of each transistor and adjust R1 and R2 for equal currents. With the current balanced, each transistor will be handling half of the power delivered to the output coupling network. C4 is a coupling capacitor and may be considered a short circuit at the operating frequency. The output coupling circuit is a pi network consisting of C5, L2, and C6. C5 and C6 are adjusted to provide the proper collector load and circuit Q for the transistors.

Another circuit employing transistors in parallel is shown in Fig.32. Besides containing three transistors instead of two, this circuit differs from the one in Fig.31 in two other important respects. First of all, load balancing is obtained by adjusting L1, L2, and L3 in the base circuits. These adjustments vary the rf drive to the transistors and equalize the load currents as previously discussed. Secondly, each of the three transistors in Fig.32 has a separate output tank circuit connected to a common load as in Fig.31.

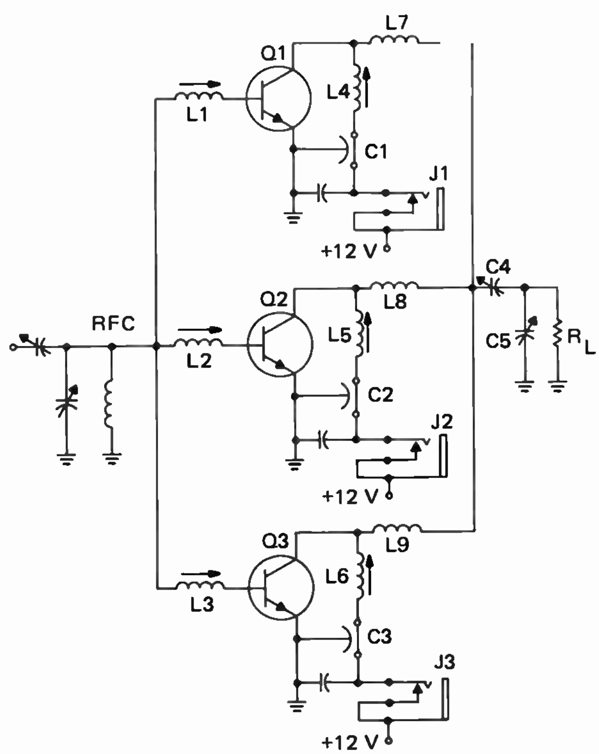


Figure 32. Three-transistor parallel-connected rf amplifier.

The tank circuit for Q1 is made up of L4, L7, and stray capacity. L5 and L8 are the tank inductances for Q2; L6 and L9 are the tank inductances for Q3. These tank circuits are also tuned by stray capacity. The right-hand ends of L7, L8, and L9 connect to C4 which, along with C5, varies the coupling from the three tank circuits to the load R_L.

There are two reasons why separate tank circuits are used for the three transistors. First, each transistor is series-fed, thus eliminating the losses and other problems of an rf choke. Second, the dc collector currents are entirely separate so each transistor can operate essentially independent of the other transistors. Thus, if some trouble developed in Q1, this stage could be disconnected and the amplifier could continue to operate at a lower power level. Directly paralleled stages would have to be completely retuned if one stage were to be removed.

Many rf power transistors have their emitters internally connected to the transistor case. This is done to eliminate the stray inductance of the emitter lead. When the case is connected to ground, as it would be in a circuit such as that shown in Fig.32, current could not be conveniently measured in the emitter circuit. The use of separate collector loads, however, enables convenient monitoring of collector current. In Fig.32, the jacks labeled J1 through J3 are provided for this purpose.

The symbol used to represent C1 through C3 may be unfamiliar to you. This is a special type of capacitor called a feedthrough and is often used for bypassing in high-frequency circuits. As the symbol suggests, one plate of the capacitor completes a dc path in the circuit; the other plate, usually connected to ground, surrounds the first much like the braided shield in a piece of coaxial cable.

FREQUENCY MULTIPLIERS

Figure 33 shows two transistor frequency multiplier stages coupled together. The output of the tripler, Q1, feeds a doubler, Q2, to give a total frequency multiplication of six. The input signal, which we've designated as Fo, is applied to the base of Q1. With the drive signal present, R1 and C1 develop a relatively high reverse bias across the emitter-base junction. The high reverse bias results in a narrow conduction angle and collector current pulses with a high harmonic content. The collector tank is tuned to the third harmonic, 3Fo, and offers a high impedance only at this frequency.

Signals at the fundamental, as well as those at other harmonics, are bypassed to ground by C2 and C3. The signal at the frequency 3Fo is inductively coupled into the base circuit of Q2. Reverse bias for Q2 is developed by the driving signal in a manner similar to that described for Q1. The collector tank for Q2 is tuned to 6Fo and inductively couples the signal at this frequency into the load. Undesired signals are again bypassed to ground, in this stage by C5 and C6.

While individual stages are seldom designed for frequency multiplications greater than three, any desired total multiplication may be obtained by connecting multipliers together. The usual arrangement in a transmitter is a straight-through class C amplifier following each one or two multiplier stages. In this manner, the relatively low output from the multiplier is built up before being applied to the next multiplier. The straight-through amplifier also offers additional suppression to the undesired harmonics generated in the multiplier stage.

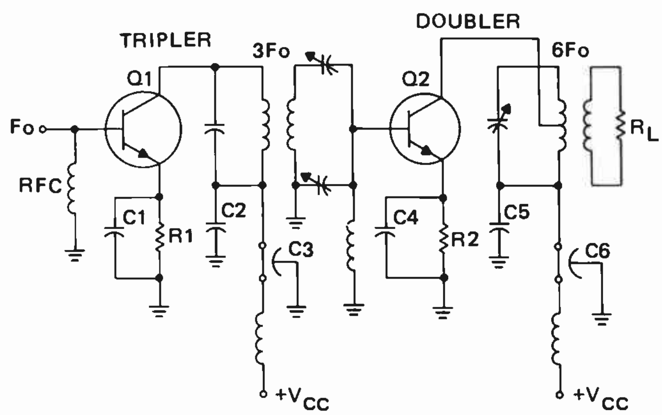


Figure 33. Two frequency multiplier stages coupled together.

AMPLIFIER STABILITY

You learned that, in the triode power amplifier, in-phase feedback through plate-to-grid interelectrode capacitance could cause the amplifier to oscillate. To prevent these oscillations from occurring, components were added to feed back an out-of-phase voltage of equal amplitude and thus "neutralize" the interelectrode capacitance of the tube.

A similar capacitance exists between the collector and base of a transistor. However, the value of this collector-to-base capacitance in power transistors is voltage-dependent. That is, as the reverse bias across the collector-base junction varies (which it normally does over the period of an operating cycle) the collector-to-base capacitance also varies. To be effective, a neutralizing circuit for a power transistor would have to continuously adjust itself to this variation. Because of this requirement, neutralization of a transistor rf amplifier is normally not practical. Instead, the need for neutralization is usually eliminated by careful circuit design using transistors with low values of interelement capacitance.

Parasitics. In the radio frequency range, the power gain of a transistor falls off rapidly as frequency increases. This characteristic of transistors works to advantage in preventing parasitic oscillations above the operating frequency. At these higher frequencies, the transistor has insufficient gain to overcome the circuit losses and sustain oscillations.

On the other hand, the power gain of a transistor is higher at the lower frequencies. To illustrate this, let's assume we have an rf power amplifier operating at 175 MHz. A typical transistor operating at this frequency might have a power gain of 5 db. This same transistor could have a gain as high as 30 db at 10 MHz. In other words, the power gain of the device is over 300 times greater at 10 MHz, the parasitic frequency, than at 175 MHz, the operating frequency. Consequently, the most common cause of instability in these power amplifiers is parasitic oscillation below the operating frequency.

The amplifier circuit shown in Fig.34 illustrates a number of techniques used to prevent low-frequency parasitics. These are discussed in the following paragraphs.

The rfc connected between base and ground (1) will at some frequency form a parallel-resonant circuit with the emitter-base capacitance. To decrease the efficiency of this parasitic tank circuit, the rfc is designed to have a very low Q (high effective series resistance). Often this rfc is nothing more than a wire-wound resistor.

The emitter bypass capacitor (2) used is the smallest value which will provide effective bypassing at the operating frequency. At frequencies below the operating frequency, the reactance of this capacitor increases, resulting in degenerative feedback at these lower frequencies. This degeneration reduces the gain of the amplifier to low-frequency parasitics.

The output coupling network is designed to include a portion of the network inductance in the collector dc supply line (3). With this arrangement, no rfc is required in the collector circuit. Elimination of the collector rfc is desirable because this component can form a parallel-resonant circuit with the output capacitance at some relatively low

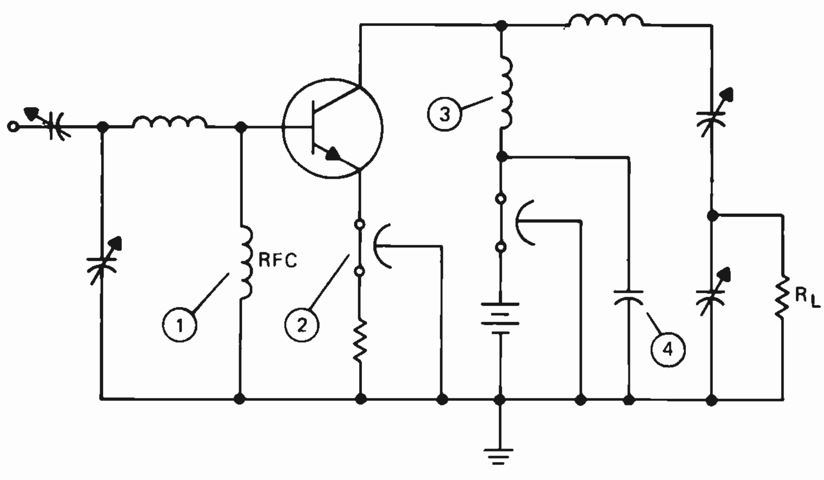


Figure 34. Transistor power amplifier showing components used to prevent low-frequency parasitics.

frequency, thus becoming a possible source of low-frequency parasitics.

In addition to the feedthrough capacitor designed to bypass the power supply at the operating frequency, a second capacitor of larger value (4) provides a short circuit to ground at lower frequencies where parasitics usually occur. You might wonder why a larger capacitor, since it bypasses well at lower frequencies, wouldn't provide an even better bypass at the operating frequency where its X_C would be even less. The reason is that at higher radio frequencies the inductive reactance of the capacitor's leads becomes significantly large. The capacitor, instead of being a short circuit to ground, becomes an impedance to ground at these higher frequencies. The feedthrough capacitor, because of its physical construction, has very low lead inductance, and therefore provides an effective short circuit to ground at the high operating frequency. Feedthrough capacitors can only be manufactured with comparatively small values of capacitance, hence the need for the more conventional larger capacitor for the low-frequency bypass.

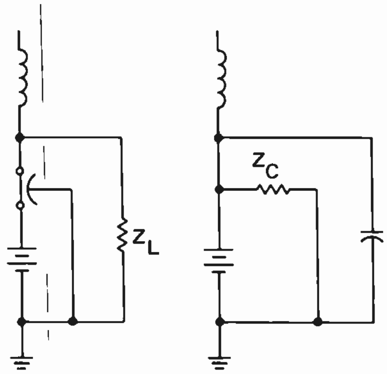


Figure 35. Equivalent circuits for the power supply bypassing arrangement shown in Fig.33.

Figure 35 summarizes what we have said about power supply bypassing. Shown in the figure are the equivalent bypass circuits for both the high operating frequency and the low parasitic frequency. At the operating frequency, the larger capacitor appears as an inductive reactance, Z_L, and the feedthrough as an ac short circuit to ground. At a low parasitic frequency, the feedthrough appears as a high capacitive reactance, Z_C, and the larger capacitor provides the ac short to ground.

SELF-TEST QUESTIONS

24 Which transistor circuit configuration has the greatest stability at radio frequencies?

25 What happens when loss of drive causes the bias on a transistor rf amplifier to drop to zero?

26 Is reverse bias on the emitter base junction necessary for class C operation in a transistor rf amplifier?

27 What characteristic of transistor rf power amplifiers accounts for the low signal voltage developed in the input circuit?

28 Why is a parallel connection of transistors favored over a push-pull connection?

29 What are two advantages of having separate collector tank circuits for a parallel-connected transistor rf amplifier?

30 Why are feedthrough capacitors used for bypassing in high-frequency circuits?

31 What characteristic of power transistors makes neutralization impractical?

32 What is the most common form of transistor rf amplifier instability?

33 Why is an additional capacitor placed in parallel with the feedthrough capacitor bypassing the power supply in Fig.32?

Adjusting Class C Amplifiers

Adjustments to class C amplifiers in transmitter stages are performed following repairs or routinely to compensate for normal circuit aging. The adjustment procedures for all class C stages, either frequency multipliers, intermediate amplifiers, or power output stages, are basically the same. There are variations, of course; when you tune a frequency multiplier, for example, you must make certain that the plate circuit is tuned to the desired harmonic frequency.

In this section, we will go through the complete adjustment procedure for both a vacuum tube and a transistor class C amplifier stage. You should realize that adjustments such as those described in this section may be performed only by a person having the necessary authority. To obtain this authority, he must hold the proper class of FCC License or be under the direct supervision of another person who does.

THE VACUUM TUBE STAGE

Figure 36 shows a typical class C amplifier circuit. Notice that there are current meters in the grid and plate circuits and that voltmeters are used to measure the bias, filament, and plate supply voltages. A power output stage using a screen-grid tube often has a current meter and voltmeter in the screen circuit. In some rf output stages, particularly in the low power exciter stages, only one voltmeter and one or perhaps two current meters are used. These meters are switched into the various stages to check the performance of the stage during operation or during the adjustment procedure.

Neutralization. The first step in the adjustment procedure is to check that the amplifier is properly neutralized. Always check for neutralization with all interstage shields in place. If the amplifier is

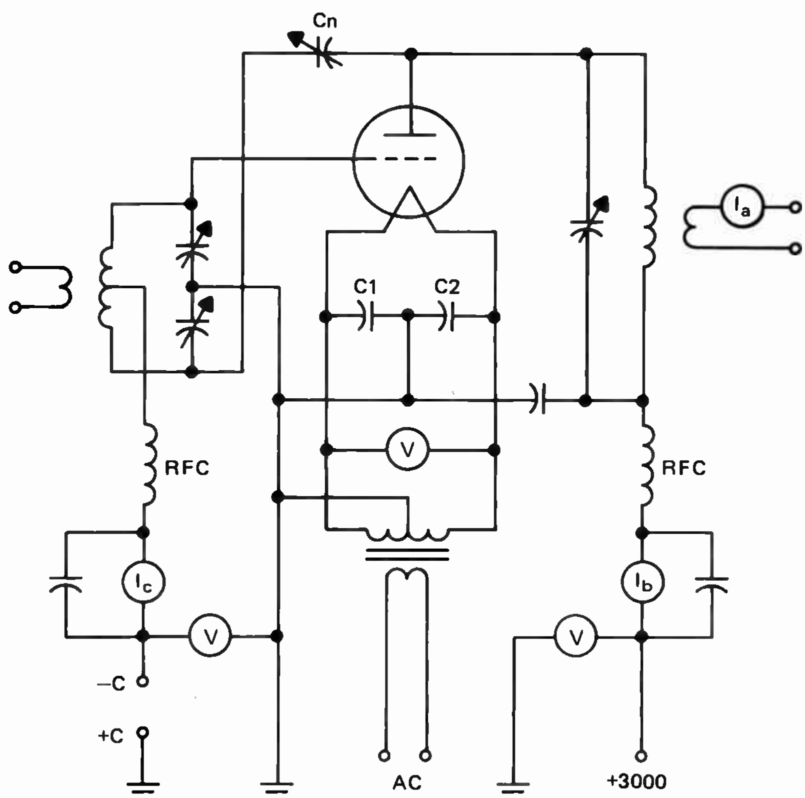


Figure 36. Typical class C amplifier.

enclosed in a separate shield box within the transmitter cabinet, check it with the shield box closed. There will probably be stray magnetic or electrostatic coupling between output and input circuits unless all shields are in place and closed.

Neutralization can correct only for capacitive coupling directly from the grid to the plate of the tube. The simplest indication which can be used to determine correct neutralization is the grid current meter. With B+ removed from the stage, the need for neutralization will be indicated by a dip in the grid current when the plate tank is tuned through resonance. The grid current dips because the power loss in the resistance of the plate tank is greatest at resonance. Since this power is fed into the plate tank through the grid-plate capacitance, it subtracts from the grid current and causes a dip.

A second, more sensitive method of checking for correct neutralization is by use of a wavemeter. As shown in Fig.37, this simple device consists of a parallel LC circuit connected to a diode and dc milliammeter. The variable capacitor, which is calibrated in units of frequency, may be adjusted to make the circuit resonant over a wide range of frequencies. Any energy coupled into the tank circuit is rectified by the diode and causes the meter to deflect. In use, the wavemeter is inductively coupled to the plate tank circuit of the stage to be checked.

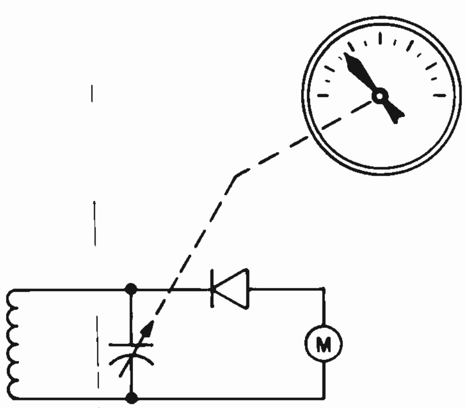


Figure 37. Simplified schematic diagram of a wavemeter.

With plate voltage removed and grid drive applied to the stage, there should be no indication on the meter as the tuning knob is adjusted near the operating frequency. An indication on the meter indicates the presence of rf in the plate tank. This rf could only have come from the grid circuit, coupled through interelement capacitance to the plate tank. Hence, the stage must be neutralized.

The procedure to be used in neutralizing an amplifier is as follows:

1. Remove the B+ from the stage. *Never attempt to neutralize an amplifier with the plate voltage connected.*
2. Set the neutralizing capacitor for minimum capacity.
3. Apply filament power and bias to the stage, and apply filament power, bias, and B+ to all stages ahead of the stage being neutralized.
4. Tune the grid circuit to resonance as indicated by maximum grid current.
5. Tune the plate tank circuit to resonance while watching the neutralization indicator. If it is a grid meter, it will dip; if you are using a wavemeter, it will peak.
6. Increase the capacity of the neutralizing capacitor slightly. Check grid and plate resonance; changing the neutralizing capacitor will sometimes detune both grid and plate circuits.
7. Continue to increase the neutralizing capacitance in small steps until there is no dip in the grid meter or no indication of power in the plate tank. The transmitter is then correctly neutralized.

As you come closer and closer to neutralization, make smaller and smaller changes in the neutralizing capacitor. There is only one correct setting; too much or too little capacitance are equally bad. Remember to retune both grid and plate each time you change the neutralizing capacitor. If the transmitter uses inductive link neutralization, start with maximum coupling to this link. Then reduce the coupling in small steps until you find the correct coupling.

Neutralization must be made as accurately as possible. Although steady oscillation will take place only when enough power is fed back from the output to overcome the input circuit losses, smaller amounts of feedback, which are not enough to cause steady oscillation, can still affect the operation of the stage. An amplifier operating like this is said to be "regenerative."

Several characteristics of an amplifier change when it is regenerative. One of the most pronounced is an increase in input impedance. This increase in impedance causes the Q of the grid and plate tuned circuits to increase also. The increase in

Q makes the circuits selective and hard to tune. To make matters worse, changes in the plate tuning change the amount of feedback and, therefore, affect the grid tuning.

A regenerative amplifier is an unstable amplifier. A slight increase in filament current or plate voltage may cause it to go into steady oscillation. Reducing the load at the output will also cause oscillation.

In a keyed transmitter, a regenerative amplifier will cause damped oscillations every time the key is closed. As a result, undesirable signals are generated. In a phone transmitter, an unstable amplifier causes still other effects.

Perfect neutralization of an amplifier is absolutely necessary. It takes time to do it right, but it does not have to be done often.

After you have completed the neutralizing procedure, apply a low plate voltage to the stage. Tune the plate tank capacitor to resonance as indicated by minimum plate current readings. You will notice that the plate current will dip sharply because the output tank circuit is not delivering power to the load. This is shown by the solid curve in Fig.38.

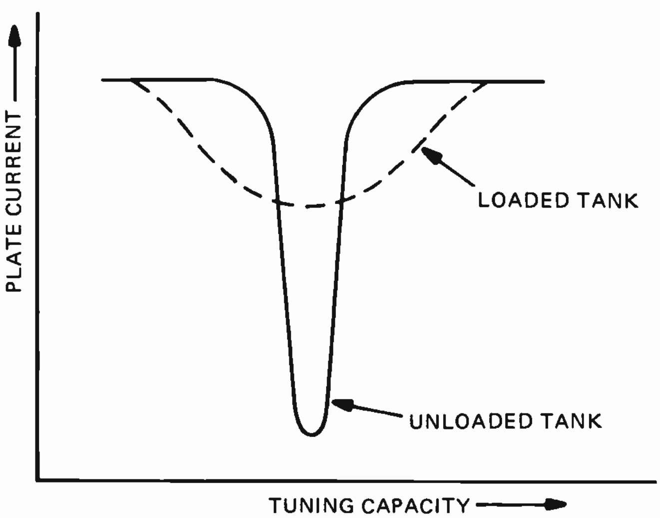


Figure 38. How plate current dips as tank capacitor is tuned through resonance; the unbroken line shows the sharp dip that occurs if the tank is not loaded; the dashed line shows the broad dip that occurs if the tank is loaded.

The grid current meter will indicate maximum at resonance. Retune the grid tank capacitor, and increase the excitation until the grid draws the rated current.

Increase the loading in the plate tank circuit until any increase in loading causes the current through the antenna meter to drop. Increase the loading in steps. Check the plate circuit tuning for resonance each time you increase the load. Now apply the plate voltage and adjust it to the rated value.

Retune the plate tank to resonance, and then advance the loading until the tube draws the rated plate current. The minimum plate current point will not be as sharp because the tuned circuit is now loaded and more power is being fed into the load circuit. The loaded plate current tuning curve is shown by dotted lines in Fig.38. Adjust the grid tank and excitation until the rated grid current is drawn.

Make final fine adjustments to the plate tuning and antenna loading. Be certain all meters show the recommended values for proper operation of the stage.

The output is indicated by the current readings on the rf antenna current meter. As the stage is resonated and the loading is increased, the antenna current increases, indicating power is being delivered to the antenna. The antenna meter is just as important as the plate ammeter when tuning. If the output current does not increase when the plate current increases, the plate circuit is not tuned to resonance or is overloaded. Reduce the coupling and retune the plate tank.

Parasitics. It is interesting to note that the wavemeter is also useful in detecting the presence of parasitics in the operating amplifier. As you know, these oscillations take place at a frequency far removed from the operating frequency. The most practical way to locate the oscillations is to reduce the bias of the stage so that the tube is no longer operated beyond plate-current cutoff. Then reduce the plate voltage so that the maximum plate dissipation of the tube is not exceeded. Disconnect the output and remove the drive from the stage. These changes make the circuit most favorable for the generation of parasitic oscillations.

With the wavemeter inductively coupled to the circuit suspected of oscillating, the wavemeter tuning knob is varied over its range. A meter deflection not only indicates parasitic oscillation, but the frequency may be approximately determined by the position of the tuning knob. Knowing the frequency at which parasitics are occurring often provides a clue to their origin.

THE TRANSISTOR STAGE

The transistor power amplifier, although used to some extent in low-level fixed station and broadcast transmitters, finds its widest application in low-powered mobile transmitting equipment. Since these units are operated largely by non-technical people working under less than ideal conditions, the emphasis in their design is on simplicity and reliability. Because of this emphasis, adjustment procedures for transistor class C amplifiers are usually simple and straightforward. The complete transmitter alignment of many of these units consists in its entirety of peaking the indication on a power output measuring device with as few as two transmitter adjustments.

Even in the more elaborate transmitters you'll seldom find more than one meter built into the equipment. This single meter is switched into the various points in the circuit where current or voltage is to be measured. Sometimes, all the monitoring points in the circuit are connected to a multipin jack on the transmitter chassis. When transmitter adjustments are to be made, an external meter equipped with a selector switch is plugged into this jack for monitoring.

Figure 39 shows a parallel-connected output stage such as might be found in one of the higher

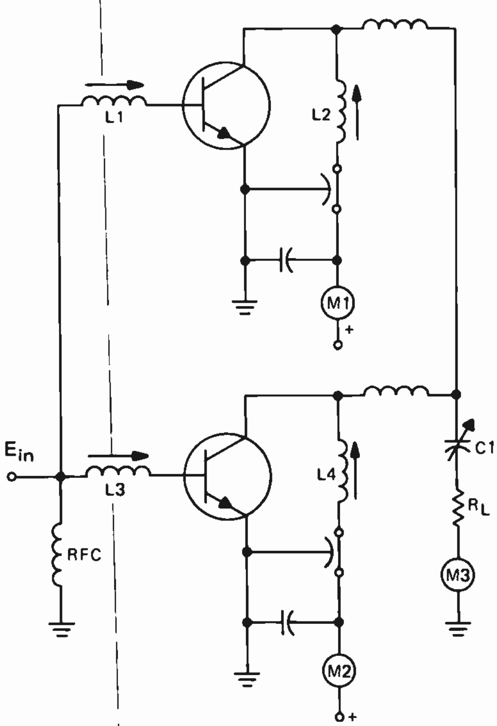


Figure 39. Parallel-connected transistor stage.

powered transistor transmitters. We have shown separate meters at the various monitoring points for clarity. Before applying power to the amplifier, L1 and L3 should be adjusted for minimum drive to the transistors (adjusted for maximum inductance). With this accomplished, apply power and adjust the collector circuit of Q1 to resonance. This is done by adjusting L2 for a dip on M1.

In like manner, adjust the collector circuit of Q2 to resonance using L4 and M2. Next, adjust the coupling to the load using C1 to obtain the rated load current as measured on M3. Finally, adjust the base drive to the transistors using L1 and L3 to obtain equal collector currents at the rated value. This completes the preliminary adjustment of the stage. Since there is some interaction between the various adjustments, recheck the setting of L2, L4, and C1. At all times maintain the collector currents at or below the rated value by adjusting L1 and L3.

In conclusion, the adjustments we've discussed in this section should not be considered as a procedure to be memorized and followed in any specific case. They are presented here to illustrate the basic approach to power amplifier adjustment. Before attempting any adjustment to a power amplifier, carefully consult and follow the manufacturer's literature. In making the adjustments, both the procedure and the sequence in which the steps are performed are of the greatest importance. An expensive tube or transistor may be destroyed as a result of any adjustments performed without complete knowledge of the correct procedure.

SELF-TEST QUESTIONS

34 How may a vacuum tube stage be checked for proper neutralization using the grid current meter as an indicator?

35 Why must B+ first be removed from a stage before checking the plate tank for the presence of rf?

36 What are some indications of a regenerative amplifier?

37 What tuning defect is indicated if the antenna current does not increase with an increase in plate current?

38 Why is the drive to the transistor amplifier in Fig.39 adjusted for minimum before collector power is applied?

ANSWERS TO SELF-TEST QUESTIONS

1 Because output current flows in the amplifier only during the relatively brief conducting period.
2 Both the bias level and the amplitude of the driving signal affect the operating angle.
3 No. Sometimes high power gain in an amplifier, obtained with a matched load, must be sacrificed for greater power output.
4 The values of L and C would both have to be changed. L would be increased and C decreased.
5 The loaded Q would decrease. The unloaded Q would be unaffected. The efficiency of the tank would increase.
6 If the loaded Q decreases, the reflected resistance in the tank must have increased. The increased resistance in the tank causes tank impedance to decrease.
7 By offering a high impedance only at the resonant frequency.
8 By decreasing the spacing between the coils.
9 Since the power supply, the tank, and the stage are in series, the amplifier is series-fed.
10 The output connection to the tank could be tapped down on the tank coil, as shown in Fig.9(C). This causes the internal resistance of the stage to appear to the tank as a higher value. This higher resistance seen by the tank reflects a higher resistance into the tank which increases the loaded Q.
11 An antenna usually loads a power amplifier more heavily than the input circuit to another class C amplifier.
12 If the drive to the stage is lost, no bias will be developed, allowing the tubes to conduct heavily. This heavy conduction could damage the tube.
13 Through the plate-to-grid capacitance.
14 Plate neutralization is effective over a range of frequencies, whereas the blocking capacitor and coil arrangement is effective only at one frequency.
15 180°. The signal fed back through a neutralizing circuit will always be 180° out of phase with the signal fed back through the grid to plate capacitance.
16 Radio frequency choke and bypass capacitor.
17 Parasitic oscillations in the amplifier.
18 Their higher power gain and reduced grid-to-plate capacitance.
19 Their higher power gain. Even a very small amount of feedback may cause the tetrode to oscillate.
20 The value of L would be lower and the value of C higher. This would provide the lower value of tank impedance necessary for the parallel-connected tubes.
21 By connecting the supply voltages to the midpoints of the tank coils.
22 The doubler. In general, the greater the frequency multiplication in a stage, the lower the power output.
23 Since the output frequency is different from the input frequency, any signal fed back would not add to the input signal.
24 The common emitter.
25 The transistor stops conducting.
26 Not always. A zero-biased transistor is already several tenths of a volt below collector current cutoff.
27 The low input impedance of the amplifier.
28 Because of the expense of the transformer required to drive a push-pull stage.
29 RF choke losses are eliminated and the transistors are electrically independent.
30 Because they have a very low lead inductance.
31 The collector-base capacitance varies over the period of an operating cycle. Thus a varying amount of signal is fed back to the input.
32 Low-frequency parasitics. The power gain of a transistor increases rapidly as frequency decreases. For this reason, a transistor amplifier is most susceptible to low-frequency parasitic oscillations.
33 Because of their physical construction, feedthrough capacitors cannot be manufactured with high values of capacitance. Proper bypass at low parasitic frequencies requires a large capacitance, so another type must be connected in parallel with the feedthrough.
34 With plate voltage removed, a dip in grid current when the plate tank is tuned through resonance indicates the need for neutralization.
35 With the B+ on the stage, rf in the tank circuit is a normal indication.
36 Changes in plate tuning affect grid tuning. Also, oscillations occur with reduced loading or slight changes in operating voltages.
37 The plate tank is not tuned to resonance or is too heavily loaded.
38 To prevent possible excessive collector current flow due to the low impedance offered by the detuned output circuit.

Lesson Questions

Be sure to mark your lesson card as shown to indicate Lesson CC204. Most students want to know their grades as soon as possible, so they mail their lesson card in immediately. Others, knowing they will finish another lesson within a few days, send in two cards at once. Either practice is acceptable. However, don't hold your answers too long; you may lose them. Don't hold answers to more than two sets of lessons at one time, or you will run out of lessons before new ones can reach you.

1. A buffer amplifier is needed in a transmitter to:
   a. Isolate the multiplier and final.
   b. Prevent parasitic oscillation.
   c. Provide proper drive for the oscillator.
   d. Present a light, constant load to the oscillator.

2. Which type of amplifier would most likely require neutralization?
   a. Tetrode.
   b. Pentode.
   c. Beam tetrode.
   d. Triode.

3. If a class C amplifier has a measured power output of 500 watts and operates with 2000 volts plate voltage and 330 ma plate current, how much power do the tube and plate tank circuit together dissipate?
   a. 160 watts.
   b. 330 watts.
   c. 420 watts.
   d. Not enough information to determine.

4. What is the approximate efficiency of the amplifier of Question 3?
   a. 25%.
   b. 50%.
   c. 76%.
   d. 85%.

5. If a transmitter output frequency of 26.4 MHz is obtained by using a tripler and a doubler, what is the master oscillator frequency?
   a. 2.2 MHz.
   b. 4.4 MHz.
   c. 8.8 MHz.
   d. 14.2 MHz.

6. With respect to a straight class C amplifier, the bias of a frequency multiplier is:
   a. Much less.
   b. Slightly less.
   c. Much more.
   d. Slightly more.

7. If the plate current of a class C amplifier using grid-leak bias suddenly rises and the power output drops, the probable cause is:
   a. Loss of grid drive.
   b. A shorted load.
   c. An open load.
   d. Poor power supply regulation.

8. In what frequency range would you expect to find parasitic oscillations in a transistor rf amplifier?
   a. Low frequencies.
   b. At the operating frequency.
   c. Frequencies higher than the operating frequency.
   d. Transistor amplifiers are not subject to parasitics.

9. A Faraday screen:
   a. Is the third grid in a pentode.
   b. Is needed only at higher frequencies.
   c. Reduces transfer of harmonics.
   d. Cannot be used with link coupling.

10. The method usually used to obtain higher output from transistor rf amplifiers is to use:
    a. Larger transistors.
    b. A push-pull connection.
    c. A common base circuit.
    d. Parallel-connected transistors.

CC205

Oscillators for Communications Equipment

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Introduction

One of the most important circuits in electronics is the oscillator. If it were not for the oscillator, radio and television and many industrial electronics applications would not be possible. An oscillator is an amplifier which generates an ac signal. The frequency of the signal is determined by the value of the components in the oscillator circuit.

In this lesson you will begin with a study of the basic oscillator circuit. You will learn the characteristics of an oscillator and how the basic oscillator works. Then, you will study applications of the oscillator, learning the details of operation of various oscillator circuits. From there you will go into methods of controlling oscillator frequency. The lesson will conclude with a brief description of nonsinusoidal oscillators.

There are many different types of oscillator circuits. For convenience in studying them, we will divide them into two types: LC oscillators and RC oscillators. LC oscillators are oscillators in which inductance and capacitance are used in the frequency-determining network. RC oscillators are oscillators in which resistance and capacitance are used in the frequency-determining network. Both types of oscillators work on the same general principle, that of feeding some of the signal from the output circuit back into the input circuit. This feedback signal enables the oscillator to go on generating its own signal. The amount of signal that must be fed back into the input depends upon a number of things, but in general it must be enough to overcome the losses in the input circuit of the oscillator.

Perhaps one of the most important considerations in an oscillator circuit is how the energy is fed from the output circuit back into the input circuit. Although it is important to feed enough signal back into the input circuit, it is even more important for the signal fed from the output back into the input to be of the correct phase. If the phase of the feedback signal is not correct, instead of aiding the input signal, it will oppose it, and the oscillator will not oscillate.

Sometimes an oscillator is considered as a converter circuit. In other words, it converts dc into ac. The dc is supplied by the power supply to the tube or transistor used in the oscillator circuit, which changes this dc energy to ac energy.

Oscillators are the only practical means of generating high-frequency radio waves. In the early days of radio, before practical oscillators were developed, rf signals were generated by means of high-frequency generators called alternators. However, there is a limit to how high a frequency a rotating machine such as an alternator can develop, and hence most radio transmission was carried out on very low frequencies.

The most important part of the oscillator is the resonant circuit, so before we begin let's review it. In a previous lesson, you learned that the resonant tank in a tuned amplifier biased class C could be made to store energy and to deliver that energy during periods when the tube or transistor was cut off. Oscillators function similarly to class C rf amplifiers in that the tank circuit must continue to supply an output once the input signal is removed.

One characteristic of a resonant circuit that we have not discussed is its ability to produce a damped wave when it is shock-excited. We will now see what we mean by a damped wave and see how it is produced by a resonant circuit.

DAMPED WAVES

Consider the circuit shown in Fig. 1. A coil and capacitor are connected in parallel and connected to a battery through a switch. For our discussion we must assume that the switch can be opened and

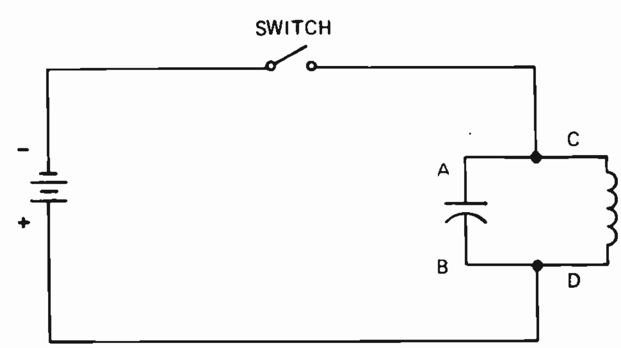


Figure 1. A simple method of producing a damped wave.

closed instantaneously. Now let's see what happens when we close the switch for an instant.

At the instant the switch is closed, electrons flow from the negative terminal of the battery into side A of the capacitor. At the same instant, electrons will flow out of side B of the capacitor to the positive terminal of the battery. If the resistance in the circuit, which includes the battery resistance, is very low, the capacitor can charge up almost instantly to a voltage equal to the battery voltage. Thus, terminal A of the capacitor will be negative and terminal B will be positive.

At the same time, when the switch is closed, there will be some tendency for current to flow through the coil from terminal C to terminal D. However, you will remember that one of the characteristics of a coil or inductance is that it opposes any rapid change in the current flowing through it. The instant before the switch is closed, the current flowing through the coil is zero. The coil would like to keep it that way. When the switch is closed, the inductance of the coil tries to prevent a current from building up in the coil. Actually, there will be some small current flowing through the coil from terminal C to terminal D, but if the switch is closed only for an instant, the current will not be able to build up appreciably. Therefore, at the instant the switch is opened again we have the capacitor charged, as shown in Fig.2(A), and a small current flowing through the coil as indicated.

When the switch is opened and we have the situation shown in Fig.2(A), we have a capacitor that is charged, and immediately starts to discharge. As a result, a current flow will be set up in the circuit as shown in Fig.2(B). Now remember that a coil opposes a change in the current flowing through it. Therefore the capacitor cannot discharge instantly through the coil, but rather must build up a current in the coil which will build up a magnetic field about the coil. Eventually, the capacitor will build up a current flow in the coil and enough electrons will leave plate A to get to plate B to discharge the capacitor.

The discharge of the capacitor removes the voltage that caused current to flow through the coil. The magnetic field around the coil now collapses. The collapsing field generates an emf in the coil, which tends to keep the current flowing in the same direction as before. This continued current causes electrons to flow onto plate B of the capacitor, giving the capacitor a charge opposite to what it had at the start. This condition is shown in Fig.2(C).

