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YOUR INSTRUCTOR'S

MODEL ANSWERS

Here are the correct answers to your FCC type Multiple Choice Examination as prepared by your instructor. Compare your answers with his. Where he felt it would be helpful, a discussion follows the correct answer giving the reasons for his answer, or how it was derived.

ANSWERS TO MULTIPLE-CHOICE EXAMINATION TC-9

1. (3) Remove any amplitude variations from the carrier before feeding it to the discriminator. The discriminator type of circuit is sensitive to amplitude variations and therefore, a limiter is used ahead of a discriminator in order to keep the signal amplitude fed to the discriminator constant and hence render the discriminator immune to noise pulses.

2. (5) Turn off the audio section of the receiver when no signal is being received. In mobile communications receivers, when no signal is being received, the automatic volume control in the receiver automatically turns up the gain of the receiver so that background noise is picked up and amplified. This can be very objectionable and there is a tendency on the part of the operator to turn down the volume control to get rid of this objectionable noise. This might cause him to miss a transmission. Therefore a squelch circuit is used that biases the audio section so that it is cut off when no signal is being received. When the signal is received, the bias disappears and the audio section amplifies the signal picked up.

3. (1) Hold the voltage produced by the FM carrier constant so that the detector is not sensitive to amplitude variations.

4. (3) The transit time is the length of time it takes an electron to travel from the cathode to the plate of the tube.

5. (4) By spacing the tube elements closely together. Since the transit time of a tube is determined by how long it takes an electron to travel from the cathode of the tube to the plate, the smaller the space between the cathode and the plate, the shorter the transit time will be. Of course, spacing the elements closely together leads to the possibility of an arc over inside the tube, so closely spaced elements can be used only in receiving tubes or small transmitting tubes, where the operating voltages are relatively low.

6. (5) Low uhf. Acorn tubes are designed for use in the uhf region up to about 600 mc. This is considered the low end of the uhf region.

7. (4) The low-frequency i-f stage.

8. (2) Better image rejection.

9. (5) Determine the attenuation in the waveguide.

10. (2) Prevent interference from low-frequency stations.

11. (5) 10 KW. The power output of an FM station remains constant at all times; it does not change with modulation.

12. (2) Strong local signals overloading the rf stage. A strong local station may block the rf stage and cause it to operate on the non-linear portion of its characteristic curve. This can cause mixing to occur in the rf stage where signals from two different stations will mix together to produce sum and difference frequencies.

13. (5) Keep the amplitude of the signal from the signal generator as low as possible. If you use too strong a signal from the signal generator you may overload some stage in the receiver and if this happens, it will be impossible to peak the various adjustments in the receiver to get maximum output.

14. (4) Its ability to pick up a weak signal. The sensitivity of the receiver, which is usually given in microvolts, tells you how strong a signal must be received by the receiver in order to produce a standard output, which is usually 50 milliwatts.

15. (2) Ceramic. Glass, paper, wood, bakelite, porcelain and rubber are often used as insulators at low frequencies, but are unsatisfactory at uhf.

16. (1) Grounding, (3) Shielding, (4) Stray capacities.

17. (4) A special uhf tube in which the cathode, grid and plate are arranged in three parallel planes.

18. (5) 1500 watts. The transmitter is rated at its maximum unmodulated output. At 100% modulation, only two thirds of the output power is in the carrier; an additional one third is in the sidebands for a total of 1.5 × the unmodulated power or 1500 watts.

19. (4) 100%. In single sideband communications, only one sideband (and no carrier) is transmitted. Thus, all of the power is in the one sideband.

20. (2) Noise generated by the receiver. With a low-level input signal, the gain of the rf and i-f stages increases. The noise inherent in these stages is greatly amplified and the output signal-to-noise ratio decreases.

21. (3) The inductance and capacitance decrease. The butterfly tank is a tunable resonator in which the position of the rotor determines both the inductance and the capacity. When the rotor is moved away from the stator, the effective capacitor plate area between the stator and rotor decreases. This results in lower capacity. At the same time, the flux path around the frame of the resonator is blocked by the stator -- resulting in lower inductance.

22. (3) Ratio of input-to-output signal-to-noise ratios. Noise figure is a measure of how much noise is added to the signal in the receiver stages.

23. (2) Skin effect. At higher frequencies, the signal has a tendency to travel near the surface of the conductor. This results in a smaller effective cross-section area at greater I^2R losses.

24. (5) Produces less harmonic distortion. In the frequency conversion method of single sideband demodulation, the demodulator is a linear stage. When both the carrier and the sideband are present, heterodyning takes place, producing the audio or difference signal at the output.

25. (3) Making the cathode, grid and plate as small as possible.

26. (1) Ten times the sideband amplitude. When the ratio of the amplitude of the carrier to the amplitude of the sidebands is 10:1, the carrier insertion mixer has a more linear response and produces less distortion.

27. (4) Reduces noise from the local oscillator. The balanced mixer cancels out the local oscillator noise so that it does not mix with the signal. Thus, the noise is not present in the output signal.

28. (3) 750 and 1750 cps. For the two-tone test using audio tones, the audio frequencies should be 1000 cycles apart.

29. (3) Tubes with oxide-coated cathodes. Contact noise results from the irregularities between the cathode and the oxide coating.

30. (5) Feeding the signal through a balanced transformer. Assuming that the noise is picked up with equal phase and amplitude on both lead-in conductors, it will cancel in the transformer primary.

31. (3) Twice the width of the guide. As frequency decreases when cutoff is approached, the rf path is more and more zigzag. When the wavelength is equal to twice the width of the guide, cutoff occurs. The rf energy then travels only from side to side and is quickly attenuated.

32. (3) Prevent the formation of moisture. Moisture can cause breakdowns by having rf arcs burn holes in the guide.

33. (4) Four horizontal quarter-wave elements arranged to form two half-wave dipole antennas at right angles to each other.

34. (2) Increases when the tube amplifies. This phenomena is often referred to as Miller Effect and produces deterioration in the high-frequency response.

35. (3) Often encountered when two stations are operating close together.

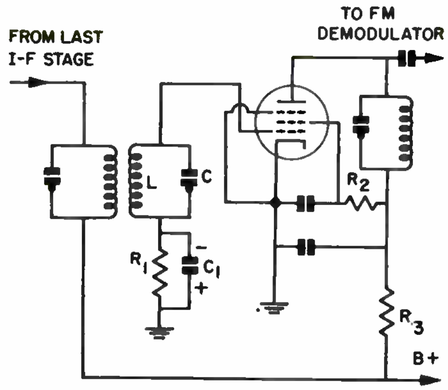
36. (4) In preventing harmonic radiation.

37. (2) Oscillator frequency variations due to poor regulation in the oscillator power supply.

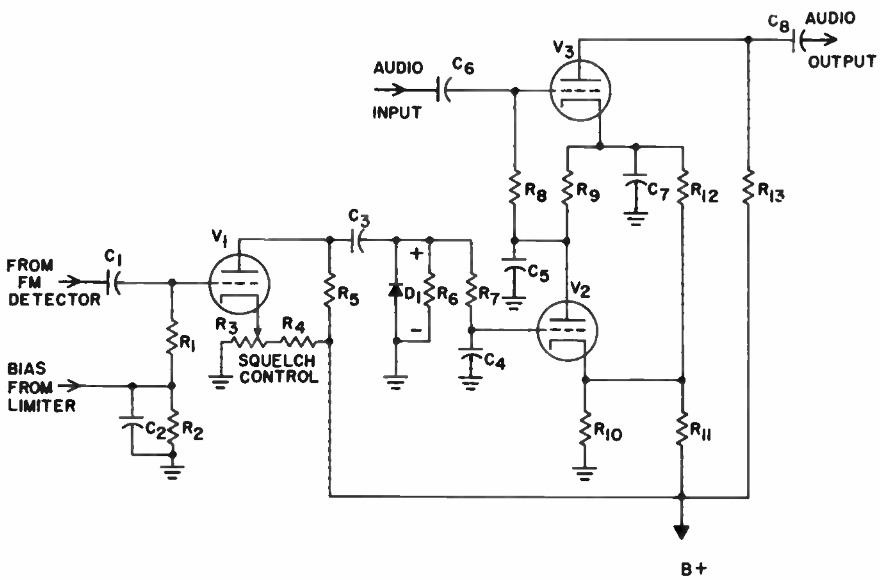
38. (3) Reinserting the missing carrier by means of a locally generated signal.

39. (4) Enables you to cut down the interference from interfering stations.

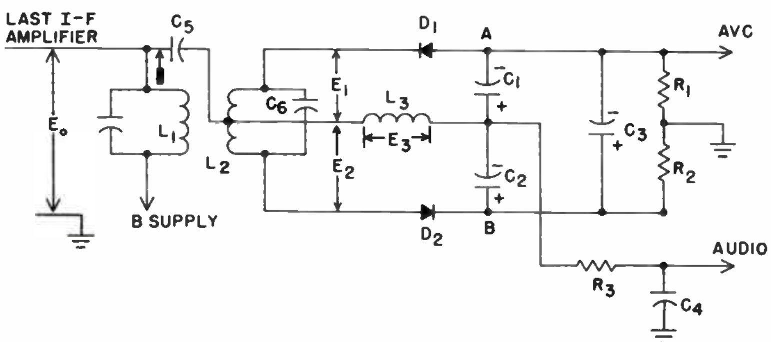
40. A limiter.



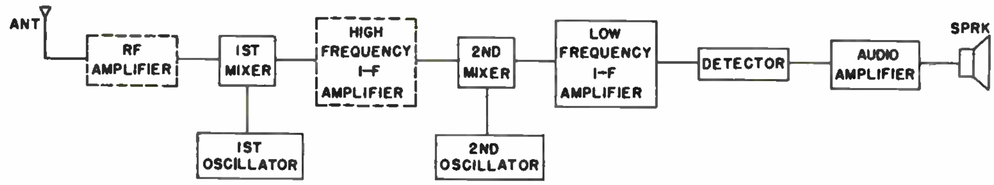
41. A differential squelch circuit.



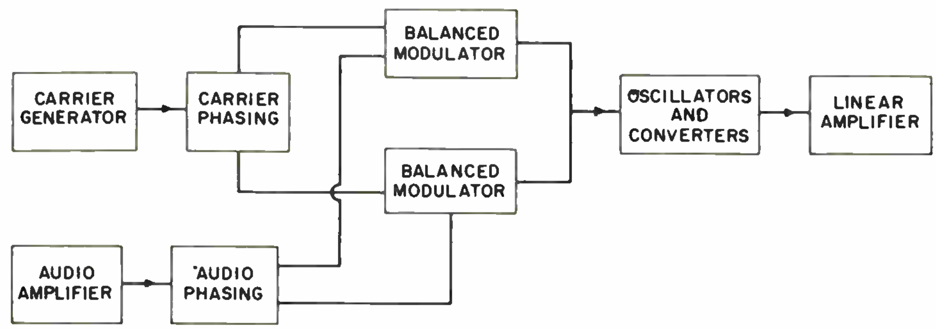
42. A ratio detector.



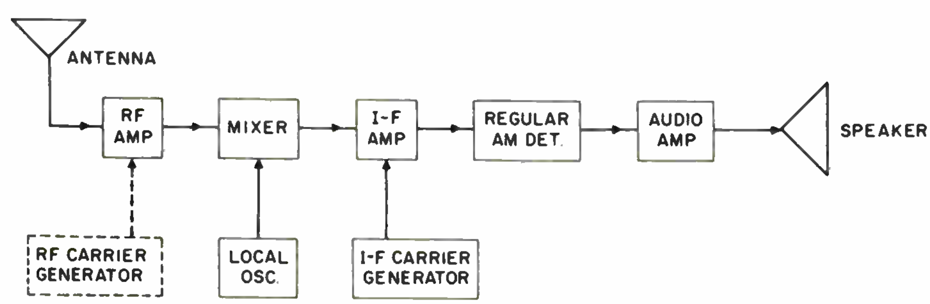
43. A block diagram of a double-conversion FM receiver.



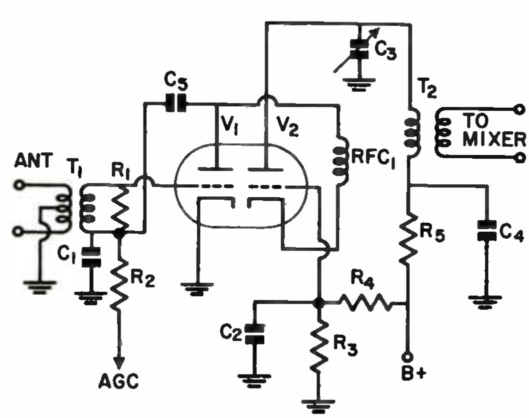
44. A block diagram of a phase-shift-type of single sideband generator.



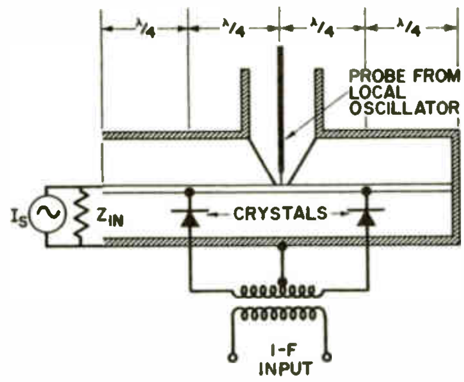
45. A block diagram of a typical carrier-insertion-type single sideband receiver.



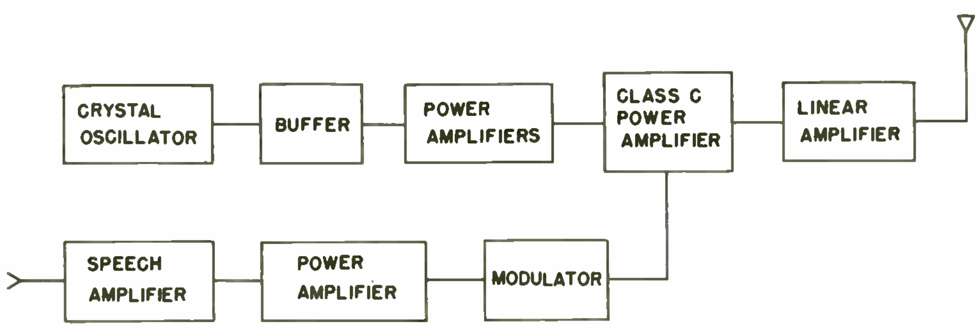
46. A cascode amplifier.