After the field around the coil has collapsed, there is no emf to hold the charge on the capacitor. The capacitor now begins to discharge back through the coil as shown in Fig.2(D). The flow of current caused by the discharge of the capacitor builds up a magnetic field around the coil until the capacitor is fully discharged; the magnetic field collapses and keeps the current flowing. This

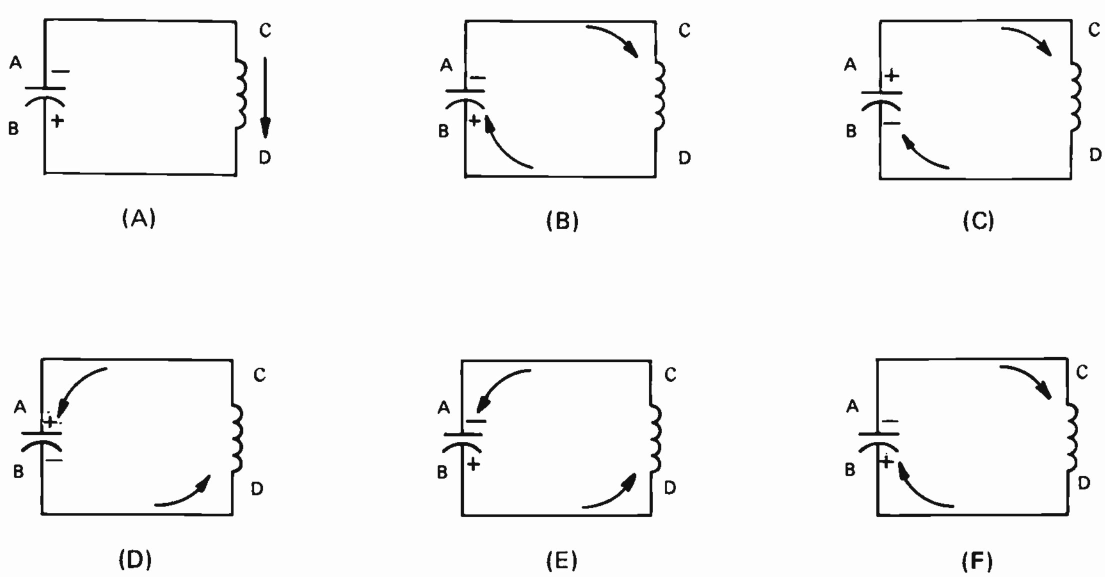


Figure 2. How oscillation takes place in a resonant circuit.

current flow charges the capacitor with the same polarity it had at the instant the switch was opened. This is shown in Fig.2(E).

Again, the current will eventually drop to zero, and then the capacitor will once again begin to discharge through the coil in the opposite direction, this time with electrons flowing from plate A to plate B as shown in Fig.2(F).

Notice that in Fig.2(F) we have the same situation as we had in Fig.2(B). In other words, we have gone through a complete cycle of events. The capacitor was charged with one polarity. This produced a current flow through the coil, which eventually charged the capacitor with the opposite polarity. The capacitor then began to discharge through the coil in the opposite direction, which built up a charge on it having the same polarity as the original charge placed on the capacitor. Once again this charge on the capacitor began the cycle of events all over again by attempting to discharge through the coil.

You might think that this oscillation, or backward and forward flow of current through the coil to charge and discharge the capacitor, would continue indefinitely. Indeed, if we had a perfect coil and a perfect capacitor, once the oscillation was started, it would continue indefinitely. However, there is no such thing as a perfect coil or a perfect capacitor. There will be some losses in both parts, so instead of having an oscillation which continues indefinitely, we will have what is called a damped wave. The damped wave of voltage across the capacitor is shown in Fig.3.

The important thing to notice in this damped wave is that the amplitude of each cycle is just a little bit less than the amplitude of the preceding cycle. In other words, the wave is slowly dying out because of losses in the resonant circuit. The lower the losses in the circuit, the greater the number of cycles that will occur before the wave disappears. On the other hand, the higher the losses in the circuit, the fewer the number of cycles.

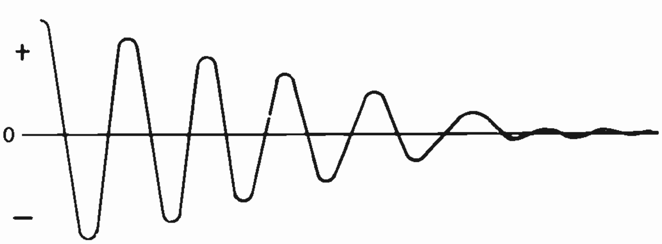


Figure 3. Voltage across the capacitor.

If we could find some way of closing the switch in Fig.1 for just an instant when plate A of the capacitor reaches its maximum negative charge, we could supply a small amount of energy to the resonant circuit to make up for losses in the circuit. If we continue to supply this small amount of energy once each cycle, then the resonant circuit will continue to oscillate indefinitely, and we could use it as a source of ac power. This is what an oscillator does – it supplies a pulse of energy at the correct time to make up for losses in the resonant circuit. We'll see how this is done later, but let's learn more about resonant circuits first.

FACTORS AFFECTING RESONANT CIRCUITS

There are several additional important things we should know about resonant circuits. For example, we should know the frequency at which oscillation takes place in a resonant circuit. We should also know what factors affect the loss of energy from cycle to cycle. In other words, how rapidly the wave train will die out.

Another term that we frequently encounter when dealing with resonant circuits is "period." We will now learn something about these factors.

Frequency. The frequency at which a resonant circuit oscillates will depend upon the inductance and capacitance in the circuit. We already know that resonance occurs when the inductive reactance of the coil is exactly equal to and canceled by the capacitive reactance of the capacitor. In other words, at resonance:

$$X_L = X_C$$

We know that the inductive reactance of a coil, X_L, is given by the formula:

$$X_L = 6.28 \times f \times L$$

and the capacitive reactance of a capacitor is given by the formula:

$$X_C = \frac{1}{6.28 \times f \times C}$$

Now, since resonance occurs when $X_L = X_C$, let's substitute the values for X_L and X_C and we will get:

$$X_L = X_C$$

$$6.28 \times f \times L = \frac{1}{6.28 \times f \times C}$$

and this can be manipulated to give us:

$$f^2 = \frac{1}{6.28^2 \times L \times C}$$

and now if we take the square root of both sides of the equation we get:

$$f = \frac{1}{6.28 \times \sqrt{L \times C}}$$

For convenience in expressing formulas of this type, the times sign is usually omitted, and in place of 6.28, the term 2π is often used. You will usually see the formula for the frequency at which a resonant circuit will oscillate expressed as:

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

You should remember this formula because it is very important; but even more important, remember what the formula tells you. The formula says that the frequency of a resonant circuit varies inversely as the square root of the LC product. Now remember we mentioned before that when one factor varies directly with another, making one bigger makes the other bigger; and when two factors vary inversely we have the opposite situation: making one bigger makes the other smaller. Here we have a situation where the frequency varies inversely as the square root of the LC product. This means that increasing the size of either L or C will reduce the frequency at which the resonant circuit oscillates, and reducing the size of either L or C will increase the frequency at which the resonant circuit oscillates. We can express this simply by saying: *Larger L or C, lower frequency; smaller L or C, higher frequency.*

In using this formula, the frequency of oscillation will be given in cycles per second and the value of L and C used must be in henrys and farads respectively.

Period. The period of a resonant circuit is the time it takes the resonant circuit to go through one complete oscillation. Thus, if we have a circuit that is resonant at a frequency of 1000 cycles per second, its period would be 1/1000 of a second, and if we have a resonant circuit that is resonant at a frequency of 1,000,000 cycles per second, the period would be 1/1,000,000 of a second.

The period of a resonant circuit is given by the formula:

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

where P represents the period of the resonant circuit in seconds and f the frequency in cycles per second.

Since in electronics we are usually dealing with comparatively high frequencies, it follows that the period of most resonant circuits will be only a very small fraction of a second. As a matter of fact, the period of many resonant circuits will be only a small fraction of a millionth of a second. Therefore, to simplify things, the microsecond is frequently used in electronics work as a unit of time. A microsecond is 1/1,000,000 (one-millionth) of a second. Thus, if a resonant circuit has a period of 5/1,000,000 (five-millionths) of a second, we can say that its period is 5 microseconds. Or, if another resonant circuit has a period of 1/10,000,000 (one ten-millionth) of a second, we can say this period is one-tenth of a microsecond.

In order to show the cycle-time relationship, the frequency of a circuit is measured in units called HERTZ. One hertz being equivalent to one complete cycle in one second, 1000 cycles in one second would then be 1000 hertz (Hz) or one kilohertz (kHz). 1,000,000 cycles in one second would be one megahertz (MHz). These terms are replacing the older terms of kilocycles (kc) and megacycles (mc) still used in many publications.

The Q Factor. The number of cycles that will occur when a resonant circuit is shock-excited depends almost directly upon the Q of the coil. The higher the Q, the more cycles will occur.

The Q of a coil tells us essentially how good a coil we have. A coil that has a high Q has a high inductive reactance compared to the resistance of the coil. A coil with a low Q has high resistance compared with the inductive reactance.

The Q of a coil is expressed by the formula:

$$Q = \frac{X_L}{R}$$

and we can express X_L as equal to:

$$6.28 \times f \times L$$

and substituting this in the formula for the Q of a coil we get:

$$Q = \frac{6.28 \times f \times L}{R}$$

If we examine this formula, we see that the Q varies directly as the frequency and inductance and inversely as the resistance. Therefore, you might think that increasing the frequency of the resonant circuit by using a smaller capacitor in conjunction with the coil will result in a higher Q. This will often happen, but the increase in Q is not as great as might be expected, because the resistance of the coil is the ac resistance rather than the dc resistance. The ac resistance of a coil actually represents ac losses in the coil and this varies directly as frequency varies. Therefore, increasing the frequency of the resonant circuit increases the inductive reactance of the coil, but at the same time it increases the losses so that the Q normally does not increase as fast as we might expect.

In a resonant circuit with a high Q coil there will be a large number of cycles in a damped wave train set up by shock-exciting the resonant circuit. In other words, the amplitude of one cycle will be very little less than the amplitude of the preceding cycle. However, if the Q of the coil is low, then the losses in the coil will be quite high so that the amplitude of each cycle will be substantially less than the amplitude of the preceding cycle. This means that the oscillation will be damped out quite rapidly and the number of cycles that occur when the circuit is shock-excited will be somewhat limited.

In most oscillator circuits a comparatively high Q coil is used. The reason for this is that if the coil has a high Q, then only a small amount of energy must be supplied by the tube or transistor in the oscillator circuit in order to sustain oscillation. On the other hand, if the coil has a low Q, the losses in the resonant circuit will be quite high, with the result that the tube or transistor used in the oscillator circuit must supply a comparatively large amount of energy in order to keep the oscillation going.

SELF-TEST QUESTIONS

Please check your answers on page 40.

1 What type of feedback is used in oscillator circuits?

2 If the inductance of an LC circuit is increased, what happens to the frequency?

3 If the Q of the resonant circuit is increased, what happens to the damped wave train?

4 If the resonant frequency of an LC circuit is 2000 kHz, what is the period of one cycle?

The Basic Oscillator

The function of the switch in Fig.1 is to supply energy of the proper phase and at the proper time to sustain oscillations in the resonant circuit. At radio frequencies it would be impossible for a mechanical switch to do this. Therefore, we must use an electronic switch such as a vacuum tube or transistor.

THE ELECTRONIC SWITCH

In order to see how the vacuum tube can be used as an electronic switch, let's go back to the basic circuit we had in Fig.1. We have repeated this circuit as Fig.4(A). It is exactly the same as Fig.1 except we have simply indicated where the battery voltage is to be connected instead of actually showing the battery in the circuit. In practice, we could use either a battery or the dc output of a suitable power supply. If we can momentarily close the switch, we will charge the capacitor C1 and produce oscillation in the parallel-resonant circuit consisting of C1 and L1. However, this oscillation will die out after a number of cycles because of the losses in a resonant circuit, unless we can find some way of supplying additional energy to the resonant circuit to make up these losses. If we could close the switch at the right instant during each cycle, we could recharge capacitor C1 once each cycle and keep the oscillation going. However, if the resonant frequency of the circuit is several hundred Hz or higher, it would be impossible to close the switch manually at the correct instant to keep the oscillation going. As a matter of fact, it would be difficult to do this mechanically except at a very low frequency.

In Fig.4(B), we have replaced the switch with a vacuum tube. The cathode of the vacuum tube is connected to the negative side of the power supply or battery, and the plate of the tube is connected to the resonant circuit. Between the cathode and the grid of the tube, we have connected a battery that will place a negative voltage on the grid of the tube. The battery voltage in the grid circuit is high enough to bias the tube beyond cutoff. Thus, with the circuit exactly as shown in Fig.4(B), the bias on the tube is so high that there will be no current flowing through the tube and hence no way to charge capacitor C1 and start the resonant circuit oscillating. We have in effect the same situation as we have in Fig.4(A) with the switch open.

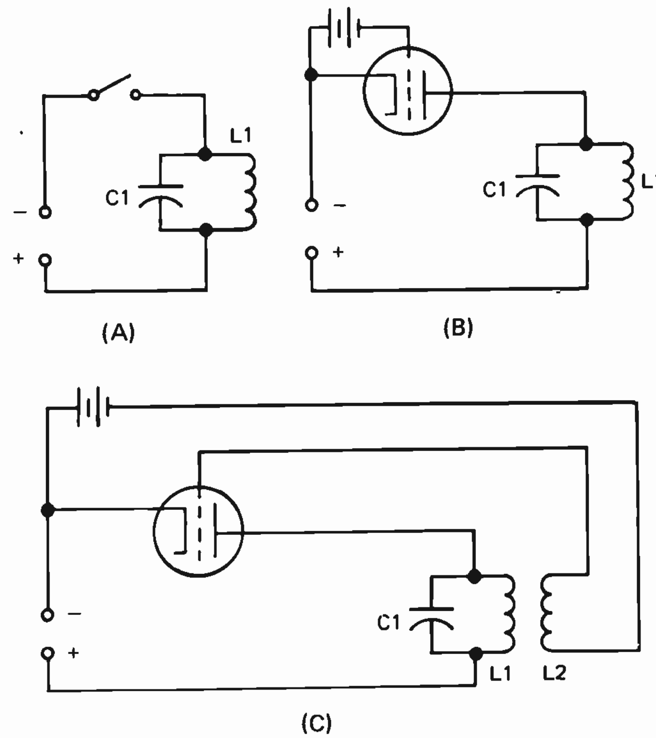


Figure 4. Using a tube as an electronic switch to supply the losses in a resonant circuit.

Now let's look at the circuit shown in Fig.4(C). Here we still have the tube connected in the circuit in exactly the same way except that we have added a coil, L2, between the negative terminal of the grid battery and the grid of the tube. This coil is placed near L1 so that it will be inductively coupled to L1. Thus, if there is any change in the magnetic field about L1, the changing flux will induce a voltage in L2.

Now let's consider what will happen if we momentarily short the plate and cathode of the tube together. If we do this, capacitor C1 will be charged. As soon as we remove the short, C1 will start to discharge through L1 and in doing so will build up a magnetic field about L1. The changing lines of flux will cut L2 and induce a voltage in it. This voltage in L2 will be in series with the battery voltage applied between the cathode and grid of the tube. If the end of L2 that connects to the grid of the tube is negative, and the other end positive, then the voltage induced in L2 will add to the grid bias, biasing the grid still further negative so that no current can flow from the cathode to the plate of the tube. However, if the voltage induced in L2 has a polarity such that the end of L2 that is connected to the grid is positive, and the other end is negative, then this voltage will oppose the battery bias voltage and reduce it so that the total grid bias will be reduced below the point where the plate current is cut off, and current can flow through the tube. Therefore, by connecting L2 with the proper polarity, we can arrange the circuit so that when the plate side of capacitor C1 reaches its negative peak, the tube will conduct, and a burst of electrons will flow through the tube, charging C1 still further. Thus, any loss in the charge across C1 due to losses in the resonant circuit will be made up for by the burst of electrons flowing through the tube.

In Fig.5, we have shown a number of sine-wave cycles such as the oscillation that might occur in the L1-C1 resonant circuit. The shaded pulses represent the bursts of current flowing through the tube that will reinforce the oscillation and keep it going. Notice in Fig.5 that the burst of current flowing through the tube occurs at the correct instant to aid the oscillation. Also, notice that the current burst occurs for only a small fraction of a cycle. The current does not flow through the tube during the entire cycle.

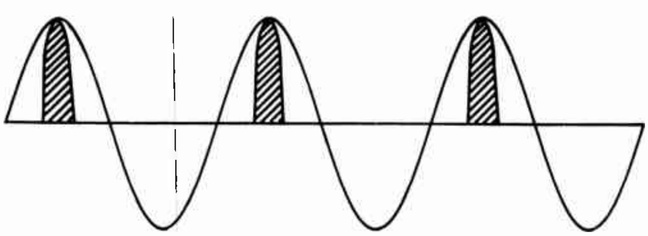
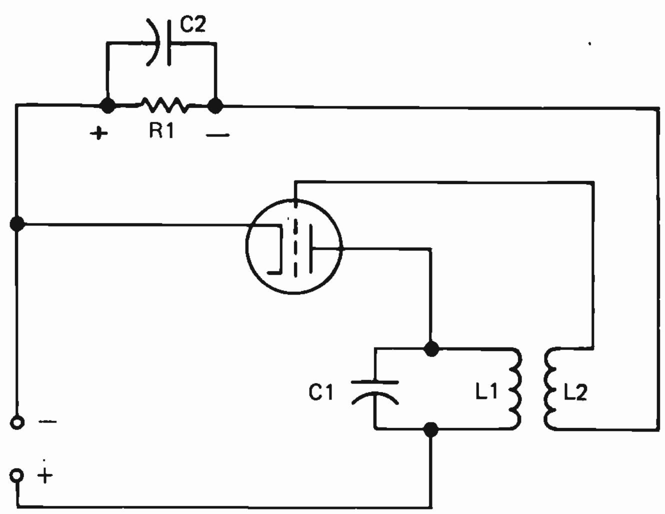


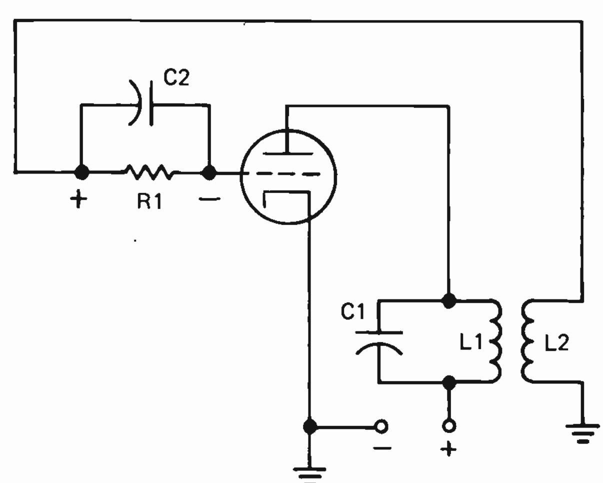
Figure 5. The oscillator pulse is timed to occur at the peak of the oscillation in the tank circuit to reinforce the oscillation.

For several reasons the oscillator circuit shown in Fig.4(C) is not a practical circuit. For one thing, the battery used to provide the negative bias on the grid is somewhat cumbersome. If we were using a power supply to furnish the voltage to operate this oscillator from a power line, we would not want to be bothered with a separate battery to supply the grid bias. Furthermore, with this type of arrangement, it would be possible to pick up such a high voltage pulse in L2 that the tube would pass an extremely high current when it was driven in a positive direction. As the grid bias battery aged and the voltage from this battery dropped, an even higher current pulse would flow through the tube. As a matter of fact, the pulse might be so high that the tube could be damaged.

Both of these objections can be overcome by modifying the circuit as shown in Fig.6. Let's look at Fig.6(A) first. In Fig.6(A), you will see that we have replaced the battery in the grid circuit by a resistor, R1, with capacitor C2 connected across it. In other respects the circuit is identical to the circuit shown in Fig.4(C).



(A)



(B)

Figure 6. A tuned-plate oscillator.

Let's see exactly how this circuit works. When voltage is first supplied to this circuit, there will be no grid bias on the tube. The tube starts to conduct and charges capacitor C1. Electrons will flow into the side of this capacitor that connects to the plate of the tube and out of the other side. At the same instant, current will start to flow through L1 and there will be a rapid change in the lines of flux about this coil. The changing magnetic lines will cut L2, and induce a rather high voltage in it. Coil L2 is connected so that the grid of the tube will be driven in a positive direction, which will result in a still further increase in current flowing from the cathode to the plate of the tube, which will charge C1 still further.

Since the grid of the tube will be driven in a positive direction, it too will attract electrons, and electrons will flow from the cathode of the tube to the grid, through L2, and then through R1 back to the cathode of the tube. In flowing through R1, they will set up a voltage drop across it and charge capacitor C2 with the polarity indicated on the diagram.

Eventually the rate at which the flux lines are cutting L2 will decrease, so the voltage induced in coil L2 will drop. The voltage across R1 will cut off the flow of plate current in the tube. Capacitor C2 starts to discharge through R1 and keeps the grid of the tube at a high enough negative potential to keep it cut off. When this happens, we have opened the switch as in Fig.1 and an oscillation starts in the tank circuit. The capacitor and coil begin exchanging energy back and forth. At the correct instant, once in each cycle, the grid of the tube will be driven positive by the voltage induced in L2 by the changing flux from L1 so that the tube will pass a burst of electrons to recharge C1 and make up any energy lost in the tank circuit.

The oscillator we have been discussing is called a tuned-plate oscillator. In actual practice, the circuit is modified and you will usually see it like Fig.6(B). Notice that the position of the grid resistor and grid capacitor, R1 and C2, have been changed with reference to L2. In other words, tracing from the grid of the tube, we come to the grid resistor and grid capacitor first and then through L2 to ground. However, regardless of how the resistor and capacitor are connected in series with L2, the action of the circuit is the same.

This type of oscillator has several disadvantages that can be eliminated by different circuitry. However, since it is a basic circuit and enables us to see exactly how the tube is acting as a switch, it is a good circuit with which to start our study of oscillators.

SELF-REGULATION

The oscillator circuits shown in Fig.6 are self-regulating. This means that they tend to control the flow of current through the tube themselves. For example, suppose the amplitude of the pulse picked up by L2 should increase for any reason; if this happens, the pulse will drive the grid even more positive than normal. With a higher positive voltage on the grid, a greater number of electrons will be attracted to it. An increase in the number of electrons reaching the grid will mean that more electrons must flow through R1. The voltage developed across R1 depends upon two things: the size of the resistor and the number of electrons flowing through it. Therefore, if the number of electrons flowing through R1 increases, the voltage developed across it will increase.

Notice the polarity of the voltage across R1. The grid end of this resistor is negative, so this bias voltage tends to reduce the flow of current through the tube. Therefore, the increase in negative voltage across R1 will subtract from the increase of positive voltage across L2 so that the net drive voltage applied to the grid remains almost the same. Thus, even though something might cause the voltage developed in L2 to increase, the tube will compensate for this change by developing an increased bias so that the burst of plate current flowing through the tube will remain essentially constant.

OSCILLATOR-AMPLIFIER

Up to this point, we have been considering the tube as a switch that closed at the proper instant to replenish the losses in the resonant circuit. We can also consider the tube as an amplifier that is amplifying part of its own output. For example, L1 and L2 in Fig.6(B) are inductively coupled together. Part of the output produced across L1 is coupled to L2, where it is fed back into the input circuit. This signal fed into the input circuit is then amplified by the tube and fed to the resonant circuit L1-C1 in the output. The cycle then continues, with part of the output being coupled to L2 and once again being fed back to the input. Thus, the oscillator can indeed be considered as an amplifier that feeds part of its own output signal back to the input, where it is amplified once again.

Of course, the signal fed back to the input must be of the proper phase to sustain oscillation. The signal must drive the grid in a positive direction when the plate current flowing through the tube should increase. Feedback of this type is called regenerative feedback. In some amplifiers a small amount of regenerative feedback is used to improve the gain of the amplifier. However, in an oscillator, enough regenerative feedback is used to start the stage oscillating, and to keep it oscillating.

OSCILLATOR FREQUENCY

You already know that the resonant frequency, f_o, of a circuit containing L and C is:

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

The resonant frequency is also often expressed in terms of resonant angular frequency, ω_o:

$$\omega_o = \frac{1}{\sqrt{LC}}$$

where $\omega_o = 2\pi \times f_o$. This expression comes from the fact that there are 2π radians in 360°. A radian is an angular measurement equal to approximately 57°. Since there are 2π radians in 360°, a vector rotating at f_o Hz travels through $2\pi \times f_o$ radians per second.

You might at first expect an LC oscillator to operate at exactly ω_o, the resonant frequency of the LC circuit. However, there is always some resistance in the circuit that affects the oscillator frequency. Furthermore, the plate resistance of the tube affects the oscillator frequency so that the actual frequency of the oscillator, ω, is:

$$\omega = \omega_o \left(1 + \frac{R}{R_p}\right)$$

where ω_o is the angular resonant frequency of the LC circuit, R represents the resistance in the resonant circuit, and R_p is the plate resistance of the tube.

In most oscillator stages the value of R will be small, because the Q of the oscillator coil will be high. At the same time, the plate resistance of the tube will be reasonably high so the term R/R_p will be small and ω will be almost equal to ω_o. However, the fact that R and R_p do enter into the frequency means that if either of these values change, the oscillator frequency will change. Thus, oscillator stability depends not only on keeping the values of L and C in the resonant circuit constant, but also the values of R and R_p must be kept constant.

OSCILLATOR STABILITY

One of the most important considerations in oscillator circuits is the stability of the oscillator – in other words, how stable the oscillator frequency is. The output frequency of a radio transmitter is controlled by the oscillator, and if the oscillator frequency does not remain constant, the transmitter output frequency will not be constant.

We have already pointed out that the oscillator frequency depends not only upon the inductance and capacitance in the resonant circuit, but also on other factors such as the resistance of the oscillator coil and the plate resistance of the oscillator tube. Now, let us consider each of these factors to see exactly what effect each has on the oscillator frequency.

Tank Inductance and Capacitance. The inductance in the oscillator tank circuit is made up of the inductance of the oscillator coil, plus any stray inductance in the circuit. The capacitance in the oscillator tank circuit is made up of the capacity connected across the oscillator coil plus any tube capacity that may be in parallel with the coil and capacitor, and the distributed wiring capacity in the circuit. The inductance in the circuit consists of the oscillator coil, the inductance of the leads connecting the coil to the tube and other parts in the circuit, and any inductance that other parts in the circuit may have. The capacity in the circuit consists of the capacity of the variable capacitor across the oscillator coil, the input capacitance of the tube, the stray wiring capacity in the circuit, plus any stray capacity the coil may have. This total inductance plus this total capacity are the major factors that determine the oscillator frequency.

When an oscillator is first turned on, the values of the inductance and capacitance in the tank circuit will usually change as the tube and other parts in the circuit heat. Therefore, the oscillator stability is usually measured in terms of the oscillator's ability to maintain a constant frequency after enough time has been allowed for the tube and parts to reach normal operating temperature. It is common practice in some transmitters to leave the oscillator on at all times to avoid any frequency drift during the warmup period. In some transmitters, the oscillator coil and capacitor are placed in an oven that is kept at a constant temperature by a thermostatically controlled heater to minimize changes in inductance and capacity due to temperature changes. In some oscillators, special temperature compensating capacitors are connected across the oscillator tank circuit to minimize frequency drift due to temperature changes. These capacitors usually have a negative temperature coefficient. This means that their capacity decreases as the temperature increases. By using a capacitor of this type with the correct temperature coefficient, it is possible to compensate for any increase in inductance or capacitance in other parts in the circuit as the temperature increases.

Changing a tube in the oscillator circuit can result in a change in oscillator frequency. The input capacity of the tube used in the oscillator circuit makes up part of the oscillator tank circuit. The input capacity of different tubes of the same type may vary appreciably, so putting a new tube in this or any other oscillator circuit may change the tank circuit capacitance, and hence the frequency. Therefore, if you replace the oscillator tube in a transmitter you should check the output frequency.

Tank Losses. Earlier we pointed out that the angular resonant frequency, ω_o, of the oscillator tank circuit is given by:

$$\omega_o = \frac{1}{\sqrt{LC}}$$

and at the same time, the actual frequency at which the oscillator oscillates is given by:

$$\omega = \omega_o \left(1 + \frac{R}{R_p}\right)$$

where R is the resistance in the tank circuit and R_p the plate resistance of the tube.

The term R represents the ac resistance of the tank circuit, and as such represents all the losses in the tank circuit. Thus, this term includes such factors as coil resistance and losses from the oscillator circuit due to loading of the circuit. Therefore, any change in the oscillator coil resistance will result in a change in oscillator frequency. Similarly, a change in the loading on the oscillator will result in a change in oscillator frequency. Thus, for maximum stability, the oscillator should be lightly loaded and the load on the oscillator must remain constant.

The Plate Resistance. Since the plate resistance of the tube enters into the frequency of the oscillator, any change in plate resistance will produce a change in the oscillator frequency. The plate resistance of the tube will change if either the plate or grid voltage is changed in the case of a triode, and if the grid or screen voltage (and to some extent the plate voltage) is changed in the case of a pentode. Thus, it is important that the voltages supplying the oscillator be kept constant. These voltages must also be free of hum, which actually is a changing voltage superimposed on the dc supply voltage, because the hum voltage could produce a constantly changing plate resistance which will result in a frequency-modulated signal being generated by the oscillator.

Changes in loading on the oscillator may affect the bias developed on the grid of the tube. When this happens, the grid voltage will change, causing the plate resistance and hence the frequency to shift.

Looking at the expression for the oscillator angular frequency, we see that the frequency is equal to the angular frequency of the tank circuit times one plus a fraction. Thus, the oscillator frequency will be higher than the resonant frequency of the tank circuit. Also, if the term R/R_p is small, which it usually is, the oscillator frequency will differ from the tank frequency by only a small percentage. However, at high frequencies, this small percentage or fraction can represent a great enough frequency change to cause concern. For example, if the resonant frequency of a tank circuit is 10 MHz and the value of R/R_p is only 0.01, 0.01 × 10,000 kHz represents 100 kHz so the oscillator frequency would be 10,100 kHz, or 10.1 MHz. If the value of R_p changed because of changing voltages on the tube, the oscillator might drift 50 kHz or more above or below this frequency.

TRANSISTOR OSCILLATORS

Of the three basic transistor configurations, the common emitter is the most frequently used in rf oscillator circuits. There are several reasons for this. The power, current, and voltage gains of the common-emitter configuration are all greater than one, and the highest possible power gain can be had. Also in the common-emitter circuit, moderate input and output impedances make less power necessary for feedback. In the common-base configuration, low input and high output impedances inherent in the circuit cause a mismatch in the feedback circuit, producing greater losses and requiring more feedback. The current gain in the common-base circuit is less than one, even though voltage and power gains are greater than unity. A somewhat similar condition exists in a common-collector circuit where high input and moderate output impedances exist. Voltage gain is less than unity, but current and power gains are greater than one.

Transistor oscillators may be designed to operate class A, B, or C depending on the desired efficiency. Since rf oscillators are also amplifiers, bias supply and temperature stabilization are similar to rf amplifiers discussed in a previous lesson.

A combined voltage divider and feedback type biasing arrangement is often used because it helps produce oscillation and at the same time establishes a stable dc bias point. Emitter biasing with a bypass capacitor is also used, the operation being similar to grid leak biasing. Usually the amplitude is regulated by driving the transistor into saturation or by using special diode limiting circuits. Either shunt or series type collector feed may be used, the shunt type being preferred for greater output efficiency.

Frequency stability of the transistor oscillator is equivalent to, and sometimes greater than, the electron tube oscillator. The use of lower voltages, currents, and power permits construction of better tank circuits. In particular, the low power used with transistors aids in stability due to the decrease in heat. One major disadvantage of transistors is their critical operating point. A slight bias change can cause a large shift in frequency.

The collector-to-emitter capacitance of the transistor also affects frequency stability. This internal capacitance will vary with changes in collector or emitter voltages and with temperature. In high-frequency oscillators it is sometimes necessary to place a swamping capacitor across the collector to emitter leads. The total capacitance of the two in parallel results in a circuit which is less sensitive to voltage changes. The added capacitor may be a part of the tuned circuit.

Partial compensation of voltage changes may be obtained by use of a common supply. Since an increase in collector voltage tends to increase oscillator frequency and an increase in emitter voltage decreases oscillator frequency, the use of a common bias source for both the collector and emitter helps stabilize the frequency. By using a common bias source, a change in one is somewhat counteracted by the change in the other.

The three basic transfigurations used for oscillators are shown in Fig.7. Bias and feed arrangements are omitted for simplification. Although any of the basic transistor configurations can

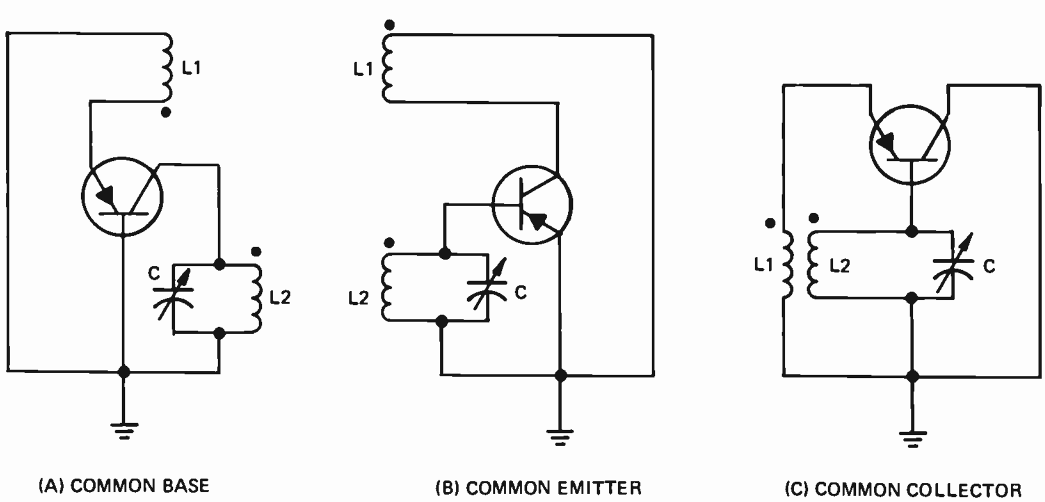


Figure 7. Tickler coil oscillators.

be used, generally only two, the common-emitter and common-base, are used in actual practice. The common-emitter configuration offers the advantages of easily matched input and output impedances and its close parallel to the electron tube.

The major advantage of the common-base circuit is that at high frequencies collector-emitter capacitance helps feed back an in-phase voltage independently of tickler coil L1, and oscillation is more easily obtained. In the common-emitter circuit, this capacitance feeds back an out-of-phase voltage which requires additional feedback from the tickler coil to overcome it. In both the common-base and common-emitter circuits, oscillation is easily sustained. This is the result of feedback provided by voltage induced through the mutual inductance of L1 and L2. In the common-collector circuit, the voltage gain is always less than unity; therefore, feedback tends to be insufficient for stable oscillations at the lower frequencies. At the higher frequenices it is assisted by the base-emitter capacitance. Sometimes, an external capacitor is added between the base and emitter to give additional feedback.

Operation of the LC circuit is similar to that of the electron tube circuit. As the oscillator is switched on, current flows through the transistor as determined by the biasing circuit. Initial current produces a feedback voltage between the collector and the emitter which is in-phase with the input circuit. As emitter current increases, collector current increases and additional feedback between L1 and L2 causes the emitter current to increase until saturation is reached. When saturation is reached, emitter current is no longer changing (increasing), and the induced feedback voltage is therefore reduced. At this time the collapsing field around the tank and tickler coils induces a reverse voltage into the emitter circuit which causes a decrease in the emitter current, thus causing a decreasing collector current. The decreasing current then induces a greater reverse voltage in the feedback loop driving the emitter current toward cutoff.

Although the emitter is cut off, a small reverse (leakage) current flows. This current has no effect on the operation of the circuit but it does represent a loss which lowers the efficiency. In this respect the transistor differs from the electron tube, which has zero current at cutoff.

The discharge of the tank capacitor through L2 will cause the voltage applied to the emitter to rise from a reverse bias to a forward bias condition. Emitter and collector current start to increase and the cycle repeats itself.

The transistor oscillator circuit that most closely resembles the tuned-plate vacuum tube oscillator is the tuned-collector oscillator. This circuit is shown in Fig.8.

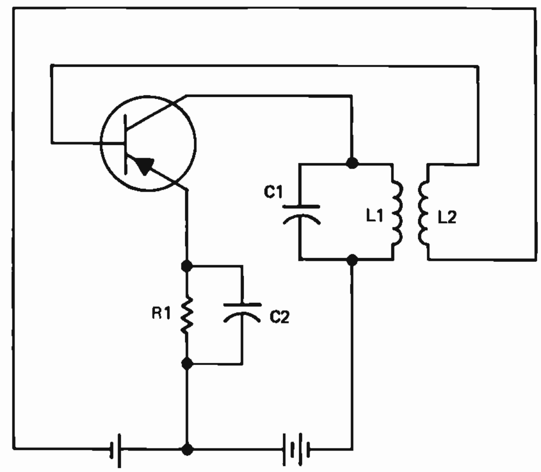


Figure 8. A tuned-collector oscillator.

Notice that in this circuit the resonant circuit consisting of C1-L1 is in the collector circuit of the transistor. L2 is inductively coupled to L1 so energy is fed from L1 to L2. The signal developed in L2 is fed back to the base of the transistor.

In the operation of this oscillator, resistor R1 and capacitor C2 develop a bias voltage sufficient to cut off the transistor. The signal needed to overcome this cutoff bias is induced in L2 and applied between the base and the emitter. Since this is a pnp transistor, the signal in L2 must make the base negative and the emitter positive at the instant that a pulse of current is needed from the collector in order to sustain oscillation in the resonant circuit consisting of L1 and C1.

SELF-TEST QUESTIONS

5 Describe the phase relationship between the plate current wave shape and the voltage wave shape developed across L2 in Fig.4(C).

6 What makes the oscillator in Fig.6(B) self-regulating in regard to the amplitude of the grid signal?

7 List three means of reducing frequency drift due to changes in temperature in a resonant circuit.

8 In high-frequency transistor oscillators, what is sometimes used to compensate for collector-to-emitter capacitance?

9 In the oscillator circuit in Fig.8, where is the bias developed and what component develops the signal that overcomes this bias?

Practical Oscillator Circuits

The oscillator circuits we have discussed up to this point weren't very practical. They were used to illustrate some of the basic characteristics of oscillators. Let us now look at some practical circuits actually found in communication equipment. These oscillators are grouped according to the type of resonant circuit used, inductance-capacitance (LC) or resistance-capacitance (RC).

LC OSCILLATORS

The LC oscillators can be placed into one of two classifications: those using inductive feedback and those using capacitive feedback. The inductive feedback oscillator uses inductive coupling to return a portion of the output back to the input. The capacitive feedback circuit uses capacitive coupling to accomplish this. Although there is some difference in the circuitry involved, both types are LC oscillators, and the net result is essentially the same.

OSCILLATORS USING INDUCTIVE FEEDBACK

One of the most important and most widely used oscillators in electronics work is the Hartley oscillator. This oscillator uses inductive feedback. The resonant circuit is placed in the grid circuit of the tube or the base circuit of the transistor instead of in the output circuit as in the case of the tuned-plate and tuned-collector oscillators. However, before we look at the Hartley oscillator let's look at another oscillator which will help you understand how the Hartley oscillator works. Let's first look at the tuned-grid oscillator.

Tuned-Grid Oscillator. Two versions of the tuned-grid oscillator are shown in Fig.9. The circuits are basically the same; the only electrical difference is in the connection of the grid resistor R1. In the circuit shown in Fig.9(A), R1 is connected directly across the grid capacitor C2, whereas in the circuit shown in Fig.9(B), R1 is connected between the grid and the cathode of the tube. The action of R1 is the same in both cases; it provides a path for the electrons striking the grid of the tube to get back to ground or the cathode of the tube. In the circuit of Fig.9(A), when C2 discharges through R1 to develop negative bias for the tube, there is no discharge through the tank circuit. In Fig.9(B), when C2 discharges through R1, the discharge current also flows through the tank circuit.

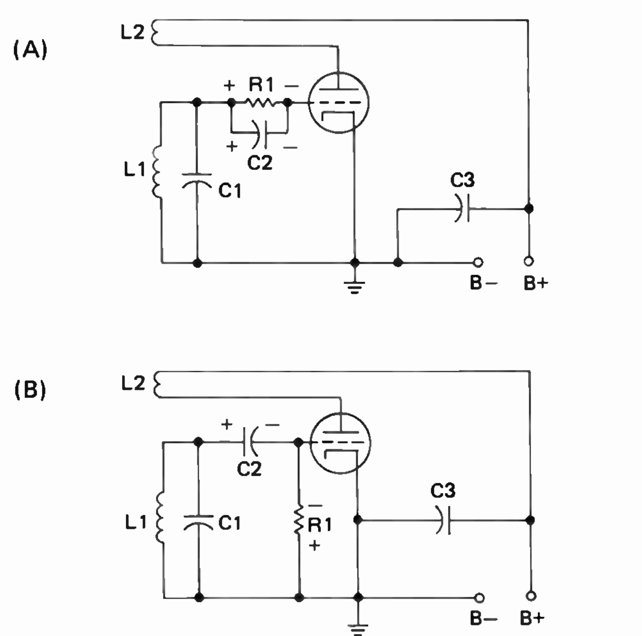


Figure 9. Two versions of the tuned-grid oscillator.

Actually, the biggest difference between this oscillator and the tuned-plate oscillator that you already studied is that the resonant circuit is in the grid circuit instead of the plate circuit. With this circuit, when the power is turned on, changes in plate current will set up a changing magnetic field about L2. L2 is inductively coupled to L1 so the changing magnetic field about L2 will induce a voltage in L1. The induced voltage charges capacitor C1, starting the oscillatory discharge in the tank circuit consisting of L1-C1. The voltage across C1 becomes the grid voltage because the value of C2 is large enough so that its reactance is so small at the frequency of oscillation, that the grid is, in effect, connected directly to C1.

Now since the increasing plate current causes the end of C1 that is connected to the grid through C2 to swing in a positive direction, the grid of the tube is driven in a positive direction. Driving the grid positive produces two effects; it increases the plate current, causing C1, and hence the grid, to be driven still further in a positive direction, and it causes grid current to flow, which charges C2 with the polarity shown on the diagram.

Now if the plate current of the tube could keep on increasing indefinitely, the grid end of C1 would be driven more and more positive. However, there is a limit to how high the plate current can become, because a balance will be reached between the positive voltage across C1 and the negative voltage across C2. When this happens, the plate current flowing through L2 will no longer change. We will no longer have voltage induced in L1, and C1 will begin to discharge through L1, setting up an oscillation in the LC circuit. As soon as this happens, the positive voltage on the grid end of C1 begins to disappear, and the plate current will be cut off by the negative voltage across C2. The LC circuit is now free to oscillate as though the tube were removed from the circuit. C2 meanwhile starts to discharge through R1, setting up the voltage drop across it as shown on the diagram. During the next half cycle when the voltage on the grid end of C1 again becomes positive, it will drive the grid in a positive direction enough to let some plate current flow through the tube; this will result in a change in the field about L2 which will induce a voltage in L1 which drives the capacitor and the grid voltage still further in a positive direction.

The important point to remember about this oscillator is that the energy needed to sustain the oscillation in the tank circuit, consisting of L1 and C1, is inductively coupled to L1 from L2. This energy comes from the plate of the tube in the form of bursts of plate current which produce a changing magnetic field about L2. These bursts of current are the result of the grid of the tube being driven positive by the voltage across C1 swinging positive once each cycle.

The Hartley Oscillator. Two Hartley oscillators are shown in Fig.10. The circuit shown in Fig.10(A) uses a vacuum tube, whereas the one shown in Fig.10(B) uses a transistor. Although the operation of the two circuits is so similar that if you understand one, you will understand the other, we will go through both circuits in considerable detail.

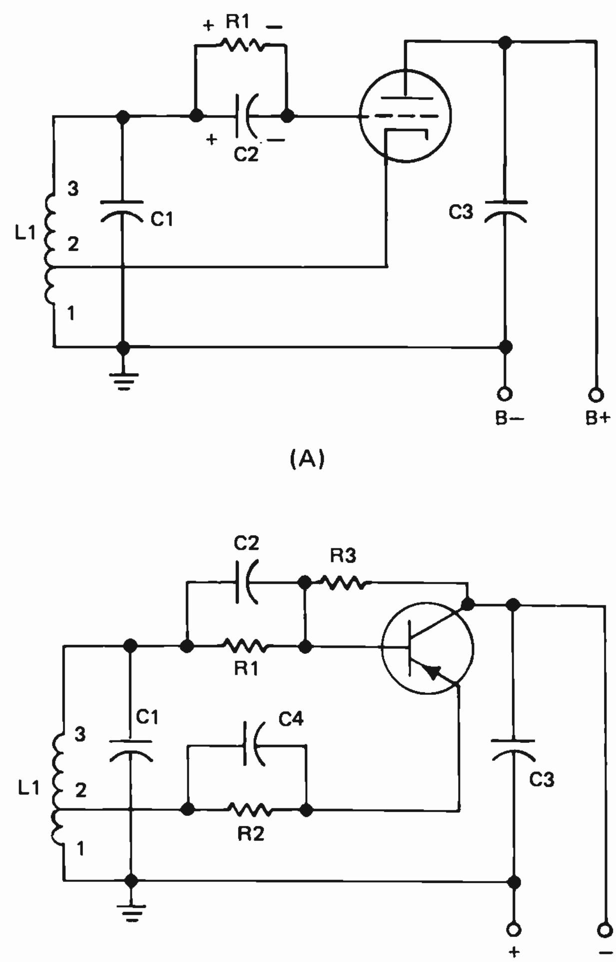


Figure 10. Typical Hartley oscillators; a vacuum tube Hartley oscillator is shown at (A), and a transistorized one at (B).

Notice the difference between the Hartley oscillators and the tuned-grid oscillators. The tuned-grid oscillator has two coils, whereas the Hartley oscillator uses only a single tapped coil. In the circuit shown in Fig.10(A), the cathode of the tube is connected to the tap, and in the circuit shown in Fig.10(B), the emitter of the transistor is connected to the tap.

In the circuit shown in Fig.10(A), when the plate current starts to flow through the tube, current must flow through the lower half of the coil between terminals 1 and 2. Since the entire coil is wound on a single form, all the various turns of the coil are inductively coupled together. Therefore the increasing plate current flowing between terminals 1 and 2 will produce a changing magnetic field which will induce a voltage in the portion of the coil between terminals 2 and 3. This voltage will charge capacitor C1 with the polarity such that the end of the capacitor that connects to the junction of C2 and R1 is positive. Again since the value of C2 is chosen so that its reactance is practically zero at the oscillation frequency, the grid of the tube is in effect connected directly to C1. This means that the increase in plate current will drive the grid of the tube in a positive direction, causing a still stronger burst of current through the coil. This in turn causes still higher induced voltage between terminals 2 and 3 which again charges capacitor C1 still further. At the same time when the grid is driven positive, it will attract electrons, and C2 will be charged with the polarity shown.

As in the tuned-grid oscillator, the point is eventually reached where there is a balance between the positive voltage applied to the grid by C1 and the negative voltage applied to the grid across C2 and R1 so that there is no further increase in plate current. This means that the magnetic field produced by the current flowing between terminals 1 and 2 becomes constant and no further voltage will be induced in the coil. C1 starts to discharge through the coil, and the oscillating cycle is started. Furthermore, the positive voltage on the end of C1 that connects to the grid of the tube through C2 disappears, and the tube stops conducting.

Again, the tube will be biased beyond cutoff by the discharge of C2 through R1. These electrons charge C2 during the portion of the cycle when the grid is conducting. When grid current stops flowing, C2 will discharge through R1, setting up a voltage drop across it such that the grid end is negative. This voltage across R1 maintains the bias on the grid of the oscillator tube.

In some cases you will see slight variations of the Hartley oscillator circuit. In some instances, R1 may be connected between the grid and cathode or from the grid of the tube directly to ground. In another variation the cathode connects directly to ground, R1 connects between the grid of the tube and ground, and then the plate of the tube connects directly back to terminal 1 of the oscillator coil. The B+ voltage is then applied to terminal 2 of the coil. This is simply a modification of the Hartley oscillator circuit; it works in exactly the same way as the Hartley oscillator shown in Fig.10(A).

In the circuit shown in Fig.10(B) we have a pnp transistor. When holes begin to travel from the emitter to the collector, electrons will flow from the emitter through R2 to terminal 2 on the coil. From terminal 2 they will flow through the coil to terminal 1 and back to the positive terminal of the battery. The electrons, in flowing through the coil from terminal 2 to terminal 1, will build up a field about this part of the coil. This field will be a changing field as the current builds up, and this will induce a voltage in the portion of the coil between terminals 2 and 3. The induced voltage will charge C1 with the polarity such that the end connecting to terminal 3 of the coil is negative and the other end is positive. This negative voltage on one end of C1 will be applied to the base of the transistor through capacitor C2 because C2 has a low reactance at the frequency of oscillation. The negative voltage on the base of the transistor will increase the forward bias across the emitter-base junction, causing an increase in the number of holes flowing from the emitter to the collector. This causes a still further increase in the electron movement from terminal 2 to terminal 1 of the coil, causing the base of the transistor to be driven still further in a negative direction.

In this circuit when the number of holes flowing from the emitter to the collector increases, terminal 3 of the coil will be driven in a negative direction, and when the holes flowing from the emitter to the collector decrease, terminal 3 will be driven in a positive direction. Remember that in a pnp transistor, driving the base in a negative direction causes the holes moving through the transistor to increase, whereas driving it in a positive direction causes the number of holes flowing from the emitter to the collector to decrease. The burst of hole movement through the transistor causes the electron movement from terminal 2 to terminal 1 of coil L1 to flow through the coil in burst, and this burst of energy makes up for any losses in the resonant circuit consisting of L1 and C1.

It is interesting to note the similarity between the circuits shown in Fig.10(A) and Fig.10(B). Although we have a vacuum tube used in one circuit and a transistor in the other, there is a great deal of similarity between the two circuits and the way they work. In each case we have energy lost in the resonant circuit being replaced by bursts of energy; from the tube in one case and from the transistor in the other case. Also notice that the energy is fed across only part of the coil in each case, but the voltage induced in the entire coil is enough to set up a current flow that will replace the capacitor charge that is lost because of resistance or other losses in the resonant circuit.

OSCILLATORS USING CAPACITIVE FEEDBACK

There are a number of different oscillator circuits in which capacitive feedback rather than inductive feedback (as in the preceding examples) is used to sustain oscillation. Let's look at some of them now.

Colpitts Oscillator. Perhaps the most important of the oscillators using capacitive feedback is the Colpitts oscillator shown in Fig.11. The one in Fig.11(A) uses a vacuum tube while the one in Fig.11(B) uses a transistor.

The operation of the two oscillators is quite similar. When the equipment is first turned on, current flows through L2, which is the small rf choke used to complete the cathode circuit in Fig.11(A) and the emitter circuit in Fig.11(B). Current flowing through the coil produces a voltage drop across the coil, and this charges capacitor C2. The charge on capacitor C2 will start an oscillation in the tank circuit, which consists of coil L1 and two capacitors, C1 and C2. Remember that when we have two capacitors connected in series they will act like one capacitor insofar as the coil is concerned, and the circuit will start to oscillate. The voltage developed across C1 is the

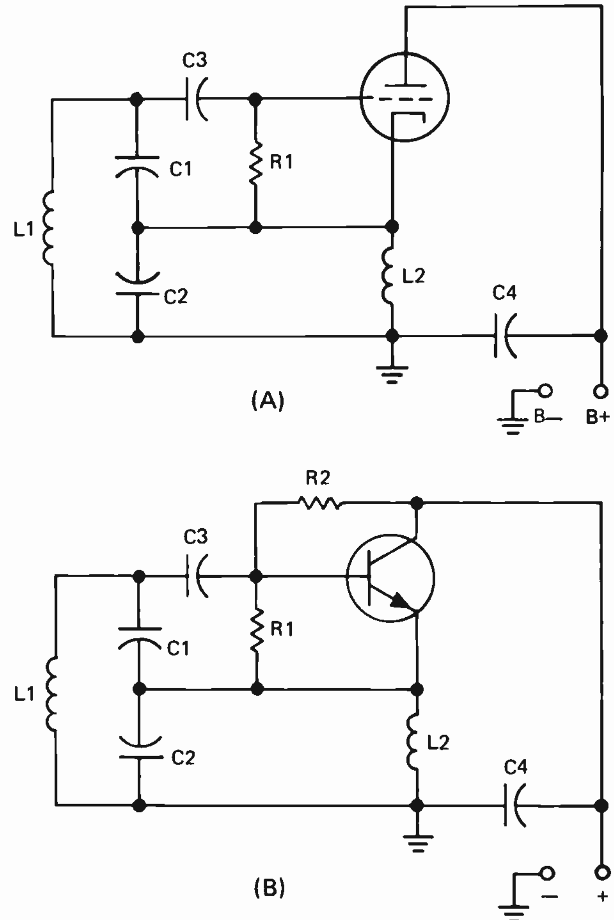


Figure 11. Two Colpitts oscillators.

feedback voltage. It is applied between the grid and the cathode in the circuit shown in (A) and between the emitter and the base of the circuit shown in (B). When this voltage swings in a direction that makes the end of C1 connected to C3 positive and the other end negative, it will increase the current flowing through the tube or transistor, causing an increase in current flow through L2, which charges C2 still further. When the polarity of the voltage across C1 reverses, the voltage will oppose the current flow and in Fig.11(A) simply add to the bias between the grid and the cathode, reducing the plate current to zero; or in Fig.11(B), put a reverse bias across the emitter-base junction, reducing the current flowing through the transistor to practically zero.

The amount of feedback voltage applied to the input of the circuit depends upon the ratio of C1 to C2. If C1 is large compared to C2, the reactance of C1 will be low and the reactance of C2 will be high. Most of the voltage developed across the capacitors will be developed across the higher reactance, in this case C2. This means that the feedback voltage applied to the input will be low. However, if C1 is small compared to C2, the reactance of C1 will be high compared to the reactance of C2 and the feedback voltage supplied to the input of the circuit will be high.

The ratio of C1 and C2 can be altered to provide the required feedback to the input circuit to sustain oscillation. If the value of C1 is increased and the value of C2 decreased by the correct amount, the total capacity in the circuit formed as the result of two capacitors in series remains the same, and hence the resonant frequency of the oscillator does not change.

In some Colpitts oscillators an additional capacitor is connected directly across L1. This is done to provide some means of changing the resonant frequency so we can vary the frequency at which the oscillator oscillates. It is impractical to try to vary both C1 and C2 at the same time, but an additional capacitor placed directly across the coil can be varied, and this will change the resonant frequency of the oscillator. At the same time, since C1 and C2 will still form a voltage divider, part of the total voltage developed across the two capacitors in series is fed back to the input circuit; this part can still be controlled by the proper selection of C1 and C2.

There are a number of variations of the Colpitts oscillator circuit. It is sometimes found in radio transmitting equipment that must be designed so that its frequency can be varied. The Colpitts oscillator can be designed with excellent frequency stability. By this we mean that once the oscillator is adjusted to operate at a certain frequency, it will not drift from that frequency very much. Some oscillators, on the other hand, do not have good frequency stability and will drift appreciably.

Another variation of the Colpitts oscillator circuit is shown in Fig.12. Here we have the capacitor C1 connected across L1 in addition to the voltage divider capacitors C2 and C3.

Notice that in this circuit the plate of the tube is fed back directly to L1, C1, and C3 and that the choke coil L2 has been moved from the cathode circuit to the plate circuit of the tube. The cathode in this oscillator circuit is connected directly to ground.

In this oscillator, when the plate current increases there will be a voltage developed across the

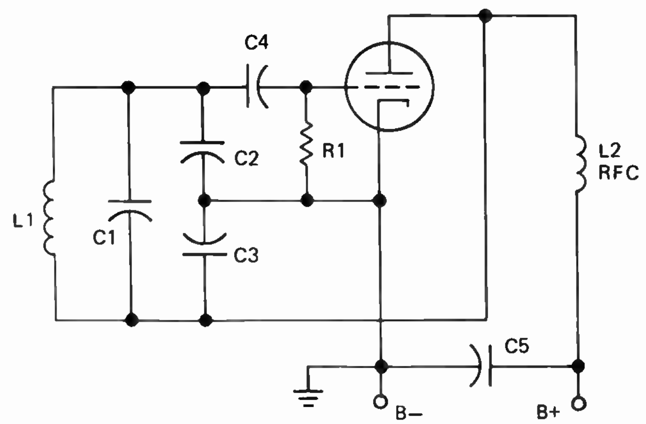


Figure 12. A variation of the Colpitts oscillator.

rf choke, L2, in the plate circuit of the tube, and this voltage will charge C3. Once this capacitor is charged, oscillation starts in the circuit just as in the Colpitts oscillators shown in Fig.11.

The Ultra-Audion Oscillator. Another oscillator that uses capacitive feedback is the ultra-audion oscillator shown in Fig.13(A). When this type of oscillator was first developed, it was considered as

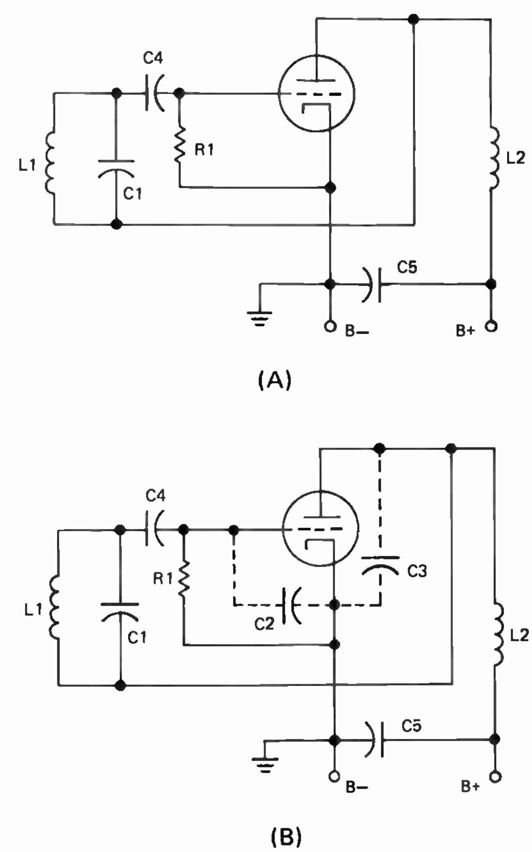


Figure 13. The ultra-audion oscillator.

a new type of oscillator. However, with careful analysis, we can see that it is actually a Colpitts oscillator, practically identical to the oscillator shown in Fig.12. We have used the same designations to identify the parts in the circuits shown in Figs.12 and 13. As you can see, the parts are all the same except for C2 and C3, which Fig.13(A) does not seem to have. However, in Fig.13(B) we have shown these two capacitors. C2 is the grid-to-cathode capacity of the tube, and C3 is the plate-to-cathode capacity of the tube. When we consider these two capacities, we have a capacitive voltage-divider network just like the one in Fig.11. C2 in Fig.13(B) is between the grid and the cathode of the tube. Notice that C2 in Fig.12 also is in effect connected between the grid and the cathode of the tube. C2 and C3 are in series in both circuits and they are connected across the tank circuit. C3 connects directly to the lead going from the plate of the tube to one side of the resonant circuit, and C2 connects through capacitor C4 to the resonant circuit. Therefore, this oscillator is simply another form of a Colpitts oscillator.

This type of circuit is frequently used in the vhf oscillators in the tuners of television receivers. Of course, it is usually shown in the schematic in the form shown in Fig.13(A). Manufacturers seldom draw in the distributed capacities; they expect the technician to know enough about oscillator circuits to recognize this as the ultra-audion oscillator and to know that this is simply a modified form of a Colpitts oscillator.

The Electron-Coupled Oscillator. So far all of the vacuum tube oscillators we have discussed have been triode oscillators. These oscillators are widely used in receiving equipment and are sometimes found in transmitters and other rf power-generating equipment. However, they have some disadvantages, one of which is the direct coupling between the output and input circuits through the grid-to-plate capacity of the tube. Loading the output circuit of the oscillator has an effect on the input circuit and hence often results in an appreciable shift in the frequency at which the oscillator is oscillating. The net result is that triode oscillators are not stable enough for some purposes.

An oscillator that overcomes this difficulty is the electron-coupled oscillator. In this circuit a tetrode or a pentode tube is used so that the only coupling between the input and output circuits is in the electron stream flowing from the cathode to the plate of the tube. Schematic diagrams of two electron-coupled oscillators are shown in Fig.14. The circuit shown in Fig.14(A) is for an electron-coupled Hartley oscillator and the one shown in Fig.14(B) is for an electron-coupled Colpitts oscillator.

The operation of these oscillators is similar to the operation of the triode oscillators, except that in the electron-coupled oscillator the screen grid of the tube acts like the plate of a triode tube. In other words, insofar as the oscillator action is concerned, we have three elements in the tube to be concerned about, the cathode, the grid, and the screen grid. The screen grid acts like the plate of the oscillator tube. The oscillation is set up in this circuit by these three tube elements. However, the electron stream flows from the cathode of the tube to the plate of the tube, and it is controlled by the grid. Since the grid is biased beyond cutoff during most of the rf cycle, but receives a strong positive pulse during a portion of the cycle, the plate current flowing through the tube flows in the form of pulses, which flow only when the grid of the tube is driven in a positive direction.

In the plate circuit of both oscillators we have shown an rf choke and a capacitor through which

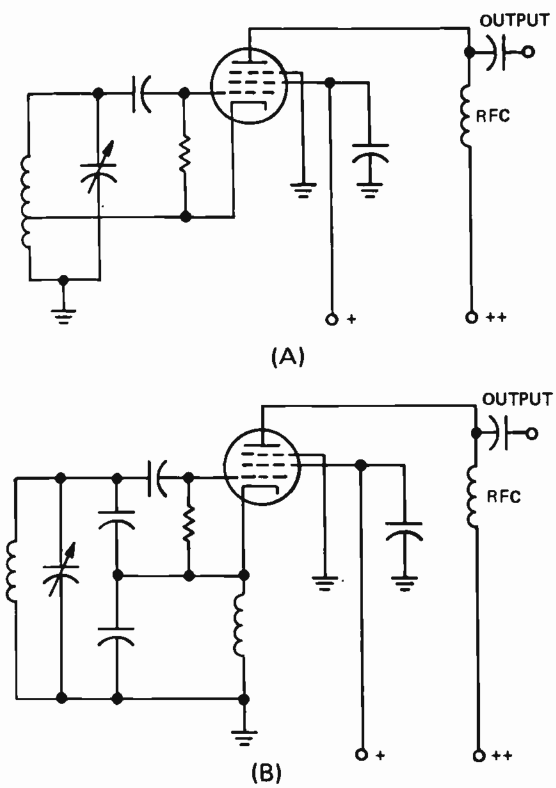


Figure 14. Electron-coupled oscillators; the circuit at (A) is a Hartley oscillator.

the oscillator can be coupled to the following stage. In some electron-coupled oscillators you will find a resonant circuit in the plate circuit of the tube instead of the rf chokes shown in Fig.14.

In some oscillators, this resonant circuit will be tuned to the same frequency as the resonant circuit in the grid circuit of the oscillator, but in other oscillators you'll find that it is tuned to a frequency equal to twice or three times the frequency to which the resonant circuit in the grid circuit of the tube is tuned. If we tune the resonant circuit in the plate circuit of the tube to twice the frequency of the resonant circuit in the grid circuit of the tube, we will have a frequency doubler.

The resonant circuit in the plate circuit of the tube is set into oscillation by the burst of plate current flowing through the tube. However, since the resonant frequency of this circuit is twice the resonant frequency of the input circuit, the circuit in the plate circuit begins to oscillate at a frequency equal to twice the frequency being generated in the grid circuit. The oscillation in the plate circuit therefore goes through two complete cycles before a second pulse is received from the plate of the tube. This is called a frequency-doubler circuit.

If the resonant circuit in the plate circuit is tuned to three times the frequency of the input circuit, the oscillation set up in the plate circuit will go through three complete cycles before it receives an additional pulse from the plate of the tube. This is called a frequency tripler. You might expect this to result in a damped wave, with the amplitude of the cycles which do not receive a reinforcing pulse from the plate of the tube being considerably less than the amplitude of the particular cycles during which the pulse is received. Of course, there will be some loss in the resonant circuit and there will be some change in amplitude. However, by the use of a high Q resonant circuit in the plate circuit, the change in amplitude that occurs each cycle is very small and for all practical purposes all the cycles of the sine wave produced in the resonant circuit will have essentially the same amplitude.

RC OSCILLATORS

Resistance-capacitance (RC) oscillators use the charging and discharging action of a capacitor and a resistor in the feedback path to cause oscillation. The RC oscillator is used in audio and low rf ranges. They offer an inexpensive method of obtaining a fairly stable sine wave within the range of their operation. Although many variations of the RC oscillators exist, there are only two basic types, the phase shift and the bridge.

The phase shift uses a series of RC phase shifting circuits between the output and the input to produce a feedback in-phase with the input. The bridge circuit usually uses an additional tube or transistor to obtain the phase shift and a bridge type circuit to control the feedback at the proper frequency.

Phase Shift. The phase shift oscillators in Fig.15 consist of a conventional amplifier and a phase shift feedback circuit. As in LC oscillators, the feedback must be positive. In the circuits in Fig.15(A) and (B), the output signal will be 180° out-of-phase with the input signal. Therefore the

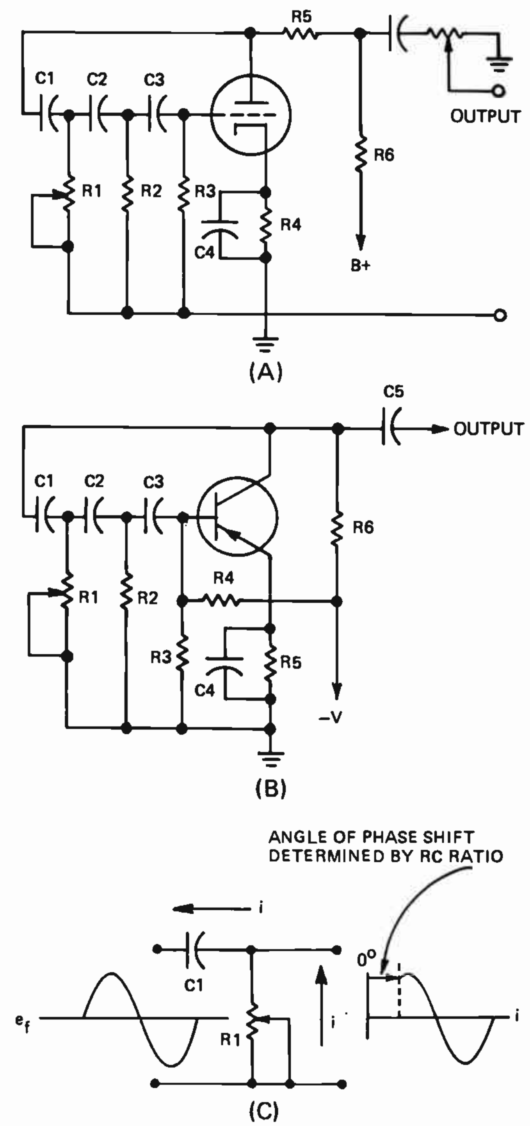


Figure 15. Phase shift oscillators.

phase shift network must shift the phase of the feedback signal 180°. This phase shift is provided by C1, C2, C3, and R1, R2, R3. One section, C1-R1, of the feedback loop is shown separately in Fig.15(C). The impedance of the circuits is capacitive and the feedback voltage, e_f, produces a current, i, through C1-R1 which leads e_f. The angle of the current lead is determined by the ratio of the reactance of C1 to the resistance R1. As the value of R1 is reduced, the circuit will become more capacitive and more current will lead the voltage up to a maximum of 90°. If resistance is increased the current lead will decrease.

By increasing the number of RC networks, the losses of the total feedback circuit will decrease and a less amount of phase shift will occur in each section of the feedback loop. For this reason some oscillators use five or six RC sections. In Fig.15 each section produces a 60° phase shift. The reactance of the capacitors is inversely proportional to the frequency; therefore, only one frequency will pass through the feedback loop. Normally the output is fixed in frequency due to the constant value of the capacitors. A variable output may be obtained by using ganged variable capacitors or resistors since an increase in the value of either R or C will decrease the frequency.

Let us examine the operation of the circuit in Fig.15(B). R3 and R4 establish the base bias while R5 and C4 furnish thermal stabilization and furnish an rf ground to the emitter. R6 is the load across which the output is taken and C5 is the coupling capacitor.

Once power is supplied, operation is started by any random noise in the power source or the transistor. This noise causes a change in base current which causes a large change in collector current. This change in collector current develops an output voltage across R6 which is 180° out-of-phase with the original change in base voltage. Part of the signal developed across R6 is returned to the base shifted 180° by the RC network. The shift in phase through the RC network causes the feedback to aid the output, resulting in positive feedback.

With fixed values of R and C, the 180° phase shift occurs at only one frequency; therefore, the output is a sine wave of fixed frequency. At all other frequencies, the reactance either increases or decreases, causing a variation in the phase relationship resulting in degenerative feedback.

The Bridge Oscillator. The Wien bridge oscillators shown in Fig.16 consist of low gain amplifier stages and a resistive-capacitive bridge circuit. The feedback is taken from the second amplifier and

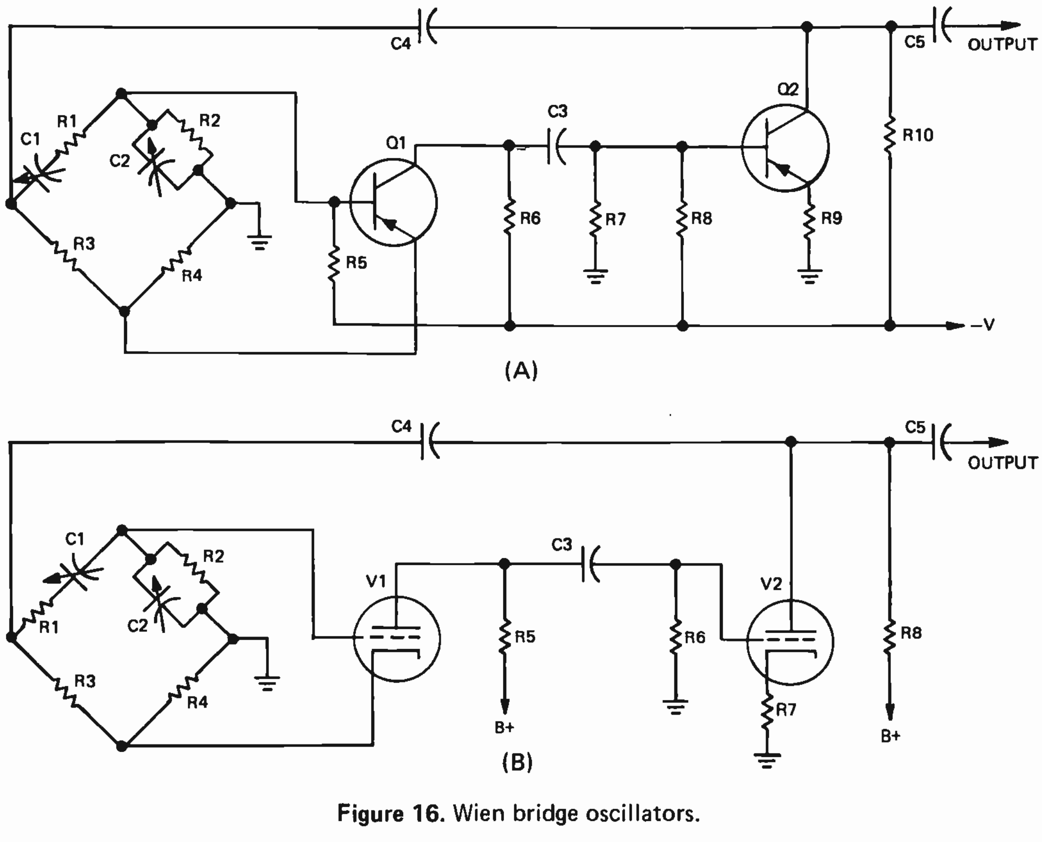


Figure 16. Wien bridge oscillators.

returned to the bridge circuit. Since there are two 180° phase shifts between the input of the first amplifier and the output of the second amplifier, the feedback will be positive. The resistive-reactive bridge circuit is designed to be balanced at the operating frequency; therefore, only feedback of the desired frequency reaches the input of the first stage. Let's examine Fig.16(A) and see how this works.

The voltage divider, R2 and R5, supplies base bias for Q1 and R7, and R8 furnishes bias for Q2. Thermal stabilization is provided by R4 and R9. R3 and R4 form a resistive leg of the bridge which is in shunt with the output of Q2. A portion of this output coupled by C4 and R3 appears across R4 as negative feedback. Since R4 is not frequency sensitive, the negative feedback is constant regardless of the frequency of the output. At frequencies other than the operating frequency, the negative feedback furnished by R4 will prevent oscillation. At the frequency of operation, which is controlled by the bridge reactive leg of R1-C1 and R2-C2, the positive feedback to the base of Q1 is maximum. This in-phase feedback signal is applied to the base of Q1 and is of sufficient amplitude to overcome the negative feedback across R4. The total feedback is therefore positive at the operating frequency.

The amplified output of Q1 is coupled to the base of Q2 by C3 and R7. Q2 further amplifies the signal and the voltage developed across R10 is coupled to the output by C5, and a feedback signal is coupled through C4 to Q1.

We have not covered all of the various LC and RC oscillator circuits you are likely to encounter. There are many different variations of the circuits we have discussed, and some entirely different circuits. However, most of the circuits you are likely to encounter will be one of the circuits we have discussed in this section of the lesson or a variation of one of these circuits. If you come across a circuit you do not recognize immediately, first determine whether it is RC or LC and, in the latter case, whether capacitive or inductive feedback is used. Once you have decided on the type of feedback that is used, you should be able to figure out how the oscillator circuit works if you keep in mind that its operation is similar to one oscillator we have described in this lesson.

SELF-TEST QUESTIONS

10 When plate current increases in the tuned grid oscillator [Fig.9(B)], what happens to the voltage at the junction of R1 and C2?

11 Where is the feedback voltage developed in the Colpitts oscillator [Fig.11(B)]?

12 The ultra-audion oscillator is modification of what type oscillator?

13 How is the 180° phase shift in feedback voltage accomplished in the phase shift oscillator?

14 What component(s) develops the feedback voltage in the Wien bridge oscillator (Fig.16)?

Crystal Oscillators

Although LC oscillator circuits have been developed that have very good frequency stability characteristics, the master oscillators in most transmitters still use crystals. Better frequency stability is the main reason for using crystals instead of LC resonant circuits in master oscillators. The frequency tolerance allowed by the FCC for most transmitting services is very small. The tolerance of a transmitter operating in the broadcast band, for example, is a frequency deviation of only 20 Hz from the assigned frequency. Some services are permitted slightly more frequency tolerance, but in all cases, the tolerance is rather strict. This restriction is necessary to keep the large number of stations operating in the frequency spectrum from interfering with each other.

Several types of materials can be used for crystals. These include quartz, Rochelle salts, and tourmaline. The most often used crystal material for generating radio frequency signals is quartz. Rochelle salts work better in low-frequency applications, such as in loudspeakers and microphones. Tourmaline will work as well as quartz, but because it is a semiprecious stone, it is more expensive.

The assembly usually referred to as a crystal is composed of a small piece of crystal material mounted in a holder. The crystal material is in the form of a small slab or wafer cut from a larger crystal. The way in which the wafer is cut from the natural crystal determines many of its electrical characteristics. In this section, we will find out how the crystal works, how the crystal is used in oscillator circuits, and finally we will discuss some of the most-used crystal oscillator circuits.

THE PIEZOELECTRIC EFFECT

A quartz crystal exhibits a property called the piezoelectric (pronounced pi-ee-zo) effect when it is compressed mechanically or when a current is applied to it. To illustrate this effect, suppose we have a small crystal wafer with leads attached to its two surfaces. If we squeeze the wafer or bend it in some way, a voltage will appear between the leads. If, on the other hand, we apply a small voltage across the leads, the crystal slab will bend, expand, or contract, depending on the polarity of the applied voltage and the crystal type. These two effects are used in many types of electronic equipment.

A crystal wafer has a definite mechanical resonant frequency at which it will vibrate most readily. This resonant frequency is determined by the physical dimensions of the wafer, particularly the thickness. The thinner the wafer, the higher the resonant frequency. Thus, when an ac voltage, whose frequency is near the crystal resonant frequency, is applied, the crystal will vibrate the greatest amount and produce the greatest output. Because of the large amount of vibration at the resonant frequency, there is a limit to how thin the wafer can be before it becomes too fragile for practical use.

CRYSTAL CUTS

The natural quartz crystal, as shown in Fig. 17, is said to have three major axes at right angles to each

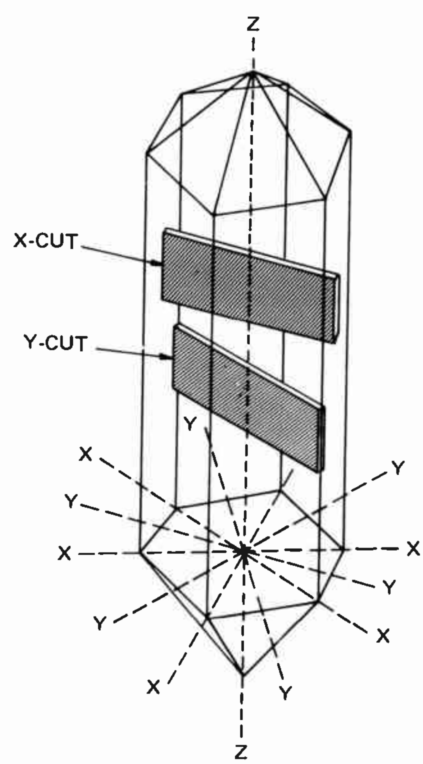


Figure 17. Major axes quartz crystal.

other. These axes, called the X, Y, and Z axes, are shown in the illustration. The X axis is called the electrical axis, the Y axis is called the mechanical axis, and the Z axis is called the optical axis. The way that the crystal wafers are cut with respect to these axes determines many electrical characteristics, including the frequency range in which the crystal will oscillate and the amount that the resonant frequency will change with changes in the temperature (the temperature coefficient).