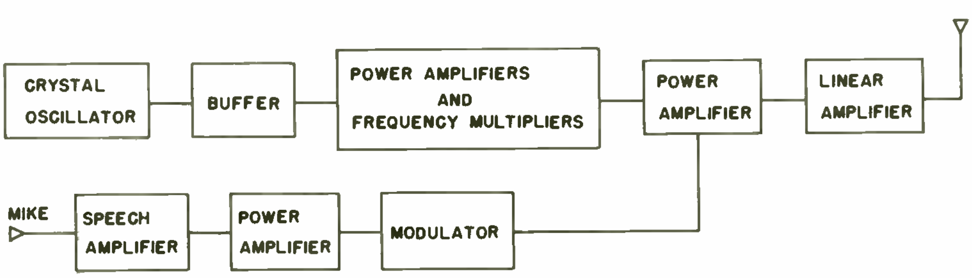
47. A balanced mixer circuit typical of those used in microwave receivers.



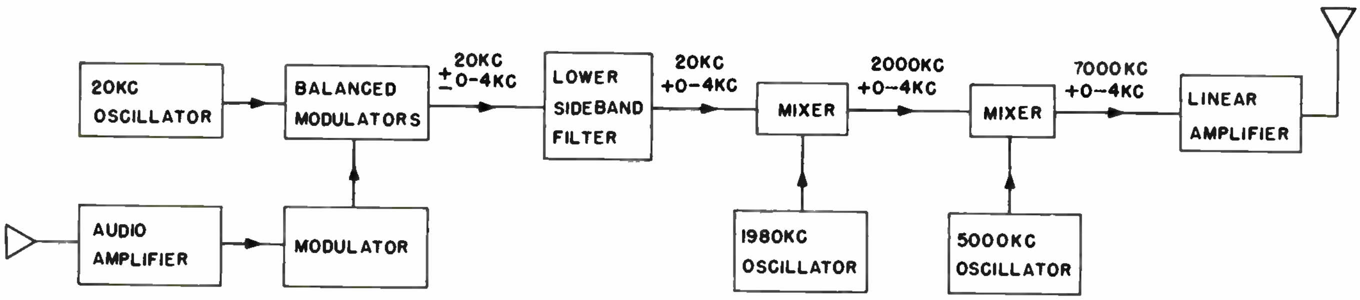
48. A block diagram of an AM broadcast transmitter using low-level modulation.



49. A block diagram of an AM transmitter.



50. A block diagram of an SSSC transmitter (filter type) with a 20 kc oscillator and an emission frequency in the range of 7 mc.



TCA-9

YOUR INSTRUCTOR'S MODEL ANSWERS

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ANSWERS TO MULTIPLE-CHOICE EXAMINATION TC-10

1. (2) The amount of feedback applied is divided by means of a capacitor-divider network.

2. (1) The forward bias higher on the Class A amplifier. A Class A amplifier is a linear amplifier and in the case of a push-pull amplifier the signal will cause the signal current to increase in one transistor and decrease in the other. In the Class B amplifier the signal drives one transistor into the conduction region while the other transistor is cut off. Thus, the Class A amplifier is biased essentially in the linear portion of its characteristic curve, while there is little or no forward bias on the Class B amplifier.

3. (4) 1.5 volts

4. (1) Lead-peroxide

5. (2) Sponge lead

6. (4) To cancel the feedback through the collector-base capacity of the transistor. If neutralization is not employed, there may be enough energy fed from the output circuit, through the collector-base capacity of the transistor, back into the input circuit to cause the stage to go into self oscillation. The feedback energy fed through the neutralizing capacitor is 180° out-of-phase with the energy fed through the collector-base capacity of the transistor, and will therefore cancel the feedback signal and provide for stable operation of the stage.

7. (3) A diluted solution of sulphuric acid.

8. (3) Can be repaired by soldering a piece of wire across the break. Sometimes it is not even necessary to use a piece of wire, you can simply bridge the break, where it is cracked, by flowing solder across the break. However, when solder won't flow across the break, and where the open in the circuit may be of some length due to the fact that the conductor has come off the etched board, you can solder a piece of wire across the open circuit and continue to use the board.

9. (3) Use a low wattage soldering iron to avoid heating the connection any more than necessary. If you apply too much heat to the connection, the bond between the copper and the board will be destroyed and the copper conductor will come up off the circuit board.

10. (2) When the vehicle is in motion. A hairline crack in a printed circuit board is often difficult to locate because the crack may be so fine that a contact is normally made across the cracked conductors. However, when the board is flexed, as it may be when it is in a piece of mobile equipment, when there is some vibration, the contacts break intermittently and cause intermittent operation of the equipment. You can usually find this type of defect by flexing the board, by pushing on it with a pencil or some other similar object that will stress the break. Sometimes you can determine which circuit is opening visually, other times you must check measurements as you flex the board. Usually the intermittent hairline crack can be repaired simply by flowing solder across the break.

11. (3) The chemical action of the electrolyte on two dissimilar metals. In effect, there is a short-circuit battery connection within the plates of the cell that does not produce useful current in the external circuit. Local action causes the battery to discharge itself more rapidly than if the grids were made of pure lead. (Local action occurs when the grids are made of an alloy.)

Local action is sometimes confused with another process called sulphation. This is the formation of lead sulphate on the positive and negative plates of the battery during discharge. It is a normal process in the lead-acid cell that is caused by sulphuric acid molecules combining with lead oxide and sponge lead to form lead sulphate. If proper charging is neglected, the sulphate eventually hardens on the surface of the plate and prevents proper contact of the electrolyte with the active material of the plate. Sulphation is increased by allowing the battery to remain in a discharged condition or by adding acid instead of properly charging cells. The effects of sulphation are reduced terminal voltage, increased terminal resistance, reduced power input, and possible buckling of the plates.

12. (5) 1.300. When completely discharged its specific gravity is about 1.100.

13. (3) To tune for a zero beat, read the dial of the frequency meter, and then refer to the calibration book to determine the frequency of the incoming signal.

14. (1) Check the calibration of the frequency meter.

15. (5) To enable the operator to take frequency measurements accurate to a few cycles per second. This is required at the low frequency measurements, in the broadcast band, for example, where the tolerance allowed is only a few cycles. The human ear cannot hear frequencies below 20 cycles per second, so when you want to take a frequency reading accurate to 1 or 2 cycles, you turn on the audio oscillator and this will produce a tone that rises and falls at a rate equal to the difference between the incoming signal and the signal produced by the variable oscillator in the heterodyne frequency meter. Thus, when the tone becomes constant, the signal produced by the frequency meter is exactly the same as the frequency of the incoming signal.

16. (1) 1.95 volts. The open-circuit voltage (no load) should be about 2.1 volts.

17. (4) Flaked-nickel and nickel-hydrate.

18. (2) Black iron-oxide.

19. (3) A 21% solution of potassium and lithium hydroxide.

20. (2) Very similar to an FM receiver.

21. (2) Produce an i-f signal.

22. (5) A peak reading meter that is calibrated to read the deviation. The reading on the meter will be a function of the deviation because the greater the deviation, the greater the output voltage that will be produced by the discriminator. The meter is simply calibrated to read directly the deviation of the incoming signal.

23. (4) A field-strength meter. The strength of the fundamental is measured as one point some distance from the transmitter and the strength of the harmonics are measured at the same point. This will enable you to determine the amount of attenuation of the second or any other harmonic of the transmitter.

24. (2) Prevent interference between stations. The harmonic of a station may interfere with another station operating on some multiple of its fundamental frequency. Therefore it is important that the second harmonic, and higher order harmonics, are kept as low as possible to prevent interference.

25. (2) Because the amount of feedback in the oscillator in the grid-dip meter decreases. This happens because the resonant circuit under test begins to take power from the grid-dip oscillator. Thus the available feedback signal goes down and the grid current flowing in the oscillator tube in the grid-dip meter decreases. This causes the meter reading to dip.

26. (3) It enables you to set the tuning in each stage to the approximate setting, so you can be sure doublers are doubling and triplers are tripling.

27. (4) Check for vhf parasitics. With the power removed from the transmitter, the grid-dip meter can be brought near the final amplifier resonant circuit and you can check for dips in the vhf region. If the grid-dip meter dips at a certain frequency in the vhf region, you can be sure that there is a vhf resonant circuit present, which will probably cause parasitics unless steps are taken to prevent them.

28. (5) Check the exact frequency of the transmitter. An absorption-type meter gives you an approximation rather than an exact reading of the transmitter frequency. Therefore it can be used to be sure that a doubler is doubling and a tripler is tripling, but you can't measure to the exact frequency output of these stages. You can also use the meter to check for vhf parasitics and determine approximately the frequency of the parasitics.

29. (3) An accurate crystal-controlled oscillator that can be used to check the accuracy of other frequency measuring equipment in the station.

30. (3) The varying pressure of the sound waves causes the resistance of the carbon button to change. The alternate increase and decrease in pressure causes the carbon particles to be first packed together more closely and lowers the resistance and then allows them to spread out slightly, increasing the resistance of the button. This causes the current flowing in the circuit due to an external battery to vary and this varying current flows through the primary winding of the microphone transformer.

31. (1) On the piezoelectric effect. The varying sound waves place a mechanical stress on a small crystal slab and this in turn generates a small electric voltage.

32. (4) It does not require an external power supply.

33. (3) A secondary cell can be recharged, but a primary cell cannot.

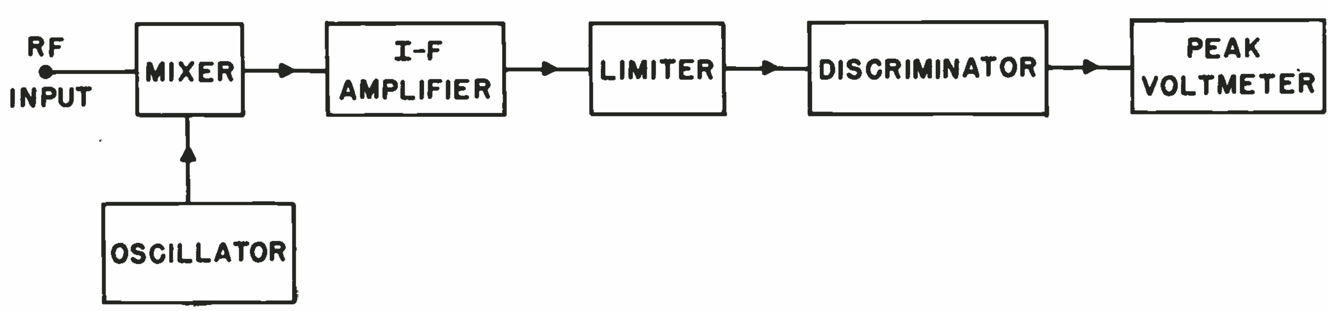
34. (3) Store the cell fully charged, and check its condition periodically, recharging it if it is necessary.

35. (5) Excessive heat may be developed causing the plates to warp and short.

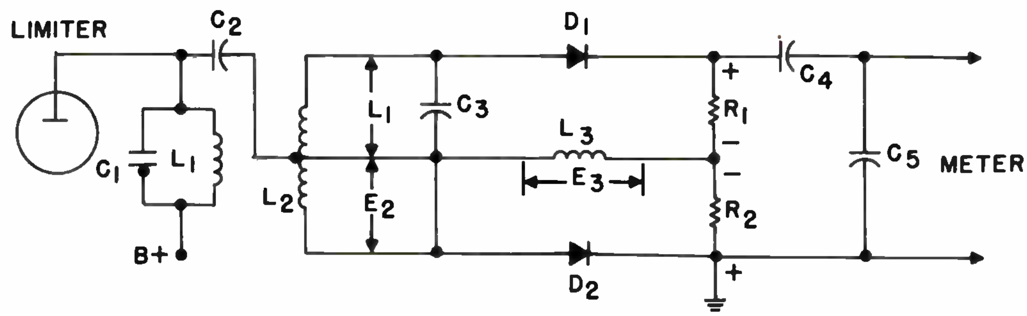
36. (1) 2.1 volts.

37. (4) It tells us how long we can draw a certain current from a battery; we can determine this by dividing the current drawn into 200. For example, in the case of a 200 ampere hour battery, if we are drawing a current of 5 amperes from the battery, we can divide 5 into 200 and find that we can draw this current for 40 hours before the battery is fully discharged. If we are drawing a current of 20 amperes, we could draw it for 200 ÷ 20 = 10 hours.

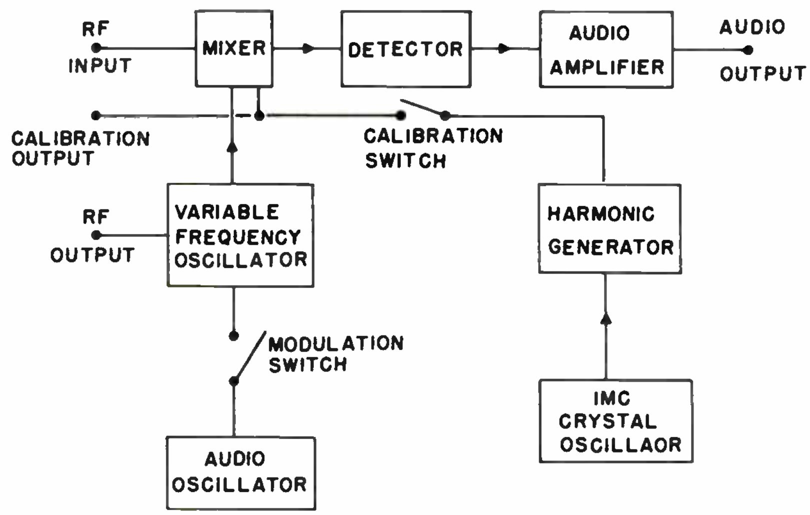
38. A block diagram of an FM deviation meter with each stage labeled.



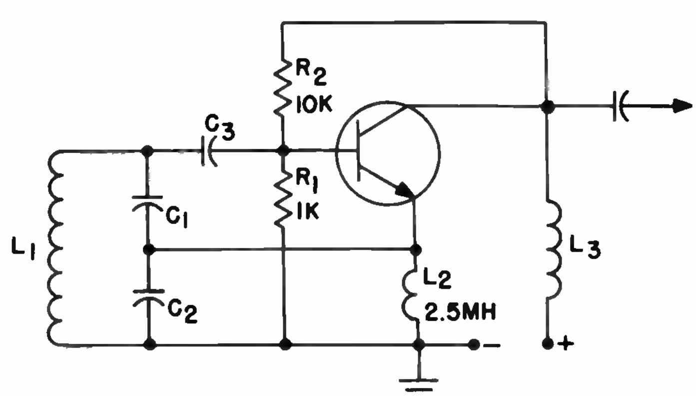
39. A schematic diagram of the discriminator circuit of an FM deviation meter showing how the discriminator is coupled to the limiter stage.



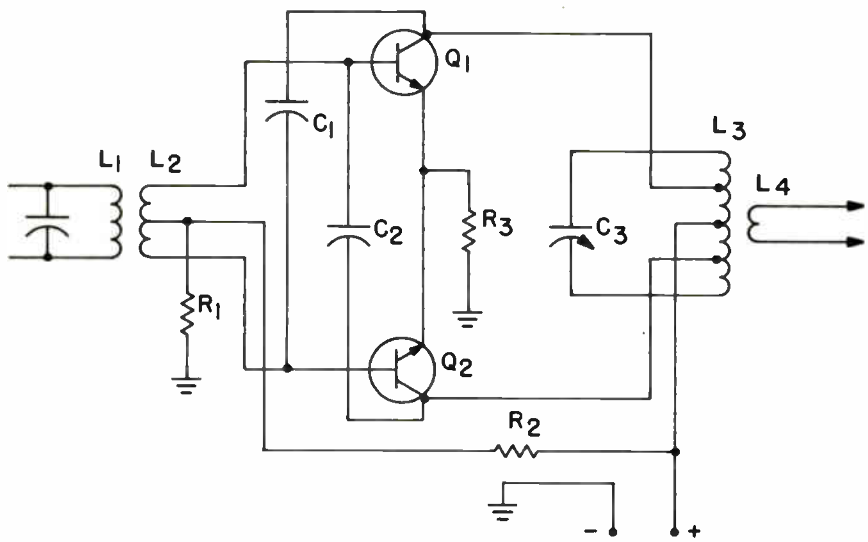
40. A block diagram of a heterodyne frequency meter with each stage in the meter labeled.



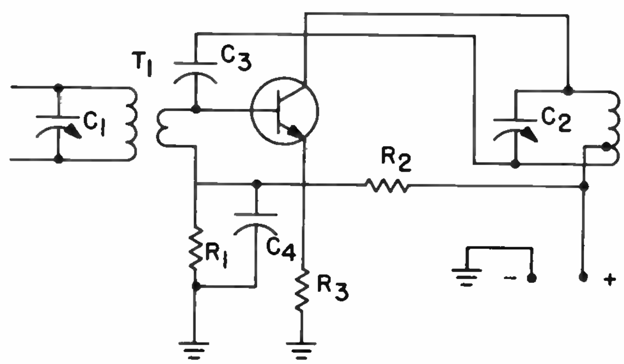
41. A schematic diagram of a Colpitt's type oscillator using a transistor.



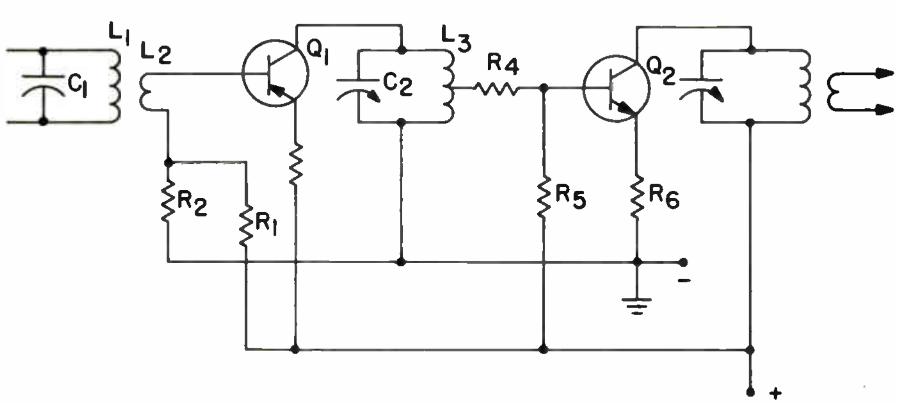
42. A schematic diagram of a push-pull Class B rf power amplifier using transistors.



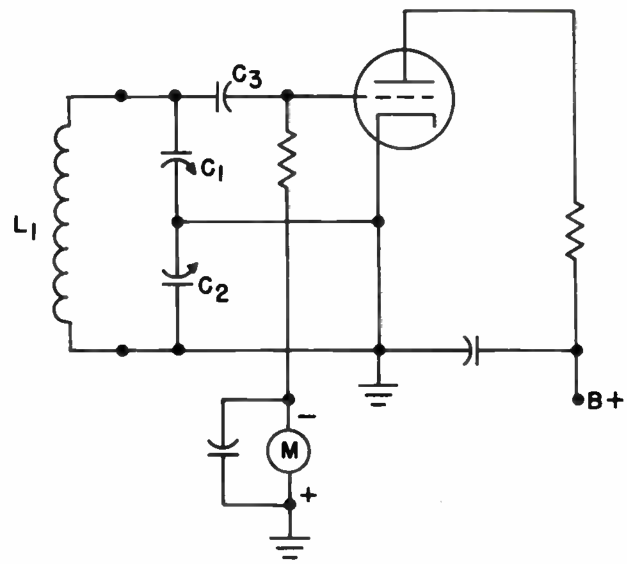
43. A schematic diagram of a common-emitter rf amplifier.



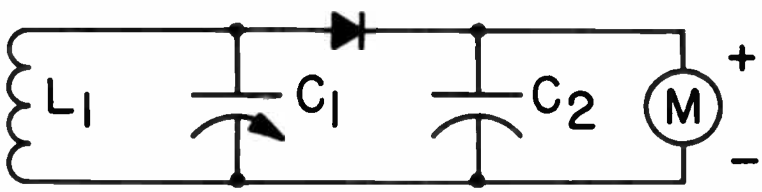
44. A schematic diagram of a PNP rf amplifier directly coupled to an NPN transistor used as an rf amplifier.



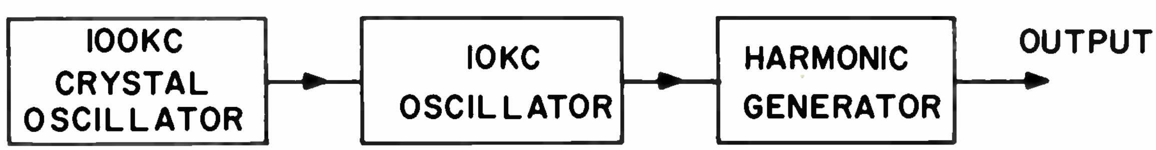
45. A schematic diagram of a grid-dip meter.



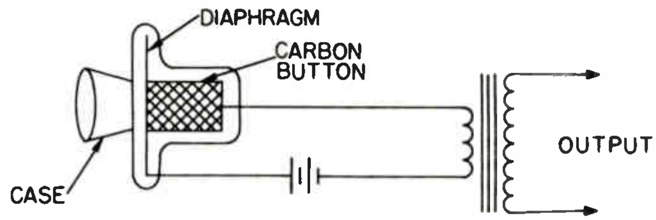
46. A simplified circuit diagram of an absorption-type wavemeter with a galvonometer-type indicator.



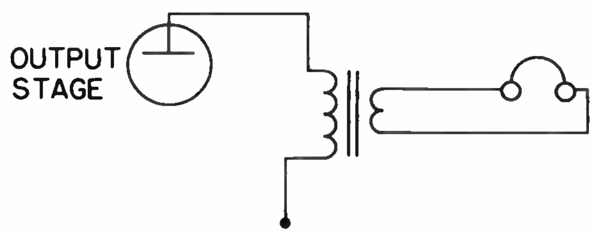
47. A block diagram of a secondary frequency standard with the various stages labeled.



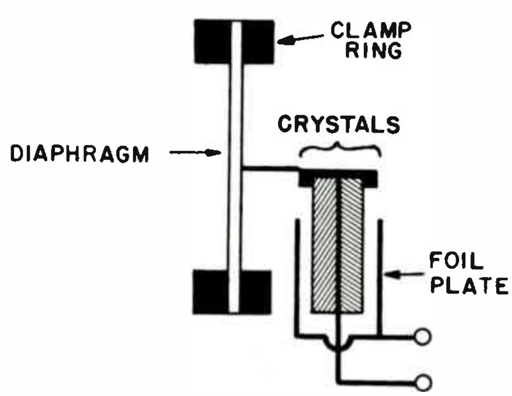
48. A diagram of a single-button carbon microphone circuit showing the source of power in the microphone transformer.



49. A schematic diagram showing how a low impedance pair of headphones can be connected to the output of a vacuum tube amplifier.



50. A simplified diagram of a crystal microphone.



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ANSWERS TO MULTIPLE-CHOICE EXAMINATION TC-11

1. (1) It directs current through each armature coil in the proper direction to provide a torque acting to turn the armature in the required direction.

2. (2) Fine sandpaper. Never use anything such as emery cloth that may short the commutator bars or get into the winding.

3. (3) Reduce brush sparking. The interpoles prevent the neutral plane from shifting away from the brush position when the load changes.

4. (4) Brushes in neutral plane.

5. (5) Increasing the field current by adjusting a rheostat in series with the shunt field increases speed. On the contrary, this decreases speed.

6. (3) It has a constant speed with widely varying loads. If the load is light or if there is no load, speed may increase to a point where the machine virtually destroys itself. As the load on the motor increases, the speed goes down.

7. (1) 3600 rpm. The motor has four poles per phase or two pairs of poles. To find the speed, we can use the formula:

$$N = \frac{f \times 60}{p}$$

where N = the synchronous speed in rpm, f is the frequency of the power line and p is the number of pairs of poles. Substituting we have:

$$N = \frac{120 \times 60}{2} = 3600 \text{ rpm}$$

8. (5) 1750 rpm. The synchronous speed of a 4-pole induction motor operating on a 60-cycle power line will be 1800 rpm. However, an induction motor operates at a speed slightly less than the synchronous speed, therefore the choice of 1750 rpm is the only applicable one.

9. (4) Residual magnetism in the generator pole pieces.

10. (4) Varying the strength of the separately excited field. The field in an ac generator is excited by dc and usually some method is provided for controlling the current flowing through the field. This enables the operator to vary the strength of the field and hence control the output voltage from the generator.

11. (5) Insufficient or incorrect lubrication of the bearing.

12. (2) The field has opened.

13. (4) It gives you the maximum rated dissipation of the transistor with the surrounding air temperature of 25°C.

14. (5) The emitter-to-base voltage must not exceed -5.0 volts and the collector current must not exceed 10 ma.

15. (3) A half-wave antenna.

16. (1) Is one-quarter wavelength long.

17. (3) Must be made of a suitable high strength material.

18. (1) Always equal to one.

19. (5) .833. The power factor is equal to the true power measured by the wattmeter divided by the apparent power as given by the product of the voltage times the current. Thus we have 500 ÷ 600 = .833.

20. (3) A dynamotor is a motor-generator combination with a single field and two armature windings.

21. (4) The dc output from the generator is comparatively easy to filter.

22. (1) It is comparatively easy to get more than one dc operating voltage from the output.

23. (5) The automobile ignition system.

24. (5) UHF

25. (4) The electrons travelling down the tube in bunches.

26. (1) Control the amplitude of the signal fed from the output back to the input.

27. (5) All four of the preceding answers are correct.

28. (2) It is used to keep the electron beam going down the exact center of the tube.

29. (4) Energy is fed from the electron stream into the helix.

30. (5) The dimensions of the anode cavity.

31. (3) One dimension must be greater than one-half wavelength and the other less than one-half wavelength.

32. (4) By both electric and magnetic fields flowing down the guide.

33. (5) Both (3) and (4) are correct.

34. (4) Moisture may collect in the guide and introduce excessive attenuation.

35. (2) Successive groups of electrons passing by the entrance of the cavity at the correct time to reinforce the oscillation.

36. (4) To prevent moisture absorption that would result in excessive attenuation in the guide.

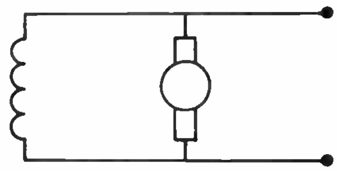
37. (5) The reading on M_1 would drop to zero and the reading on M_2 would increase.

38. (2) The reading on M_1 would remain the same, the reading on M_2 would decrease.

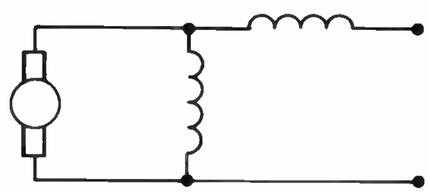
39. (3) The collector voltage will increase. If R_1 opens, the forward bias placed across the emitter-base junction will be removed. This will cause the current flow through the transistor to decrease and therefore the current flowing through the collector resistor will decrease. This will cause the voltage between the collector of the transistor and ground to increase.

40. (4) It is used to stabilize the operation of the transistor with changes in temperature.

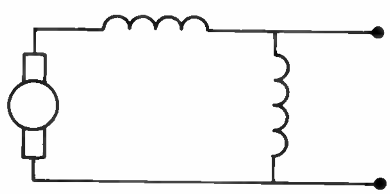
41. A schematic diagram of a shunt generator.