If the crystal wafer shown by the shaded section in Fig.17 is cut so that its face is perpendicular to a Y axis, it is called a Y-cut crystal. Similarly, if its face is perpendicular to an X axis, it is called an X-cut crystal. Crystal wafers cut perpendicular to the Z axis have no piezoelectric properties.

Crystal cuts can be made at different angles to the major axes to produce crystals having slightly different electrical characteristics. The type of cut depends on the purpose for which it is to be used.

One of the most important characteristics of a quartz crystal used in oscillator circuits is the amount the crystal frequency varies with variations in temperature. We call this the temperature coefficient of the crystal. Y-cut crystals have a range of about -25 to +100 Hz/°C/MHz (Hz per degree centigrade per MHz). In other words, a one-degree centigrade increase in temperature may cause the frequency to decrease as much as 25 Hz, or increase by as much as 100 Hz for each MHz of the frequency for which the crystal is ground. To take an extreme case, the frequency of a 4-MHz crystal might increase by 400 Hz for each degree of temperature change. Because of this rather high temperature coefficient and also because Y-cut crystals have a tendency to oscillate at a second frequency near the design frequency, they are seldom used.

X-cut crystals have a range of about - 10 to - 25 Hz/°C/MHz. The minus sign means that the crystal frequency decreases with an increase in temperature. We call this a negative temperature coefficient. Even though the frequency variation is less, close temperature control is still required to keep the crystal oscillating at the correct frequency.

As an example of how temperature affects the crystal frequency, consider an X-cut crystal that has an operating frequency of 4650 kHz at 50°C. If it has a temperature coefficient of - 20 Hz per degree centigrade per MHz, let's determine what its operating frequency would be if the temperature changes 10°.

$$4650 \text{ kHz} = 4.65 \text{ MHz}$$

Therefore, the change in frequency per degree centigrade is:

$$4.65 \times 20 = 93.00 \text{ Hz}$$

The change in frequency with a temperature change of 10° will be:

$$93 \times 10 = 930 \text{ Hz}$$

and

$$930 \text{ Hz} = 0.930 \text{ kHz}$$

Thus we see that with a temperature change of only 10°, the frequency will change almost 1 kHz. With a 20° change, the frequency change would be almost 2 kHz. If the temperature increases 10°, the frequency will decrease to about 4649 kHz (4649.07 kHz), and if the temperature drops 10°, the frequency will increase to almost 4651 kHz (4650.93 kHz). Where there are many stations operating on frequencies close together, this shift in frequency could be enough to cause interference.

We can reduce the temperature effect on the resonant frequency of the crystal by cutting the crystals at angles to the major axes. Examples of such crystal cuts, called the AT-cut and the BT-cut, are shown in Fig.18. These crystals are really Y-cut

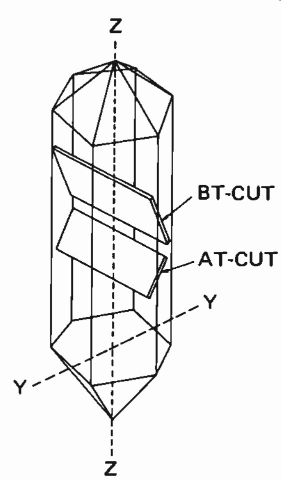


Figure 18. AT-cut and BT-cut crystals are cut on an angle to the Z axis as shown.

crystals with the face of the wafer at an angle of about 39° to the Z axis instead of parallel to it in the case of the AT-cut crystal, and about 45° to the Z axis in the case of the BT-cut crystal. Notice that the angle of the AT-cut crystal is opposite to that of the BT-cut crystal with respect to the Z axis. The temperature coefficient of these cuts is about ±2 parts per million at a temperature of 40°C to 50°C. Thus, the angle cut practically cancels the effects of temperature variation on the frequency of oscillation if it is operated within the temperature range.

Other angular cuts can be made to get other electrical characteristics and effects. Some examples are the CT and DT cuts used for lower frequency operation below 500 kHz. A CT-cut crystal is cut perpendicular to the BT-cut crystal, and the DT-cut crystal is cut perpendicular to the AT-cut crystal. Another cut is the GT-cut. This crystal is cut on an angle of about 45° to either the CT- or DT-cut crystals.

Figure 19 illustrates the variation of the resonant frequency with temperature for various crystal cuts. Notice that the frequency of the GT-cut crystal is practically constant from 0° to about 100°. This cut, therefore, is the best to use in equipment that will be subjected to wide temperature variations. The other cuts have zero temperature coefficients at or near specific temperatures.

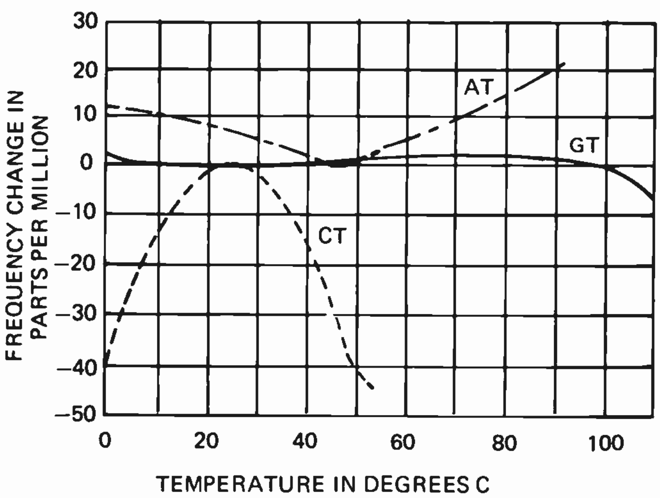


Figure 19. Frequency changes of different crystal cuts with changes in temperature.

CRYSTAL HOLDERS

After the crystal has been ground, or etched by means of chemicals, until it is of the proper thickness, it is placed in a holder. The holder is then sealed so that no air, dirt, or oil can reach the crystal wafer. These can cause erratic operation of the crystal. A crystal in its hermetically sealed holder is shown in Fig.20.

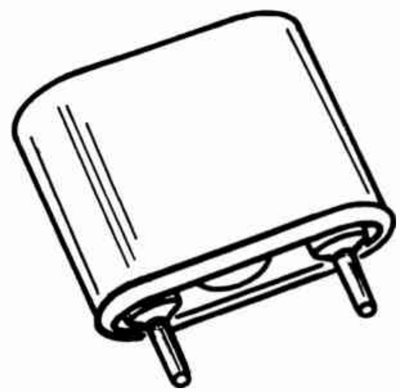


Figure 20. A quartz crystal in its holder.

There are two ways of mounting the crystal in the holder. In one way, a very thin film of metal is formed directly on the surface of the crystal by spraying or firing. The metal can be either silver, gold, or aluminum. The crystal is supported by flexible wires fastened to this metallic film with solder having a low melting point. The leads are attached to a node or non-oscillating portion of the crystal. The node can be either in the center of the crystal face or at an edge where inhibiting the vibrations will do no harm.

Another method of mounting the crystal in a holder is to arrange it so that it is pressed between two metal plates. The metal plates make electrical contact with the crystal surfaces. Another type uses an air gap 0.001 to 0.005 inch thick between the crystal and the upper plate. This air gap produces a damping effect on the amount of crystal vibration. Often, however, the crystal holder is evacuated so that the damping will be reduced.

Regardless of the type of mounting used, the holder itself is designed to keep the crystal free of grit, dirt, and oil film. Even a speck of dirt or a greasy film can change the characteristics of the crystal. Therefore, the crystal holder should never be opened, nor should the crystal be handled. If it is ever necessary to take one apart, the crystal may be cleaned with a good nonflammable commercial cleaner. The crystals should be handled with clean, lint-free cloths, not with bare fingers (bare fingers leave traces of perspiration on any object they touch).

Equivalent Circuit. The symbol in Fig.21(A) is used in schematic diagrams to represent the crystal in its holder. The equivalent electrical circuit of the crystal and holder assembly is shown in Fig.21(B). As you can see, we have here a series-parallel circuit composed of the series components L, R, and C1 shunted by capacity C2. The crystal, therefore, acts electrically as an LC circuit; as an inductance at frequencies above the resonant frequency and as a capacitance at frequencies below resonance. The apparent inductance L is due to the mass of the crystal, resistance R is the result of internal mechanical losses, and capacity C1 is the stiffness (piezoelectric properties) of the crystal. Capacity C2 is the capacity between the electrode plates, with the quartz crystal acting as the dielectric.

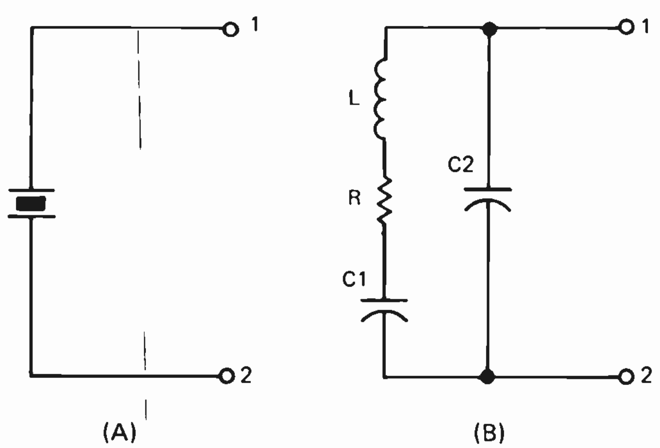


Figure 21. The schematic symbol for a quartz crystal in its holder is shown at (A). The equivalent electrical circuit of a quartz crystal is shown at (B).

Since the crystal acts as an electrical resonant circuit, we would expect it to be frequency selective; that is, it will oscillate more vigorously at its natural resonant frequency than at any other frequency. This is true. When an ac voltage is applied to its electrodes, the crystal generates an alternating potential of its own.

Because of its electrical characteristics, the crystal can be used in an oscillator circuit. The electrical properties of the crystal are somewhat different from those of a typical coil and capacitor. The mechanical properties of the crystal produce a very high apparent inductance, L. Also, since the mechanical losses during vibration are small, the electrical equivalent resistance is also very small. When we have an LC circuit containing a large inductance and a very low resistance, the Q of the circuit will be very high. This is the case with the crystal used in oscillator circuits.

Practical crystals have effective Q values which are about 100 times as great as that ordinarily obtainable with the usual inductance coil and tuning capacitor. Crystals, therefore, have extremely good frequency selectivity. The higher the Q, the better the frequency stability. Thus, if we substitute a crystal for the ordinary LC tank circuit, we can make an oscillator that has good frequency stability.

CRYSTAL OVENS

The purpose of a crystal oven is to maintain the crystal at a constant temperature to prevent frequency drift. Some time ago, it was common practice to put the crystal, the entire master oscillator of the transmitter, and often even the buffer stage in a heat-controlled chamber. This prevented temperature variations from affecting the physical dimensions of the coil and capacitor in the plate circuit, and thus changing the oscillator frequency. This is seldom done in modern transmitters; generally only the crystal is temperature-controlled.

A typical modern crystal oven is shown in Fig.22. The overall unit is about the size of a metal receiving tube – about 4 inches high and 1-1/4 inches wide. Smaller units are also available. The crystal oven unit fits into an octal tube socket. The unit in Fig.22 is guaranteed to hold the transmitter frequency within 10 Hz at any point on the broadcast band.

The crystal is contained in a ceramic holder attached to a copper support. The heater is also wrapped around this support. Thus, the metal support conducts the heat to the crystal and maintains it at a constant temperature.

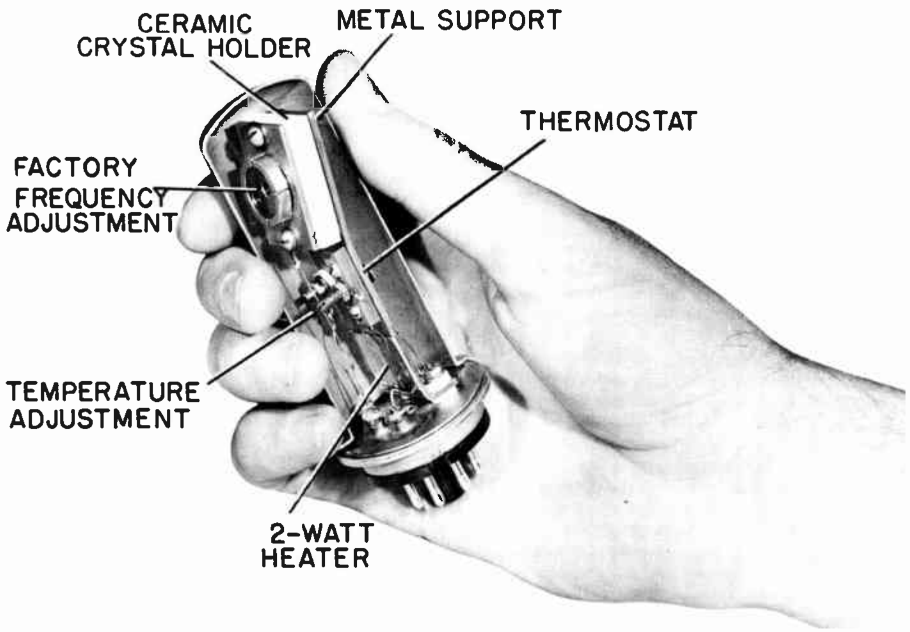


Figure 22. A cutaway view of a crystal oven.

The heat is controlled by a bimetallic thermostat attached to the metal support. When the temperature of the support drops a very small amount, the thermostat contacts close, and current is applied to the heater. When the support reaches a certain temperature, the thermostat contacts open. The difference between the on and off temperatures of the thermostat is very small; that is, the temperature must drop only a small amount before current is again applied to the heater coil.

The crystal frequency and thermostat temperature adjustments shown in Fig.22 are set at the factory when the unit is assembled. It is impossible to change the adjustments because the unit is hermetically sealed. Thus, when ordering such a crystal unit, you must specify the exact crystal frequency you want and the circuit in which the crystal will be used. Crystal manufacturers will not guarantee the operating frequency of a crystal unless the crystal is adjusted in the circuit in which it will be used.

CRYSTAL OSCILLATOR CIRCUITS

To help see how the crystal oscillator works, let's look at another LC oscillator. This oscillator is shown in Fig.23, and is called a tuned-grid, tuned-plate oscillator. It is easy to see where this oscillator gets its name, since there are resonant circuits in both the grid circuit and the plate circuit of the tube.

The tuned-grid, tuned-plate oscillator works because of the capacity between the plate and grid of the triode tube. When the resonant circuit in the plate circuit of the tube is tuned to a frequency slightly lower than the operating frequency, it will act like an inductance. Under these conditions, the phase of the signal voltage fed from the plate of the tube back to the grid of the tube is correct to aid the ac grid voltage, and oscillation occurs.

The crystal oscillator shown in Fig.24 is simply a modification of the tuned-grid, tuned-plate oscillator shown in Fig.23. Here a crystal has been substituted for the resonant circuit in the grid circuit of the oscillator, and a milliammeter is shown in the plate circuit of the stage to measure plate current. Actually, almost all oscillators have some provision made for measuring plate current.

Since the crystal is the equivalent of the circuit that has been removed, the crystal oscillator operates in exactly the same way as the resonant circuit in the grid of the tuned-grid, tuned-plate oscillator. The plate circuit must be tuned so that it presents an inductive load, and energy will be fed from the plate of the tube back to the grid in the correct phase to aid the ac grid voltage so oscillation will occur.

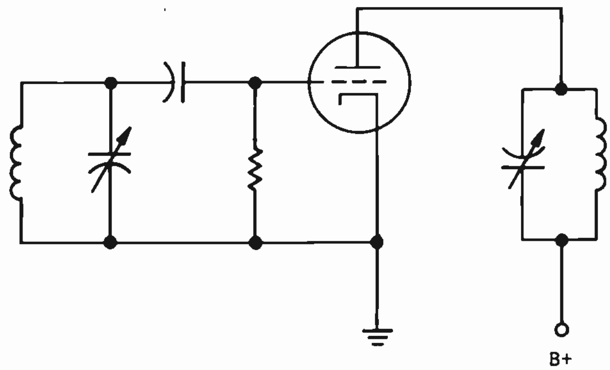


Figure 23. A tuned-grid, tuned-plate oscillator.

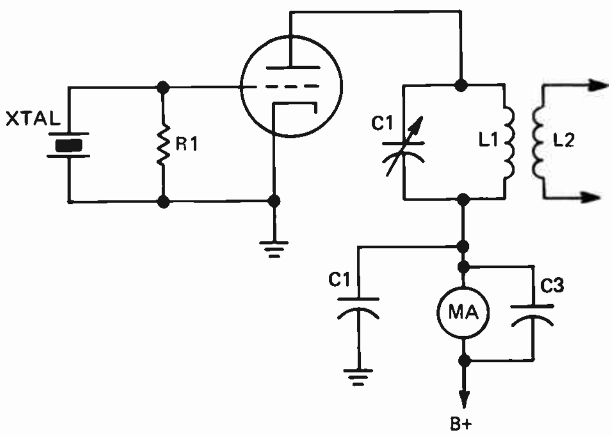


Figure 24. A simple crystal-oscillator circuit.

In the input circuit, the rf current flowing through the crystal itself is limited only by the resistance of R1. This resistance is usually low enough to result in a fairly high crystal current. If the resistance is made too high, the crystal may be somewhat erratic as it starts to oscillate when power is applied to the stage. On the other hand, if the current becomes higher than about 100 ma, the crystal vibrations may be so violent that the crystal may shatter, and thus destroy itself. Of course, the thinner the crystal and the higher its resonant frequency, the lower the safe current limit becomes.

Many transmitters have an rf current meter in series with the crystal to indicate this current. To prevent possible damage to the crystal, the amplitude of oscillation in a crystal oscillator circuit must be kept at a safe level. Usually this is accomplished by keeping the plate voltage on the oscillator tube low. This low plate voltage reduces the maximum output power that can be obtained from such oscillators.

Pierce Oscillator. It is possible to place the crystal between the plate and grid circuits, as shown in Fig.25, instead of between the grid and cathode. In this case the circuit is similar to the ultra-audion circuit. This circuit is called the *Pierce* oscillator.

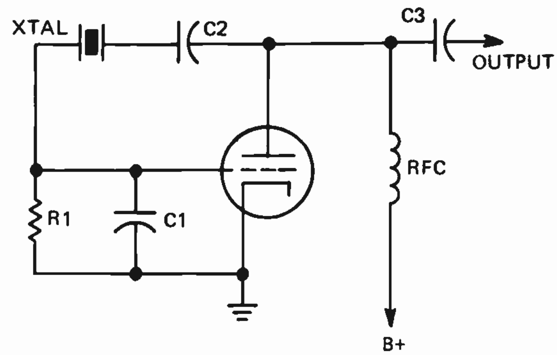


Figure 25. The basic Pierce oscillator.

Notice that the circuit contains no tank inductance or tuning capacity. The amount of feedback and the grid excitation can be controlled to some extent by adjusting the capacity of C1. The larger this capacity, the less the feedback. The exact capacity of C1 in most cases is not critical. Usually, when the best value is determined, crystals of slightly different frequency can be switched into the circuit without further adjustment. Again the plate voltage must be low to prevent damaging the crystal.

Crystal Oscillators Using Multi-Grid Tubes. Tetrode and pentode tubes also may be used in crystal-oscillator circuits. These tubes have less plate-to-grid capacity than do triodes, and therefore, there is less feedback current to the grid when they are used as crystal oscillators. The lower feedback means that these tubes can be operated at a higher output power than a triode without excessive rf crystal current.

A crystal oscillator using a pentode tube is shown in the diagram of Fig.26. Because of the limited amount of plate-grid feedback, the plate voltage can be considerably higher than for the triode oscillator. This, of course, increases the output power. The circuit may also be used for high-frequency crystals having fundamental resonant frequencies of about 10 MHz. The output in this case can be high, but the small amount of current through the crystal protects it from excessive vibration. Sometimes extra feedback is needed

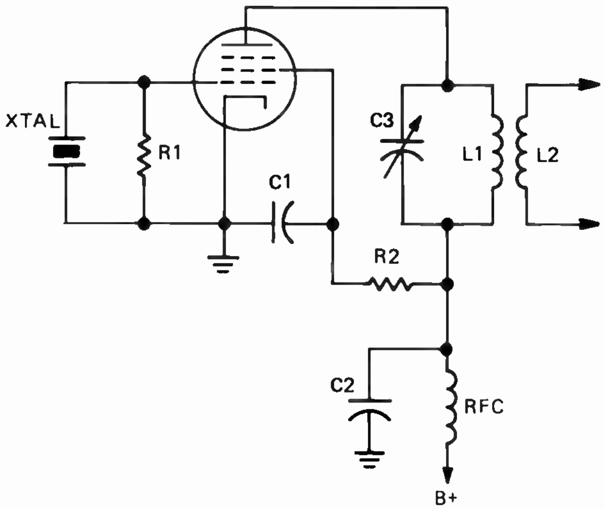


Figure 26. Crystal oscillator using a pentode tube.

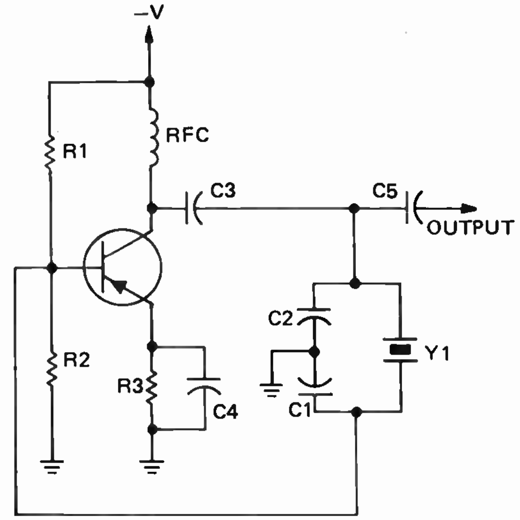


Figure 28. Colpitts crystal oscillator.

to produce oscillation in the circuit. This is done by connecting a small capacitance between the plate and the control grid.

Crystal Control Transistor Oscillators. A transistor tickler coil oscillator using a crystal to control feedback is shown in Fig.27. Positive feedback from collector to base is provided through the mutual inductance between the windings of T1. This provides the 180° phase shift necessary to sustain oscillation. In this circuit, the crystal acts as a series-resonant circuit. At frequencies other than the resonant frequency the crystal acts as a high impedance, blocking the feedback path. At the resonant frequency the crystal offers minimum impedance to the feedback signal. The operating frequency may be varied by the use of different crystals and tuning the tank to the frequency of the crystal with C1.

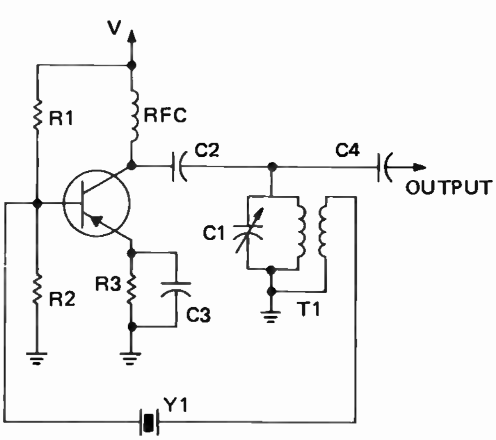


Figure 27. Crystal tickler coil oscillator.

The crystal oscillator shown in Fig.28 is a variation of the Colpitts oscillator. In this circuit the crystal acts as a parallel tuned circuit and replaces the LC tank. Operating frequency is determined by the crystal and the capacitance in series with it. At the resonant frequency, the crystal and capacitors in parallel with it form a high impedance tank circuit. Capacitors C1 and C2 form a voltage divider that is center-tapped to ground. The voltage developed across C1 is applied between the base and ground providing a 180° phase shift in the feedback path.

OVERTONE OPERATION

Most crystals will oscillate not only at their fundamental resonant frequencies, but also at odd overtones of the fundamental. A crystal with a fundamental of 4 MHz, for example, will oscillate also at frequencies near 12 MHz, 20 MHz, 28 MHz, etc. The oscillation frequency will not be an exact multiple of the fundamental oscillation frequency. Exact multiples would mean harmonic operation rather than overtone operation. Thus, if you use a crystal designed for fundamental operation as an overtone crystal, you can use the frequency markings on the crystal holder only to get an approximate idea of the actual frequency at which the oscillator is working. To determine the exact frequency, you will have to measure it with a frequency meter.

Although any crystal can be used in overtone operation, it is best to use one that has been ground specifically for this purpose. The ordinary fundamental type of crystal is somewhat unstable and hard to adjust when operated on an overtone. Most of the overtone crystals are designed to operate on the third overtone; operation at higher overtones is possible, but the stability and ease of adjustment becomes more critical at the higher overtones.

The chief advantage of overtone operation is that we can generate frequencies in the vhf range without the use of doubler stages. The highest fundamental oscillation frequency of a crystal is about 10 MHz. If one is ground to operate on a higher frequency, it is so thin that it breaks easily. Thus, by operating it on the third or fifth harmonic, we get a much higher frequency than we could on the fundamental. Another advantage of overtone operation is that no frequencies are generated that can cause interference with other channels. Because the number of frequency multiplier stages is reduced or completely eliminated, overtone crystals are used often in mobile, marine, and aircraft transmitters in which compactness is important.

The oscillator circuit using an overtone crystal is similar to an ordinary crystal oscillator circuit of the type you will study later. The overtone at which the crystal will operate is determined by the plate circuit resonant frequency. The circuit of an overtone oscillator is shown in Fig.29. The oscillator uses a crystal having a fundamental frequency of about 8 MHz to give a 24 MHz output when operated on the third overtone.

In Fig.29, the circuit made up of C1 and the upper end of L1 is resonant at 24 MHz. The tap on L1 is held at rf ground potential by C2. The lower end of L1 is inductively coupled to the upper end

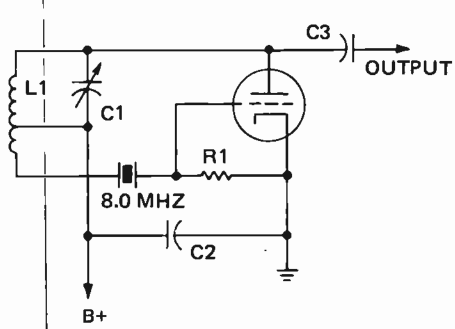


Figure 29. An overtone oscillator circuit.

(usually this is a simple one-tapped coil) so energy is fed from the plate circuit back to the grid circuit to sustain oscillations. The rf signal fed back to the grid circuit will cause the crystal to oscillate on its third overtone frequency. Output from the oscillator is taken from the plate circuit through C3.

Modern overtone crystals are capable of oscillating as high as 100 MHz on higher order harmonics, so the output of a crystal oscillator could be up in this region. However, when such high-frequency signals are needed, you will often find a crystal oscillator operating at 50 MHz followed by a doubler to increase the frequency to 100 MHz, or an oscillator operating at about 33.3 MHz followed by a tripler.

An oscillator circuit that can be used to generate both even and odd harmonic frequencies of the fundamental is the tri-tet circuit shown in Fig.30. The output signals are harmonics rather than overtones.

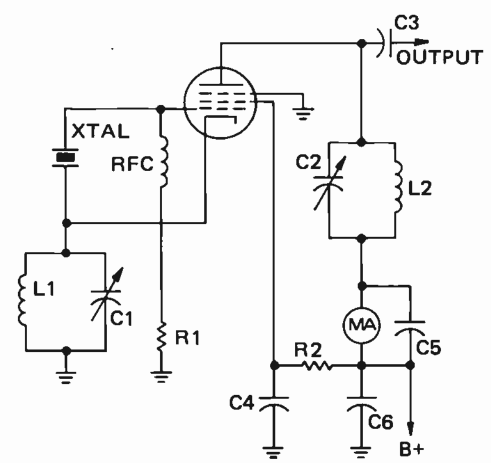


Figure 30. The tri-tet oscillator circuit.

The tri-tet circuit is actually the crystal version of the electron-coupled circuit described earlier in this lesson. The crystal and the resonant tank L1-C1 are connected to the control grid, the cathode, and through C4 to the screen grid (which acts as the oscillator plate), to form a modified tuned-plate, tuned-grid oscillator. The cathode tank circuit L1-C1, therefore, must be tuned to a frequency slightly lower than that of the crystal in order to be inductive.

Since the screen grid is bypassed to ground by capacitor C4, there is very little direct coupling between the actual tube plate and the oscillator portion of the circuit. The electron stream reaching the plate, however, arrives in the form of pulses that contain relatively large amounts of harmonic energy. If we tune the plate tank circuit L2-C2 to a frequency twice that of the crystal, a considerable amount of output power at the second harmonic frequency will be obtained. Even if the plate is tuned to a frequency three times that of the crystal we will get a fair amount of third harmonic output power.

Therefore, the tri-tet circuit not only behaves as a crystal-controlled oscillator, but also performs as a frequency doubler or tripler at the same time. The additional feature of the electron coupling prevents load variations from reaching the crystal and influencing the oscillator frequency. A disadvantage of this circuit is that both sides of the crystal are above rf ground potential. Another is the fact that a cathode coil is needed.

CRYSTAL-OSCILLATOR ADJUSTMENT

The tuning procedures for crystal oscillator circuits of all types are similar. The curves in Fig.31 show how the plate current of an oscillator tube varies with changes in the tuning capacity. The solid curve is for an unloaded oscillator circuit, and the dashed curve represents the circuit loaded.

When you adjust a triode or Pierce oscillator like those shown in Figs.24 and 26, begin first with the plate tank capacitor in the minimum capacity position. Then rotate the capacitor toward the maximum capacity position. As soon as oscillation begins, the plate current will begin to decrease. It will decrease more and more, going through points 3 and 2 in Fig.31 as oscillation becomes stronger. If the plate capacitor is turned too far, however, oscillations will stop abruptly as at point 1 in Fig.31. In practice, it is best not to approach point 1 too closely because minor voltage or current variations may stop the oscillator. Instead, adjust the plate tuning capacitor so that operation will be somewhere in the stable region between points 2 and 3.

When a load is placed on the oscillator circuit, the plate current dip of the oscillator will not be as pronounced. It will follow the dashed curve in the diagram. As before, however, too much capacity will cause the oscillator to stop. The operating point again should be somewhere between points 2 and 3.

The transistor oscillator in Fig.27 would be adjusted in a similar manner, C1 being tuned to a point where the current through the tank is between points 2 and 3 in Fig.31. C1 is first tuned for minimum capacitance. Then the capacitance is increased until current is between the stable points of operation.

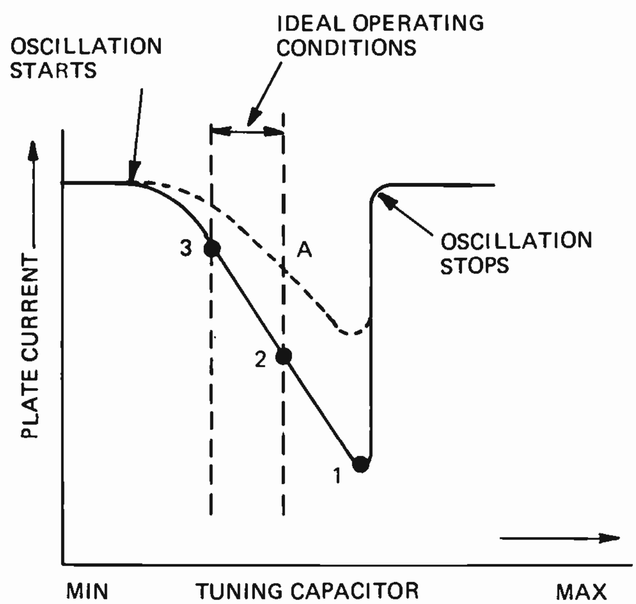


Figure 31. Curves showing how the plate current in a crystal-oscillator circuit varies with tuning.

When adjusting the tri-tet circuit in Fig.30, first set the cathode tank capacitor C1 to a frequency higher than resonance (minimum capacity) so that oscillations will occur. In this circuit, the usual parts values are such that oscillation will be maintained over a fairly wide range of the C1 adjustment. However, the crystal current increases very rapidly as the capacity is increased. Start tuning C1 for the minimum capacity position and progress only to the point of normal crystal current. Usually an rf milliammeter is placed in series with the crystal to measure this current. The current should be kept below 100 ma.

With no load connected, the plate tank capacitor C2 should be adjusted for a minimum value of plate current as indicated by a sharp dip in the meter reading. Now, connect the load to the circuit. This is usually the grid of the following buffer amplifier stage. With the load attached, capacitor C2 may need readjustment to bring the plate current back to minimum. This time, however, the current value will be somewhat higher because of the loading.

Finally adjust the cathode tank capacitor C1 for maximum harmonic power output which will be indicated by a maximum current flow in the following amplifier grid current. Also watch the crystal current to be sure that it does not rise above the safe limit.

SELF-TEST QUESTIONS

15 Why is plate voltage kept low in the Pierce oscillator?

16 Describe how frequency is controlled in the oscillator in Fig.27.

17 What is the advantage of overtone operation?

18 In the multiple crystal frequency synthesizer in Fig.32, what is the output of the mixer if S1 is in the 80 kHz position and S2 is in the 1300 kHz position?

19 In addition to performing as an oscillator, what other function is accomplished by the tri-tet oscillator?

Frequency Synthesizers

Several communications services require equipment capable of operation on many different frequencies. Two good examples are the class D citizens band and the aircraft navigation and communication band. If this equipment used the crystal oscillators discussed previously, the crystal would have to be changed each time operation on a different frequency was required. In most cases, the equipment is a transceiver which allows both transmitting and receiving on a channel. This equipment would require two crystals for each channel, one to control the frequency of the transmitter and another to control the frequency of the receiver local oscillator. This method is satisfactory as long as operation on only a few channels within a band is desired. However, in the case of a class D CB transceiver, forty-six crystals would be needed for operation on all the channels! The cost of the crystals alone would make the unit quite expensive. A frequency synthesizer provides a good solution to this problem.

A frequency synthesizer is a circuit which produces a sizable number of frequencies, each having the stability and accuracy that would normally be obtained only by using individual crystal oscillators. There are two major types of frequency synthesizers in use today. They are the multiple crystal type and the digital phase-locked loop frequency synthesizers. The multiple crystal synthesizer currently enjoys the widest use; but, the digital frequency synthesizer is rapidly gaining in popularity due to the greater number of frequencies available from still fewer crystals.

MULTIPLE CRYSTAL SYNTHESIZER

Figure 32 shows a multiple crystal frequency synthesizer used in the Johnson Messenger 123A class D CB transceiver. Circuit details have been omitted for simplicity. This type of synthesizer operates on the same principle that you studied earlier in the basic superheterodyne receiver in which an incoming signal beats with the local oscillator to form two new frequencies. These new frequencies are the sum and the difference of the incoming signal frequency and the local oscillator frequency, the difference frequency being the intermediate frequency.

In Fig.32 you see that two oscillator circuits are used with a mixer which will beat the outputs of the oscillators together. A double-tuned circuit is used at the output of the synthesizer mixer to select the difference frequency and reject the sum of the two frequencies. Each of the oscillators is tunable to different frequencies by rotating the channel selector switch, S2A and S2B. The switch has 24 positions; however, only 23 of the positions connect the crystals to their respective oscillators to allow the transceiver to operate on one of the 23 class D CB channels. S2A connects one of the six 32 MHz crystals to the high oscillator. S2B connects either a 5 MHz crystal or a 6 MHz crystal to the low oscillator, depending on whether the transceiver is in the transmit or receive mode.

With S2 in the position shown, a 5.735 MHz crystal is selected for transmit. This signal mixes with the 32.7 MHz signal from the high oscillator to produce an output frequency of 32.7 - 5.735 = 26.965 MHz. This is CB channel 1. In the receive mode, S2B selects the 6.190 MHz crystal while the 32.7 MHz crystal still controls the high oscillator. This time the output

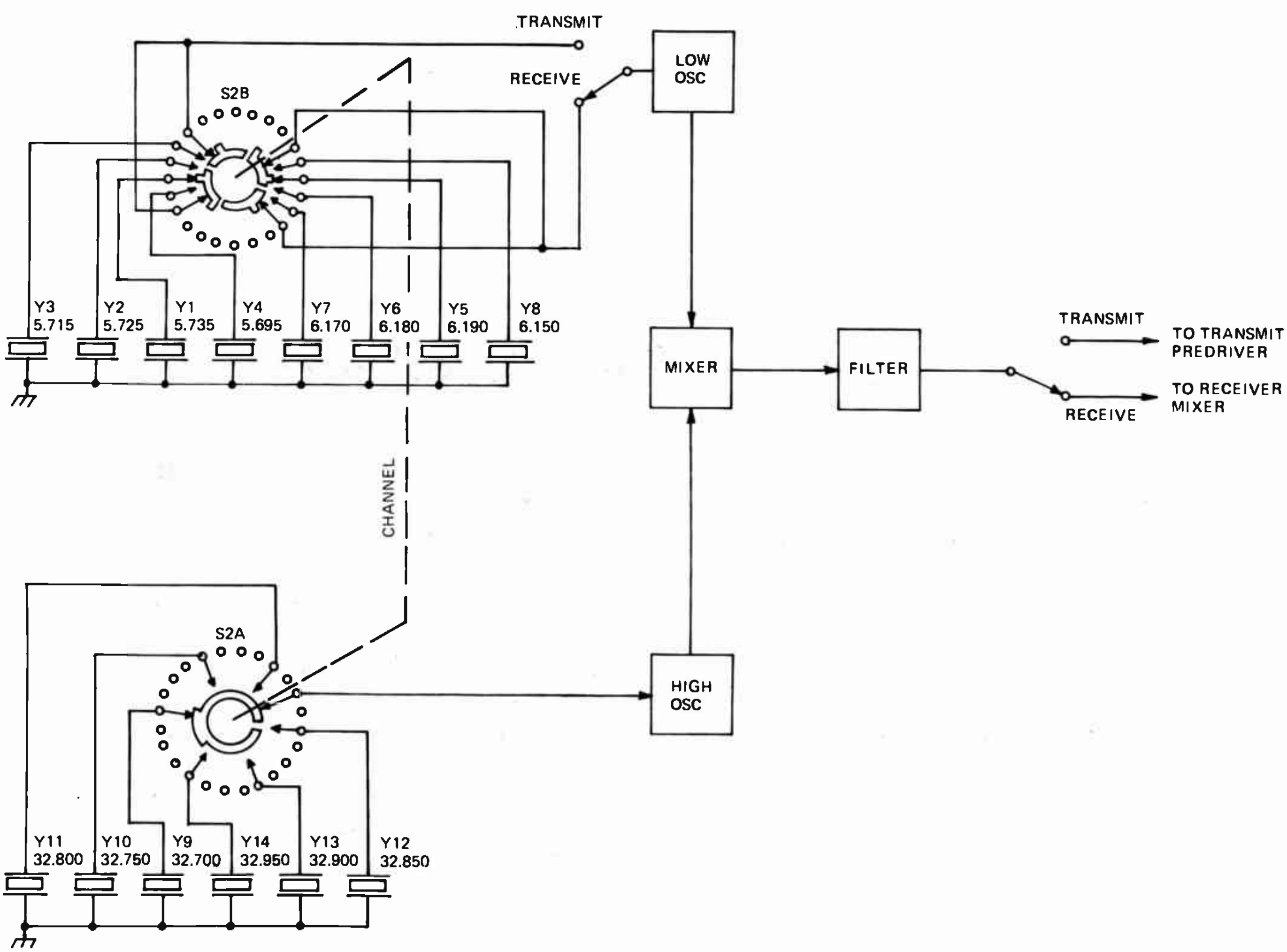


Figure 32. A multiple crystal frequency synthesizer used in the Johnson Messenger 123A.

of the mixer is 32.7 - 6.190 = 26.510 MHz and is fed to the receiver mixer as the local oscillator frequency. Since the transceiver uses an i-f of 455 kHz, a signal on channel 1 could be received:

$$26.510 + 0.455 = 26.965 \text{ MHz}$$

If you wish to see how the transmit and receive local oscillator frequencies for any of the other channels are generated, you could do so by mentally rotating the selector switch to see which crystals are selected. For example, imagine the switch rotors are moved one position clockwise. S2A still selects the 32.7 MHz crystal. S2B selects the 5.725 MHz and the 6.180 MHz crystals to produce the correct transmitter frequency of 26.975 MHz and the correct receive local oscillator frequency of 26.520 MHz for operation on channel 2.

This synthesizer produces all the necessary 46 specific frequencies to allow the transceiver to cover the full 23 channels which make up the class D citizens band. Instead of requiring 46 separate crystals, only 14 are needed.

DIGITAL FREQUENCY SYNTHESIZERS

Multi-channel communications equipment being designed today seldom includes a synthesizer of the multiple crystal type. While considerably fewer crystals are needed, this system still requires far more crystals than digital phase-locked loop (PLL) frequency synthesizers. A PLL synthesizer can be designed using only one crystal in the system, yet offering an almost unlimited number of output frequencies.

These systems were not designed into earlier equipment due to the lack of integrated circuit "building blocks" containing the various circuit functions necessary for constructing a PLL system. Without them, the cost of constructing even a simple system using individual transistors and other components would have been too high. This is especially true in the case of a unit requiring only 46 specific frequencies, such as in the unit we previously discussed.

The basic components involved in a digital phase-locked loop frequency synthesizer, in its simplest form, are shown in Fig.33. There are four major circuits in a PLL system. They are the VCO, the programmable dividers, the frequency/phase detector, and the loop filter. Notice how the circuits are connected to each other. The signal path forms a loop from the output of the VCO, through the programmable divider to the phase detector, through the phase detector to the loop filter, and through the filter back to the VCO. Thus, you can see how the system got the "loop" part of its name.

You may have already noticed that the phase detector has a second input which we have labeled reference frequency. It was not included in our discussion of the loop components because it is not "in the loop." In other words, it is not affected or changed in any way by changes in any of the loop components. The loop components affect each other directly. A change in any one of them will cause a change in the other three.

Let's examine each of the loop components a little more closely. The block in Fig.33 labeled VCO is the frequency generating component of the PLL system. VCO stands for voltage-controlled oscillator. The frequency of the oscillator's output changes in proportion to the applied control voltage. Let's see how such a circuit might be implemented.

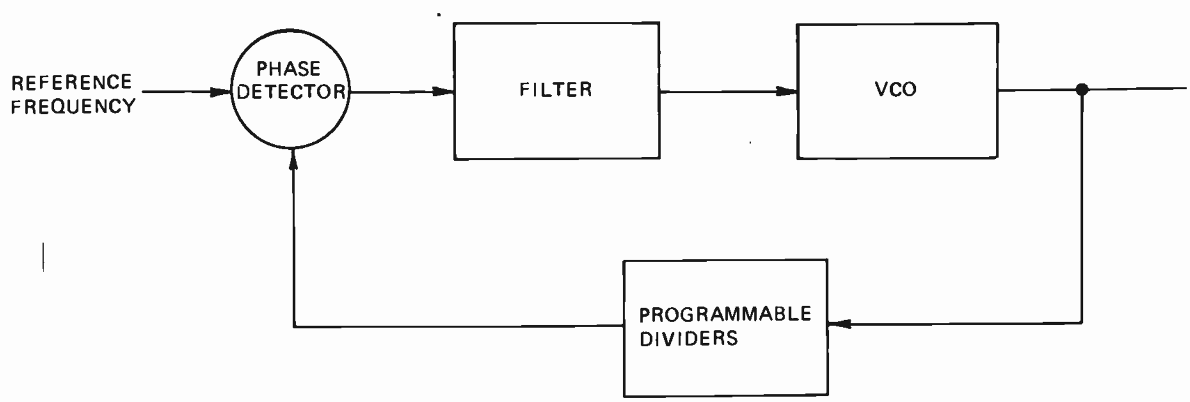


Figure 33. The basic digital phase-locked loop frequency synthesizer.

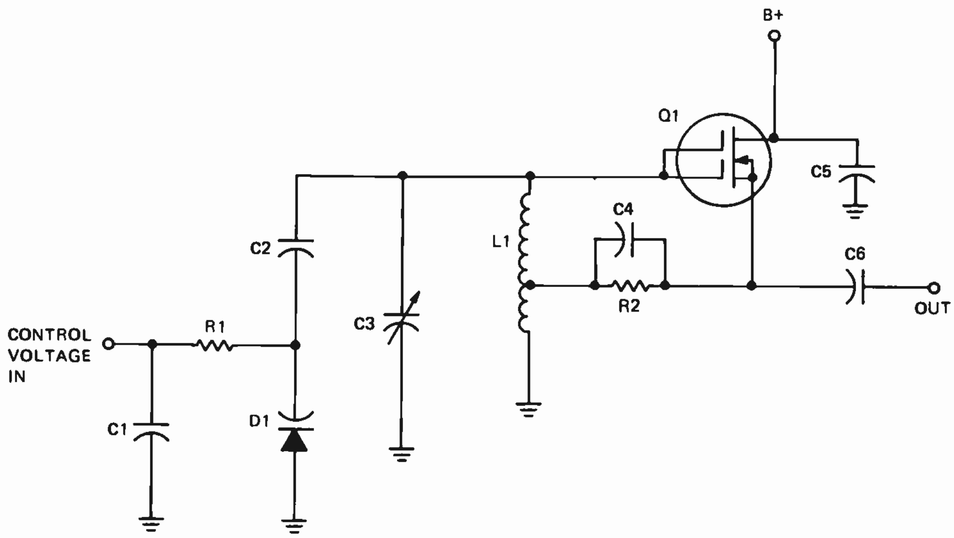


Figure 34. A Hartley voltage controlled oscillator.

A Voltage Controlled Oscillator. A dual-gate MOSFET is used as the active device in the Hartley oscillator circuit shown in Fig.34. Since you have already learned about the operation of a Hartley oscillator, we will not cover it again here. The important component to notice in Fig.34 is the diode D1.

You will recall from a previous lesson that the capacitance of a diode junction will decrease if the reverse bias voltage across the junction increases. D1 is a special junction diode called a varactor diode whose capacitance versus voltage characteristic is linear and predictable. Since the diode is connected in series with C2, across the tank circuit, the frequency of the oscillator will increase as the applied voltage increases. The lower frequency limit of the tank circuit is adjusted by C3 with zero volts, or the minimum expected voltage, applied to D1 through R1.

Multivibrators. The oscillators you have studied up to this point have all been sine wave oscillators; that is, their output is a sine wave. The output of the oscillator we are about to discuss is not a sine wave and therefore is referred to as a nonsinusoidal oscillator. An oscillator of this type is generally classified as a relaxation oscillator. A relaxation oscillator uses a regenerative circuit in conjunction with an RC circuit to provide a switching action. The RC time constants control the shape and frequency of the output waveforms.

Let's examine the circuit in Fig.35. This is an astable or free running multivibrator. A multivibrator is a two-stage oscillator in which one stage conducts while the other is cut off, until a point is reached at which the stages reverse their conditions. The output of each stage is coupled to the input of the opposite stage. The output signal of the stage is a square wave whose frequency is determined by the RC time constants in the feedback loops.

In the circuit in Fig.35, one of the transistors will conduct a little harder than the other when power is applied due to the slight differences that exist between any two transistors, even of the same type. Let's assume that Q1 conducts hardest. The collector current flowing through R1 will cause the voltage at the junction of R1 and C1 to go in a negative direction. This negative pulse is fed through C1 to the base of Q2, cutting off its conduction. This causes the voltage at the junction of R4 and C2 to increase. The positive-going pulse is coupled through C2 to the base of Q1, causing it to go into saturation.

Once Q1 is saturated, there is no change in the voltage level at the junction of R1 and C1; therefore, the base of Q2 begins to go positive as C1 charges up through R3. Soon, Q2 begins conducting, producing a negative-going pulse at its collector. This pulse is fed through C2 to the base of Q1 and starts reducing the transistor's conduction. The resulting positive-going pulse at the collector of Q1 is coupled through C1 to the base of Q2, driving it into saturation. Q1 is driven to cutoff.

This process continues until the power is removed from the circuit. The rate at which the

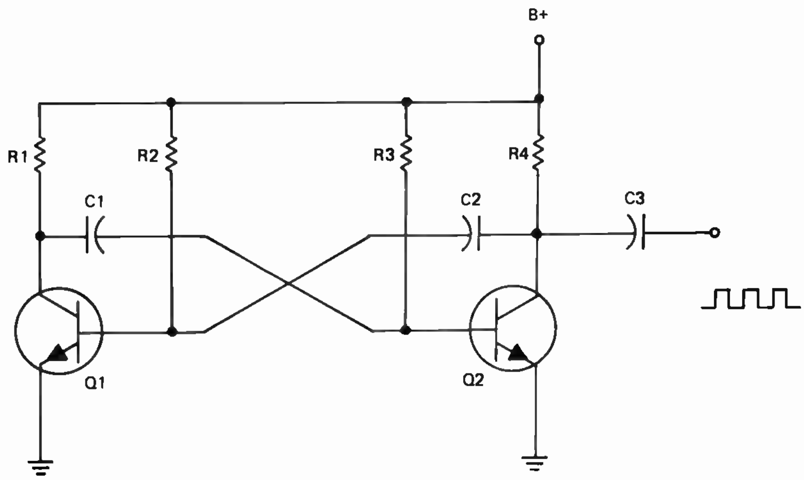


Figure 35. An astable multivibrator.

circuit changes state is controlled by the time constant of R3-C1, R2-C2, and the B supply voltage. If the two time constants are exactly equal, the output signal will be a symmetrical square wave.

There are variations of the astable multivibrator called the monostable and the bistable. Trigger pulses are required for these two circuits. When the monostable is triggered, it will change state, remain in the new state for a short time, then return to the state it had before the arrival of the trigger pulse. The bistable has two stable states. It will change state upon the reception of a trigger pulse and remain in the new state until a second pulse is received. This circuit is used in digital flip-flops such as those you studied in an earlier lesson.

Voltage-Controlled Multivibrator. The circuit in Fig.36 is one which can readily be used as a low-frequency VCO. The circuit could more accurately be referred to as a voltage-controlled multivibrator, VCM. The only difference between this circuit and the one we just discussed is the addition of the two junction field-effect transistors Q2 and Q3. They allow the frequency of the multivibrator to be controlled by an external voltage. R2 and R3 are current limiting resistors, used to ensure that the FET's are not destroyed by excessive current. The applied control voltage changes the channel resistance of the FET's and causes them to perform as though they were variable resistors. Since they are the timing resistors for the RC circuits, the frequency of the multivibrator changes. Thus we could use this circuit if we wished to construct a low-frequency VCO for a phase-locked loop frequency synthesizer.

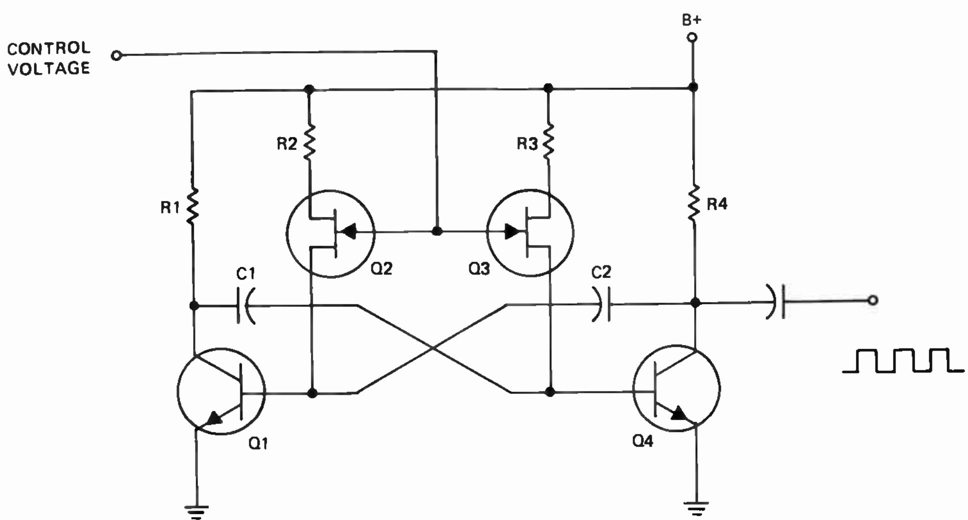


Figure 36. A voltage-controlled multivibrator (VCM).

Programmable Dividers. Let's examine the next component in the PLL, the programmable divider. This circuit allows the output frequency of the synthesizer to be changed according to the number on the divider's programmable inputs.

We have shown a diagram of a type D (data) flip-flop in Fig.37 together with its associated truth table. The table tells how the flip-flop will perform, depending upon the condition of its inputs. It will divide an incoming signal by two, the same as other flip-flops you studied in an earlier lesson. The thing which makes the type D flip-flop particularly useful in a programmable divider is its preset and clear inputs. These allow the state of the outputs to be changed, or set up, prior to the dividing operation. Let's go over the operation of the type D flip-flop.

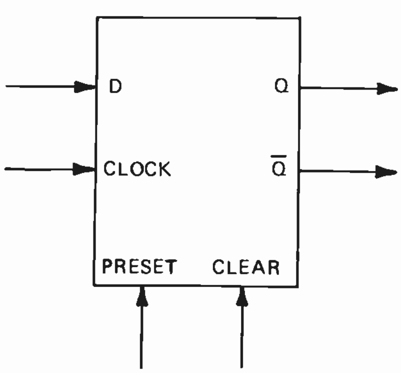
Remember, the 0 condition means a low voltage level and the 1 condition means a high voltage level. Further, you will see both X and an arrow pointing up in the INPUTS column of the truth table shown in Fig.37. The X means the input "doesn't care" whether it is high or low. The outputs will be as shown regardless of the condition of this input. The upward pointing arrow at the bottom of the CLOCK column means that the outputs change to the state shown on the positive-going or leading edge of the clock pulse.

The preset and clear inputs of the type D flip-flop are "active low." In other words the flip-flop is set when the preset input is pulled low, toward ground. The clear input must remain high. The state of the clock and D inputs may be either high or low. The Q output will be set to a logic 1 and the $\overline{Q}$ to a logic 0. When the preset input remains high and the clear line is pulled low, the outputs will be set to the opposite state; Q will be low and $\overline{Q}$ will be high. It is only necessary to pull the preset or clear inputs low momentarily to set or clear the flip-flop. The outputs will remain set or cleared when the selected input returns to its inactive or high state.

An illegal set operation is possible when both inputs are pulled low at the same time. This operation is not normally used since both the Q and $\overline{Q}$ outputs will be set high but they will not remain in this state when the inputs return high. Further, it is not known exactly what state the outputs will assume once the preset and clear inputs return to their inactive condition.

When the preset and clear inputs remain inactive, the logic level appearing at the D or "data" input of the flip-flop will be transferred to the Q output on the positive-going edge of the clock signal.

Four type D flip-flops can be connected together using additional logic to form a programmable decade counter which can be made to divide by any whole number from 1 to 10. Such a divider is called a programmable decade counter and is available as a single integrated circuit. The incoming signal is applied to the clock input of the first flip-flop. The D inputs are connected to cause the flip-flops to divide by 10 through the use of feedback connections. If four stages were merely connected with the Q output of one feeding the clock input of the following flip-flop, the divider would divide by 16 instead of 10. This is the difference between a straight binary counter and a decade or binary coded decimal (BCD) counter. Four-bit (four stage) binary counters are sometimes used as programmable dividers, but most PLL systems use decade counters for easier programming.



INPUTS				OUTPUTS	
PRESET	CLEAR	CLOCK	D	Q	$\overline{Q}$
0	1	X	X	1	0
1	0	X	X	0	1
0	0	X	X	1*	1*
1	1	↑	1	1	0
1	1	↑	0	0	1

* NOT ALLOWED

Figure 37. A type D flip-flop and its truth table.

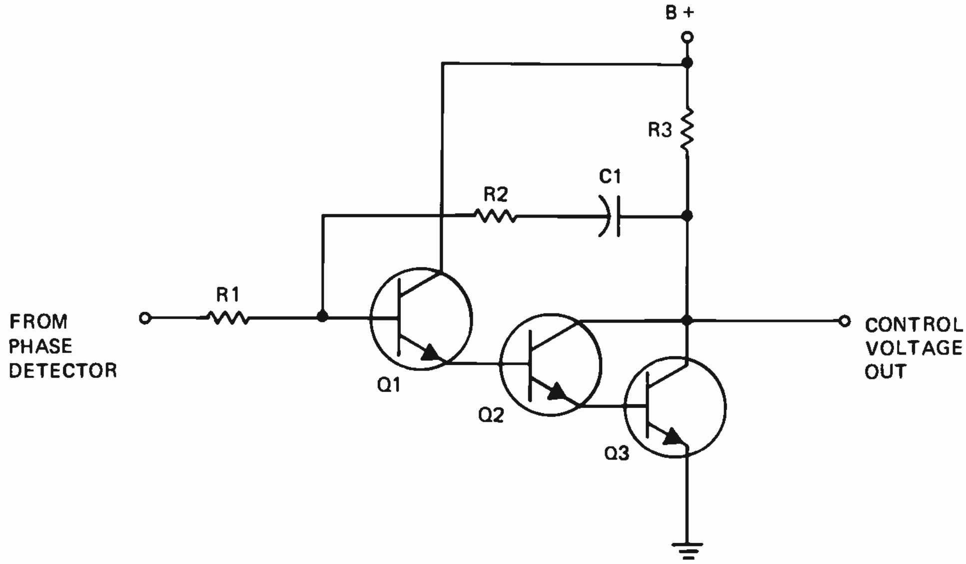


Figure 38. A typical loop filter.

The programmability comes about through additional logic which allows the counter to be preset to any number between 0 and 9. This will cause the counter to count down from the preset number, effectively dividing the incoming number by the preset. The programmable inputs are also masked to allow the counter to divide by 10 or by the preset number.

Phase Detectors and Loop Filters. The next component in the phase-locked loop is the phase detector. You will recall from Fig.33 that the phase detector has two inputs and one output. One input receives a signal from the programmable divider. This signal is compared with the frequency and phase of a signal entering the second input of the phase detector called the reference frequency. If there is a frequency or phase difference between the two signals, a series of pulses appears at the output of the phase detector. The pulses occur at the rate of the reference frequency, but vary in width, or duration. These pulses are fed to the next component in the PLL system, the loop filter.

A schematic diagram of a simple loop filter is shown in Fig.38. The purpose of the filter is to smooth out the pulses from the phase detector until they become essentially a pure dc voltage. This control voltage is then fed to the VCO to control its frequency.

Now that you have seen each of the major components involved in a basic digital phase-locked loop frequency synthesizer, let us see how they work together.

The VCO is responsible for generating the output signal. The programmable divider divides the VCO frequency so that it is within the range of the control system. If the number (N) programmed into the dividers is changed, the phase detector generates a voltage of the proper polarity and amplitude to steer the VCO frequency in the direction that will bring the frequency coming from the dividers back to the reference frequency. Programming is usually accomplished with BCD thumbwheel or rotary switches.

The reference frequency is usually derived from a stable, crystal-controlled oscillator whose frequency is divided down to the desired reference frequency. For example, the reference frequency used in a synthesizer designed for the aircraft band would be 25 kHz, since that is the channel spacing on that band. The channel frequency is displayed, at the control panel of the unit, on the programming switches. The unit would be capable of covering 118-136 MHz in 25 kHz steps for a total of 720 channels. Imagine how many crystals that would require – even for a multiple crystal frequency synthesizer!

Direct Frequency Synthesizer. There are two major types of digital PLL frequency synthesizers. They are generally classified as direct or indirect. The classification refers to the manner in which the output of the synthesizer VCO is used. If the VCO output is fed to a multiplier or mixer before it is actually used, the synthesizer is considered to be indirect. If the output of the VCO is used directly,

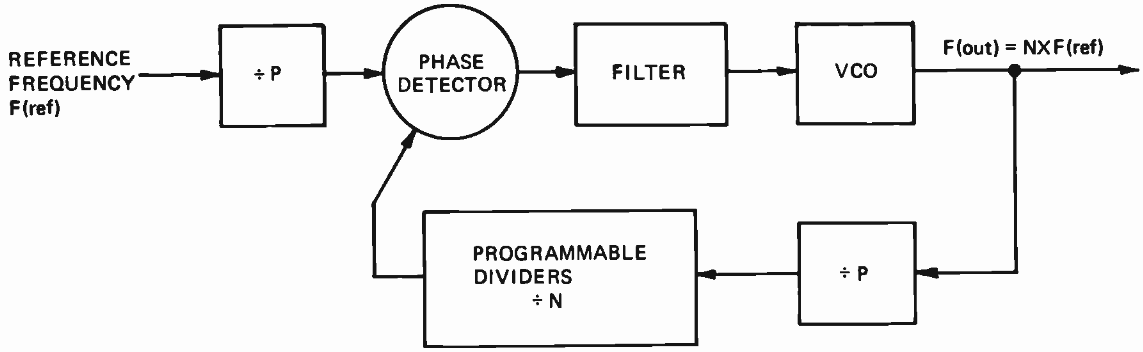


Figure 39. A PLL synthesizer using prescaling.

it is of the direct type. The synthesizer we have been discussing is one of these. The output frequency is equal to the number programmed (N) times the reference frequency [F (ref)].

There are several ways to implement both direct and indirect synthesizers. They may be classified more specifically. We will briefly discuss three more direct synthesizers.

The frequency synthesizer shown in Fig.39 uses a fixed divisor (÷ P) prescaler between the VCO and the programmable dividers. This technique is used to increase the upper frequency limit of the synthesizer. The programmable dividers are not capable of working up into the vhf range. Fixed dividers are readily available up to 1 GHz (1000 MHz).

Unfortunately, the reference frequency must also be divided the same amount to maintain the same frequency steps of the basic direct synthesizer discussed earlier. Since a lower frequency is used as the reference, it is more difficult to filter and slows down the response of the loop to frequency changes.

Another type of direct synthesizer is shown in Fig.40. It is called a dual modulus prescaler PLL. Modulus is just another way of saying divisor. In other words, the prescaler used in this synthesizer is capable of dividing by two different divisors. The divisor is selected by another programmable divider called the A counter.

Divisors of 10 and 11 are frequently used in a dual modulus prescaler. The number programmed into the A counter tells the prescaler how many times it has to divide by 11. This technique allows the synthesizer output to be changed in steps equal to the reference frequency. Its advantage over the prescaler synthesizer previously discussed is that the reference frequency is not divided down to a lower frequency. The loop response is fast and performs just like the basic PLL synthesizer discussed first.

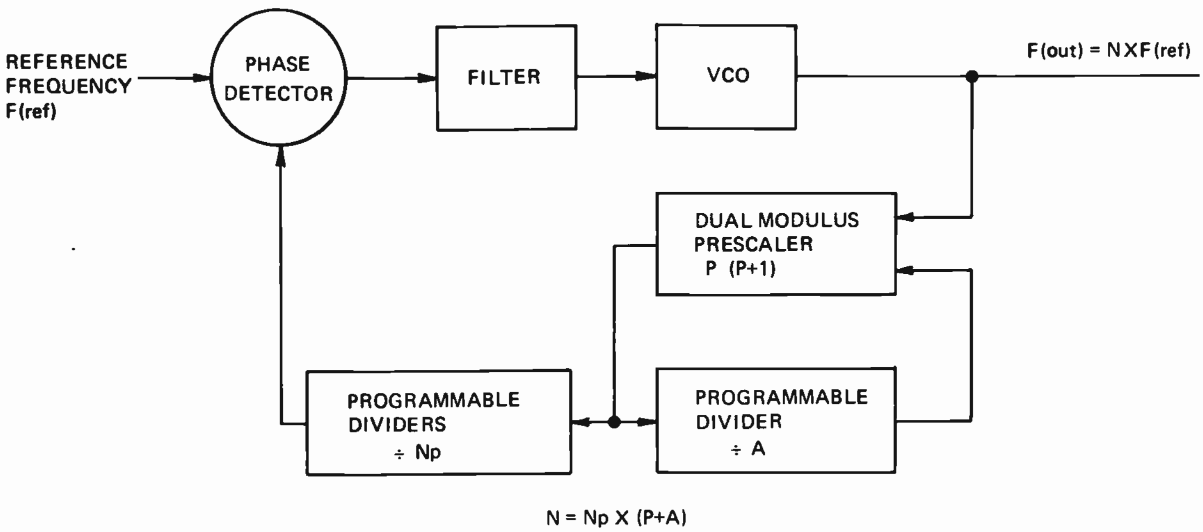


Figure 40. A dual modulus prescaler synthesizer.

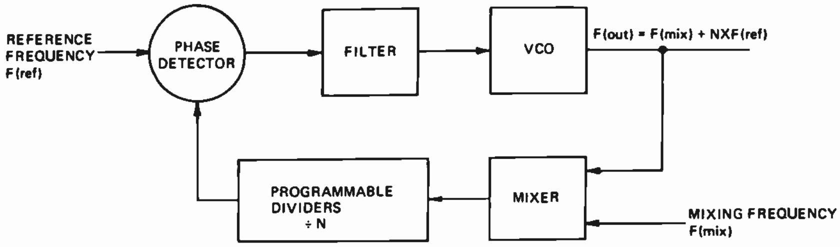


Figure 41. A synthesizer using down mixing.

There is an alternate method of getting the output of a vhf synthesizer down to a frequency the programmable dividers can handle. This involves using a mixer to heterodyne the output of the VCO down to a lower frequency, as shown in Fig.41. The difference frequency is selected by tuned circuits at the output of the mixer.

This method has two disadvantages, both caused by the presence of tuned circuits in the system. If a synthesizer such as this were used commercially, the manufacturer's costs would be increased by the necessity of having to align the tuned circuits. Also, the tuned circuits limit the range over which satisfactory mixing can take place. This method would be unsuitable for a synthesizer covering more than a few megahertz.

Notice that the mixer is in the loop. Therefore, any tendency toward frequency drift or instability in the mixer oscillator would be compensated for by loop action. Nevertheless, the mixer oscillator is usually crystal controlled. This is a third disadvantage over the previous synthesizers, which required only one crystal.

Indirect Synthesizers. Finally, let's take a look at a couple of indirect frequency synthesizers. They represent two more ways in which the requirements of a vhf output from a synthesizer can be achieved while still keeping the signal fed to the programmable dividers within their range of operation.

The block diagram of a multiplier PLL synthesizer is shown in Fig.42. In this circuit, the relatively low frequency output of the VCO is multiplied nine times, up to the desired frequency. The output of the VCO is fed directly to the programmable dividers as it is well within their range of operation. The reference frequency must be divided the same number of times the VCO output is multiplied, if steps in the amount of the reference frequency are to be maintained.

In this respect, this system is like the fixed prescaler PLL. Again, the lower effective reference frequency will slow down the response time of the loop. This system has a couple of additional disadvantages caused by the multipliers. The multipliers include tuned circuits which must be aligned. Further, the Q of the tuned circuits must be high enough for good multiplying action, yet low enough to achieve a reasonable bandwidth.

Another disadvantage of the multiplier type synthesizer is that any subaudible disturbances of the VCO frequency may be multiplied up to the audible range. This means that a cleaner VCO and improved loop filter would be required.

Another type of indirect synthesizer is shown in Fig.43. This synthesizer uses a mixer at its output to beat the VCO frequency up to the band of interest. The unfortunate disadvantage of this technique is that the mixer is not in the loop. Therefore, a highly stable heterodyne oscillator must be used in addition to the stable reference frequency oscillator. Even then, the stability and accuracy of the output signal will depend upon the worst case conditions of both oscillators.

There are probably many more ways in which a digital phase-locked loop frequency synthesizer

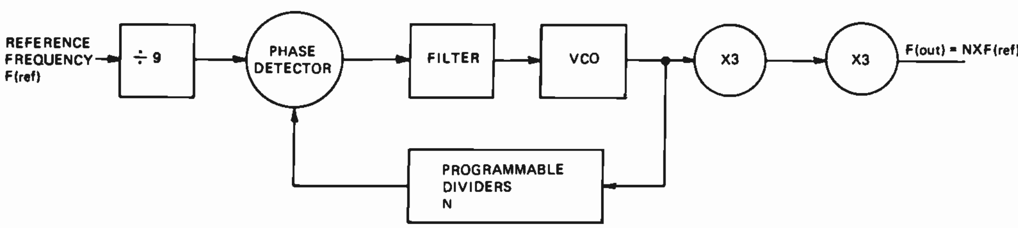


Figure 42. A multiplier PLL synthesizer.

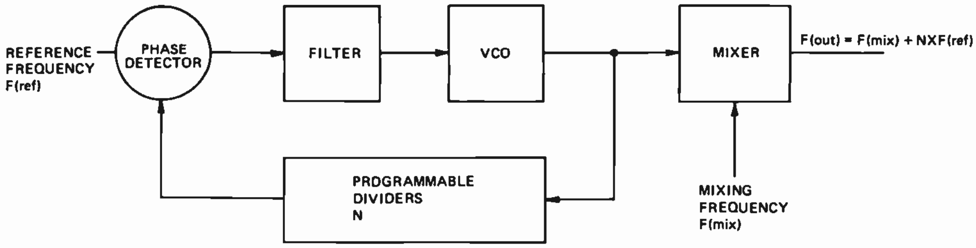


Figure 43. A synthesizer using UP mixing.

could be implemented. We have shown several proven methods for accomplishing this task and discussed the advantages and disadvantages of each. While we have not been able to give you an in-depth technical discussion, because it would be beyond the scope of this lesson, we have presented sufficient information to allow you to feel comfortable in working with communications equipment containing a digital PLL synthesizer.

SELF-TEST QUESTIONS

20 What effect would doubling the value of C1 and C2 in Fig.36 have on the output frequency?

21 What is D1 in Fig.34 called?

ANSWERS TO SELF-TEST QUESTIONS

1 Regenerative or positive feedback. If positive feedback is not present, oscillations will be damped.

2 Decrease. Frequency is equal to:

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

Therefore, if inductance increases, the frequency will decrease.

3 There will be more cycles in the wave train. Losses in the coil of a high Q circuit are low so more cycles occur before the wave train is damped.

4 0.0000005 second or 0.5 microsecond. The formula is:

$$P = \frac{1}{f} = \frac{1}{2{,}000{,}000}$$

$$= 0.0000005 \text{ second}$$

5 They are in-phase. The voltage developed across L2 is the feedback voltage and must aid plate current.

6 The bias voltage developed across R1. If the oscillator output increased, C2 would be charged to a higher potential on positive peaks and would produce a larger bias voltage across R1. The larger bias would reduce the amplitude of the oscillator output.

7 1. Oscillator remains energized at all times.
2. Oscillator coil and capacitor are placed in an oven.
3. Temperature compensating capacitors are placed across the tank.

8 A swamping capacitor is placed across collector-to-emitter junction.

9 R1 and C2 develop the negative bias. L2 develops the signal.

10 It goes positive. The reactance of C2 is small at the resonant frequency of the tank; therefore, the positive voltage induced in the tank by L2 (voltage across C1) is coupled to the grid.

11 C1 develops the feedback voltage. It is applied to the base through C3 and across R1.

12 The ultra-audion oscillator is a modification of the Colpitts circuit.

13 Through the RC phase shift network in the grid. Each pair of RC components shifts the phase a definite number of degrees. Total shift through all stages must equal 180°.

14 R2 and C2 develop the feedback voltage.

15 To prevent damage to the crystal.

16 Frequency is controlled by the crystal, Y1. At the resonant frequency, Y1 offers minimum impedance to the feedback voltage. At other frequencies, impedance increases.

17 A higher frequency can be obtained without the use of frequency multipliers.

18 1380 kHz.

19 The tri-tet oscillator functions as an oscillator and a frequency multiplier.

20 The output frequency would be halved.

21 A varactor diode.

Lesson Questions

Be sure to mark your lesson card as shown to indicate Lesson CC205. Most students want to know their grades as soon as possible, so they mail their lesson card in immediately. Others, knowing they will finish another lesson within a few days, send in two cards at once. Either practice is acceptable. However, don't hold your answers too long; you may lose them. Don't hold answers to more than two sets of lessons at one time, or you will run out of lessons before new ones can reach you.

1. The time it takes a resonant circuit to go through one complete cycle is:
   a. Its period.
   b. Its frequency.
   c. Its Q.
   d. 0.001 μs.

2. The tube or transistor used in an rf oscillator acts effectively as:
   a. An electronic switch.
   b. A damper.
   c. A constant voltage source.
   d. A modulator.

3. If the operating voltage of an oscillator tube is changed so that the plate resistance of the tube increases, what will happen to the operating frequency?
   a. Remains the same.
   b. Cannot tell.
   c. Increases.
   d. Decreases.

4. The quartz crystal cut which shows the greatest stability with temperature variations is:
   a. AT.
   b. BT.
   c. CT.
   d. GT.

5. When adjusting the capacitance of the tank circuit in a crystal oscillator, what happens to the plate current as oscillations start?
   a. It rises rapidly.
   b. It starts to decrease.
   c. Nothing.
   d. It drops sharply.

6. The temperature coefficient of an X-cut crystal is -20 Hz/°C/MHz. What will its oscillating frequency be at 65°C if its frequency is 5250 kHz at 50°C?
   a. 5251.575 kHz.
   b. 5248.323 kHz.
   c. 5248.425 kHz.
   d. 5250.157 kHz.

7. In Fig.11(B), the ratio of what two parts determines the amount of feedback applied to the base?
   a. R1 to R3.
   b. C3 to L1.
   c. C1 to C2.
   d. C3 to R1.

8. For what part of the cycle does plate or collector current flow in an LC oscillator circuit?
   a. Less than one-half cycle.
   b. One-half cycle.
   c. Three-quarters of a cycle.
   d. One full cycle.

9. An LC oscillator whose frequency can be varied by changing a control voltage is a:
   a. VCM.
   b. Hartley oscillator.
   c. VCO.
   d. Colpitts oscillator.

10. The four major circuits in a basic phase-locked loop (PLL) are the programmable divider, loop filter, and:
    a. Frequency modulator, VCO.
    b. Phase detector, VCO.
    c. Lock detector, VCO.
    d. Phase detector, multiplier.

CC206

Amplitude Modulation and Demodulation

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Introduction

The rf signal that is amplified by the class C power amplifier stages in a transmitter can be fed to an antenna and transmitted through space to the receiving location. This signal by itself, however, is of little value. It has a single, constant frequency and a constant amplitude. Since the signal does not change in any way it cannot transmit intelligence from one point to another; but by adding an intelligence signal that changes the rf signal in some way the signal can be used to carry intelligence such as code, sound, or TV. Any process that changes or varies the rf signal so that it carries intelligence is called modulation. The intelligence signal is called the modulating signal, and the rf signal is called the carrier.

At the receiver, the modulated rf signal is amplified, and then the intelligence signal is removed from the carrier. This process is called demodulation.

There are several ways of adding intelligence to or "modulating" a radio frequency carrier. Perhaps the simplest form of modulation is code modulation, shown in Fig.1. With code modulation, messages are sent by switching the carrier on and off at prescribed rates and durations. The most common method of code transmission uses the International Morse Code, in which dots and dashes represent letters and numerals.

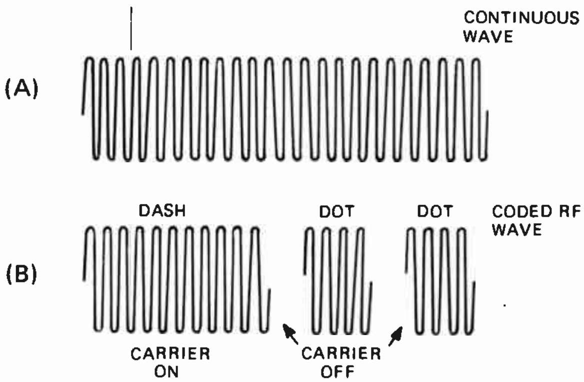

Figure 1. A code-modulated wave.

In more complex forms of modulation, sound or picture information is used to vary the amplitude, the frequency, or the phase of the carrier signal. These three forms of modulation are compared in Fig.2. The rf carrier signal is shown in Fig.2(A) and the intelligence to be modulated on it is shown in Fig.2(B). We have used a sine wave merely for convenience; the intelligence signal actually transmitted by a radio or TV station has a very complex waveform.

In amplitude modulation (abbreviated AM) the modulating signal causes the carrier amplitude to vary in accordance with the strength of the audio signal as shown in Fig.2(C). Notice that a steady carrier is transmitted when the modulating signal is zero. When the modulating signal is applied, it causes the amplitude of the carrier to vary in step

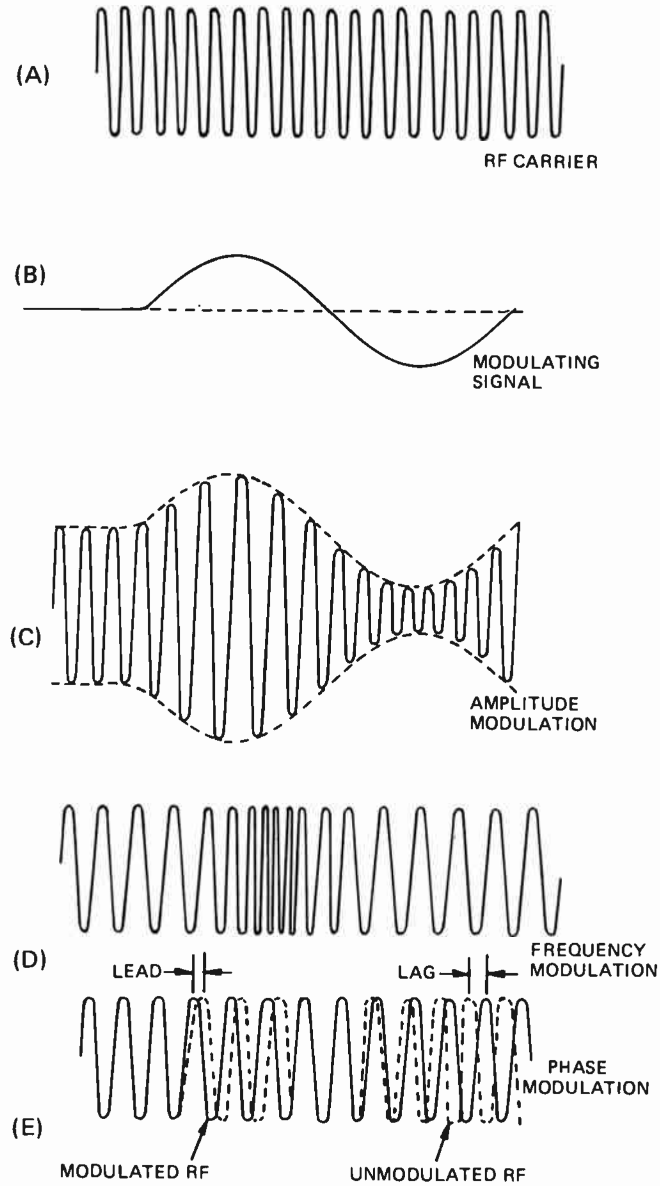

Figure 2. An rf signal can be modulated by varying its amplitude, frequency, or phase.

with the strength or amplitude of the modulating signal. In other words, the radio frequency output exactly follows the variations of the modulating signal.