42. A schematic diagram of a compound generator. Either diagram is correct.

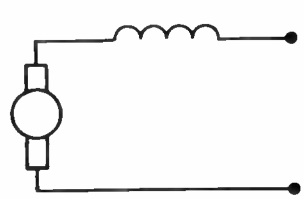


SHORT
SHUNT

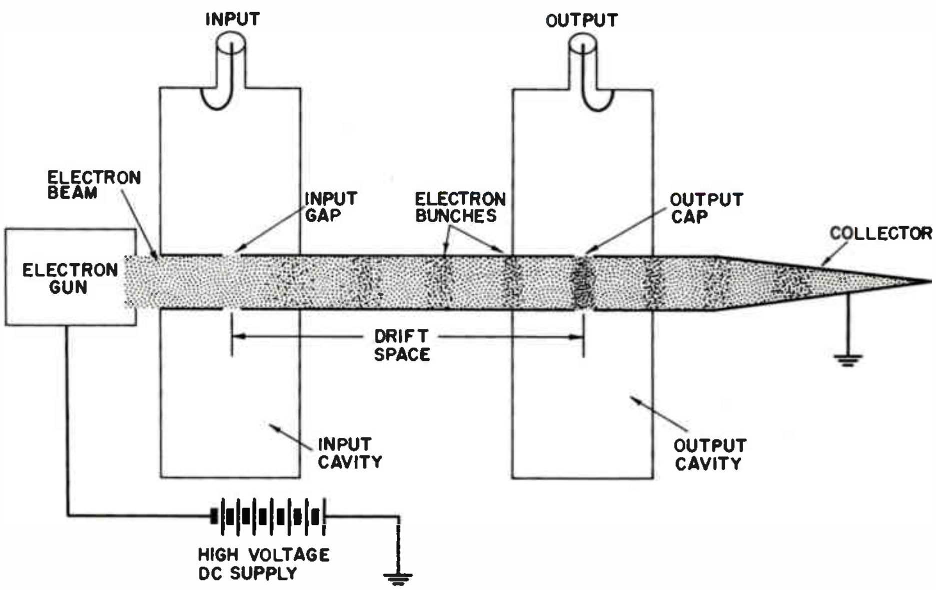


LONG
SHUNT

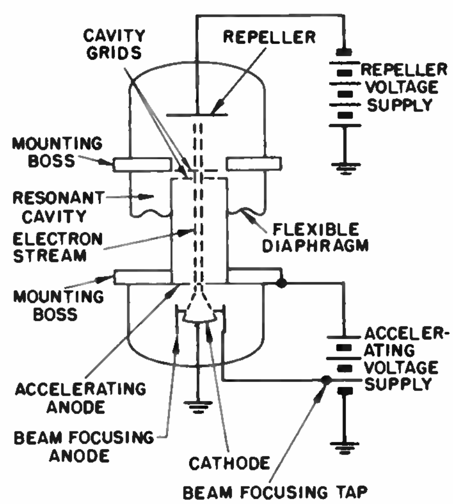
43. A schematic diagram of a series motor.



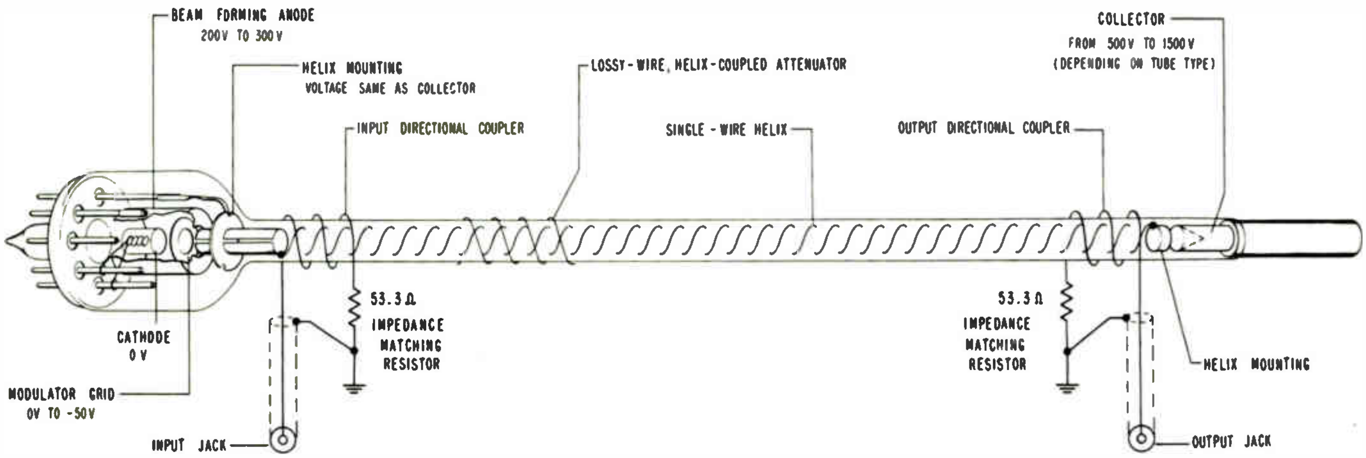
44. A diagram of a basic two-cavity klystron tube.



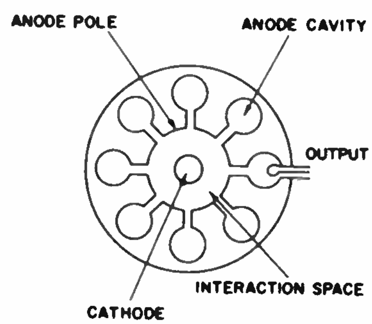
45. A diagram of a reflex klystron showing the polarity of the voltage applied to the repeller.



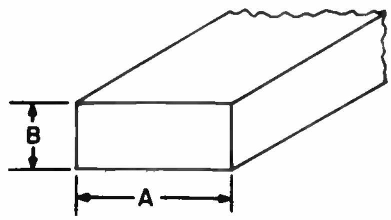
46. A sketch of a travelling-wave tube.



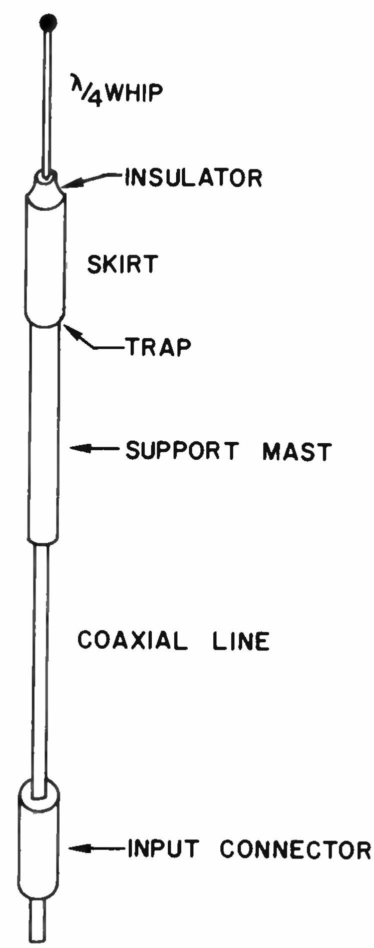
47. A sketch showing a cross-section of a magnetron.



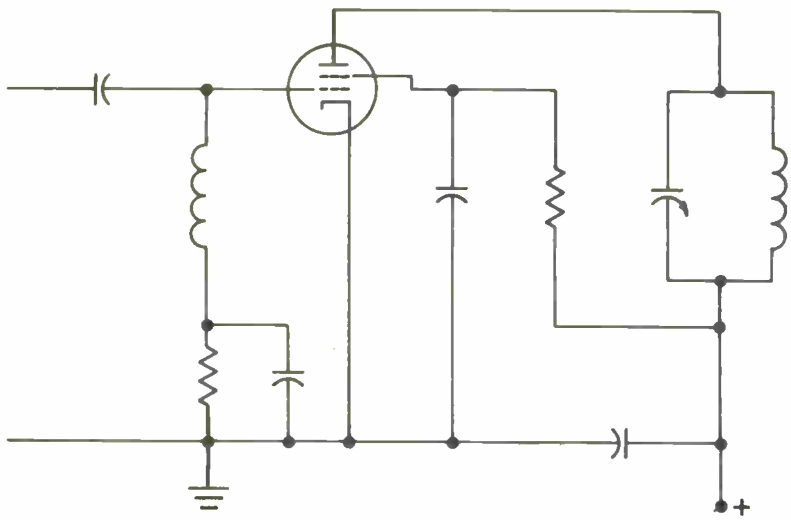
48. A sketch of a waveguide with the critical dimensions labeled.



49. A sketch of a coaxial-whip antenna identifying the various parts.



50. A schematic diagram of a Class B rf power amplifier using a vacuum tube.



YOUR INSTRUCTOR'S MODEL ANSWERS

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ANSWERS TO MULTIPLE-CHOICE EXAMINATION TC-12

1. (3) The voltage leads the current by 90°.

2. (1) 12 ohms. To find the impedance of a circuit of this type we must first find the impedance of the parallel branch. To do this we assume any convenient voltage across the capacitor and the coil. In this case, 20 volts is a convenient voltage to assume. With this assumption, we can then find that the current through the capacitor is:

$$I_C = \frac{20}{20} = 1 \text{ amp,}$$

and the current through the inductance can be found by:

$$I_L = \frac{20}{5} = 4 \text{ amps.}$$

We know that the capacitive current will be 180° out-of-phase with the inductive current. Therefore the actual current flowing in the parallel circuit will be the difference between these two currents. Thus we have the current $I_t = 4 - 1 = 3$ amps. Now that we know the current flowing to the parallel circuit we can find the impedance of the parallel circuit by dividing the current into the voltage. Thus we get:

$$Z = \frac{20}{3} = 6.67 \text{ ohms.}$$

To find the total impedance of the circuit we use the formula:

$$Z = \sqrt{R^2 + X^2}$$

and substituting for R and X we have:

$$Z = \sqrt{10^2 + 6.67^2}$$

$$Z = \sqrt{100 + 44.4}$$

$$= \sqrt{144.4}$$

$$= 12 \text{ ohms.}$$

3. (2) 1620 kHz.

4. (1) 20 volts. To find the voltage across the capacitor, you must first find the impedance of the circuit and then find the current flowing in the circuit. The reactance of the capacitor and inductance in series, since they are opposites, will be equal to the difference between the two. Thus the reactive component has a reactance of 10 – 6 = 4 ohms. Now to find the impedance we substitute in the formula:

$$Z = \sqrt{R^2 + X^2}$$

and substituting for R and X in the formula we get:

$$Z = \sqrt{3^2 + 4^2}$$

$$Z = \sqrt{9 + 16}$$

$$= \sqrt{25}$$

$$= 5 \text{ ohms.}$$

Thus with a voltage of 10 volts across the circuit, the current flowing in the circuit will be 10 volts divided by 5 ohms equals 2 amps. This means that the voltage across the capacitor will be 10 ohms X 2 amps = 20 volts.

5. (3) A noise caused by irregular current flow through a diode.

6. (4) The gain of a transistor in a common-base circuit. Alpha is given by the formula:

$$\alpha = \frac{\Delta I_C}{\Delta I_E}$$

7. (5) 910,021 Hz. The authorized frequency tolerance of a station in the broadcast band is 20 Hz. Thus the station may operate as much as 20 Hz above 910 kHz or 20 Hz below 910 kHz and still be within its assigned tolerance. The only one of the frequencies listed that is more than 20 Hz from 910 kHz is 910,021 Hz.

8. (5) They are susceptible to damage by voltage transients.

9. (2) 30 ohms. If R_1 has one-third the resistance of R_2, then the ratio of R_2 over R_1 must be 3. Substituting this in the formula for:

$$R_4 = \frac{R_2}{R_1} \times R_3$$

we will get:

$$R_4 = 3 \times 10 = 30 \text{ ohms.}$$

10. (5) It is reasonably rugged.

11. (5) All of the preceding answers are correct.

12. (2) 3.18 mfd. If the cathode bias resistor is 5000 ohms, then we assume that the reactance of the cathode bypass capacitor should be one-tenth this value or 500 ohms. We know that the reactance of a capacitor is given by the formula:

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

This formula can be arranged so that we have:

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

and substituting in this formula we get:

$$C = \frac{1}{6.28 \times 100 \times 500}$$

$$C = \frac{.159}{50{,}000}$$

$$C = \frac{15.9 \times 10^{-2}}{5 \times 10^{4}}$$

$$= 3.18 \times 10^{-6}$$

$$= 3.18 \text{ mfd}$$

13. (4) It is less expensive.

14. (3) It has a higher output.

15. (4) Picks up equally well in all directions.

16. (1) 8.3 The voltage gain of an amplifier can be found using the formula:

$$VG = \frac{\mu \times R_L}{R_P + R_L}$$

and substituting for μ, R_L and R_P in the formula we have:

$$VG = \frac{10 \times 50{,}000}{10{,}000 + 50{,}000}$$

$$VG = \frac{500{,}000}{60{,}000}$$

$$= 8.3$$

For further information on this problem see Example 258 in Study Guide 12.

17. (5) 13 henrys. The inductance of series-connected coils whose fields aid can be obtained from the formula:

$$L_T = L_1 + L_2 + 2M$$

and substituting for L_1, L_2 and the mutual inductance we get:

$$L_T = 10 + 2 + 2 \times .5$$

$$L_T = 10 + 2 + 1$$

$$= 13 \text{ henrys.}$$

18. (2) Beta. The beta of a transistor is given by:

$$\beta = \frac{\Delta I_C}{\Delta I_B}$$

19. (3) The audio frequency range.

20. (4) 3 db. You can find the loss using the formula:

$$db = 10 \log \frac{P_1}{P_2}$$

and substituting P_1 and P_2 we have:

$$db = 10 \log \frac{100}{50}$$

$$db = 10 \log 2$$

$$db = 10 \times .3 = 3$$

21. (3) Send the completed application and fee to the nearest FCC Field Office.

22. (1) So that the transistor can be soldered in the circuit quickly. This is important because if it takes too long to solder the transistor into the circuit there will be considerable heat transmitted up the transistor leads to the junction, and the possibility that one of the junctions may be damaged by the heat.

23. (4) Solder a piece of hookup wire across the break in the conductor.

24. (2) By variations in air pressure.

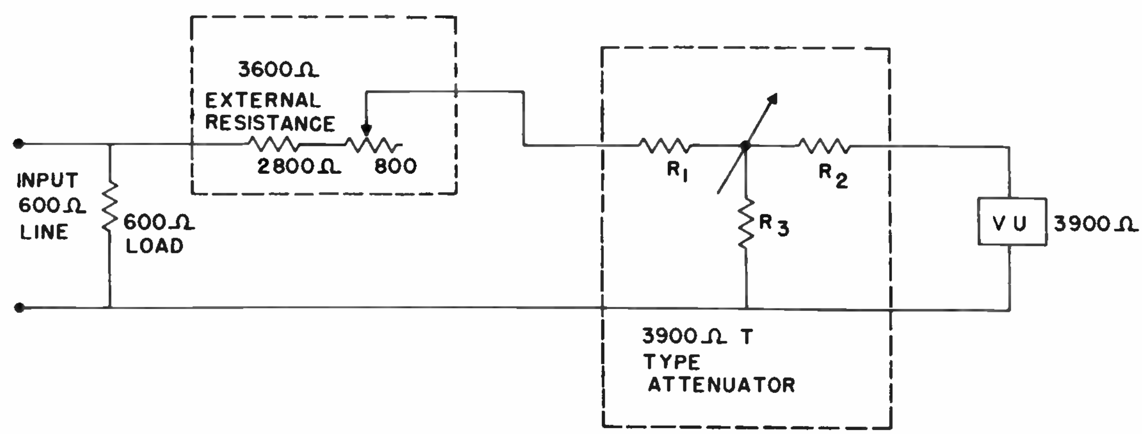
25. (5) A small permanent-magnet dynamic speaker.

26. (5) Five years from date of issuance.

27. (3) Normal carbon hiss from the microphone.

28. (1) Because they are designed for use with 600-ohm transmission lines.

29.



30. (3) To prevent hum and noise pickup in the transmission line.

31. (5) The original must be posted at duty point or kept on operator's person according to class of station requirements.

32. (3) By connecting them so their output signals are in-phase.

33. (3) Reduce distortion and improve stability.

34. (2) Voltage and current feedback.

35. (1) Send a written reply within ten days to the notifying FCC office. Reply should explain the situation, remedial action taken to prevent recurrence and name and license number of operator.

36. (4) The feedback voltage must be 180° out-of-phase with the input voltage.

37. (4) 3.33%. To find the percentage feedback we use the formula:

$$a = \frac{A}{1 + BA}$$

where:

a = gain with feedback
A = gain without feedback
B = fraction of output voltage fed back.

We rearrange this formula by cross-multiplying and get:

$$A = a + BAa$$

Rearranging we have:

$$BAa = A - a$$

Dividing through by Aa we get:

$$B = \frac{A}{Aa} - \frac{a}{Aa}$$

$$B = \frac{1}{a} - \frac{1}{A}$$

and substituting the gain with feedback 15 for "a" and the gain without feedback, 30 for "A", we get:

$$B = \frac{1}{15} - \frac{1}{30}$$

and bringing these to the common denominator which is 30 we have:

$$B = \frac{2 - 1}{30}$$

$$B = \frac{1}{30}$$

Now to express this as percentage we multiply by 100 and we have:

$$B = \frac{1}{30} \times 100$$

$$= 3.33\%$$

38. (1) 8.33. To find the gain with feedback we use the formula:

$$a = \frac{A}{1 + BA}$$

where:

a = gain with feedback
A = gain without feedback
B = fraction of output voltage fed back

Substituting 50 for A and 1/10 for B we have:

$$a = \frac{50}{1 + 1/10 \times 50}$$

$$a = \frac{50}{1 + 5}$$

$$a = \frac{50}{6}$$

$$= 8.33$$

39. (2) 16.7. To solve this problem we first convert the percentage feedback into a fraction. 4% feedback equals:

$$\frac{4}{100} = \frac{1}{25}$$

Now we substitute the gain of the amplifier without feedback and the percentage of feedback into the formula:

$$A = \frac{A}{1 + BA}$$

where:

a = gain with feedback
A = gain without feedback
B = fraction of output voltage fed back

and substituting 50 for A and 1/25 for B, we get:

$$a = \frac{50}{1 + 1/25 \times 50}$$

$$a = \frac{50}{1 + 2}$$

$$a = \frac{50}{3}$$

$$= 16.7$$

40. (3) Produce a low-frequency distortion referred to as "wow".

41. (2) Within fifteen days send an application for a hearing to the FCC.

42. (4) The ability of the amplifier to produce an amplified version of the input without distortion.

43. (4) It provides good reproduction with a minimum of distortion.

44. (5) a high-frequency distortion that may be caused by dirt on a tape playback head.

45. (3) A low-frequency distortion caused by turntable vibration.

46. (4) Incorrect bias current in the tape head or a defect in the amplifier.

47. (1) 13 db. To find the gain we use the power formula:

$$db = 10 \log \frac{P_2}{P_1}$$

and substituting 100 for P_2 and 5 for P_1 we get:

$$db = 10 \log \frac{100}{5}$$

$$db = 10 \log 20$$

$$db = 10 \times 1.3$$

$$= 13$$

48. (3) 100 ohms. To find the impedance of this circuit we use the formula:

$$Z = \sqrt{R^2 + X^2}$$

$$Z = \sqrt{60^2 + (100 - 20)^2}$$

$$Z = \sqrt{3600 + 6400}$$

$$Z = \sqrt{10{,}000}$$

$$= 100 \text{ ohms}$$

49. (2) The reading on M_2 will increase. The bias for the stage is developed across the resistor in the grid circuit. If the rf input disappears, there will be no bias developed and therefore the current through the tube will increase to a high value.

50. (2) R_2 provides forward bias across the emitter-base junction and a small amount of degeneration.

National Radio Institute

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ANSWERS TO MULTIPLE-CHOICE EXAMINATION TC-13

1. (1) The operator on duty at the transmitter.

2. (4) The turntable speed is varying slightly above and slightly below the correct rate.

3. (3) A test record and measure the output from the amplifier at various frequencies.

4. (4) An amplifier that has a variable gain which prevents overmodulation at high amplitude levels.

5. (2) It compensates for unequal frequency attenuation in the line.

6. (4) A fine of not more than $10,000 and/or not more than one year in prison or both.

7. (1) A high Q series-resonant circuit because the equivalent inductance is high compared to the equivalent resistance.

8. (3) They are amplifiers that automatically compensate for signals of different amplitude to prevent overmodulation of the transmitter.

9. (3) By means of a biased diode which conducts when the signal exceeds a certain level and using the signal developed to control the gain of an amplifier.

10. (5) The dimensions and type of cut.

11. (3) An amplifier containing a diode that develops the signal which is used to compress the dynamic range of a broadcast.

12. (4) It is a very sensitive type of thermostat used to control the temperature of a crystal oven.

13. (1) It is less expensive.

14. (3) They should be painted with alternate equal bands of aviation surface orange and white paint, with widths ranging between 1-1/2 and 40 feet.

15. (4) 50%. The maximum theoretical efficiency of any Class A power amplifier is 50%. However, in practical applications the efficiency is usually considerably less than this and often runs as low as 10 to 12%.

16. (4) Their operating efficiency is too low.

17. (5) If the drive disappears, the plate current will rise to a high value and destroy the tube.

18. (3) A push-push frequency doubler.

19. (1) It usually does not require neutralization.

20. (3) 12-1/2 watts. To find the power required we use the formula:

$$P = \frac{m^2}{2} \times P_C$$

where P is the audio power required, m is the modulation index and P_C is the rf power.

To find the modulation index, we divide the percentage of modulation by 100, therefore the modulation index in this case is 50 divided by 100 which equals .5. Now substituting this in the formula we have:

$$P = \frac{.5 \times .5}{2} \times 100$$

$$= \frac{25}{2}$$

$$= 12\text{-}1/2 \text{ watts}$$

21. (5) The maximum plate current flow through the tube occurs at the time of minimum plate voltage and hence the tube losses are low.

22. (5) From the General Services Administration, Washington, D. C. 20407.

23. (1) 15,000 ohms. The size of the bias resistor can be found using Ohm's Law:

$$R = \frac{E}{I}$$

We are told that the grid-leak bias should be –45 volts and that the grid current is 3 ma. Converting 3 ma to amps we get .003 amps and substituting these values in the formula we have:

$$R = \frac{45}{.003}$$

and multiplying the top and the bottom by 1000 to get rid of the decimal we have:

$$R = \frac{45{,}000}{3}$$

$$= 15{,}000 \text{ ohms}$$

24. (2) A high operating bias is used to reduce the operating angle of the stage.

25. (3) The plate current increases during modulation indicating nonsymmetrical modulation.

26. (2) 5000 watts. Using the indirect method, we determine the power output by multiplying the input plate voltage of the final stage times the plate current, times the efficiency. Thus we have:

$$P = 10{,}000 \times 2 \times .25$$

$$= 5000 \text{ watts.}$$

27. (3) Call sign of the station tested, then the word "testing" followed by the count 1, 2, 3, 4, etc.

28. (4) They are safety devices used to protect the operator from the high voltages of the transmitter.

29. (4) The plate-current overload relay would function and remove the plate voltage from the output stage.

30. (4) The higher frequency signal will cause the resting frequency to change at a faster rate.

31. (5) 150 kHz. The maximum deviation is 75 kHz. This means that the transmitter frequency may swing 75 kHz below and 75 kHz above the resting frequency for a total frequency swing of 150 kHz.

32. (5) It improves the signal-to-noise ratio at the high audio frequencies.

33. (4) 50 kw. If the antenna gain is 14 db and the line loss is 4 db, the overall gain of the transmission line and antenna will be 14 - 4 = 10 db. Now using the formula:

$$db = 10 \log \frac{P_2}{P_1}$$

and substituting 10 for db we have:

$$10 = 10 \log \frac{P_2}{P_1}$$

or,

$$\log \frac{P_2}{P_2} = 1$$

Now we need to determine what number has the logarithm of 1. The answer to this is 10. Therefore the antenna has a gain of 10 so that the effective radiated power will be 5 kw times 10 equals 50 kw.

34. (3) As often as necessary to maintain good visibility.

35. (4) On the top of the highest building or hill in the area.

36. (1) In the direction along the line A-B. The signal applied to the antenna element nearest A will be radiated and will travel through space as the signal travels down the transmission line to the radiating element B. The two signals will arrive at the same time so they will be in-phase and reinforce each other. Thus maximum radiation will occur along the line A-B. Signals traveling from the antennas A and B in the direction C-D will be 180° out-of-phase with each other and therefore will cancel each other.

37. (2) Construction of a high-rise apartment building with a steel frame.

38. (4) Adjusting the power and phasing of the energy fed to each element.

39. (2) They are used to improve the ground conductivity.

40. (3) By using an adjustable T-type network between the transmission line and the antenna.

41. (3) 50%

42. (2) 15 microvolts per meter

43. (3) North

44. (1) It adds reactive current to the oscillator tank circuit that either increases or decreases the effective inductance or capacity of the circuit.

45. (4) By using a low-frequency oscillator followed by frequency multipliers then a converter which reduces the center frequency without reducing the deviation and then feeding the signal to additional frequency multipliers.

46. (3) There is little or no difference between frequency and phase modulation.

47. (5) The electric field radiated by the antenna is vertical.

48. (2) To get a maximum life from the relay contacts.

49. (4) It isolates the oscillator from the power amplifiers.

50. (4) The final amplifier grid circuit is tuned for minimum grid current.

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ANSWERS TO MULTIPLE-CHOICE EXAMINATION TC-14

1. (4) 5.00 volts. Since the peak-to-peak value of the voltage is 14.14 volts, then the peak value must be 7.07 volts. To convert the peak value to rms value, you multiply by .707; .707 times 7.07 is equal to 5.00 volts.

2. (3) 57.7 volts. In a wye-delta connection, with a 1:1 turns-ratio, there will be a step-down in the voltage in the order of $1 \div \sqrt{3}$ which is .577. Therefore, with the 1:1 turns-ratio, we would have 577 volts across the secondary. However, since we have a 10:1 turns-ratio, the secondary voltage would be 57.7 volts.

3. (4) 10.0 mh. In a parallel-resonant circuit the inductive reactance will be equal to the capacitive reactance. Therefore the inductive reactance must be 100 ohms. To find the value of inductance we use the formula:

$$L = \frac{X_L}{6.28 \times f}$$

and substituting 100 ohms for X_L and 1590 Hz for f we have:

$$L = \frac{100}{6.28 \times 1590}$$

L = .01 henry = 10 mh.

4. (2) A lower turns-ratio is required.

5. (5) Plate modulation is not more desirable; grid modulation is used.

6. (5) 1399 volts. To find the peak-inverse rating of the rectifier tube you must first find peak voltage across the entire transformer winding. Since the voltage across one-half is 500 volts rms, the rms voltage across the entire winding will be 1000 volts. To convert this to the peak value we multiply by 1.414 and get 1,414 volts. From this value we subtract the drop in the rectifier tube, which is 15 volts, and get 1399 volts.

7. (1) To reduce line losses.

8. (2) They introduce high line losses.

9. (4) 1 kHz.

10. (1) Secondary emission.

11. (5) By using an electron multiplier inside of the tube.

12. (2) 160 watts. To find the power in the sidebands, you use the formula:

$$P_{sb} = \frac{m^2 P_C}{2}$$

where m^2 is the modulation index and in this case it will be .8 and P_C is the power in the carrier. Substituting these values in the formula we get:

$$P_{sb} = \frac{.8^2 \times 500}{2}$$

$$= \frac{.64 \times 500}{2}$$

$$= 160 \text{ watts.}$$

13. (5) It is less complex.