The frequency of the carrier can also be made to vary in accordance with the strength of the modulating signal to produce frequency modulation (abbreviated FM) as shown in Fig.2(D). The amount of carrier frequency variation increases as the amplitude of the modulating signal increases, and decreases as the amplitude of the modulating signal decreases. For phase modulation (abbreviated PM), shown in Fig.2(E), the modulating signal causes the phase of the carrier, or the time interval between peaks of the carrier cycles, to vary according to the amplitude sine wave signal. The unmodulated rf carrier leads the modulated carrier at the peak of the sine wave signal, but lags the modulated carrier during the troughs. In both frequency modulation and phase modulation, the amplitude of the carrier remains constant; the intelligence to be conveyed causes only the frequency or phase of the carrier to vary.

Frequency modulation and phase modulation will be taken up later in the course. In this lesson, you will learn how the intelligence signal amplitude modulates the carrier, and study the circuits in which amplitude modulation takes place. After you have studied AM systems, you will learn how the modulated signal is demodulated at the receiver.

The Amplitude-Modulated Signal

You have learned that an amplitude-modulated signal is made up of two parts: one is a constant-amplitude, constant-frequency rf signal called the "carrier," and the other is the intelligence signal – either audio or TV picture information. When the two signals are combined the strength of the carrier is made to vary according to the amplitude of the modulating signal.

Before we study the modulated wave itself, let us find out how the two signals are combined.

In a radio transmitter, the modulating or combining of the two signals is done in a power amplifier stage similar to those you have already studied. You will remember that signal amplification in a class C stage is nonlinear. Only a part of the positive alternation of the input signal causes plate current to flow. Thus, a change in the signal on the grid does not cause a corresponding change in the plate current. When you feed the audio and rf carrier signals into a class C stage, the two will beat together or "heterodyne." The two signal components add and subtract from each other to produce the waveform shape shown in Fig.2(C) and the output.

The overall outline, shown by the dashed line in Fig.2(C), is called the "envelope." Notice that each half of the modulation envelope has the same shape as the modulating audio signal. When the amplitude of the modulating audio signal increases, the amplitude of the carrier signal increases. Likewise, when the amplitude of the audio decreases, the amplitude of the carrier decreases. The shape of the audio modulating signal, however, is not changed; it is actually superimposed on the carrier and causes the amplitude of the carrier to vary.

MODULATION PERCENTAGE

The strength of the audio modulating signal applied to the class C stage determines the amount that the carrier signal will vary. Figure 3 shows an unmodulated carrier signal and several examples of various degrees of modulation. In Fig.3(B), the audio signal increases the carrier 50% above its unmodulated value (E) during the positive modulating alternation, and decreases the carrier 50% below its unmodulated value on the negative modulating alternation. The maximum and minimum values are labeled E_{MAX} and E_{MIN} in the illustration.

Now suppose the audio signal is strong enough to modulate the carrier completely; that is, it is able to drive the carrier to zero on the negative modulation alternation. The carrier is then said to be modulated 100%, as shown in Fig.3(C). At the peak of the modulating signal cycle the carrier amplitude is twice its unmodulated value, as shown in Fig.2(E). This condition represents the maximum amount the carrier can be modulated without seriously distorting the desired audio information

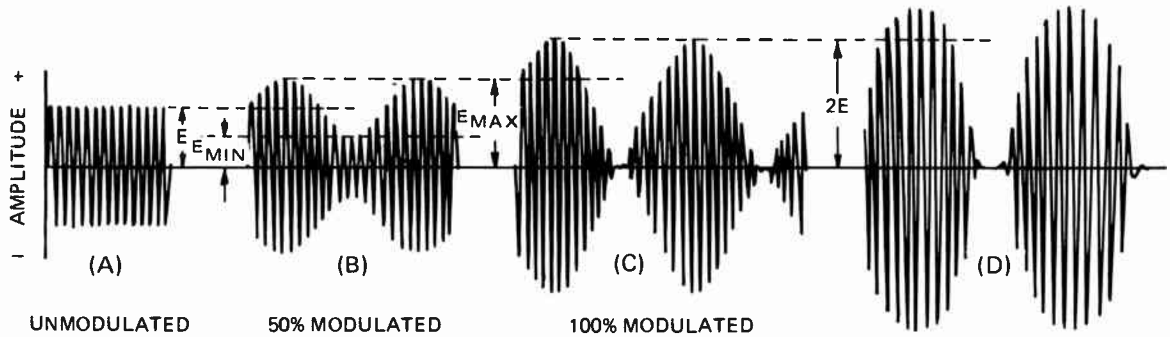

Figure 3. An unmodulated carrier is shown at (A), and the same carrier with different degrees of modulation is shown at (B), (C), and (D).

and generating strong harmonics of the carrier signal.

The result of overmodulating the carrier by applying an excess amount of audio signal is shown in Fig.3(D). Here the audio has driven the carrier to zero for a large part of the negative modulating alternation. Thus, the envelope shape is no longer exactly the same as the audio modulating signal. The largest amount that the carrier can be modulated without introducing distortion, therefore, is 100%.

The modulation percentage is important because it determines the amount of power contained in the modulated signal, and thus the amount of audio signal present at the output of the audio detector in a receiver. The higher the percentage, up to 100%, the higher the output. Let us see how we can compute the percentage of modulation on the carrier.

If we know the value of the unmodulated carrier [E in Fig.3(A)] and the maximum value, E_{MAX}, at the modulation peak, we can use the following formula: percentage modulation equals

$$\frac{E_{MAX} - E}{E} \times 100$$

For example, suppose the value of E_{MAX} is 400 volts and the value of E is 200 volts: percentage modulation equals

$$\frac{400 - 200}{200} \times 100 = 100\%$$

A factor that is often used to indicate how much the carrier is modulated is the degree of modulation (m). This is expressed as a decimal instead of a percentage.

$$\text{(Degree of Modulation) } m = \frac{E_{MAX} - E}{E}$$

Notice that the formula is similar to the one given for percentage modulation. Thus for 100% modulation, the value of m is 1; for 50% modulation, the value of m is 0.5, etc.

This method of computing modulation percentage is true only if the modulating signal is a sine wave. Complex sound and picture (TV) waveforms will not have the same modulation percentage over a complete cycle. The modulation of speech, music, or image signals will reach 100% only on the high-amplitude peaks. However, even the most complex sound is made up of sine wave signals. Since it is easier to visualize the modulation process with sine wave signals, we will continue to use them in our discussion.

HOW SIDEBANDS ARE FORMED

At the beginning of this section we said that the carrier and the audio-modulating signals beat together or heterodyne in a class C amplifier stage to produce the modulated waveform. Whenever two signals beat together in a nonlinear device, such as a class C amplifier, two new signal frequencies are produced. One of these signal frequencies is equal to the sum of the two frequencies beating together and the other is equal to the difference of the two frequencies beating together. Thus the heterodyning process that occurs in a class C amplifier will produce signal components that are the sum of the audio and carrier frequencies and the difference between the audio and carrier frequencies. The sum and difference frequencies are located above and below the carrier frequency and are called the "sidebands."

To illustrate what these sidebands are, let us again use a pure sine wave as the modulating signal. Suppose the carrier frequency is 1000 kHz and the modulating signal is 5000 Hz (5 kHz). When these signals are heterodyned in a properly-designed and adjusted class C amplifier, two new signals will be produced, one equal to the sum of the two signal frequencies which is 1000 + 5 = 1005 kHz, and the other equal to the difference between the two signal frequencies which will be 1000 - 5 = 995 kHz. The signal at the output of the modulated class C amplifier then contains three frequencies: the original carrier frequency of 1000 kHz, an upper sideband frequency of 1005 kHz, and a lower sideband of 995 kHz. These upper and lower sideband frequencies are very important because they contain all the intelligence that is in the modulated wave. The carrier signal contains none of the intelligence and can actually be eliminated as is done in some special transmitting systems. However, in standard AM transmission it is transmitted along with the sidebands and used in the detector to extract the original modulating information. You will see how this is done later.

The strength of the upper and lower sidebands depends on the strength of the modulating sine wave signal. The higher the percentage of modulation, the greater the amplitude of the two sidebands. With 100% modulation, the amplitude of each sideband is equal to one-half the amplitude of the unmodulated carrier, a ratio of 2 to 1, as shown in Fig.4. Again E is the amplitude of the

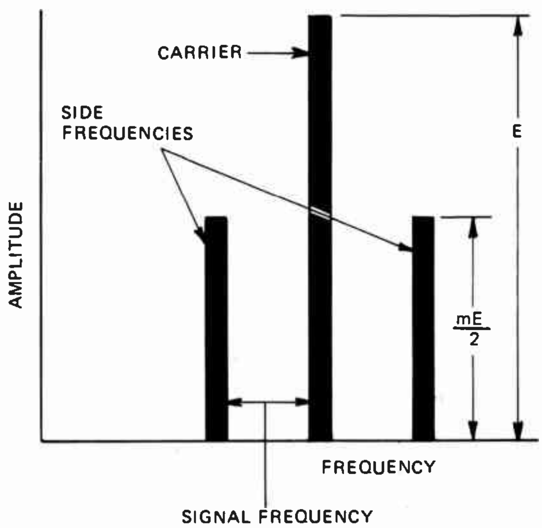

Figure 4. Relationship between carrier and side frequencies at 100% modulation with a pure sine wave signal.

unmodulated carrier. The amplitude of each sideband can be computed by multiplying the values of m and E, and then dividing the product by 2. Of course, as the percentage modulation decreases, the amplitude of the sidebands also decreases.

Bandwidth. The frequencies of the upper and lower sidebands determine the bandwidth, or the space the transmitted signal occupies in the broadcast, i.e., whether the signal falls in the vhf or uhf band. As shown in Fig.5, the higher the highest frequency audio components to be transmitted, the greater the required bandwidth. To transmit a 5000 Hz (5 kHz) sine wave tone, the bandwidth must be 10 kHz; an 8000 Hz tone requires a 16 kHz bandwidth.

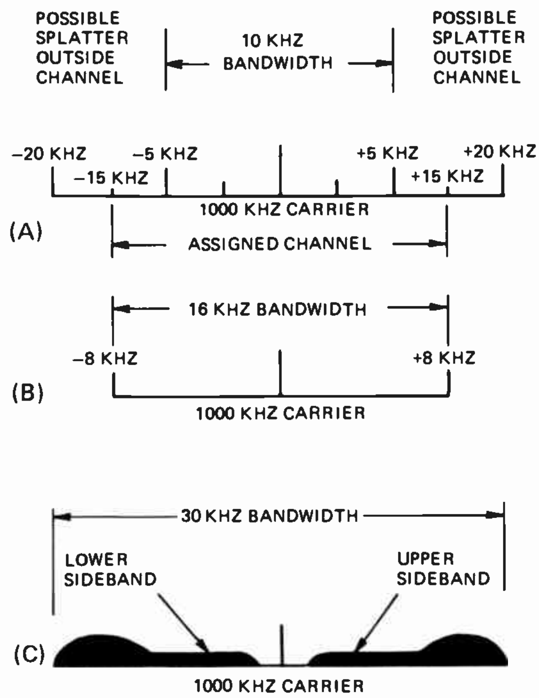

Figure 5. (A) A 1000 Hz carrier modulated with a 5 kHz wave requires a 10 kHz bandwidth; (B) an 8 kHz modulating signal requires a 16 kHz bandwidth. (C) The sidebands for a complex modulating signal.

The waveform of a complex signal is made up of one or more fundamental frequencies and many harmonics and overtones. Therefore, when this kind of signal is used to modulate the carrier, bands of frequencies will be produced above and below the carrier, as shown in Fig.5(C). Most of the standard AM broadcast stations are assigned a bandwidth or channel 30 kHz wide, 15 kHz above and 15 kHz below the carrier. TV stations require a much wider band.

Broadcast stations are compelled by law to confine their radiation within the assigned channels. An AM broadcast station, for example, must not broadcast sideband frequencies or spurious radiations outside its channel. For example, a 5000 Hz modulated sine wave can produce spurious sidebands as much as 10 kHz, 15 kHz, or 20 kHz above and below the carrier frequency. Some of the spurious or "splatter" signals shown in Fig.5(A) will be outside the assigned channel. There is usually a close space between broadcast stations in the frequency band. Thus, the undesirable radiation could interfere with stations transmitting on adjacent channels. However, no undesired radiation will be transmitted if the transmitter is properly designed and adjusted.

SIDEBAND POWER

When the carrier is modulated in a class C stage, the total power contained in the modulated wave at the output is the sum of the carrier power and the sideband power. The sidebands, as you have learned, contain all the intelligence. It is important for the power in the sidebands to be as high as possible to get a high signal-to-noise ratio. Let us see how the voltage, current, and power output of the class C stage varies during modulation.

With 100% modulation, at the positive peak of the modulating cycle, the modulation envelope of the carrier is twice the amplitude of the unmodulated carrier; at the negative peak of the modulating cycle, the modulation envelope is at zero. The average amplitude over the entire cycle, however, is equal to the amplitude of the unmodulated carrier.

Similarly, when the modulating signal is supplied to a class C stage, although the effective value of the power input to the stage increases and decreases with the modulation, the average dc plate power input, as indicated by the panel meters, does not change. Each increase in plate voltage and plate current is followed by an equal and opposite decrease on the next half-cycle of the modulating signal. Thus, in a properly adjusted and operated transmitter, the dc plate current is always constant, with or without modulation.

Now let's see what happens to the power output of a class C stage when a modulating signal is applied to produce 100% modulation.

As you learned, power varies in accordance with the square of the current, $P = I^2 R$. Therefore, since the plate current on the positive peak of the modulated carrier is twice what it is for the unmodulated carrier, the power will be four times as great on the positive peak, and the average power will increase when modulation is applied.

With 100% sine wave modulation, the amplitude of each sideband is half the amplitude of the carrier. Therefore, the power in the carrier is four times the power in each sideband, or in other words, power in both sidebands is one-half the power in the carrier.

For example, if the plate input power for a stage that is unmodulated is 100 watts, then 50 watts of sine wave audio power must be added for 100% modulation. The total average input power to the transmitter, therefore, will be 150 watts. One-third of the total average power is in the sidebands, and two-thirds is in the carrier. Thus, when 100% modulation is applied to the stage, the total power output will increase 50%.

As the degree of modulation decreases, the sideband power also decreases. This is shown by curve 1 in Fig.6. At 100% modulation ($m = 1$), each sideband contains 25% of the carrier power; at 50% modulation ($m = 0.5$), each sideband contains 6% of the carrier power; and at 20% modulation ($m = 0.2$), each sideband contains only 1% of the carrier power.

Suppose the carrier power is 100 watts. As you learned earlier, 50 watts of modulating power must

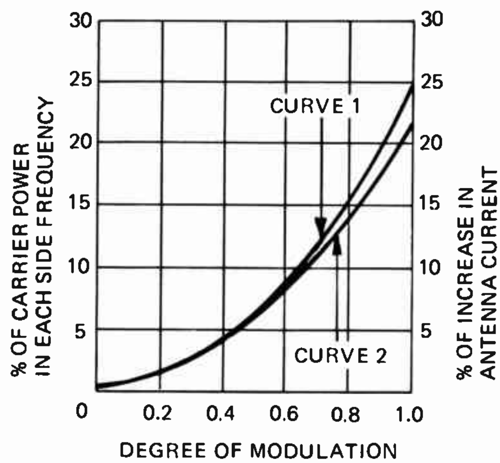

Figure 6. The effects of various degrees of modulation on the antenna current (curve 2) and on the power in each side frequency (curve 1).

be added to modulate the carrier 100%. This can be computed in the following way, where P_{sb} is the sideband power, P_c is the carrier power, and m is the degree of modulation:

$$P_{sb} = \frac{m^2}{2} \times P_c$$

$$= \frac{1^2}{2} \times 100 = 50 \text{ watts}$$

For 50% modulation, the sideband power would be only 12.5 watts.

$$P_{sb} = \frac{0.5^2}{2} \times 100 = 12.5 \text{ watts}$$

Thus, the power in the sidebands has decreased 75% because 12.5 watts is one-quarter of the sideband power at 100% modulation (50 watts).

It is the power in the sidebands alone that determines the amount of useful signal at the receiver, because only the sidebands contain the intelligence. Thus, you can see that it is very important for the carrier to be modulated as near 100% as possible to get the greatest amount of power in the sidebands.

When a complex waveform is transmitted, such as speech or music, the relationship between the power in the carrier and the power in the sidebands is not the same. The waveforms do not usually contain as much average power as is contained in the sidebands for a sine wave signal. In fact, the average power for speech is about one-half that of a sine wave. Although the average power in the sideband for speech modulation is less than with sine wave modulation, the peak of the modulating audio signal must be the same for 100% modulation.

ANTENNA CURRENT

When the carrier is modulated by a sine wave, the additional power in the sidebands causes the total power output of the transmitter to increase. The current meters in the antenna circuits will indicate an increase in the antenna current.

Curve 2 of Fig.6 shows the percentage of increase in antenna current for different degrees of modulation. For 100% sine wave modulation, the antenna current will increase 22.5% over the unmodulated carrier value. Again this is for sine wave modulation; the percentage of increase in antenna current is less for speech or music modulation, usually from one-quarter to one-half the sine wave value.

We can also figure the increase in antenna current by rearranging the formula for power, $P = I^2 R$, as follows:

$$I = \sqrt{\frac{P}{R}}$$

where I is the antenna current with an unmodulated carrier.

For example, suppose a transmitter with no modulation is supplying 100 watts to a 100-ohm antenna system. We will have:

$$I = \sqrt{\frac{100}{100}} = 1 \text{ amp}$$

Now, if 100% modulation is added, P becomes 150 watts, and using I_1 for the antenna current with a modulated carrier, we have:

$$I_1 = \sqrt{\frac{150}{100}} = \sqrt{1.5} = 1.225 \text{ amps}$$

To find the % of increase, you use the formula:

$$\% = \frac{I_1 - I}{I} \times 100$$

$$= \frac{1.225 - 1}{1} \times 100$$

$$= \frac{0.225}{1} \times 100$$

$$= 22.5\%$$

We can also calculate the antenna current for various degrees of modulation by using the formula:

$$I_1 = \sqrt{1 + \frac{m^2}{2}} \times I_c$$

As you have learned, the modulation factor, m, is 1 for 100% modulation, 0.5 for 50%, 0.2 for 20%, etc. Using the same problem we have:

$$I_1 = \sqrt{1 + \frac{m^2}{2}} \times 1$$

$$= \sqrt{1 + 0.5} \times 1$$

$$= 1.225 \times 1$$

$$= 1.225 \text{ amps}$$

As you can see this is the same figure you got using the other formula.

The percentage of increase in antenna current with sine wave modulation is a pretty good indication of the percentage of modulation. If the antenna current increases only 5%, you know you are nowhere near 100% modulation. If the increase is 20% you know you are close to 100% modulation. Final adjustments to get 100% modulation, however, are usually performed using indications other than the antenna current.

MODULATION DISTORTION

Any change in the modulation waveform that is not directly due to the modulating signal will introduce distortion. If the modulation signal is distorted, the sound output from a radio or the image on a TV screen will also be distorted. A distorted modulated signal can also cause spurious signal components to be radiated outside the assigned channel. Spurious radiations represent wasted power which might otherwise be used as part of the desired signal energy. Improper modulation also lowers the efficiency of the transmitter. Let us look at some possible causes of modulation distortion.

Sideband Clipping. One source of distortion is the loss of useful sideband power when high-frequency audio information is to be transmitted. The loss of sideband energy at high modulating frequencies is often the result of insufficient bandwidth in the modulator and associated audio amplifier circuits. Poor quality audio transformers and modulation transformers can cause both high-frequency and very low-frequency losses.

The Q of the tank circuit is also important to obtain the proper bandwidth. If the Q is too high, the circuit will become too selective, and some of the useful sidebands may be cut off or clipped. Too low a Q circuit is also undesirable. A low Q circuit is difficult to load properly. This will also influence the bandwidth. Likewise, proper matching of the transmission line system to the antenna is an important consideration. Bandwidth is especially important when it is necessary to convey the very wide band of frequencies required in the transmission of a television signal.

Phase Shift. An improper phase shift in the audio amplifier, modulator, or modulated amplifier can cause a particular type of audio distortion. Phase shift causes a nonuniform time delay of various audio components. For example, let us assume that the crest of a 200 Hz sine wave and a 10,000 Hz sine wave occur at the same instant during the pickup of a particular program.

This is shown by the waveform in Fig.7(A). No attempt has been made in the diagram to show the exact frequency ratio between the two waveforms. If phase shift is present in the transmitting signal, the peak of the 10,000 Hz tone as reproduced at the receiver loudspeaker will be time-displaced with respect to the reproduction of the peak of the 200 Hz tone. In other words, the 10,000 Hz tone will reach its crest after the crest of the 200 Hz tone as shown in Fig.7(B). The sound from the loudspeaker will be blurred.

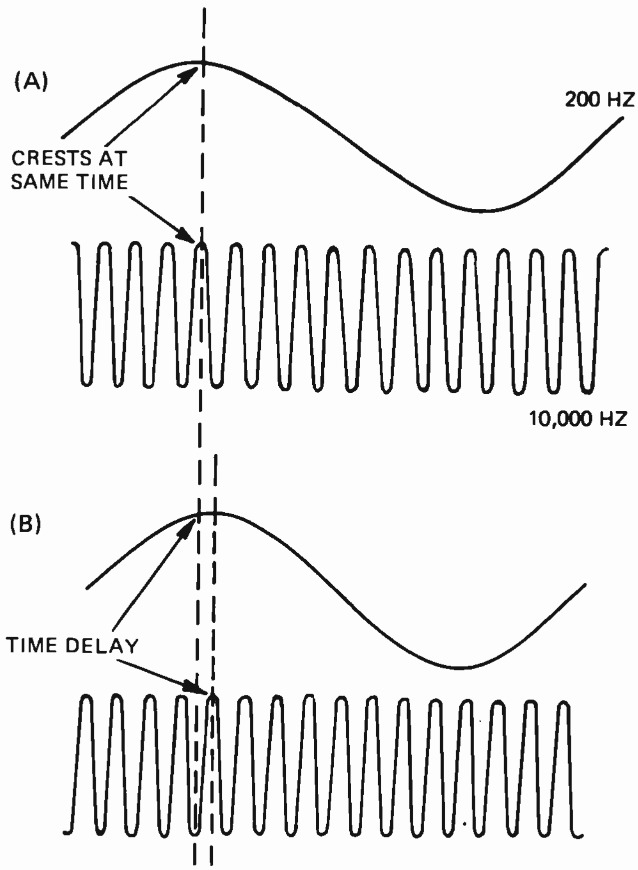

Figure 7. Phase shift between two sine waves in a transmitting system.

Phase shift in audio systems can be caused by inadequate amplifier frequency response, mismatches, incorrectly designed filters and compensators, and improper bandwidth of the modulated class C amplifier or antenna system. The problems of phase shift are also magnified in television modulating systems, and especially in the transmission of a color picture. This type of distortion is difficult to correct; it can be prevented only by proper design of the signal amplifiers and modulators.

Overmodulation. As you know, overmodulation occurs when the transmitter modulation is over 100%. When this happens, one side of the modulated signal becomes compressed. This not only causes distorted audio output at the receiver, but also produces sidebands that can interfere with stations on adjacent channels.

When a standard AM broadcast band transmitter is modulated correctly, the reading on the dc plate current meter will not vary. However, if the transmitter is overmodulated, the reading on the meter will vary, instead of remaining constant as it should. The way to remedy overmodulation is to decrease the degree of modulation. However, in most transmitters, there are circuits that are designed to limit the peak amplitude of the modulating signal and thus prevent overmodulation.

Carrier Shift. Another type of distortion is called carrier shift. With this type of distortion, the modulated signal is asymmetrical. In other words, the amplitude of the positive and negative modulation peaks are not the same. Carrier shift can be either positive or negative as shown in Fig.8(C) and (D). Let us see what happens in each case.

As you have learned, in a class C stage with 100% or less modulation, the average plate current is the same as with no modulation. However, if the modulation is not symmetrical, the average plate current will vary. If the negative part of the modulation is greater than the positive as shown in Fig.8(C), there is a decrease in the average power output and a decrease in the plate current drawn by the stage. This is called negative carrier shift. It is quite easily detected by watching the plate current meter on the modulated stage. As the stage is modulated, the meter will kick down slightly.

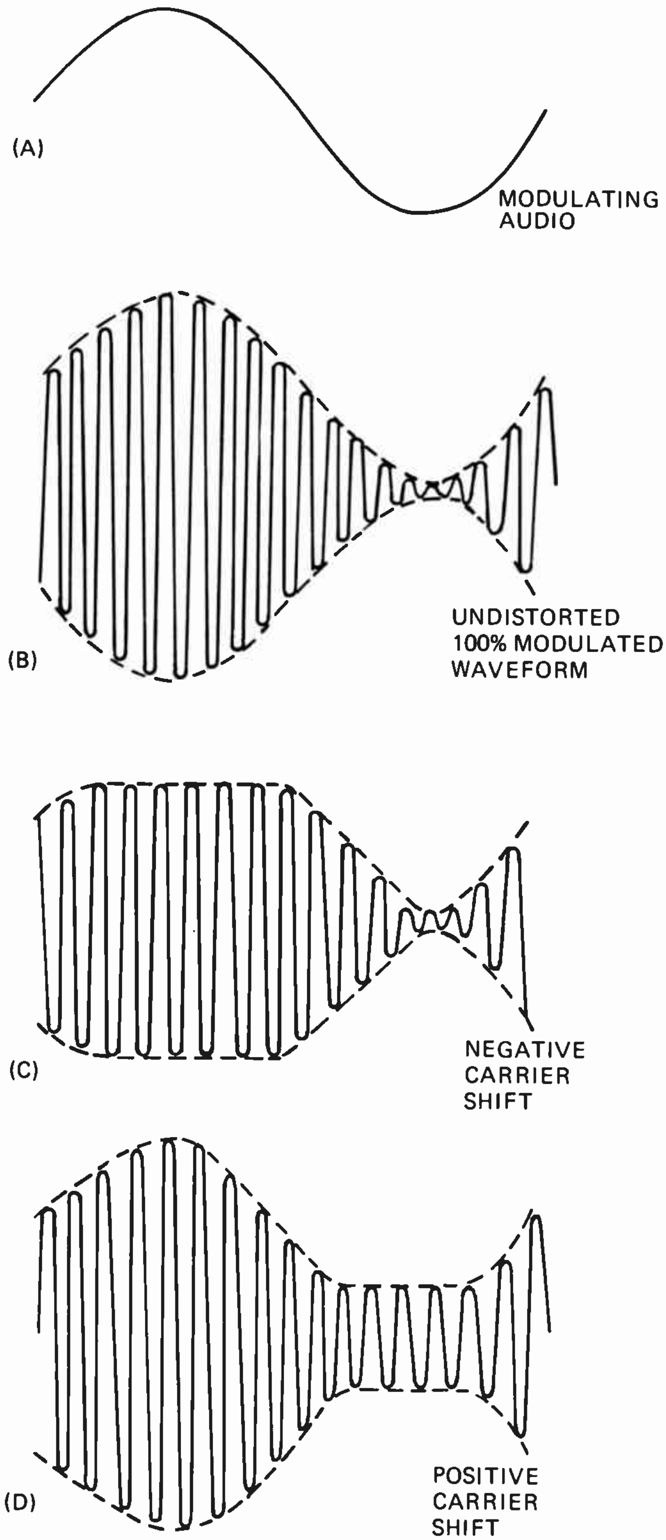

Figure 8. Result of carrier shift on modulation envelopes.

If the positive part of the modulation is greater than the negative, as shown in Fig.8(D), there is an increase in the average power output, and an increase in the plate current; this is called positive carrier shift. It can be detected by an upward kick in the plate current meter reading as the carrier is modulated.

Negative carrier shift may be due to a distorted modulating wave from the modulator and its audio system; overmodulation; improper matching between modulator and modulated stage; an improperly tuned class C stage; insufficient radio frequency excitation to the modulated stage; or poor power supply regulation of the modulator or modulated stage. Positive carrier shift is most often caused by parasitic oscillations, incorrect neutralization, or incorrect bias. It can also be the result of improper tuning or loading of a class C stage, or distortion of the modulating signal.

Almost any type of asymmetrical modulation can cause carrier shift. If the asymmetrical modulation is the result of an asymmetrical modulating signal, the defect is in the modulator or its associated audio amplifier chain. The asymmetrical modulation can also be caused by a defect that arises in the actual modulation process, such as parasitic oscillations, incomplete neutralization, or the inability of the power output of the class C stage to follow the modulating wave. In other words, incorrect tuning, insufficient excitation, incorrect bias, or inability of the power supply to furnish the necessary current on modulation peaks can cause distorted modulation even though the applied modulating signal is not distorted.

It is quite important that you remember what carrier shift is, and know how to identify both positive and negative carrier shift. You should also know the causes of each type so that when you run into it you will be able to remedy it. Also, you can be pretty sure when you go for your FCC License Examination that there will be one or more questions on carrier shift.

AMPLITUDE-MODULATION SYSTEMS

In an AM transmitter, the carrier signal is generated in the master oscillator and amplified class C amplifier stages. These stages are shown in Fig.9. The audio signal from a microphone or other signal source is amplified in the audio amplifier stages and the modulator, and it is then applied to the class C power amplifier. This is the stage in which the mixing or heterodyning of the two modulation signal components (the carrier and the audio) takes place.

In low-power transmitters, the plate of the final or power output stage is usually modulated, and the modulated signal is fed directly to the transmitting antenna. This is called "high-level" modulation. More powerful transmitters have additional power amplifiers following the modulated stage. This is called "low-level" modulation.

When the transmitter is used for voice, music, or TV signals, the following amplifier must be operated either in class A or class B (usually class B as in Fig.9) because it gives a higher power output. Sometimes you will find beam power tetrodes operated in class AB in this type of stage. A class C stage cannot be used; it will distort the modulated-signal waveform. An exception to this is the system that combines phase modulation and amplitude modulation. You will learn about it in a later lesson.

The primary requirement of the modulator stage is to produce an undistorted output signal at a high enough power level to modulate the carrier properly. The modulator stage in a transmitter is basically a high-powered audio amplifier, operated in class A or class B. You have studied audio

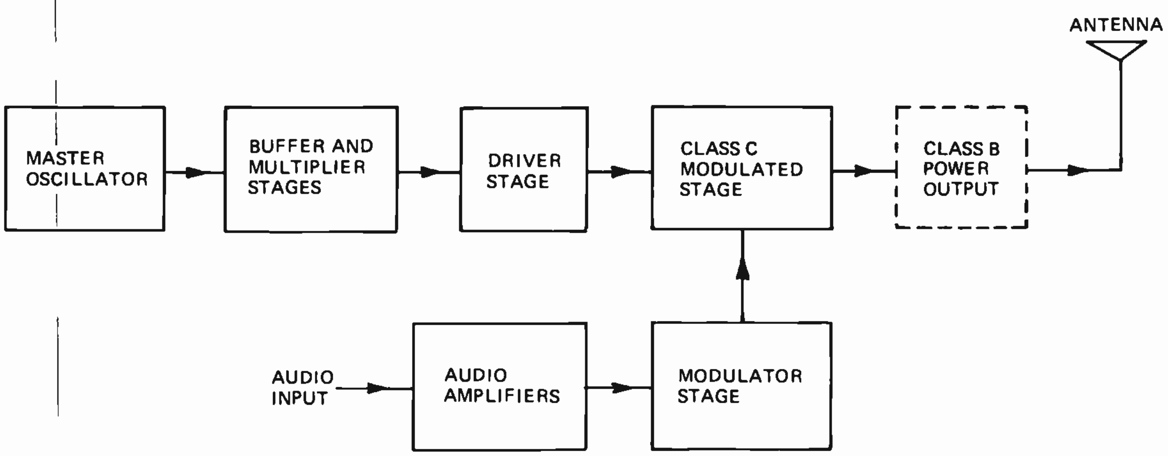

Figure 9. Block diagram of a typical amplitude-modulated transmitter.

amplifiers earlier, so we will not give a complete discussion of this circuit here. We will be concerned with the operation of the modulated class C stage and the methods used to modulate the rf carrier signals.

There are a number of ways that you can amplitude modulate a class C stage. Two of these are: plate modulation when a vacuum tube is used, and collector modulation when a transistor is used. We will discuss these two types in the next two sections of the lesson and then other types of modulation that you might encounter in a following section.

SELF-TEST QUESTIONS

Please check your answers on page 29.

1 What is the simplest form of modulation?
2 What type of modulation is used in the standard radio broadcast band?
3 If the output power in the carrier of a standard broadcast station is 1000 watts with no modulation, what will the power be with 100% modulation?
4 If the modulation index is 0.5, what is the percentage of modulation?
5 What is the minimum bandwidth required in AM modulation to transmit a 15 kHz audio tone?
6 If the unmodulated power in a carrier is 100 watts, what will the power be in one sideband with 50% modulation?
7 If the antenna current of a transmitter is 10 amps with no modulation, and it increases to 11 amps with modulation, would you say that the modulation is 100% or less than 100%?
8 If when you modulate a transmitter the plate current of the modulated stage kicks upward, what does this indicate?

Plate Modulation

Plate modulation can be used in a class C amplifier stage using vacuum tubes because of the relationship between the plate voltage and power output. Changing the plate voltage changes the mutual conductance of the tube. When the plate voltage is doubled, the power output will increase four times. This output variation, with variations in plate voltage, provides linear amplitude modulation.

A basic plate-modulation system is shown in Fig.10. The modulating audio signal is inserted, by means of the modulation transformer, in series with the plate-supply line. Variations in the audio voltage add to and subtract from the dc voltage applied to the plate of the class C stage. In this way the rf carrier applied to the grid of the modulated stage and the audio variations on the plate are mixed in the tube to produce the modulated output waveform. To get 100% modulation, the highest amplitude audio component must be able to vary the class C plate voltage from approximately zero to twice its normal unmodulated dc value.

An undistorted sine wave signal will cause the plate voltage to increase and decrease the same amount, so the average dc plate voltage and plate current remain constant. In other words, the reading on the dc plate current meter in the class C stage should remain constant and have the same value as for the unmodulated carrier. Any variation in the dc plate current with modulation indicates improper adjustment or a circuit defect.

As you have learned, for 100% modulation of a class C stage, the power of the modulating signal must be 50% of the carrier power. This means that the modulator must add audio power to the stage equal to one-half the dc plate input power. This increases the power input to the stage, and therefore increases the plate dissipation. The stage must be designed so that the plate dissipation rating of the tube is not exceeded. This means that with no modulation the plate dissipation should be only two-thirds of the maximum rated plate dissipation of the tube. Thus, when the input power to the stage increases 50% with 100% modulation, the peak plate dissipation will be equal to the maximum rated plate dissipation of the tube.

The modulation transformer must act as an impedance-matching device between the plate output impedance of the modulator and the plate input impedance of the modulated stage.

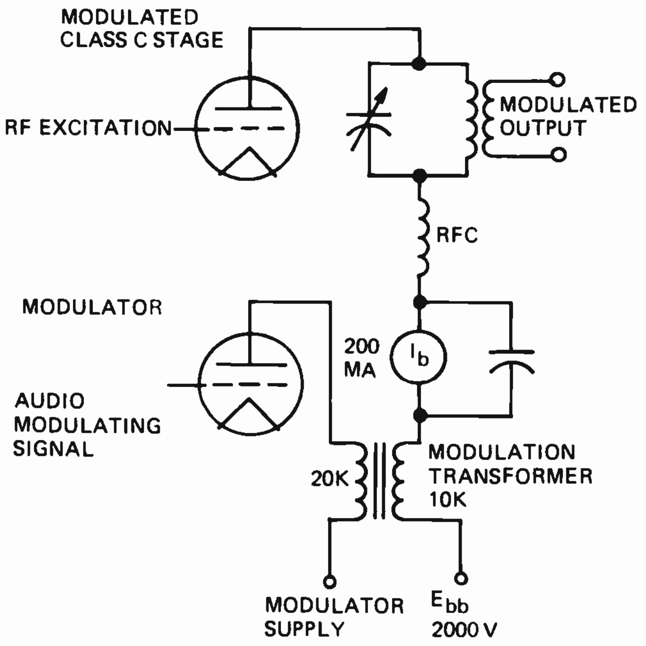

Figure 10. Basic plate-modulation system.

The plate input impedance of the modulated stage can be found by dividing the plate supply voltage by the dc plate current in amperes. For example, suppose that the plate supply voltage (E_{bb}) of a class C modulated stage is 2000 volts, and the dc plate current (I_b) is 200 ma (0.2 amp) as in Fig.10. The impedance (Z_s) would be:

$$Z_s = \frac{E_{bb}}{I_b} = \frac{2000}{0.2} = 10{,}000 \text{ ohms}$$

If we find from a tube manual that the modulator must have a load impedance of 20,000 ohms, we can find the turns ratio, n, of the modulation transformer as follows, where Z_p is the primary impedance and Z_s is the secondary impedance:

$$n = \sqrt{\frac{Z_p}{Z_s}} = \sqrt{\frac{20{,}000}{10{,}000}} = \sqrt{2} = 1.4$$

Thus the primary of the transformer must have 1.4 times as many turns as the secondary.

Now let us find the power required to modulate the class C stage. The formula for finding the input power, P_{in}, of the class C stage is:

$$P_{in} = E_{bb} \times I_b$$

so we have:

$$P_{in} = 2000 \times 0.2 = 400 \text{ watts}$$

The maximum power required from the modulator (P_m) is half of the carrier power for 100% modulation, and we have:

$$P_m = \frac{P_{in}}{2} = 200 \text{ watts}$$

This makes a total of 600 watts of input power in the class C modulated stage on modulation peaks when 100% modulation is reached. If we know the efficiency of the modulated stage, we can find the output power, P_o:

$$P_o = \text{efficiency} \times \text{input power}$$

If the efficiency of this stage is 70% when no modulation is applied, we would have:

$$P_o = 0.7 \times 400 = 280 \text{ watts}$$

With 100% modulation, we would have:

$$P_o = 0.7 \times 600 = 420 \text{ watts}$$

A PLATE-MODULATED TRIODE

A triode tube can be plate-modulated 100% by using the circuit shown in Fig.11. The modulator in this circuit is a push-pull class B stage. The output of the modulator is transformer-coupled into the plate supply circuit of the class C rf amplifier stage. The audio voltage developed across the secondary of the transformer adds to and subtracts from the plate supply voltage for the modulated stage. This changing plate voltage, when applied to the class C amplifier, forms a modulated rf output.

The dc component of plate current for the modulated stage flows through the secondary of the modulation transformer T2, the meter M1, and the rf choke. The choke is used to block the rf energy from the metering circuit, the modulator, and the power supply. The rf choke in no way impedes the audio signal because its reactance is extremely low at audio frequencies. A small capacitor shunts the dc plate current meter to provide a low impedance rf path around the meter.

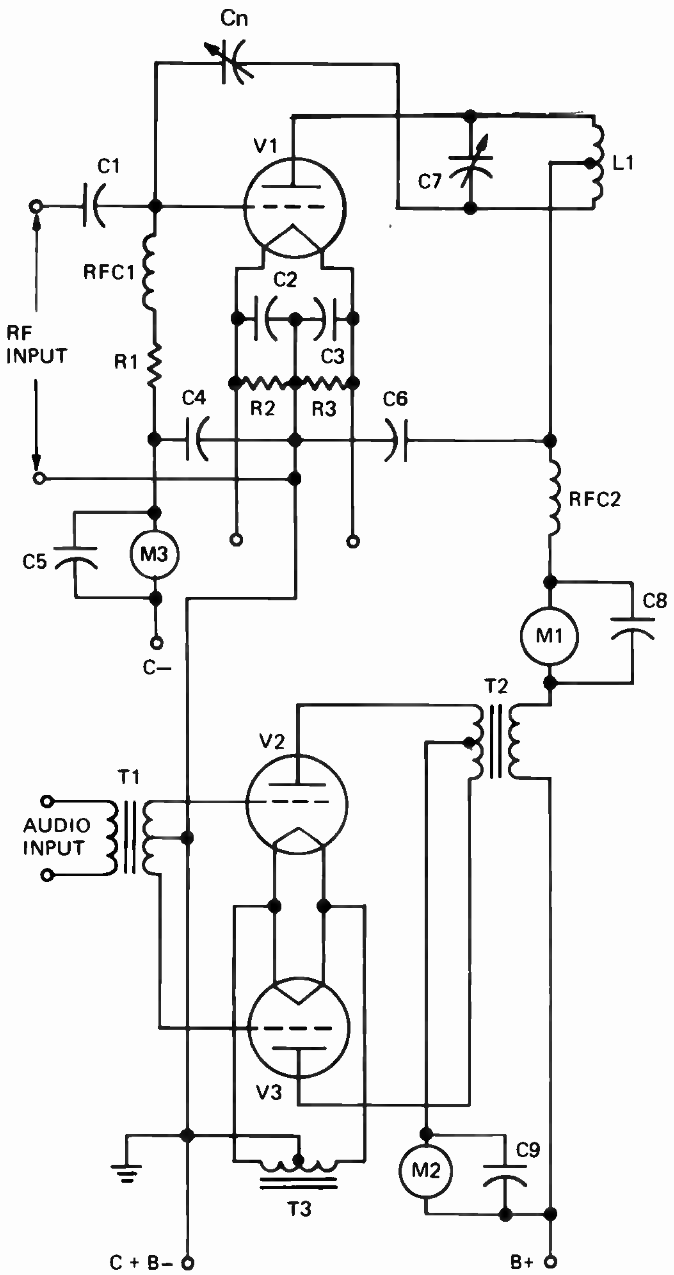

Figure 11. A plate-modulated class C triode rf amplifier.

The class C modulated stage is a plate-neutralized triode. The supply voltage and modulation are applied to the plate circuit at the tap on the tank circuit coil, L1. The tube shown is a filament type. Notice in the filament circuit that a true ground point for low frequencies as well as radio frequencies is obtained with a center-tapped connection to a pair of capacitors and resistors shunted across the filament line. This method of connection prevents the ac-powered filament circuit from hum modulating the carrier.

Grid bias is obtained by using a combination of external and grid leak bias. The rf amplifier must self-bias itself somewhat to prevent excessive grid current during the time the modulation drives the plate voltage to very low values. An increase in the grid current flow through resistor R1 automatically causes an increase in the grid bias, which stabilizes the grid circuit. The grid leak arrangement also improves the linearity of the circuit by preventing the grid from robbing the plate of electrons during the modulation troughs. The plate current can then follow the plate voltage variations more closely.

PLATE-MODULATED PUSH-PULL STAGE

Another circuit using plate modulation is shown in Fig.12. This circuit uses a push-pull class B modulator stage, which is transformer-coupled to the plate circuit of the push-pull class C stage. Notice that there is no bias on the modulator stage.

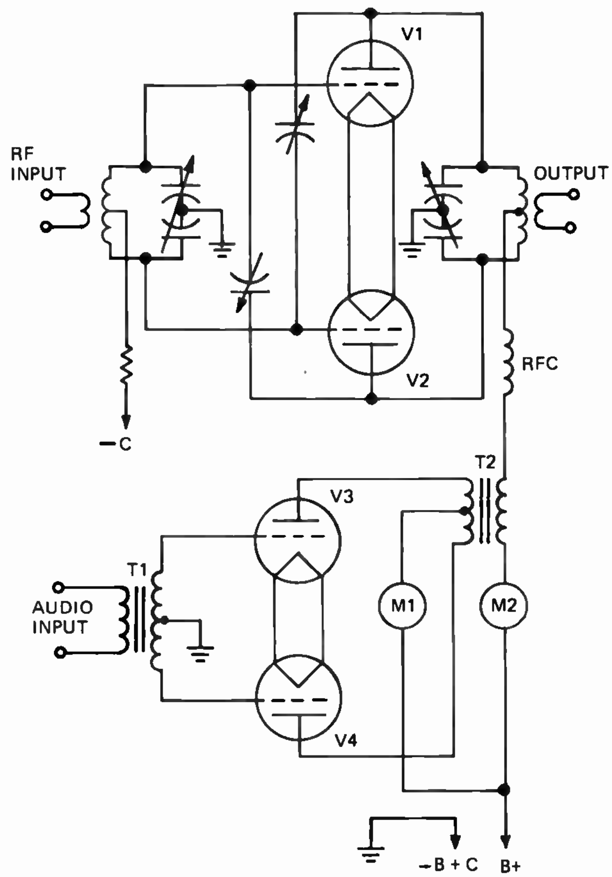

Figure 12. Push-pull zero bias class B modulator and push-pull output.

This is because the tubes used in this stage are a special type of zero bias tube. They are designed and built so that with zero bias applied to them there will be little or no plate current flow through the tube. Thus the tubes operate in class B without any external bias; this eliminates the need for a special bias supply for the class B modulator.

Notice also that in the circuit shown in Fig.12 (and Fig.11 also) the supply lines to the class B and class C modulator amplifiers are metered separately. With modulation, the class C plate current should remain constant, indicating symmetry of modulation. However, from your knowledge of class B operation, you can expect that the plate current meter for the modulator will fluctuate with the audio information. This is normal. In fact, with the class B stage biased to near cutoff, as required, the plate current reading will be nearly zero with no modulation. The meter reading will increase with modulation – the greater the amplitude of the modulation, the higher the meter reading.

The connection for the high-voltage plate voltage for the class C stage is through the meter M2, the secondary of the modulation transformer, the rf choke, and to the center tap of the tank circuit coil. Notice that a bypass capacitor need not be connected from the center tap of the rf coil to ground because of the balanced split-stator circuit and the fact that the center tap of the capacitor is at rf ground potential. As a result, the high-frequency audio signals are not bypassed by a capacitor as they are by capacitor C6 in Fig.11. A split-stator capacitor is one with two separate stators and a common rotor.

TETRODE OR PENTODE PLATE MODULATION

You will remember that there is a linear relationship between the plate current and the plate voltage in a triode tube. By this we mean that if you double the plate voltage, the plate current will double. This is a requirement that must be met for linear modulation. When we modulate a triode 100% and double the plate voltage, the plate current doubles so that we have four times the peak power output. This happens because the plate current for triode tubes depends primarily on the plate voltage.

To obtain 100% plate modulation of a tetrode or pentode tube, both the plate voltage and the screen voltage must be modulated. The reason for this is that the plate current in a tetrode or pentode does not primarily depend upon the plate voltage, but rather on the screen voltage. Therefore, in order to double the plate current of a tetrode or pentode, when we double the plate voltage we must also double the screen voltage. Similarly, to drive the plate current to cutoff we must drive the screen voltage essentially to zero.

There are several methods used to produce simultaneous changes in the plate and screen voltage of tetrode and pentode tubes. Three of the most widely used methods are shown in Fig.13.

In Fig.13(A) we have shown a circuit using a modulation transformer, T1, with two secondary windings. The voltage developed in one winding is used to modulate the screen, and the voltage developed in the other is used to modulate the plate. Thus both the screen and plate voltages are varied by the modulating signal. This circuit is often used to modulate high-power tetrode class C amplifiers. With this type of arrangement, both secondaries must have the correct number of turns. In other words, it is not sufficient simply to have the plate circuit of the class C stage matched to the plate circuit of the modulators; the screen circuit must also be matched. Thus a rather carefully designed and frequently quite expensive modulation transformer is required for this arrangement.

Another circuit used with high-power class C stages is shown in Fig.13(B). Notice that in this circuit the plate supply voltage from the class C stage is fed through the secondary of the modulation transformer. There is no connection from the modulation transformer to the screen. The screen voltage is supplied to the screen of the tube through the audio choke L1. This choke has a high impedance at audio frequencies.

The success of this system depends on the fact that the plate will attract many more electrons when the modulating signal causes the plate voltage to increase above the no-modulation level. When the plate voltage increases, the plate begins attracting electrons that would normally flow to the screen. This causes the screen current to drop. The change in screen current induces a voltage in the choke L1 that is in series with the screen supply voltage. Thus the screen voltage increases when the plate voltage increases. The increase in screen voltage causes the plate current to increase

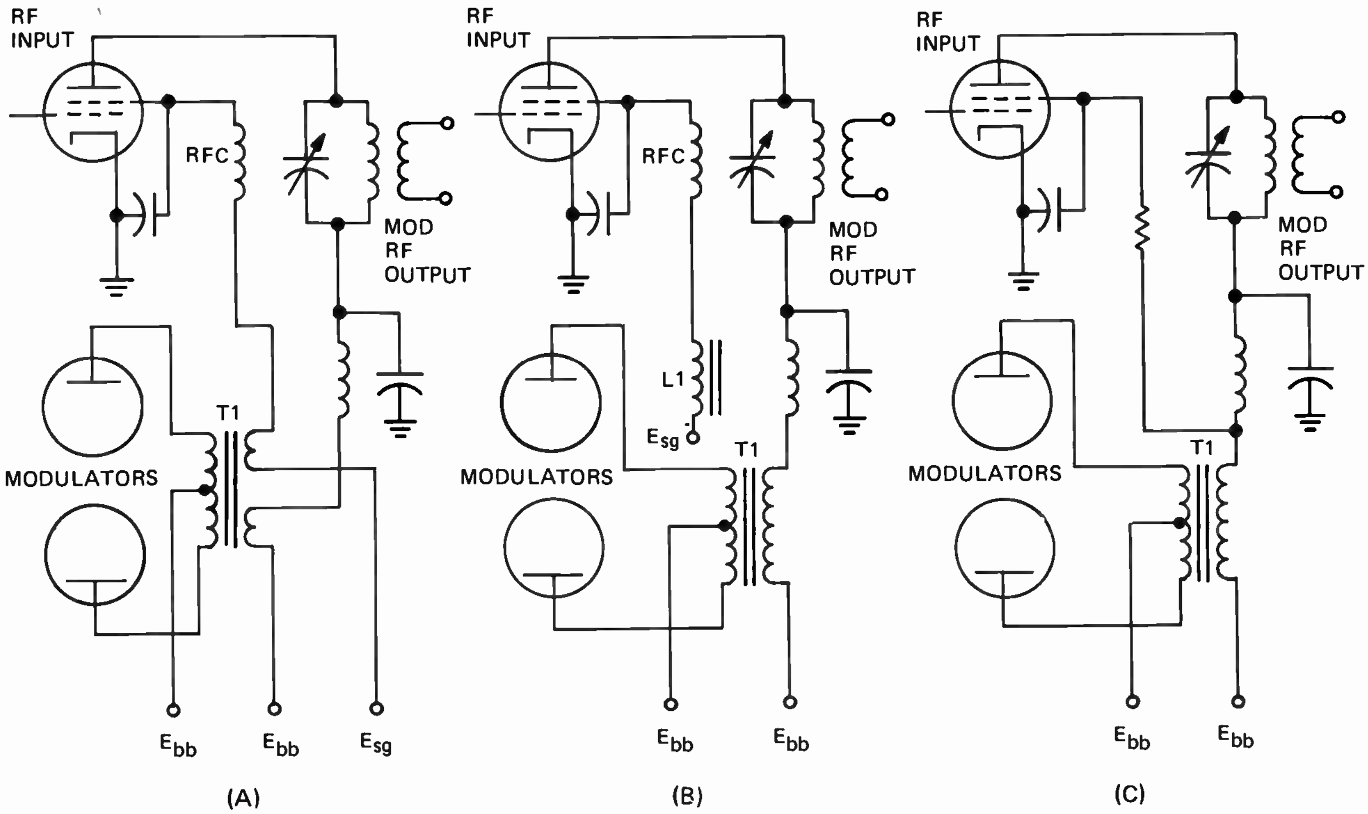

Figure 13. Methods of modulating plate and screen grid of tetrode or pentode tube.

still further. When the modulating signal reverses polarity and reduces the plate voltage on the tube, the plate will attract fewer electrons. This means that there will be more electrons available to flow to the screen so the screen current will increase. The increase in current through L1 will induce a voltage in L1 which opposes the screen voltage supply so that the screen voltage decreases. When the screen voltage decreases, the plate current will decrease still further.

A third method of modulating the screen is shown in Fig.13(C). Here the screen voltage is fed through a series resistor from the modulated plate supply voltage. Thus if the plate voltage varies, the screen voltage will also vary because the voltage to the screen circuit varies. The big disadvantage to this system lies in the fact that in big tetrode tubes the screen voltage is usually much lower than the plate voltage. In a tube that operates with a plate voltage of 2000 volts, the screen voltage may be as low as about 400 volts. Thus we would have to drop and waste 1600 volts in the screen-dropping resistor. This scheme can be used in low-power tetrode stages, but it is too wasteful to use in high-power stages.

If a class C stage using a multigrid tube is plate modulated alone, there is a nonlinear relationship between the plate voltage and the power output because the plate current does not vary linearly with the plate voltage. The modulation is compressed in the high-amplitude ranges before 100% modulation can be reached. Thus a compressed and distorted modulation envelope is produced.

HEISING MODULATOR

An older and seldom used type of modulation is the Heising system shown in Fig.14. In fact, you could probably forget about this type of modulator completely except that questions frequently appear about it on the FCC License Examination.

In the Heising modulator, the audio information is developed across coil L1 through which the class C stage also receives its voltage. The plate-supply current divides between the modulator and the class C stage in such a way that the audio voltage developed across L1 adds to and subtracts from the supply voltage. This varies the plate voltage to the class C amplifier.

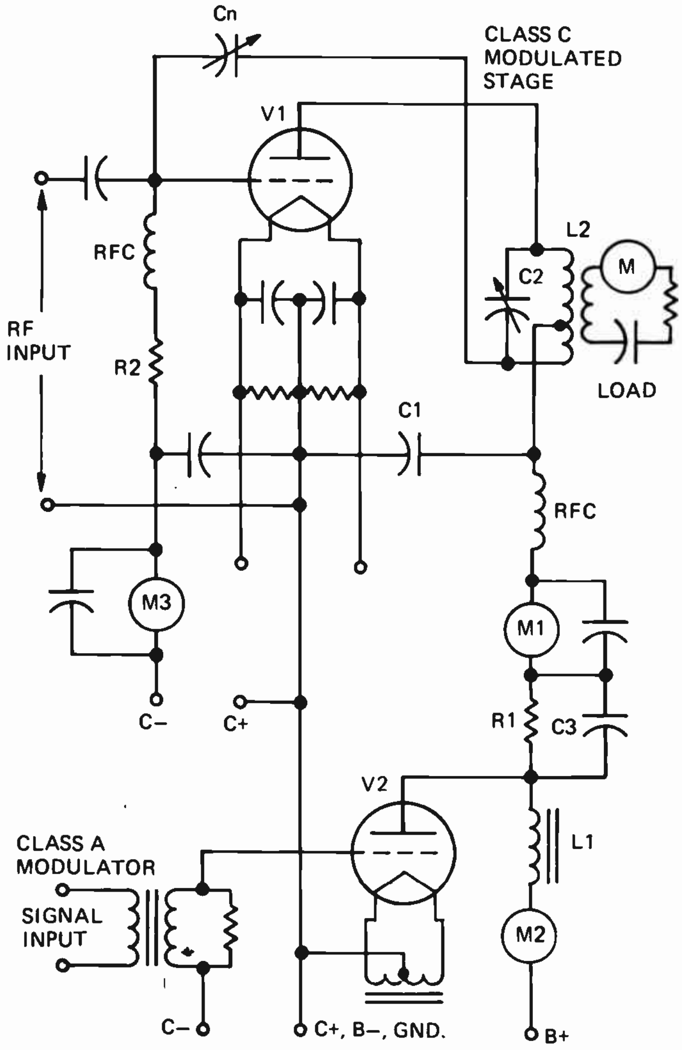

Figure 14. The Heising or "constant-current" method of plate modulating a class C amplifier.

One advantage of the Heising system is that the same power supply can be used for both the modulator and the modulated amplifier. To get 100% modulation, the plate current drawn by the modulator should vary between 0 and an extremely high value. However, you cannot operate a class A amplifier stage this way and obtain a linear output. Consequently, the class C amplifier must be operated with a plate voltage 30 to 40% lower than the modulator plate voltage. With this reduced voltage and high enough variation in the audio output, voltage of the modulator can be obtained to reach 100% modulation. Resistor R1 acts as a voltage-dropping resistor and reduces the dc plate voltage supplied to the modulated stage. Capacitor C3 bypasses the audio modulating signal around the resistor.

This modulation system is undesirable except in very low power equipment because of the large amount of power lost in the resistor. Furthermore, it is not practical to try to develop large amounts of audio power with a class A amplifier.

ADJUSTING A PLATE-MODULATED AMPLIFIER

The initial adjustment procedure for a modulated class C amplifier is the same as for a nonmodulated (cw) class C amplifier. With the modulator turned off (no audio being fed to the class C stage), go through the neutralizing, grid excitation, and plate tuning and loading procedures as recommended by the transmitter manufacturer. Make sure that the plate voltage and current, the excitation, and the grid bias are correct.

It is important that the excitation be adjusted correctly to get linear modulation. Insufficient excitation can reduce the amplifier efficiency and the power output, produce overmodulation, and cause negative carrier shift.

Increase the excitation until there is no further increase in the antenna current. Then, decrease it slightly to see if there is an immediate decrease in the antenna current. When the excitation is set correctly, there will be a slight increase and decrease in antenna current when you vary the excitation slightly. This adjustment makes certain that the class C stage is driven into saturation.

The plate voltage and plate current must be set to rated values because they determine the impedance into which the modulator must work – the impedance that the modulator was designed to match. The antenna coupling must be adjusted so that, with the plate circuit tuned exactly to resonance, the plate draws the rated current.

The next step is to adjust for 100% sinusoidal modulation. You can do this by feeding an audio signal from an audio oscillator to the modulator input. The audio signal must have an amplitude equal to the signal that will normally be fed to the modulator. Increase the gain of the modulator until the antenna current rises slightly less than 22.5%; the stage is then modulated close to 100%. You should then use a modulation monitor, which we will discuss later, to make the final modulation adjustment for maximum modulation.

As you increase the modulation to bring the modulation level close to 100%, there should be no change in the class C modulated amplifier dc plate current. Any variation in this plate current reading indicates a carrier shift and a nonsymmetrical modulation characteristic.

One type of rf current meter used to measure antenna current uses a device called a thermocouple. This type of meter depends on the current flowing through the thermocouple producing heat which in turn generates a voltage which causes the meter to operate. When this type of rf meter is used in the antenna circuit, the heat produced by the current flowing through the thermocouple must become stable before the meter pointer will come to rest. Time is required to heat the junction and generate the voltage to operate the meter. Therefore, a thermocouple meter does not respond immediately to changes in circuit adjustments; the adjustments must be made slowly to compensate for the lag in the meter itself.

COLLECTOR MODULATION

Class C transistor rf power amplifiers can be modulated using methods similar to plate modulation for vacuum tubes. The solid-state equivalent of plate modulation is collector modulation.

Most AM transmitters that you are likely to run into will be fairly high-powered units and in most cases will use vacuum tubes rather than transistors. Therefore you are not likely to run into too many transistorized modulated-amplifiers except in the case of low-power equipment such as citizens band equipment, portable equipment for police or emergency work, or amateur equipment. However, you should know how a transistor amplifier can be modulated. You should be familiar with this type of equipment because you may have to repair some citizens band equipment or other low-power equipment using transistors at some time.

Transistor Characteristics. You will remember from your earlier study of transistors that the characteristic curve showing the collector voltage against collector current in many respects resembles the characteristic curve of the plate voltage versus plate current for a tetrode or a pentode tube. There is not a linear relationship between collector voltage and collector current. By that we mean that if you double the collector voltage, the collector current will not double; as a matter of fact, in some instances it will increase very little.

Because of this nonlinear relationship between collector voltage and collector current, modulating the collector alone will not produce distortion-free amplitude modulation. We would run into a situation similar to that encountered in a tetrode or pentode tube where only the plate is modulated. The modulation would be compressed at high-volume levels.

A Typical Collector-Modulated Amplifier. Figure 15 is a schematic diagram showing how collector modulation of a transistor amplifier is accomplished. Notice that the rf input is fed into the base of Q1. In the collector circuit of Q1 we have a parallel-resonant circuit consisting of L1 and

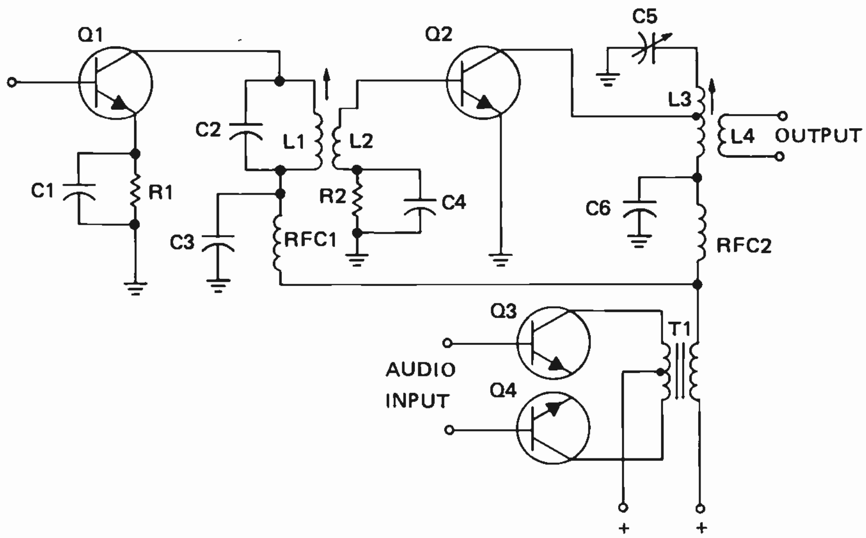

Figure 15. A collector-modulated amplifier.

C2. L1 is inductively coupled to L2 so the rf signal is induced in L2 and applied between the base and ground of Q2. Q2, the final rf power amplifier, is connected to the rf output tank circuit consisting of L3 and C5.

In order to obtain 100% collector modulation we modulate both the collector of Q1 and the collector of Q2. Modulating the collector of Q1 will produce some amplitude modulation of the rf signal at the output of Q1. This signal in turn is fed to the base of Q2 resulting in considerable change in the base current with the amplitude-modulation variations. This will produce the change in collector current. The modulation is also applied to the collector of Q2 and this, along with the partially modulated rf carrier which is fed to the base of Q2, results in reasonably effective collector modulation.

In the modulator we have shown push-pull transistors feeding a modulation transformer with a center-tapped primary. In some circuits a single-ended push-pull arrangement might be used and in this case the primary of the modulation transformer probably would not be tapped.

In some applications applying the modulation directly to the collector of Q1 may result in overmodulation. In this case, the amount of modulation fed to the collector of Q1 can be controlled by replacing the rf choke RFC1 with a resistor. This would reduce the dc voltage and also the modulated voltage fed to the collector of Q1. It will also reduce the amplitude of the partially modulated signal fed to the base of Q2. However, usually it is not necessary to do this. Simply adjusting the level of the audio gain correctly in the audio amplifier will provide the correct amount of voltage at the output to produce 100% modulation without overmodulation.

SELF-TEST QUESTIONS

9 When the plate voltage on a triode tube is doubled, what happens to the power output?

10 What must the turns ratio be on a modulation transformer if the class C stage is operated at 1000 volts and draws a plate current of 125 ma and the modulator stage requires a plate-to-plate impedance of 16,000 ohms?

11 Why are the tubes in the push-pull class B modulator shown in Fig.12 operated with no bias?

12 Why can we not obtain 100% modulation of a tetrode or pentode tube by modulating the plate alone?

13 Why is the screen modulation arrangement shown in Fig.13(C) not practical in high-power class C amplifiers?

14 What is the disadvantage of Heising modulation?

15 What characteristic of a transistor makes it impossible to obtain 100% modulation if only the collector is modulated?

16 How is 100% collector modulation obtained?

Grid Modulation Systems

The modulation in an amplitude-modulation system using vacuum tubes can be applied to the control grid of the modulated stage instead of the plate. These two forms of amplitude modulation, plate modulation and grid modulation, are the most common.

However, a class C amplifier can also be modulated in other ways. Stages using multigrid tubes can be modulated by applying the modulating signal to the screen grid or the suppressor grid. In this section we will take up control-grid modulation first, then we will study screen-grid and suppressor-grid modulation. Another form of grid modulation that we will study is called "suppressed-carrier" modulation. We will also study transmitter-keying, which is usually done in the grid circuit.

CONTROL-GRID MODULATION

In a control-grid modulation system, the plate input power and the rf grid excitation are constant. The modulation is applied to the control-grid circuit.

The rf stage used for grid modulation is not truly class C, but is biased somewhere between class B and class C. Figure 16(A) shows the tube characteristic curve and the waveforms with no modulation. Notice that the peak plate current drawn is at the center of the linear portion of the curve, point B.

The modulating voltage is superimposed on the fixed grid bias, and the effective grid voltage varies with the modulating signal. The amplitude of the modulating signal is chosen so that the limit of modulation extends over the linear part of the curve, between points A and C. This is shown in Fig.16(B). The plate current pulses that excite the tank circuit also vary in peak amplitude according to the modulation. As a result, a modulation envelope is formed in the tank circuit; the tank circuit voltage and the power output follow the modulating signal variations.

The rf excitation must be adjusted so that when it is 100% modulated, at the crest of the modulation, the rf cycles reach up to the saturation point (C in Fig.16) but do not go beyond it.

As you can see, the settings of the grid excitation, the amplitude of the modulating signal,

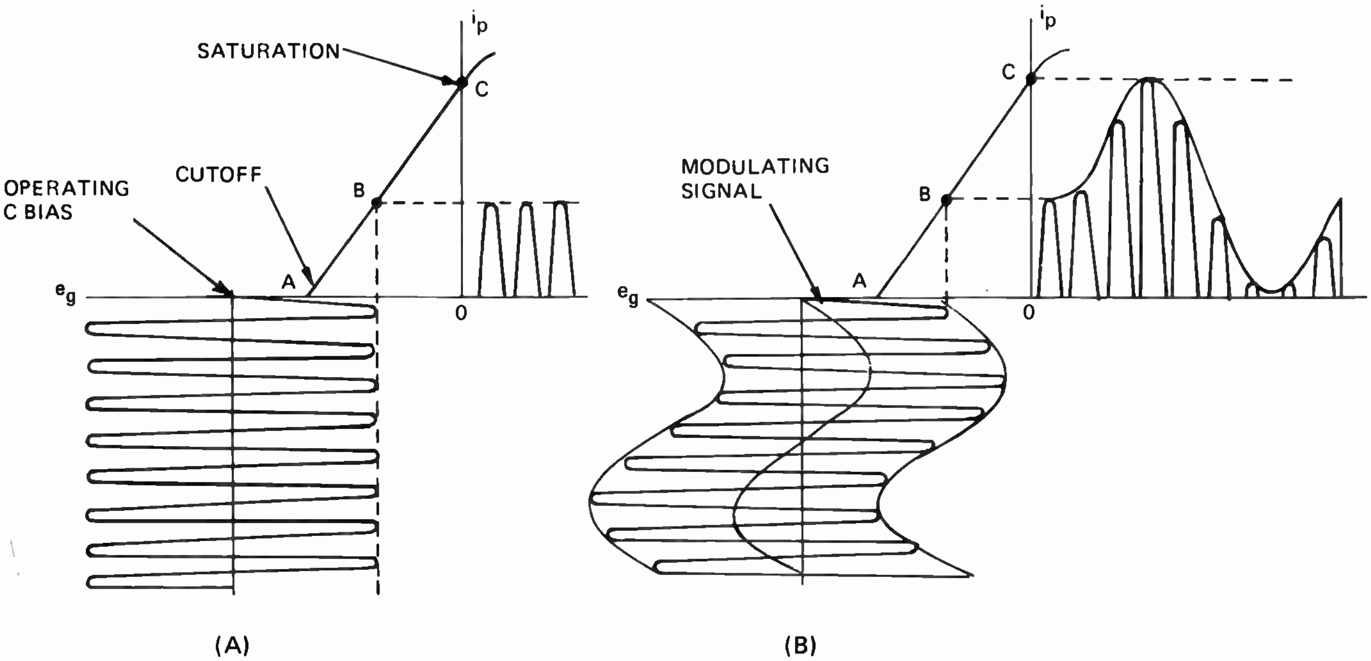

Figure 16. Grid modulation waveforms.

and the grid bias are important for linear and efficient grid modulation. If the excitation is excessive, the dc plate current will vary.

The efficiency of a grid-modulated amplifier when no modulation is being applied is approximately half that of a plate-modulated amplifier – approximately 35%. However, when 100% modulation is applied, the positive alternations of the excitation reach up nearly to saturation, and therefore the efficiency rises to nearly 70%. This rise in efficiency is the extra power necessary to form the sidebands of the modulated signal. In plate modulation, the power for the sidebands is supplied by the modulator; in grid modulation, this extra power is provided by the rise in efficiency with modulation. For this reason, all forms of grid modulation are frequently referred to as efficiency modulation.

The maximum output with grid modulation is much less (approximately one-quarter) than that obtainable when using the same tube with plate modulation. The major advantage of grid modulation is that less modulator power is required than for plate modulation. With grid modulation, the modulated stage itself supplies the sideband power; the audio signal on the grid changes the efficiency of operation. Thus, the power required from the modulator is very small. For example, it is possible to modulate a 1000-watt carrier fully with as little as 20 watts of audio power. A bulky, expensive, and high-powered modulation system is not required.

To get the least distortion in a grid-modulated system, the modulator and driver must have good regulation, so that they will supply the necessary driving energy as the grid impedance of the modulated stage changes throughout the modulation cycle. Generally the grid input circuit is loaded or a limit is placed on the amount of grid current. Then the input impedance variations are not as great as when the stage is operated as an unmodulated class C amplifier. Figure 17 shows a grid modulation circuit. In this circuit, the output of the modulator is loaded by resistor R1 to maintain a more constant load on the modulator. In practice the exciter and the modulator are designed to deliver two or three times as much power as is necessary so that the modulating system can be loaded and thus give more favorable regulation.

Adjustments. When tuning a grid-modulated amplifier, you must first run through the same

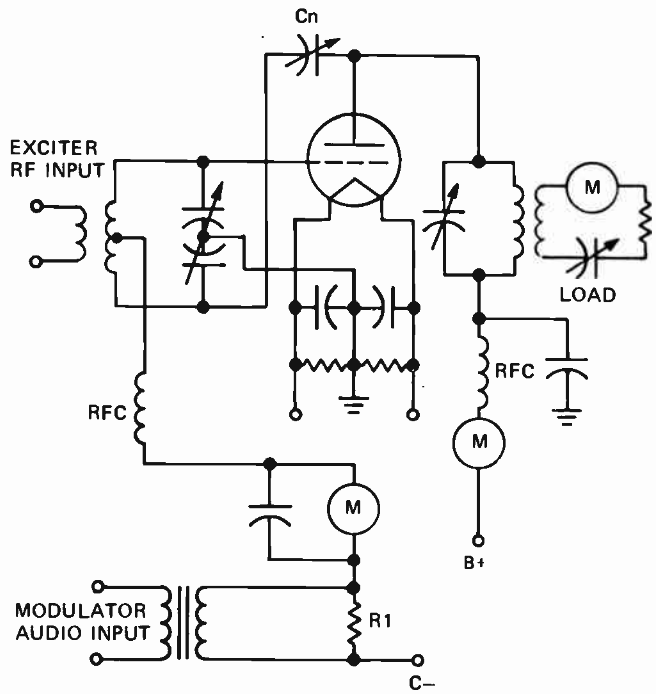

Figure 17. Circuit for grid modulation.

preliminary steps as for tuning an unmodulated class C stage. After you have made these preliminary adjustments, you must set the class C bias, the rf grid excitation, and the loading presented by the output tank circuit to get a linear modulation characteristic.

One method of adjusting the grid circuit is to set the class C bias (with no rf excitation applied) to a value that just begins to draw plate current when the plate voltage is set to the recommended value. As you increase the excitation, watch the meter reading for the plate current and the antenna current. Increase the excitation until the plate current continues to rise sharply but there is no further proportional increase in the antenna current reading. At this point, the class C stage is being driven to the saturation level and the output is maximum. This is the level of operation that represents the crest of 100% modulation.

Increase the grid bias until the antenna current is reduced to half the peak value. This step will increase the bias the proper amount to allow the crest of the unmodulated carrier to occur at the center of the linear portion of the transfer characteristic, point B of Fig.16. The dc component of the plate current should also be at the recommended value. If it is not, a loading adjustment must be made so that the proper load is being presented to the tube.

With the operating voltages and currents set correctly, if a sine wave signal is fed to the audio system to modulate the class C stage, and the antenna-current meter has increased 22.5% above its "no modulation" value, you have 100% modulation. Additional information on modulation adjustments will be given when you study modulation monitors.

Distortion. Some causes of carrier shift are poor regulation of the rf exciter and distortion in the modulator and associated amplifiers. Proper loading of the tank circuit is important because the most linear modulation is obtained when the tube works into a rather high value of tank circuit impedance. The output coupling must be adjusted until 100% modulation can be obtained without carrier shift.

Two of the causes of downward modulation in a grid-modulated stage are too low an operating bias and excessive rf grid excitation. Poor power supply regulation, improper loading of the class C tube, or a defective tube itself can also be suspected.

SCREEN AND SUPPRESSOR-GRID SYSTEMS

A class C amplifier stage using a tetrode tube can be modulated by applying the modulating signal to the screen of the tube. A class C amplifier stage using a pentode tube can be modulated by applying the modulating signal to either the screen grid or the suppressor grid, as shown in Fig.18. Suppressor-grid and screen-grid modulation systems, however, cannot be modulated to 100% because voltage changes on these elements do not have a linear effect on the plate current. The limit of undistorted modulation is about 80%. A few special tubes that have been designed for this type of modulation can reach 100% modulation without serious distortion. The carrier efficiency of both types of modulation is approximately 35%, about the same as for control-grid modulation.

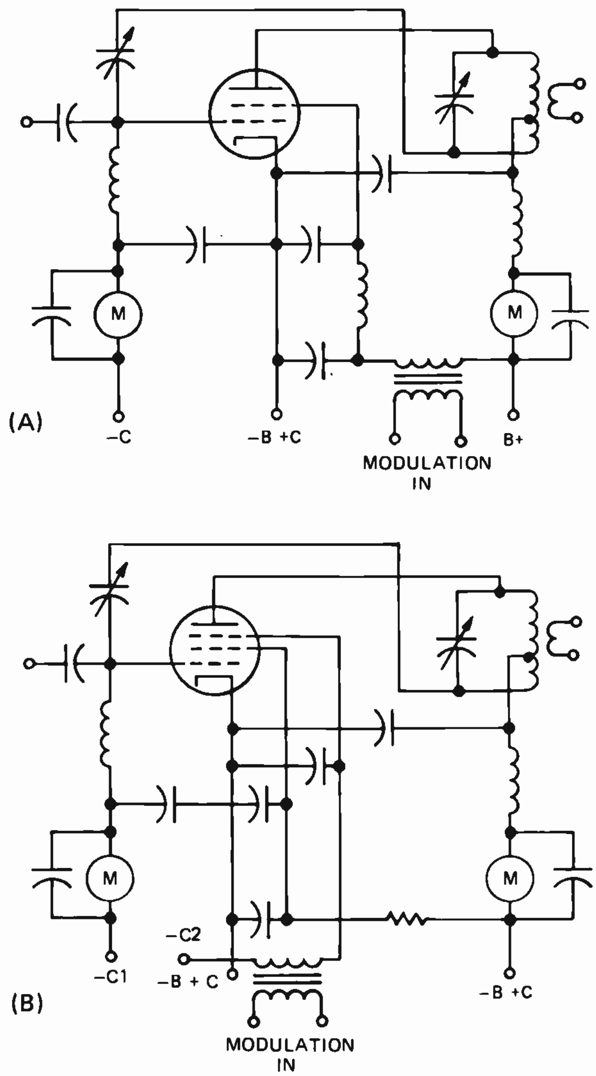

Figure 18. Typical circuits for (A) screen and (B) suppressor modulation.

Screen-Grid Modulation. In a screen-grid modulated system, a modulating signal voltage is used to vary the screen-grid voltage of the modulated stage. The dc screen voltage must be adjusted so that at the peak of the modulation it will be the same as that for a stage operating as a class C amplifier without modulation. Therefore if the stage is to be screen modulated, the dc screen voltage supplied is usually about half the rated value for normal operation. The modulating signal then varies the screen voltage between zero on the modulation troughs and the rated value on the peaks.

The modulating power required for screen-grid modulation is slightly more than is required for control-grid modulation. However, there is less distortion on the peaks of the modulated signal because the load presented by the modulated stage to the modulating voltage does not vary with the signal-voltage variation. In some circuits, part of the modulation voltage is applied to the screen grid of the driver stage. This is done to permit a higher modulation percentage without distortion.

Suppressor-Grid Modulation. The suppressor grid of a receiving-type pentode tube is generally connected directly to the cathode so that the potential between the cathode and suppressor is zero. The suppressor grid of a transmitting tube is usually operated at zero voltage also, but sometimes it is operated with either a small positive or a small negative voltage applied to it.