14. (2) 49 kHz. The deviation with 100% modulation on an FM station is 75 kHz. Therefore with 65% modulation the deviation should be 75 X .65 = 49 kHz.

15. (3) To reduce flicker.

16. (5) 2000 Hz.

17. (2) 20 kW. The power output of an FM station does not change with modulation.

18. (3) Two years.

19. (3) The station may be operated for 60 days without FCC authority.

20. (3) The color synchronizing pulse has an eight-cycle color burst on the back porch that is not present on the monochrome synchronizing pulse.

21. (4) 75%.

22. (1) Because a sawtooth waveform moves the beam across the face of the tube at linear rate.

23. (5) A studio-to-transmitter link by means of radio transmission.

24. (2) To check the standing wave ratio on the transmission line.

25. (3) To feed the sound and video carrier signals to a single antenna.

26. (4) Four elements, a horizontally polarized pattern.

27. (1) L_1 – 59 ohms, L_2 – 89 ohms. To find the reactance of L_1, we take the square root of the product of the antenna resistance and the characteristic impedance of the transmission line:

$$X_{L1} = \sqrt{R_a Z_o} = \sqrt{77 \times 45} = \sqrt{3465} = 59 \text{ ohms}$$

L_2 must compensate for the 30 ohms of capacitive reactance of the antenna, so it is equal to:

$$X_{L2} = 59 + 30 = 89 \text{ ohms}$$

28. (2) Both the plate and screen.

29. (5) 400 ohms. The reactance of an inductor varies directly with the frequency. Therefore the reactance of the inductor will be five times at 3500 kHz what it is at 700 kHz.

30. (3) 1515 kHz.

31. (1) To isolate the crystal oscillator from the intermediate power amplifier.

32. (3) They help obtain the required frequency deviation.

33. (2) 67 kHz.

34. (4) the Y signal.

35. (1) At least one operator shall be on duty who has a Radiotelephone First Class License.

36. (3) The ratio of picture width to picture height.

37. (5) 10 ohms per volt.

38. (2) 30.

39. (5) 30 microseconds. The time-constant of an R-C circuit is defined as the time it takes the capacitor to charge to 63% of the applied voltage, or in the case of an R-C circuit that is already charged, the time it takes to discharge to 37% of the applied voltage. In this case, since the applied voltage is 100 volts, 37% of the voltage would be 37 volts. The R-C time-constant is obtained by multiplying the capacity of the capacitor in microfarads times the resistance of the resistor in megohms. In this case, we have 3×10^{-6} microfarads $\times$ 10 megohms, so the time-constant is $3 \times 10^{-6} \times 10 = 30$ microseconds.

40. (5) As often as necessary to insure that no deterioration of station performance has occurred.

41. (3) Still resonant at 1000 kHz. Placing two identical inductors in parallel will give an equivalent inductance of one-half that of either inductor. Placing two identical capacitors in parallel gives an equivalent capacitance of twice that of either capacitor. Thus, since the inductance is halved and the capacitance is doubled, the combination is still resonant at 1000 kHz.

42. (1) At intervals not to exceed three months.

43. (2) The station is operating within assigned frequency tolerance established by the FCC.

44. (4) The characteristic impedance of the transmission line will not be affected, so it should be terminated correctly by the antenna.

45. (4) 576 watts. With a power input to the station of 400 watts, and an efficiency of 60%, the rf power output from the transmitter will be 240 watts. So if the efficiency of the transmission line is 80%, 80% of the 240 watts, which is 192 watts, will be delivered to the antenna. Since the antenna has an effective gain of 3, the effective radiated power will be three times 192 watts, which is 576 watts.

46. (3) 2.3 inches.

47. (3) It provides a better signal-to-noise ratio.

48. (1) .1°C.

49. (2) 6 volts.

50. (1) 0 watts.

MULTIPLE-CHOICE EXAMINATION TC-13
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CITY ____________ STATE ________ ZIP ________ GRADE ______________

Read Instructions Carefully Before Answering Questions

FILL OUT AND SEND IN THIS EXAMINATION AS SOON AS YOU HAVE COMPLETED PART XIII OF THE NRI STUDY GUIDE WHICH YOU RECEIVED WITH THIS EXAMINATION

There are 50 problems in this examination which are multiple-choice. These problems are typical of the questions you will be given in Element IV of FCC License Examinations that you must pass for your First-Class Radiotelephone Operator's License. You should be able to answer all of the questions without referring to your texts or FCC Study Guides. If you cannot answer them, do whatever additional studying may be necessary. Select the answer you believe is most correct and mark the number representing that answer in the space provided.

Definitions are also part of the Element IV examination. Review thoroughly the definitions contained in texts C111X and C112X. Understanding the terms will make it easy for you to choose the most correct optional answer.

Be sure to write your name, address and student number on the top of this page in the space provided, and also on the attached label. Send the label to NRI along with your completed examination.

1. Who keeps the keys to the fence which surrounds the antenna base at a standard broadcast station?

(1) The operator on duty at the transmitter.
(2) They are kept in the office of the station licensee.
(3) They are kept in a special location at the antenna base.
(4) They are kept in a key closet with other important keys.
(5) They are kept in the possession of the custodian of the station.

Ans. ____________

2. If in checking the speed of a turntable with a stroboscope disc, you discover that the bars first move slowly in the direction of rotation and then slowly in the opposite direction, you know that:

(1) the turntable is operating at the correct speed.
(2) the turntable is running too fast.
(3) the turntable is running too slow.
(4) the turntable speed is varying slightly above and slightly below the correct rate.
(5) none of the preceding answers is correct.

Ans. ____________

3. To test the frequency response of a turntable pickup and its associated amplifier, you should use:

   (1) a test tape.
   (2) an audio oscillator with a constant amplitude output and feed the output into the amplifier input.
   (3) a test record and measure the output from the amplifier at various frequencies.
   (4) a stroboscope disc.
   (5) a constant amplitude variable-frequency oscillator connected directly to the stylus and measure the output from the amplifier.

Ans.____________

4. A limiting amplifier is:

   (1) an amplifier that limits the length of a program.
   (2) an amplifier that limits the range of frequencies that are amplified.
   (3) an amplifier that has a very high gain.
   (4) an amplifier that has a variable gain which prevents overmodulation at high amplitude levels.
   (5) an amplifier that has very little gain and hence prevents undermodulation at low signal levels.

Ans.____________

5. What is the purpose of a line equalizer?

   (1) It compensates for lines of different lengths so that signals arriving at the studio from two different sources will arrive at the same time.
   (2) It compensates for unequal frequency attenuation in the line.
   (3) It compensates for amplitude variations due to lines of different lengths.
   (4) It prevents overmodulation of the transmitter.
   (5) It compensates for poor frequency response in the studio audio equipment.

Ans.____________

6. What is the penalty for violating a provision of the Communications Act of 1934?

   (1) None. A satisfactory explanation to the FCC is sufficient.
   (2) Loss of license for one year.
   (3) Revocation of operator license and two years in prison.
   (4) A fine of not more than $10,000 and/or not more than one year in prison or both.
   (5) A fine of $500 a day for each day of violation plus other penalties.

Ans.____________

7. A quartz crystal is equivalent to:

   (1) a high Q series-resonant circuit because the equivalent inductance is high compared to the equivalent resistance.
   (2) a low Q series-resonant circuit because the equivalent inductance is high compared to the equivalent capacitance.
   (3) a high Q parallel-resonant circuit because the equivalent inductance is high compared to the equivalent resistance.
   (4) a low Q parallel-resonant circuit because the equivalent inductance is low compared to the equivalent resistance.
   (5) a high Q series-resonant circuit because the capacity of the holder has little effect on the circuit.

Ans.____________

8. What are agc amplifiers that are used in broadcast studio amplifiers?

(1) They are amplifiers whose gain is controlled by the operator rotating the gain control at the console.
(2) They are amplifiers that automatically compensate for variations in frequency response.
(3) They are amplifiers that automatically compensate for signals of different amplitude to prevent overmodulation of the transmitter.
(4) They are amplifiers that automatically compensate for unequal frequency response in transmission lines.
(5) They are amplifiers whose gain can be varied automatically by the studio operator.

Ans.__________

9. In a limiting amplifier, overmodulation is prevented:

(1) by clipping the modulation peaks of the diode.
(2) by varying the gain of an amplifier at the studio console.
(3) by means of a biased diode which conducts when the signal exceeds a certain level and using the signal developed to control the gain of an amplifier.
(4) by means of a biased amplifier that operates at low gain.
(5) by means of a biased diode that develops a signal which signals the operator at the console to adjust the amplifier gain.

Ans.__________

10. The resonant frequency of a crystal is primarily affected by:

(1) the operating temperature of the crystal.
(2) the type of circuit in which the crystal is used.
(3) the type of holder in which the crystal is operated.
(4) whether or not the crystal is operated in an oven.
(5) the dimensions and type of cut.

Ans.__________

11. A compression amplifier is:

(1) an amplifier containing a diode that clips the modulation peaks.
(2) an amplifier containing a biased diode that controls the gain of a variable gain stage.
(3) an amplifier containing a diode that develops the signal which is used to compress the dynamic range of a broadcast.
(4) an amplifier that clips the positive peaks of the audio signal to prevent overmodulation.
(5) none of the preceding is correct.

Ans.__________

12. What is a mercury thermometer crystal-heater control?

(1) It is a thermometer used to tell the temperature of the crystal oven.
(2) It is a thermostat that makes use of a bimetal strip.
(3) It is a thermostat that controls the amplitude of the crystal oscillator.
(4) It is a very sensitive type of thermostat used to control the temperature of a crystal oven.
(5) It is a sensitive mercury-type thermostat used to control the frequency of an oscillator.

Ans.__________

13. What is the advantage of the bimetallic strip thermostat over the mercury thermometer thermostat?

(1) It is less expensive.
(2) It is more accurate.
(3) It is more sensitive.
(4) It is more stable.
(5) It is more rugged.

Ans.______________

14. When required, how should antenna structures be painted?

(1) There are no requirements in areas well lighted 24 hours each day.
(2) Alternate bands of aluminum and luminous orange paints are required whenever tower structures are lighted and have flashing beacons.
(3) They should be painted with alternate equal bands of aviation surface orange and white paint, with widths ranging between 1-1/2 and 40 feet.
(4) The top 200 feet of the structure must be painted in alternate equal bands of aviation surface orange and white paint with widths ranging between 1-1/2 and 40 feet.
(5) Alternate equal band widths, ranging between 1-1/2 and 40 feet of black and white paint.

Ans.______________

15. What is the maximum theoretical efficiency of a linear Class A power amplifier?

(1) 10 to 12%
(2) 25%
(3) 30 to 35%
(4) 50%
(5) 66%

Ans.______________

16. Why are tubes which are used in linear power amplifiers not normally biased Class A?

(1) The distortion is too high.
(2) They are usually biased Class A.
(3) They will produce excessive harmonics.
(4) Their operating efficiency is too low.
(5) None of the preceding answers is correct.

Ans.______________

17. What is the disadvantage of grid-leak bias on a Class C power amplifier using a vacuum tube?

(1) The efficiency of such a stage is low.
(2) This type of stage generates a large number of harmonics.
(3) If the drive disappears, the plate's current will drop to zero and destroy the tube.
(4) If the drive is excessive, the grid bias developed will be low.
(5) If the drive disappears, the plate current will rise to a high value and destroy the tube.

Ans.______________

18. In what type of circuit would you expect to find the grids of two tubes operated in push-pull and the plates in parallel?

(1) A push-pull Class C power amplifier.
(2) A push-pull Class B linear amplifier.
(3) A push-push frequency doubler.
(4) A push-pull Class A audio power amplifier.
(5) A push-pull frequency tripler stage.

Ans. ______________

19. What is the chief advantage of a grounded-grid power amplifier?

(1) It usually does not require neutralization.
(2) It has a higher efficiency than a grounded-cathode power amplifier.
(3) It makes a good frequency doubler.
(4) Its harmonic output is low.
(5) It is easier to drive than a grounded cathode amplifier.

Ans. ______________

20. What is the audio power required to provide 50% modulation if the rf power input to the final stage is 100 watts?

(1) 50 watts (2) 25 watts (3) 12-1/2 watts (4) 10 watts (5) 5 watts

Ans. ______________

21. Why is it possible to obtain such high efficiency from a Class C rf power amplifier?

(1) A high negative bias on the grid limits the plate through the tube and hence limits the losses.
(2) The high negative bias applied to the grid of the tube permits operating the tube at a very high plate voltage.
(3) Plate current flows through the tube only during a small part of the cycle thus limiting losses.
(4) Driving the grid positive causes the tube to conduct current in the form of a series of short pulses.
(5) The maximum plate current flow through the tube occurs at the time of minimum plate voltage and hence the tube losses are low.

Ans. ______________

22. Where can specifications for the paint used to paint antenna structures be obtained?

(1) From the local hardware store.
(2) From any paint manufacturer.
(3) From a nearby Sears, Roebuck store.
(4) From the nearest Field Office of the Federal Communications Commission.
(5) From the General Services Administration, Washington, D. C. 20407.

Ans. ______________

23. A tetrode rf power amplifier is operated with the grid-leak bias. If the correct bias is –45 volts and the dc grid current is 3 ma, what size grid-leak bias resistor is required?

(1) 15,000 ohms (2) 10,000 ohms (3) 18,000 ohms (4) 45,000 ohms (5) 5000 ohms

Ans. ______________

24. Which of the following statement(s) about a triode frequency multiplier is correct?

(1) A high operating bias is used to improve the efficiency of the stage.
(2) A high operating bias is used to reduce the operating angle of the stage.
(3) A high operating bias is used because the stage draws a high grid current.
(4) A low operating bias is used to provide an output rich in harmonics.
(5) All of the preceding statements are correct.

Ans.____________

25. What is meant by positive carrier shift?

(1) The carrier frequency increases when the transmitter is modulated.
(2) The carrier frequency decreases when the transmitter is modulated.
(3) The plate current increases during modulation indicating nonsymmetrical modulation.
(4) The plate current decreases during modulation indicating nonsymmetrical modulation.
(5) None of the preceding answers is correct.

Ans.____________

26. Using the following data about a radio transmitter, calculate the power output of the transmitter, using the indirect method. The plate voltage into the final amplifier stage is 10,000 volts, the plate current is 2 amps, the efficiency is 25%, the antenna current is 10 amps and the antenna impedance is 52 ohms.

(1) 5200 watts (2) 5000 watts (3) 5500 watts (4) 4500 watts (5) 6000 watts

Ans.____________

27. If an operator wants to make a brief test of the transmitter, what would be a good choice of words to use in his test?

(1) Call sign followed by a familiar quotation, or latest weather report, etc.
(2) Call sign followed by any transmission which gives you sufficient information to complete the test.
(3) Call sign of the station tested, then the word "testing" followed by the count 1, 2, 3, 4, etc.
(4) Testing, testing, 1, 2, 3, 4, etc. is sufficient.
(5) Call sign followed by "testing", with a question to be answered by a distant receiving station.

Ans.____________

28. What is the purpose of interlocks on a radio transmitter?

(1) They prevent any unauthorized person from opening a transmitter door.
(2) They are doors that are locked so that only the operator in charge, who has the key, can open them.
(3) They are locks designed to prevent anyone from stealing expensive tubes or transistors from the transmitter.
(4) They are safety devices used to protect the operator from the high voltages in the transmitter.
(5) They are devices used to prevent a fire in case of an overload in the transmitter.

Ans.____________

29. In a radio transmitter that is fully protected by relays in the final amplifier using grid-leak bias, what would you expect to happen if the excitation to the final amplifier failed?

(1) The time-delay relay would fail to operate.
(2) The time-delay relay would operate to protect the final stage.
(3) Power would be removed, but a recycling relay would put the power back on to the stage after a two minute delay.
(4) The plate-current overload relay would function and remove the plate voltage from the output stage.
(5) The loss of excitation of the final stage would not cause any of the protective relays to function.

Ans. ______________

30. A frequency-modulated transmitter is first modulated with a 400-cycle note and then with an equal-amplitude 1000-cycle note. Which of the following statements is true?

(1) The higher frequency signal will produce a greater number of sidebands.
(2) The higher frequency signal will produce a greater frequency deviation.
(3) The lower frequency signal will produce a greater frequency deviation.
(4) The higher frequency signal will cause the resting frequency to change at a faster rate.
(5) The lower frequency signal will cause the resting frequency to change at a faster rate.

Ans. ______________

31. The maximum frequency swing that occurs in FM broadcasting is:

(1) 5 kHz (2) 10 kHz (3) 25 kHz (4) 75 KHz (5) 150 kHz

Ans. ______________

32. What is the purpose of pre-emphasis in an FM transmitter?

(1) It improves the coverage of the station.
(2) It prevents loss of low frequency audio signals.
(3) It prevents loss of high frequency audio signals.
(4) It improves the signal-to-noise ratio at low audio frequencies.
(5) It improves the signal-to-noise ratio at the high audio frequencies.

Ans. ______________

33. If the power output of a transmitter is 5 kw, the gain of the antenna is 14 db, and the line loss is 4 db, what is the effective radiated power of the transmitter?

(1) 5 kw (2) 10 kw (3) 25 kw (4) 50 kw (5) 100 kw

Ans. ______________

34. How often should antenna towers be painted?

(1) Regularly once each year.
(2) When advised during FCC inspection.
(3) As often as necessary to maintain good visibility.
(4) When advised by the Federal Aviation Agency (FAA).
(5) Not less than once each five years.

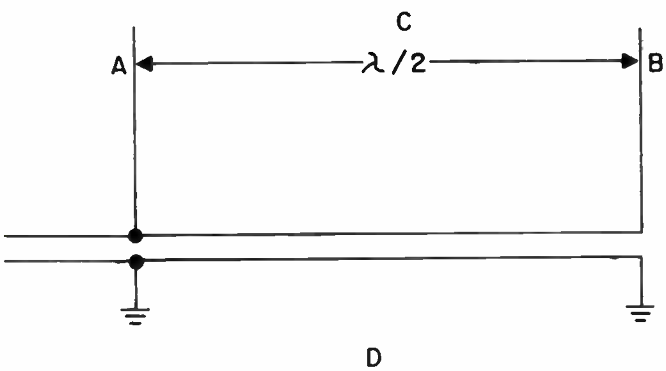
Ans. ______________

35. Which of the following would be the best location for the antenna for an FM broadcast station?

(1) In the center of a swampy meadow.
(2) In the middle of the downtown business district.
(3) As close as possible to a large body of water.
(4) On the top of the highest building or hill in the area.
(5) At a location selected by the FCC inspector.

Ans.________________

36. In the figure shown, in which direction will maximum radiation occur?



(1) In the direction along the line A-B.
(2) In the direction along the line C-D.
(3) In both directions along the line A-B and the line C-D.
(4) Neither along the line A-B or along the line C-D.
(5) There will be no radiation at all from the system.

Ans.________________

37. Which of the following might change the radiation pattern of a broadcast station?

(1) Construction of a large wooden garage nearby.
(2) Construction of a high-rise apartment building with a steel frame.
(3) Paving a highway nearby.
(4) Construction of a new single-family residence nearby.
(5) None of the preceding.

Ans.________________

38. The directional pattern of an antenna system is usually controlled by:

(1) adjusting the height of the towers.
(2) watering the grass around the towers to keep the ground conduction good.
(3) keeping the transmitter on frequency.
(4) adjusting the power and phasing of the energy fed to each element.
(5) doesn't change once the antenna towers are in place and hence no adjustments are needed.

Ans.________________

39. What is the purpose of ground radials used with an AM broadcast antenna?

(1) They are used to help support the antenna tower.
(2) They are used to improve the ground conductivity.
(3) They are used to keep harmonic radiation at a minimum.
(4) They are used to make it easier to tune the transmitter.
(5) They serve little or no useful purpose.

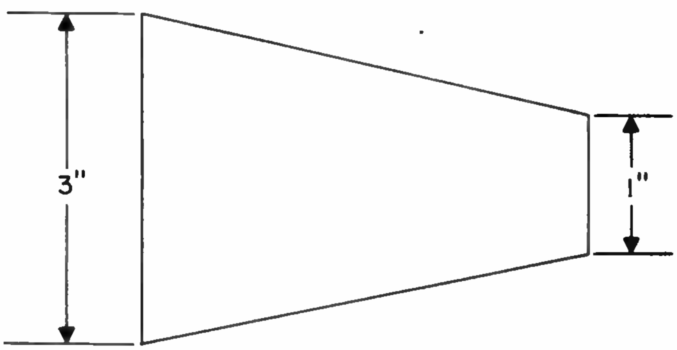
Ans. ______________

40. How can the antenna of a broadcast station be matched to the transmission line and at the same time attenuate the harmonics?

(1) By connecting the antenna transmission line to the proper tap of a tank coil.
(2) By using a push-pull amplifier in the final power amplifier stage of the transmitter.
(3) By using an adjustable T-type network between the transmission line and the antenna.
(4) By selecting a transmission line whose impedance matches the resistance of the antenna.
(5) By using a parallel-resonant circuit, tuned to the harmonic, between the transmission line and the antenna.

Ans. ______________

41. What percentage of modulation is indicated by the trapezoidal pattern shown?



(1) 25%
(2) 33-1/3%
(3) 50%
(4) 66-2/3%
(5) 100%

Ans. ______________

42. An antenna two meters long picks up a signal of 30 microvolts. What is the field strength of the signal?

(1) 10 microvolts per meter
(2) 15 microvolts per meter
(3) 20 microvolts per meter
(4) 30 microvolts per meter
(5) 60 microvolts per meter

Ans. ______________

43. In what direction should the automatic control for tower lights face?

(1) East
(2) West
(3) North
(4) South
(5) According to geographical location, that direction which gives the most light.

Ans.________

44. How is a reactance tube used to produce frequency modulation?

(1) It adds reactive current to the oscillator tank circuit that either increases or decreases the effective inductance or capacity of the circuit.
(2) It controls the frequency of the oscillator in the AM transmitter.
(3) It adds dc to the oscillator tank circuit and hence controls the inductance or capacity of the circuit.
(4) It produces amplitude modulation in the oscillator which is converted to frequency-modulation in a mixer.
(5) It is used primarily to produce phase modulation.

Ans.________

45. How is the required 75 kHz frequency deviation obtained in FM transmitters using Armstrong phase modulation?

(1) Phase modulation is the same as frequency modulation, so the deviation is obtained simply by using a low-frequency oscillator followed by frequency multipliers.
(2) Phase modulation provides a wide frequency deviation hence the oscillator is simply followed by the required number of frequency multipliers.
(3) The Armstrong FM transmitter does not use phase modulation; it uses a reactance tube modulator.
(4) By using a low-frequency oscillator followed by frequency multipliers, then a converter which reduces the center frequency without reducing the deviation, and then feeding the signal to additional frequency multipliers.
(5) It cannot be used to get 75 kHz deviation used in FM broadcasting, but is used only in TV where 25 kHz deviation is required.

Ans.________

46. What is the difference between frequency and phase modulation?

(1) Phase modulation provides greater deviation of the oscillator.
(2) Frequency modulation provides greater deviation of the oscillator.
(3) There is little or no difference between frequency and phase modulation.
(4) Frequency modulation can only be detected by FM detectors.
(5) A special phase detector is required to detect phase-modulated signals.

Ans.________

47. What do we mean when we say broadcast band signals are vertically polarized?

(1) The towers are vertical.
(2) The signal travels in a vertical direction.
(3) The signal radiated is a ground wave.
(4) The magnetic field radiated by the antenna is vertical.
(5) The electric field radiated by the antenna is vertical.

Ans.________

48. Why must the original shape of relay contacts be maintained?

(1) To keep the relay looking like new.
(2) To get maximum life from the relay contacts.
(3) To prevent amplitude modulation of the transmitter.
(4) To prevent standing waves on the transmission line.
(5) To prevent excessive voltage drop in the relay contacts.

Ans.______________

49. Why is a buffer amplifier frequently used between the crystal oscillator and the following power amplifiers?

(1) It gives maximum possible amplification of the oscillator signal.
(2) It makes it easier to tune the final amplifier.
(3) It provides frequency multiplication.
(4) It isolates the oscillator from the power amplifiers.
(5) It is required by FCC Regulations.

Ans.______________

50. In the tuning of the transmitter, which of the following statements is not true?

(1) The oscillator is tuned for minimum plate current.
(2) The buffer is tuned for minimum plate current.
(3) The intermediate power amplifier is tuned for minimum plate current.
(4) The final amplifier grid circuit is tuned for minimum grid current.
(5) The driver is tuned for maximum final amplifier grid current.

Ans.______________

NRI Study Guide For FCC License Examination

PART I

The Federal Communications Commission which gives the examinations for Radiotelephone and Radiotelegraph operators' licenses has published a study guide to enable candidates for licenses to review and prepare for their examination. You will receive the various parts of this special NRI Study Guide with your graded answers. They will be sent to you at appropriate places in your course and will supplement the material in your lesson texts or put a different emphasis on some of the material you have already covered in order to get you used to the way the questions will be asked. This part of the Study Guide deals with direct current and alternating current. You should study this Study Guide now. If you find that you do not understand any of the questions be sure to make use of one of your consultation blanks to write in requesting additional assistance. Failing to clear up small difficulties, particularly in the beginning of your course, can lead to a great deal of difficulty later on.

As you study this Study Guide, read the question first and try to answer it yourself. Then refer to our answer to be sure that you are correct. If necessary, go back and restudy your lesson text. Time spent on basic fundamentals will be very worthwhile; a good solid understanding of basic fundamentals will make later lessons that much easier for you.

1. By what other expression may a "difference of potential" or "electromotive force" be described?

   The electromotive force of a battery is the voltage produced by a battery. The unit of electromotive force is the volt. Sometimes in a circuit there will be a potential between two parts of the circuit. For example, a difference of potential can exist across the two ends of a resistor. We can refer to this as the voltage across the resistor or a voltage drop across the resistor.

2. By what other expression may an "electric current flow" be described?

   An electric current flow is a movement of electrons. The unit of electric current flow is the ampere, which we usually abbreviate as amp.

3. Explain the relationship between the physical structure of the atom and electric current flow.

   The atom is made up of a nucleus in the center which is surrounded by electrons arranged in one or more shells. Some of the electrons in the outer shell of some atoms are not too closely bound to the nucleus so when an external force is applied, the electrons will move from the outer shell of the atom. This movement of electrons is a current flow. In a conductor the atoms are so solidly packed that when a force is applied to the two ends of a conductor electrons start moving instantaneously throughout the entire conductor. Electrons will move from the outer shell of one atom to the outer shell of the next atom, from there to the outer shell of still another atom. This movement is instantaneous along the entire length of the wire so that the same number of electrons are in motion at different points along the wire.

4. With respect to electrons, what is the difference between conductors and non-conductors?

   In a conductor, the electrons in the outer shell are not firmly bound to the nucleus so they can easily be moved from one atom to another when an external force is applied to the nucleus. In a non-conductor, the electrons in the outer shell are firmly bound to the nucleus and cannot normally be removed from the atom and made to move to another atom.

5. What is the difference between electric power and electrical energy? In what units is each expressed?

   You have already seen many examples of electric power. You know that when a resistor is connected across a battery the battery voltage will cause a current to flow through the resistor. The product of the voltage times the current will give you the power being dissipated or used in the resistor. This power is measured in watts. Sometimes we say that the electrical energy is being converted into heat. This is due to the fact that the power dissipated in the resistor causes the resistor to get hot and the heat is absorbed by the air surrounding the resistor. But this power came from the battery originally. The power stored in the battery is referred to as energy. The energy is in the battery and when you connect the resistor across the battery the energy is released in the form of power. The unit of energy is the joule and when energy is used at a rate of one joule per second we say that the power being dissipated is one watt. Power can be described as energy being used.