In the suppressor-grid modulation system, the suppressor grid is operated with a negative voltage applied to it. The modulation signal, which is fed to the suppressor grid in series with the negative voltage, varies the voltage on the suppressor grid. This, in turn, causes the plate current to vary. The negative bias on the tube and the modulation signal are adjusted so that the plate current is swung from zero to the rated value for the tube when it is operated as a class C amplifier without modulation. The power required to modulate this stage is very low, about the same as required for control-grid modulation. At the modulation peaks, the power output is about the same as for a normal class C stage, but the overall efficiency of the system is about the same as for a grid-modulated signal.

Adjustments. The adjustments that you will normally have to make on either a screen-grid or a suppressor-grid modulator are quite simple. The most important thing is to get the class C stage operating correctly first. The transmitter manufacturer will supply information on what the plate current of the final stage should be. You should load the transmitter to get the rated plate current and then feed a sine wave signal to the input and adjust the gain of the modulator to give as near 100% modulation as possible without distortion. The modulated output should be checked with a cathode-ray oscilloscope. You will learn how to do this later. If you detect any nonlinearity you may find an adjustment that can be used to vary the screen voltage on a screen-modulated stage or the suppressor voltage on a suppressor-modulated stage. Varying this voltage may enable you to improve the linearity. We will go into nonlinearity in a later lesson on oscilloscopes.

SUPPRESSED-CARRIER SYSTEMS

In the modulation systems we have discussed so far, the modulated signal, composed of both the carrier and the sidebands, was fed to the transmitting antenna. It is also possible to remove the carrier completely from the transmitted signal and transmit only the sidebands because all of the information being transmitted is contained in the sidebands. This is called suppressed-carrier modulation. The carrier must be added again by a low-powered local generator in the receiver itself for demodulation. The inserted carrier takes the place of the carrier that is removed by the suppressed-carrier modulation system.

The basic circuit for a suppressed-carrier system is shown in Fig.19. This system uses grid modulation. The modulated stage has two tubes, and the grids of the two tubes are excited out of phase by the modulating information. Two transistors could be used in place of the tubes, but the circuit would be essentially the same.

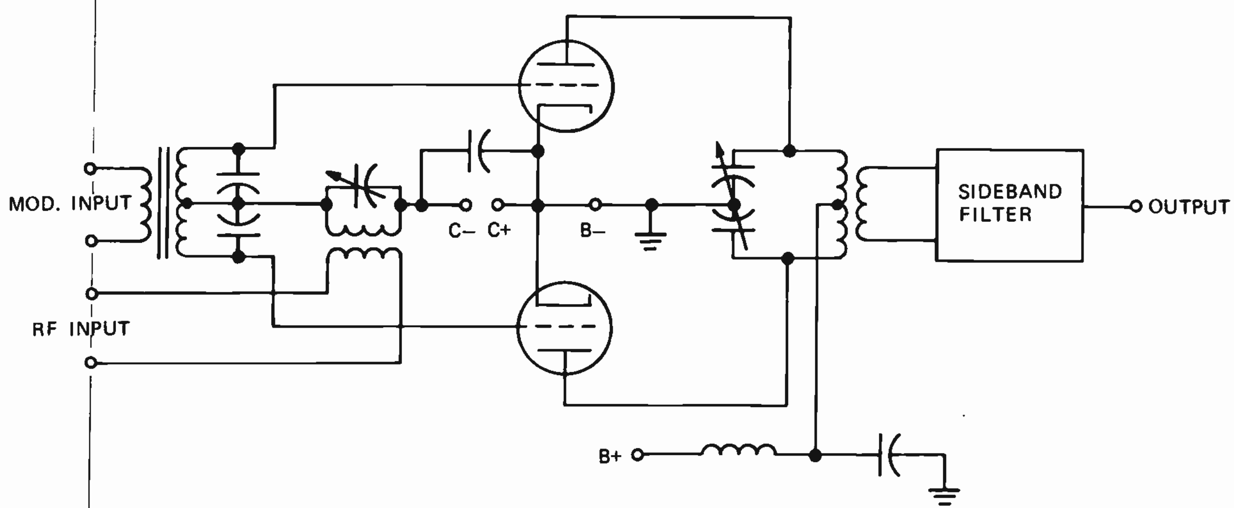

Figure 19. Circuit for suppressed-carrier single-sideband circuit.

The rf carrier signal is fed to a center tap on the grid-modulation transformer so that the grids receive the rf signal in phase. The plates are connected in push-pull.

Since the input grids are driven in phase by the rf signal, the rf signals at the plates of the tubes are also in phase. This signal will therefore be canceled in the push-pull tank circuit when the signals at the two plates are equal because the signal currents will flow through the two halves of the tank circuit in opposite directions. The field produced by one current will cancel the field produced by the other, so the net pickup by the loop coupled to the tank coil will be zero. With a perfect signal balance the output of the modulated amplifier with no modulation is zero.

The modulating signal is fed to the two grids in push-pull, which means that the modulating signals fed to the two grids will be 180° out of phase; that is, when the signal at one grid is positive, the signal at the other grid will be negative. The modulation is fed to the grids in series with the rf signal so that two sidebands are formed in the grid circuit, one equal to the sum of the rf plus the modulating frequency and the other equal to the difference of the rf minus the modulating frequency. These two sidebands are out of phase on the grids of the tubes and hence the signal currents produced by them will be out of phase. The amplified sideband signal currents which are out of phase at the tube plates are fed to the opposite ends of the plate tank, and they add in the tank circuit. Thus they produce magnetic fields which add and induce a voltage in the pickup loop placed near the tank coil. The carrier is canceled in the plate tank circuit and only the sidebands appear in the output.

Single-Sideband Transmission. Since one sideband contains all of the necessary intelligence, it is possible to go a step further and eliminate one of the sidebands as well as the carrier. This system of transmission has been in use for some time. It is correctly called single-sideband suppressed-carrier, but because this is a long name it is usually shortened to simply single-sideband transmission. With this system, the only energy radiated is that contained in one sideband. The sideband on the other side of the carrier frequency can be eliminated without destroying the intelligence. This reduces the bandwidth required to transmit the signal, and makes it possible to operate more stations in a given frequency band. For example, if a station is transmitting both sidebands and has modulation frequencies as high as 5000 Hz (5 kHz), a bandwidth of 10 kHz is needed. If single-sideband transmission is used, a bandwidth of only 5 kHz would be needed. Thus in a band of 100 kHz there would be room for a maximum of 10 stations transmitting both sidebands, but there should be room for a maximum of 20 single-sideband stations.

KEYING A TRANSMITTER

When code is transmitted, the carrier is cut on and off in a manner corresponding to the dots and dashes that form coded information. This is called continuous wave (cw) keying. It might at first seem that keying a transmitter would be a simple matter of turning the radiated energy on and off, and that the only signal frequency transmitted would be the carrier frequency. However, the simple act of turning the transmitter off and on by keying it produces sidebands, since this is effectively 100% amplitude modulation using a square wave modulation signal. Thus, if the transmitter is keyed at a rate of 10 times a second, there are two sidebands, one 10 Hz above the carrier frequency and the other 10 Hz below the carrier frequency. A 20 Hz bandwidth would be a very narrow band; however, when a carrier is keyed, it forms a square-top type of signal as shown in Fig.20(A). The harmonic content of such a pulse-like waveform is high. In fact, the amplitude of the second harmonic may be

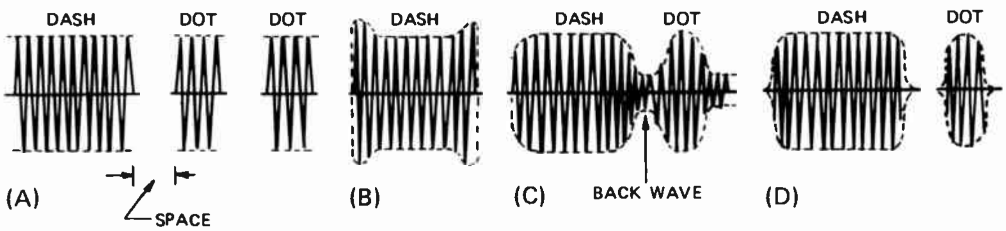

Figure 20. Envelope shapes of dots and dashes that may be radiated by a code transmitter. The form at (D) is best because it creates the least interference.

as much as one-half the amplitude of the fundamental frequency. Harmonic components up as high as the 50th can be present in the output signal. As a result, the bandwidth must be wider than the rate of code interruption.

When the transmitter is keyed very sharply, it generates random current and voltage surges, called transients, which produce clicks and thumps in the demodulated sound, and also cause the carrier amplitude to increase at the beginning and end of the dot or dash as shown in Fig.20(B). The very sharp surges of power are also capable of setting up parasitic oscillations, and can cause interference well outside of the assigned channel.

Clicks or thumps heard at the end of each dot or dash can also indicate that the amplifier is not properly neutralized. The amplifier may continue to oscillate and radiate a signal for a short time after the keying has cut off the carrier. This is called a back wave and is shown in Fig.20(C). It can cause the dots and dashes to run together.

Ideal keying occurs as shown in Fig.20(D), when the beginning and end of each individual dot or dash is rounded slightly to prevent harmonics of the keying frequency from being generated. Likewise, the space between the dots and dashes should be completely free of carrier and spurious radiation. This can be done by properly neutralizing the amplifier stage, eliminating all parasitic oscillations, and using a well-regulated power supply.

Keying Methods. Keying in a transmitter can be in the final stage of the transmitter or in one of the low-power stages. If it is in a low-power stage, enough fixed bias is applied to the following stages to keep the plate current cut off or at a low value when there is no excitation to the stage. Although it is possible to key a stage by opening and closing the circuit to any element in the tube, the only practical circuits to key are the cathode or grid circuits. Of the two, the grid circuit is preferable.

An ideal way to key a powerful cw transmitter is to key the grid of a low-level stage as shown in Fig.21. When the key is open, the grid bias is increased to a very high negative value, and the plate current is cut off. With the key closed, a normal bias is applied and the stage operates at maximum output. When the key is open, there will be no rf output from the stage. With sufficient bias on the following stages, their plate current will drop to zero when the rf excitation is removed.

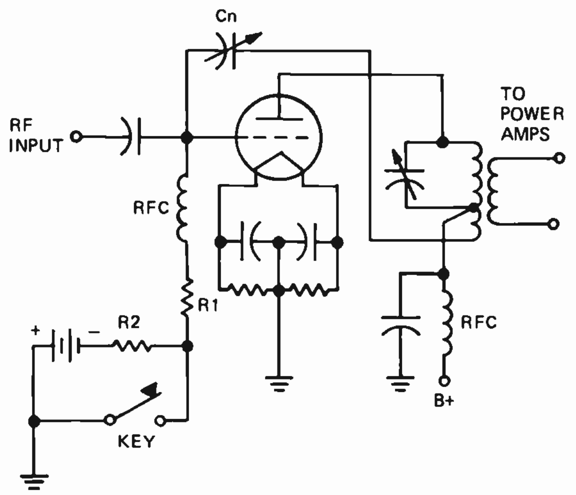

Figure 21. Using key to change bias applied to rf amplifier tube.

Another keying method is shown in Fig.22. Here the key controls the power relay in the cathode circuit of the stage. The plate current is cut off and on when the relay opens and closes. This method has the disadvantage that if the stage is handling any appreciable amount of power, the relay contacts must be very large to handle the high current, and the relay must be insulated to take the full plate voltage.

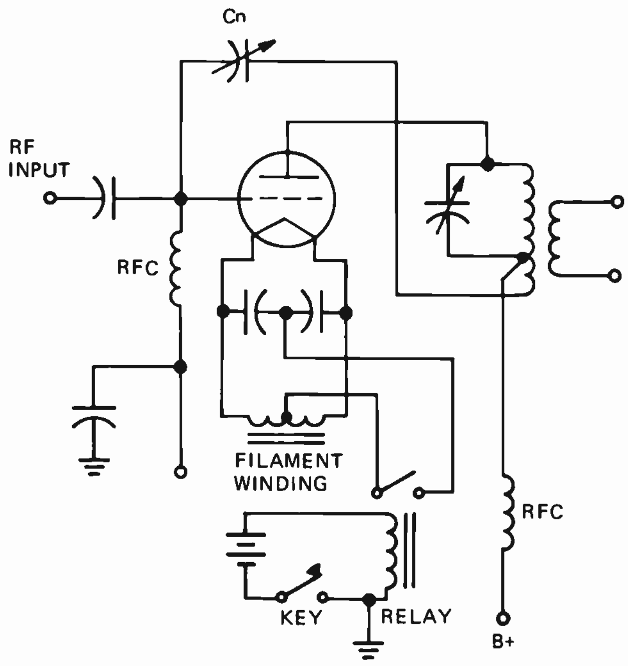

Figure 22. Keying in center-tapped filament circuit.

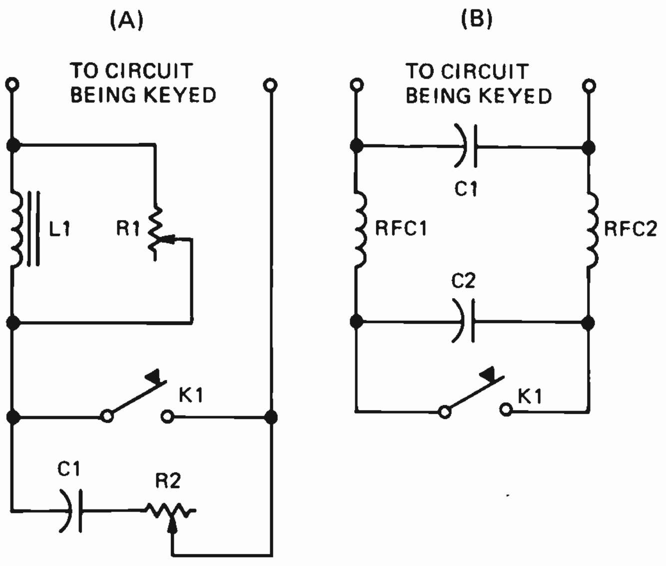

Figure 23. (A) Keying circuit for shaping signal to eliminate thumps; (B) rf filter to remove clicks.

Shaping Dots and Dashes. Proper shaping of dots and dashes is accomplished by means of a keying filter or lag circuit. Figure 23(A) is an example of one type of lag circuit. Choke coil L1 slows the buildup of current in the keyed circuit. Variable resistance R1, shunting the coil, controls the current through the coil and changes the speed with which the current builds up.

When the key is opened, the magnetic field around the choke coil collapses. The collapsing field causes a current to flow through C1 and R2 that rounds off the trailing edge of the character. C1 also prevents arcs at the key contacts when the key is opened.

In practice, resistor R1 is adjusted to the lowest value that will prevent thumps when the key is closed. R2 is adjusted to the highest value that will prevent thumps and clicks when the key is opened. The lag circuit is built into the transmitter at the keyed circuit. When a keying relay is used, the lag circuit is connected at the relay contacts.

The leads from the transmitter to the key may pick up some rf. Using a shielded lead between the transmitter and the key, with the shield grounded, will keep this pickup at a minimum, but in some cases there may be considerable pickup even with shielded leads. This rf pickup will cause rf currents to arc across the key contacts when the key is opened, and cause key clicks. The rf filter circuit of Fig.23(B) is connected at the key terminals to prevent rf arcing at the contacts.

Vacuum Tube Keyers. Many of the disadvantages of using a keying relay can be overcome by using a vacuum tube as the relay. One type of vacuum tube keying circuit is shown in Fig.24. A triode tube is connected between the filament center tap of the rf stage and B–. When this triode tube is cut off, no plate current can flow through the amplifier. Removing the bias from the keyer tube by closing the key allows both tubes to conduct.

Bias for the keyer tube is supplied by a separate power supply to resistors R1 and R2. When the key has been open for some time, capacitor C1 is charged to the level of the bias supply. Closing the key shorts the output of the bias supply through resistor R1 and removes the bias from the tube. However, before the tube can conduct, this charge must drain off C1 through resistor R2. As the charge drains off C1, the tube begins to conduct; the current through the tube increases as its grid approaches zero bias. The action of C1 and R2 shapes the leading edge of the code character.

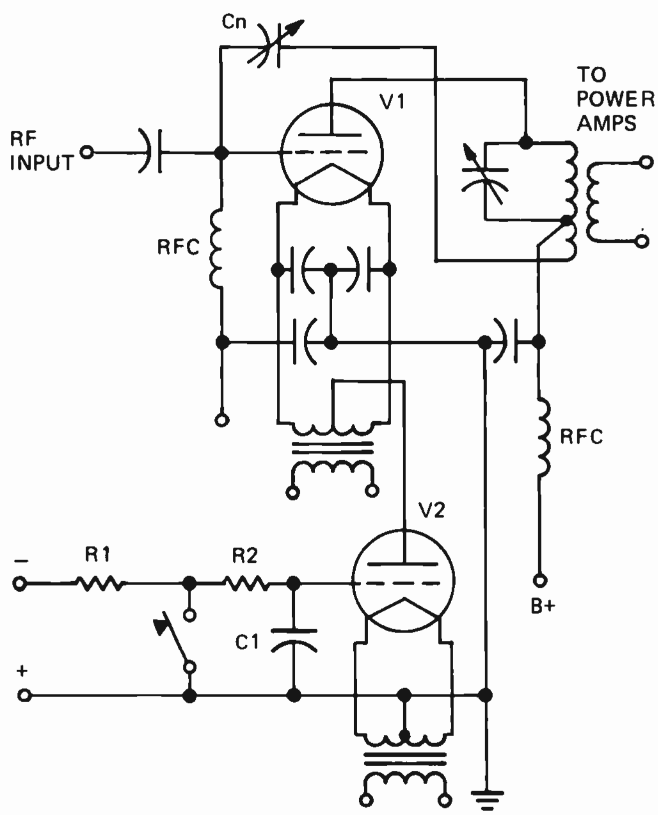

Figure 24. Vacuum tube keying circuit with key-click filter.

When the key is opened, capacitor C1 charges through R1 and R2, and gradually cuts off the current through V2. This shapes the trailing edge of the code character. The entire plate and grid currents for tube V1 must flow through tube V2. At the same time, the voltage drop across V2 must be kept small. For this reason, V2 is usually two or more parallel-connected tubes which can pass very large currents with small plate-to-cathode voltage drops. Connecting several tubes in parallel increases the current-handling capability of the keyer without increasing the voltage drop.

SELF-TEST QUESTIONS

17 What is one disadvantage of a grid-modulated class C amplifier?

18 Does the plate power input to a grid-modulated class C stage change with modulation?

19 Is 100% modulation possible with screen-grid and suppressor-grid modulation without distortion?

20 Is it necessary to transmit the carrier in an amplitude-modulation system?

21 What is the chief advantage of single-sideband suppressed-carrier modulation?

22 Can a code signal which turns the transmitter on and off 10 times a second be transmitted in a 20 Hz bandwidth?

23 What is the advantage of keying a transmitter in the grid circuit in preference to the cathode circuit?

24 Is it desirable to have the dots and dashes transmitted by a cw transmitter sharp or slightly rounded?

25 What is a vacuum tube keyer?

Amplitude Demodulation

Earlier in this lesson, you learned that a modulated wave is formed when low-frequency information is superimposed on a higher frequency carrier. The modulated wave then can be transmitted through space. At the receiver the modulation must be removed from the carrier before it can be used. This process of removing the modulation is called "demodulation." The stage in which demodulation takes place is called a demodulator or a detector.

There have been a number of different types of detectors used for amplitude modulation, but all have disappeared from general use except the diode detector. Both vacuum tube and solid-state diode detectors have been used; the two work in exactly the same way. However, all modern radio and television receivers use either silicon or germanium diodes for amplitude modulation. These detectors are frequently called "linear detectors" because the original modulating signal is produced linearly without distortion.

We are going to confine our discussion of detectors to diode detectors because this is the only type you are likely to encounter.

DIODE DETECTORS

The diode detector is a linear, large-signal detector. This means that the original modulating signal is reproduced linearly without distortion and also that the detector works best on reasonably strong signals. The modulated signal must be amplified by the rf and i-f stages of the receiver until it has a high enough level to operate the diode detector. With the proper signal amplitude fed to the detector input, the output signal is quite linear and reasonably free from distortion. However, when a weak signal is applied to a diode detector, the amount of distortion increases.

Two simple diode detector circuits are shown in Fig.25(A). Notice that the two are identical except in one case where we have shown a vacuum tube diode and in the other case a solid-state diode. The operation of the two is identical. You may run into a vacuum tube diode detector in an older radio or TV receiver, but you will find solid-state diodes in all modern equipment.

The modulated signal from the rf and i-f amplifier stages is applied between the anode and

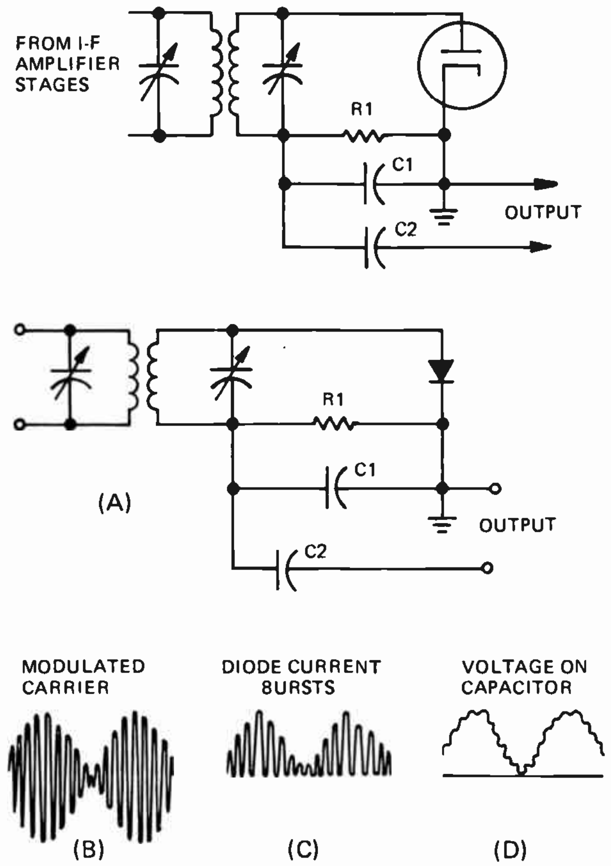

Figure 25. Diode detectors and waveforms.

cathode through the diode load resistor R1 and its filter capacitor C1. The demodulated signal is developed across resistor R1 and capacitor C1. The demodulated signal is fed through the coupling capacitor C2 to the audio amplifier. The operation of these detectors is shown in Fig.25.

When a sine-wave modulated carrier signal, shown in Fig.25(B), is fed to the diode detector, the diode conducts only on the half of each cycle that drives the anode positive.

During the other half-cycle, when the anode is driven negative, the diode does not conduct. As a result, the negative half-cycles are cut off. The output current flows in bursts that occur at the same rate as the carrier frequency. The current flow from the cathode to the anode of the diode is shown in Fig.25(C).

The peak of each positive alternation causes enough current to flow to charge the output capacitor C1 almost to the peak value of the applied signal. When the applied signal drops below the most positive part of the positive alternation, the diode ceases to conduct. Then the capacitor begins to discharge through the output resistor R1. However, the time constant of the resistor and capacitor combination is so long that the capacitor discharges only a small amount. By this time, the next alternation of the incoming signal reaches its peak value and draws another burst of diode current that recharges the capacitor.

Therefore, because of the time constant of the RC network, the diode does not conduct for the entire positive alternation of the applied signal, but only during its positive peak. Also, the output voltage no longer follows the carrier variations. It is now a dc voltage that follows the modulation variations, as shown in Fig.25(D). This is the original audio information.

The time constant of the output capacitor and resistor must not be too great or the audio information will also be filtered out. Thus, the time constant of the output circuit must be chosen to act as an effective filter at the carrier frequency, but it must not be long enough to filter the highest frequency component of the modulating information.

The length of time that the diode conducts determines the amount of loading that the diode detector circuit places on the preceding radio frequency amplifier. The less the diode conducts, the lighter the load that the detector places on the preceding stage. Likewise, the efficiency of the detector is higher because the loss in the diode itself occurs only during the interval that the diode conducts. Hence, with a limited conduction time, a greater percentage of the applied modulator energy is developed across the output circuit than is dissipated across the diode.

The ratio of the voltage developed across the output resistor compared to the modulated signal voltage indicates the effectiveness of detector action. The ratio can be made high by making certain that the output resistor has a value much higher than the internal resistance of the diode. The higher the value of the diode load resistor, the higher the detector efficiency becomes. However, keep in mind that there is a limit to the peak value of the load resistor because of the increase in the output time constant. When very high modulating frequencies are present, such as in high-fidelity and video detectors, the value of the diode load resistor must be kept quite low to prevent loss of the higher modulating frequencies.

Distortion. Diode detectors can introduce frequency, phase, or amplitude distortion into the demodulated signal. Frequency distortion occurs when some of the modulating frequencies are not developed in the output circuit, or do not have the proper amplitude relationship to other frequency components that were present in the original modulating signal. If the output time constant is too long, for example, the high-frequency modulation components may be attenuated with respect to the middle- and low-frequency components.

Phase distortion can also occur in the output circuit and can cause certain modulating frequencies to be delayed with respect to other frequencies contained in the original modulating signal information. Both types of distortion are due to incorrect values of diode load resistor and output capacitor in the diode circuit. They can be reduced by making the detector circuit present a uniformly resistive load to the preceding stage over the band of frequencies contained in the modulation envelope. In addition, the output impedance should be uniform over the entire range of modulating frequencies.

Amplitude distortion occurs when the output voltage does not exactly follow the variations of the modulation envelope. One type of amplitude distortion can be produced when a weak modulated signal is applied to the detector. Notice in Fig.26 that the lower part of the diode characteristic curve is nonlinear. Thus, when a weak signal is applied, the signal is demodulated in this nonlinear portion of the curve. A strong modulated signal, on the other hand, will swing up into the linear part of the curve, and the distortion will be considerably less. Thus, you can see why a high-amplitude signal must be fed to the diode detector for detection with a minimum of amplitude distortion.

Another factor that contributes to amplitude distortion in diode detectors is reverse current flow. In a diode detector using a vacuum tube, this is no problem because when the anode is negative there is no current flow through the tube. However, in a diode detector using a germanium or a silicon diode as the detector, there may be an appreciable reverse current. In addition, once this reverse current begins to flow it remains almost constant regardless of the amplitude of the input signal, unless the breakdown voltage of the diode is exceeded. This reverse current tends to discharge the output capacitor on each negative half-cycle.

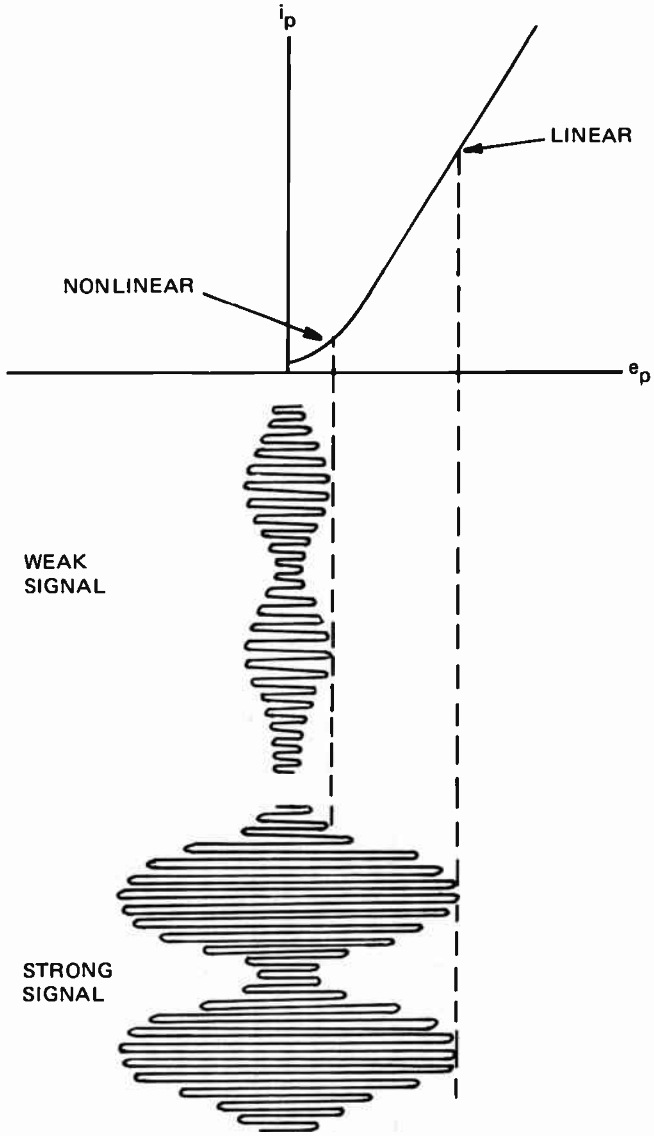

Figure 26. Diode transfer characteristic.

Since the reverse current remains essentially constant regardless of the amplitude of the signal, this means that you will have a greater percentage of discharging of current on low-amplitude signals than on high-amplitude signals. The reverse current flow of the diode therefore distorts the amplitude of the audio signal and introduces amplitude distortion.

Amplitude distortion can also be produced when the output capacitor is not able to discharge rapidly enough as the modulation envelope falls into its trough. This defect, as shown in Fig.27, actually cuts off a part of the negative alternation of the demodulated signal. The value of the output capacitor should be small enough to present a high

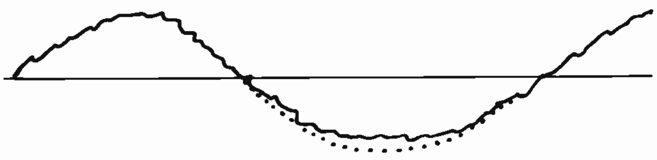

Figure 27. The wavy line shows the distorted waveform; the dotted line the undistorted waveform.

reactance to the modulating frequency, and at the same time, its value should be larger than the anode-to-cathode capacity of the diode. A greater part of the applied rf signal voltage will then develop across the diode as it should, and not across the diode output capacitor (which is in effect also shunted across the applied signal). The detector would have low efficiency if a large part of the rf voltage were dropped across the output capacitor.

This type of amplitude distortion is more noticeable on signals having a high percentage of modulation because of the greater voltage change between the crest and trough of the demodulated signal. A greater voltage must be discharged by the output capacitor in order to follow the modulation envelope. For lower percentages of modulation, the amount of voltage that must be discharged between the crest and the trough of the wave is smaller, and the output circuit is able to follow the change.

CW DETECTION

You will remember that when code is transmitted we referred to the wave as cw (continuous wave). The carrier is modulated by interrupting it to form a series of dots and dashes. When a signal of this type is fed to a diode detector such as shown in Fig.25, the capacitor C1 will be charged, but the amplitude of the charge will be constant. Thus we would have a dc voltage built up across the capacitor by each dot and dash. Sometimes when a cw signal is fed to this type of detector you can hear a small hiss but you do not have an audible tone. We overcome this in cw detectors by mixing a second signal with the incoming signal. The second signal is generated by a second oscillator called a beat-frequency oscillator (abbreviated bfo). The beat-frequency oscillator operates at a frequency close to the i-f. Usually a means is provided to vary the frequency of bfo a small amount. The signal from the bfo beats with the incoming i-f signal in the detector and, in effect, amplitude modulates the signal. An audio tone which is equal in frequency to the difference in frequency between the i-f and the frequency of the bfo is produced.

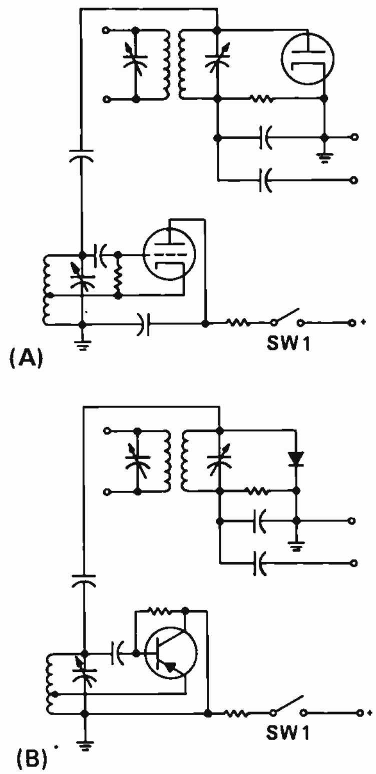

Figure 28. Two detector and bfo circuits.

Figure 28 shows two typical diode detectors along with beat-frequency oscillators. The circuit shown in Fig.28(A) uses vacuum tubes whereas the circuit shown in Fig.28(B) uses a solid-state detector and a transistor as the bfo. The operation of the two is essentially the same. With an incoming i-f of 455 kHz, if the bfo operates at 454 kHz, the two signals will be in phase and add 1000 times each second and they will be out of phase and subtract 1000 times each second. Thus we will have an amplitude-modulated signal that is modulated at a frequency of 1000 Hz. By increasing the frequency of the bfo slightly we can produce a lower frequency audio tone, and by decreasing the frequency we can produce a higher frequency audio tone. The switch SW1 in each circuit provides a means of turning the bfo on for cw reception and off for a standard amplitude-modulation reception.

SELF-TEST QUESTIONS

26 Is a diode detector primarily a large-signal or a small-signal detector?

27 What type of diode detector would you expect to find in a modern TV receiver?

28 If too large a diode load resistor is used in the video detector of a television receiver, what is the likely result?

29 How is cw detected with a diode detector?

ANSWERS TO SELF-TEST QUESTIONS

1 Code modulation.

2 Amplitude modulation.

3 1500 watts.

4 50%.

5 30 kHz. You will need 15 kHz to accommodate the upper sideband and another 15 kHz to accommodate the lower sideband. Since these two sidebands are 30 kHz apart, the minimum bandwidth is 30 kHz.

6 6 watts.

7 Substantially less than 100%. At 100% modulation, the antenna current would be 12.25 amps.

8 Positive carrier shift.

9 It increases four times. The power output increases this much because when the plate voltage doubles, the plate current also doubles so the power output increases four times.

10 1.4 to 1. The place circuit impedance of the class C stage can be found by dividing the plate current in amps into the plate voltage. The plate current in amps is 0.125 amp and this when divided into 1000 volts will give you a plate circuit impedance of 8000 ohms. The turns ratio can then be found by taking the square root of the ratio of the two impedances. You divide 8000 into 16,000 and get 2 and then take the square root of 2 which gives you 1.4 as the turns ratio.

11 These tubes are special zero bias tubes. They are built so that with zero bias very little plate current flows; the tubes are designed especially for class B operation.

12 The plate current does not vary linearly with the plate voltage. In a tetrode or pentode tube, changes in plate voltage have very little effect on the plate current. The plate current is controlled primarily by the screen voltage.

13 Too much power would be wasted in the screen voltage-dropping resistor.

14 It is very inefficient; too much power is lost in the resistors required to reduce the plate voltage of the modulated stage in order to get 100% modulation.

15 The collector current does not vary linearly with the collector voltage.

16 By modulating the collector of the class C amplifier and the collector of the driver driving the class C amplifier.

17 The power output is much lower than can be obtained from a plate-modulated stage.

18 No. The efficiency of this stage and hence the power output changes with modulation.

19 Not in most cases. Close to 100% modulation can be obtained with some special tubes designed specifically for this type of modulation.

20 No. All the intelligence is contained in the sidebands.

21 It produces a narrower bandwidth than a regular AM signal.

22 No. Turning the transmitter on and off produces sidebands far beyond the rate at which the transmitter is turned on and off.

23 Lower powers are handled in the grid circuit than in the cathode circuit and hence it is easier to control clicks and thumps.

24 They should be rounded. This will prevent the clicks and thumps and reduce the bandwidth required to transmit the signal.

25 A vacuum tube keyer is the tube used to key the transmitter. The keyer tube is frequently in the cathode circuit and a high negative bias is placed on the grid when the key opens so the tube is cut off. The transmitter is keyed by removing the grid bias through the keyer tube and hence the class C power amplifier tube can conduct.

26 A large-signal detector. It will produce considerable distortion on small signals.

27 A solid-state diode, either a germanium or a silicon diode.

28 Frequency distortion.

29 By feeding a signal from a bfo into the diode detector along with the i-f signal.

Lesson Questions

Be sure to mark your lesson card as shown to indicate Lesson CC206. Most students want to know their grades as soon as possible, so they mail their lesson card in immediately. Others, knowing they will finish another lesson within a few days, send in two cards at once. Either practice is acceptable. However, don't hold your answers too long; you may lose them. Don't hold answers to more than two sets of lessons at one time, or you will run out of lessons before new ones can reach you.

1. The easiest way to modulate an rf carrier is to use:
   a. Morse keying.
   b. AM.
   c. FM.
   d. PM.

2. Amplitude modulation primarily affects which characteristic of an rf signal?
   a. Phase.
   b. Amplitude.
   c. Frequency.
   d. Time.

3. If a 1500 kHz carrier is amplitude modulated by a 2000 Hz sine wave, what frequencies are present at the output of the modulated amplifier?
   a. 1300 kHz, 1500 kHz, and 1700 kHz.
   b. 1490 kHz, 1500 kHz, and 1520 kHz.
   c. 500 Hz, 2000 Hz, and 3500 Hz.
   d. 1498 kHz, 1500 kHz, and 1502 kHz.

4. When a sine wave signal amplitude modulates a properly adjusted class C amplifier, what is indicated by the grid and plate current meters?
   a. Grid increases; plate decreases.
   b. Plate increases; grid decreases.
   c. Both remain the same.
   d. Both increase.

5. During 100% sinusoidal amplitude modulation, what part of the total average output power is in the sidebands?
   a. 100%.
   b. 50%.
   c. 33-1/3%.
   d. 25%.

6. If a transmitter has a modulation index (m) of 1.0, what percent increase in antenna current would you expect?
   a. 10%.
   b. 22.5%.
   c. 33-1/3%.
   d. 50%.

7. What is the minimum audio power required to fully plate modulate a class C amplifier that is operating at 3000 volts plate voltage and 400 ma plate current?
   a. 600 watts.
   b. 750 watts.
   c. 900 watts.
   d. 1200 watts.

8. What should the turns ratio be of a modulation transformer to match the class C amplifier of Question 7 to a modulator having an output impedance of 1875 ohms?
   a. 0.82.
   b. 0.71.
   c. 0.50.
   d. 0.35.

9. The circuit of Fig.17 uses:
   a. Low level modulation.
   b. Efficiency modulation.
   c. Control grid modulation.
   d. All of the above.

10. The purpose of C1 in Fig.25 is to:
    a. Match the detector to R1.
    b. Couple the audio signal.
    c. Bypass the audio signal.
    d. Filter the i-f signal.

THE VALUE OF KNOWLEDGE

Knowledge comes in mighty handy in the practical affairs of everyday life. For instance, it increases the value of your daily work and thereby increases your earning power. It brings you the respect of others. It enables you to understand the complex events of modern life, so you can get along better with other people. Thus, by bringing skill and power and understanding, knowledge gives you one essential requirement for true happiness.

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