6. What is the relationship between resistance and conductance?

Conductance is the opposite of resistance. Resistance is the opposition to current flow whereas conductance is the ability to pass current. The conductance of a circuit is the reciprocal of its resistance. In other words $C = \frac{1}{R}$. Since conductance is the opposite of resistance the unit of resistance is turned around and used as the unit of conductance. The unit of conductance is the mho.

7. A relay with a coil resistance of 400 ohms is designed to operate when 0.2 ampere flows through the coil. What value of resistance must be connected in series with the coil if it is to be energized by a 110-volt dc source?

In this problem you have to connect a resistor in series with a relay coil so that the total resistance of the coil and resistor will limit the current flow to 0.2 ampere. Since we know what the current flow must be and we know the voltage, which is 110 volts, we can find the total resistance simply by using Ohm's Law:

$$R = \frac{E}{I}$$

and substituting 110 volts for E and 0.2 ampere for I we get:

$$R = \frac{110}{0.2} = 550 \text{ ohms}$$

Since the coil has a resistance of 400 ohms then we need to add an external resistor having a resistance of 150 ohms so that the total resistance in the circuit will be 550 ohms. This will limit the current flow in the circuit and through the coil to 0.2 ampere.

8. What is the relationship between wire length and the resistance of the wire?

The resistance of a wire of a given size will depend upon the length of the wire. In other words a piece of wire two feet long will have twice the resistance of a piece of wire one foot long. Similarly, a piece of wire five feet long will have five times the resistance of a piece that is only one foot long.

9. What is the relationship between wire size and resistance of the wire?

When we refer to wire size we mean the diameter of the wire. If we cut a piece of wire in half and look at the end you will see that the end of the wire is a circle. The circle has a certain area which can be found from the formula $A = \pi R^2$ where π is equal to 3.14 and R is the radius of the circle or one half the diameter. The resistance of the wire will vary inversely with the area. In other words, if the cross sectional area or the area of the circle doubles then the resistance of the wire would be only one half. Since the cross sectional area of the wire depends upon the square of the radius and the radius is one half the diameter, doubling the diameter of the wire will reduce the resistance to one-fourth of its original value. In other words, the resistance of the wire will vary inversely as the square of the diameter of the wire.

10. What effect does an increase in temperature have on the resistance of a wire?

The resistance of a wire depends on the temperature of the wire. If the temperature increases, then the resistance increases. If the temperature decreases, then the resistance decreases.

11. What is "skin effect"? How does it affect the resistance of conductors at the higher radio frequencies?

When a direct current or a low-frequency alternating current flows through a wire, the movement of electrons is uniform throughout the entire cross section of the wire. In other words, there will be electrons moving in the center of the wire, close to the center of the wire, near the outside edge of the wire and along the outside surface of the wire. However, at high radio frequencies the tendency for the electrons is to move along the outer surface of the wire so that practically all of the current flows on or near the surface of the wire. As far as the resistance of the wire at radio frequencies is concerned we can actually cut out the center of the wire and use a tube instead of a solid wire without appreciably affecting the rf resistance. This tendency of the current to flow along the surface of the wire is called skin effect, and it increases the resistance of conductors at the higher radio frequencies.

12. A certain power company charges seven cents per kilowatt-hour. How much would it cost to operate three 120-volt lamp bulbs, connected in parallel, each having an internal resistance of 100 ohms, for 24 hours?

Electric power is sold by power companies by the kilowatt-hour. A kilowatt-hour represents a power of one kilowatt for a period of one hour. In other words, if it takes 1000 watts, which is equal to a kilowatt, to operate a certain appliance and the appliance is turned on for one hour then the power consumed would be one kilowatt-hour. If it was turned on for three hours, the power consumed would be three kilowatt-hours.

In the example given we can find the power consumed by one light bulb by using the power formula:

$$P = \frac{E^2}{R}$$

and substituting 120 volts for E and 100 ohms for R we will get:

$$P = \frac{120 \times 120}{100}$$

$$= 144 \text{ watts}$$

Since there are three bulbs and each has the same resistance and is operated from the same voltage then each will draw the same power. Therefore the total power consumed by the three bulbs is 432 watts. When the bulbs are left on for one hour they will use 432 watt hours. When they are left on for 24 hours they will use twenty four times this much and multiplying 432 by 24 we get 10,368 watt-hours. Since a kilowatt-hour is equal to 1000 watt-hours we divide by 1000 to convert this to kilowatt-hours. We do this simply by moving the decimal point three places to the left and we get 10.368 kilowatt-hours. The electric company charges 7 cents per kilowatt-hour so we find the total charge by multiplying the number of kilowatt-hours used by 7 cents and this works out to 72.56 cents which would be rounded off to 73 cents.

13. What is the value and tolerance of a resistor which is color-coded (left-to-right): RED, BLACK, ORANGE, GOLD?

    20,000 ohms, ±5%. This may also be expressed as 20K-ohms, ±5%. You should know the standard EIA resistor color code so you can identify any resistor you are likely to encounter. Remember the fourth band gives the tolerance. A silver band indicates a tolerance of 10% and a gold band a tolerance of 5%. If there is no fourth band, then the tolerance is 20%.

14. List three precautions which should be taken soldering electrical connections to insure a permanent junction.

    The parts to be soldered together must be clean. If there is any dirt on a lead or terminal it should be removed either by scraping it or by rubbing it with sandpaper. The leads to be connected together should form a firm mechanical connection. If you are connecting two leads together they should be twisted together and if you are connecting a lead to a terminal it should be wrapped around the terminal so that a firm mechanical connection is formed. When the parts are actually soldered you must use rosin core solder and then use sufficient heat to enable the solder to melt completely and flow freely over the leads or parts to be connected together. Insufficient heat will result in a cold solder connection which will not make a good electrical contact.

15. What is the relationship between impedance and admittance?

    Admittance is the reciprocal of impedance. In other words, admittance is equal to 1/Z. Admittance is measured in mhos.

16. Define the term "reluctance".

    Reluctance is the opposition to the flow of magnetic lines of force. You will remember that magnetic lines of force leave the north pole of the magnet and flow through either space or surrounding material back to the south pole of the magnet. In flowing from the north pole back to the south pole they encounter opposition and this opposition or resistance is called reluctance. Reluctance in a magnetic circuit is often compared to resistance in an electrical circuit.

17. In what way does an inductance affect the voltage-current phase relationship of a circuit? Why is the phase of a circuit important?

    The amount of inductance in a circuit affects the relationship between the phase of the voltage and current in the circuit in such a way that if the only component in the circuit is an inductance the voltage will lead the current by 90°. This is the same as saying that the current lags the voltage by 90°. In a circuit that is purely resistive, the voltage and current will be in phase. When a circuit contains both resistance and inductance, the voltage will lead the current by some angle between 0° and 90° depending upon the relationship between the resistance and the inductance. If the opposition to current flow is due almost entirely to inductive reactance, the phase displacement between the voltage and current will be almost 90°. On the other hand, if the entire opposition to current flow is almost purely resistive, the phase difference between the voltage and current will be some small angle.

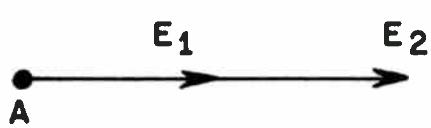
    When we speak of phase we are referring to the timing or relationship between the time when the voltage and current go through each cycle. If in an ac sine wave when the voltage starts at 0 the current is at 0 and they both increase and reach their maximum value at the same instant and then drop back to 0 so that they are continually in step with each other we say that they are in phase. On the other hand, when the current does not increase exactly in step with the voltage, there is a phase difference between the voltage and the current. If the current follows the voltage, we say it lags the voltage, but if the current goes through its cycle ahead of the voltage, we say it leads the voltage.

The phase relationship between voltage and current in a circuit is important for several reasons. One reason is that the power being consumed by the circuit depends upon the phase relationship. You will remember that in a dc circuit the power can be found by multiplying the voltage times the current. We can find the power in the same way in an ac circuit when the voltage and current are in phase. However, when the voltage and current are not in phase, the true power being dissipated in the circuit is equal to the voltage times the current times the power factor. The power factor corrects for the phase difference between the voltage and current in the circuit.

In coupling circuits phase relationship between voltage and current is important. You will find out later that in some circuits, particularly in television circuits, if the voltage and current are not in phase, there will be some distortion or smearing in the TV picture. You'll study this later, but keep it in mind now.

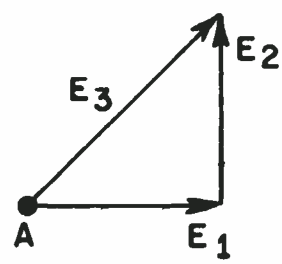
18. Explain how to determine the sum of two equal vector quantities having the same reference point but whose directions are 90° apart; 0° apart; 180°. How does this pertain to electrical currents or voltages?

Vectors are used to show the phase relationship between the voltage and current in a circuit. They are also used to add equal quantities and they are particularly useful for adding quantities that are not in phase. For example, a vector can be used to represent a voltage. A second vector could be used to represent another voltage and by means of vector addition these two voltages can be added together whether or not they are in phase. For example, suppose we have the problem of adding two voltages which we will call E1 and E2 and each voltage is equal to 10 volts. If the two voltages are in phase, we can simply add them together arithmetically and we know that the total will be 20 volts. This can also be done vectorially by starting at point A as shown below and by laying vector

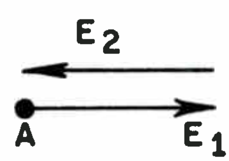


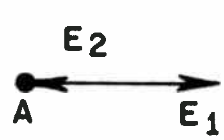
E1. We have used a scale of 10 volts equal to one inch so the vector representing E1 is one inch long. At the end of the vector representing E1, we have drawn a second vector with exactly the same phase relationship to represent E2. Since the two are in phase the two vectors are pointed in exactly the same direction and the total length will be two inches which would represent 20 volts.

On the other hand, suppose we have two voltages, E1 and E2, 90° apart and want to add them. We can't add them by means of simple arithmetic because they are not in phase. However, we can still use vector addition to add these two voltages.



We draw vector E1 equal to one inch, as shown above starting at point A. At the end of vector E1 we add vector E2. Since we have not indicated whether the voltage E2 is leading or lagging we could point it either up or down. We have drawn the vector representing E2 in a vertical direction and equal to one inch since it also is equal to 10 volts. Now we join point A to the end of vector E2 and the third vector which we mark E3 will be the vector sum of the two. You will find it is about 1.4 inches long indicating that the sum of the two voltages is approximately 14 volts.





Now if we have to add the two vectors when they are 180° apart we start at point A and draw vector E1 equal to one inch. Now from the end of vector E1 we start and draw vector E2 180° out-of-phase with vector E1. The vector is moved up a little bit so you can see it. In the actual vector diagram it would coincide as shown in the above illustration. The end of vector E2 arrives exactly back at point A so that the net sum of the two voltages 180° apart is 0. You will run into this situation later when we feed a voltage from one circuit into a second circuit in order to cancel another voltage. Here we want the two voltages to be 180° out-of-phase.

Vectors can be used as shown in the preceding example to add voltages. We can also use them to add currents that are not in phase. However, we can't add voltages and currents because they are not like items. We have the same situation where you might have two bags of apples and you could take the apples out of the two bags and simply add them together and say that you have so many apples. On the other hand, if you had one bag full of apples and the other bag full of lemons you couldn't dump them together in a single bag and simply add the total together and say that you had so many apples or so many lemons because the two are unlike, they are different items and you can't add them together in that way.

19. Explain how the values of resistance and capacitance in an RC network affect its time constant.

The time constant of an RC network is considered as the time it takes for the capacitor to charge to 63% of the voltage applied to it through a resistor. It is also equal to the time it takes a charge on a capacitor to discharge 63% of its value through a resistor. The time in seconds is equal to the capacitance of the capacitor in microfarads times the resistance of the resistor in megohms. In other words, $T = R \times C$, where T is in seconds, R is in megohms and C is in microfarads. From the formula you can see that any increase in either R or C will increase the time constant and any decrease in either R or C will decrease the time constant.

20. Explain how the values of resistance and inductance in an RL network affect its time constant.

The time constant of an RL circuit is equal to 63% of the time it takes the current to reach its steady-state value. In other words, if a resistor and a coil are connected in series and the two are suddenly connected across a battery, the inductance of the coil will prevent the current from instantly increasing to a value that will be limited only by the resistance in the circuit. The inductance of the coil offers inductive reactance to the sudden change in current through it. The time it will take for the current to build up to its final value will depend upon the inductance in the circuit and the resistance. The time it takes to build up to 63% of the total current is the time constant and is given by the formula $T = L \div R$, where T is in seconds, L is the inductance of the coil in henrys and R is the resistance in ohms.

If we examine the formula we see that the time constant is equal to the inductance divided by the resistance. Therefore if the inductance in the circuit increases the time constant will increase and similarly, if the inductance decreases the time constant will decrease. The time is inversely proportional to the resistance in the circuit. That means the larger the resistance the shorter the time constant. Therefore if the resistance in the circuit increases the time constant of the circuit will decrease and similarly if the resistance in the circuit decreases then the time constant of the circuit will increase.

21. What is the relationship between the inductance of a coil and the number of turns of wire in the coil; the permeability of the core material used?

The inductance of a coil depends upon the square of the number of turns of wire. In other words, if you have a coil with ten turns on it and you double the number of turns to twenty turns, the inductance of the coil will be 2^2 or four times the original inductance.

The permeability also has an effect on the inductance of the coil. The higher the permeability of the core material the higher the inductance of the coil. The inductance of the coil varies directly as the permeability of the material used in the core.

22. What factors influence the direction of magnetic lines of force produced by an electromagnet?

The directions of the magnetic lines of force produced by an electromagnet will depend on the polarity of the magnet. You will remember that in the external circuit the magnetic lines of force will flow from the north pole to the south pole. In an electromagnet you can reverse the polarity of the magnet by reversing the direction of current flow through the coil. Therefore one of the factors that will affect the direction of magnetic lines of force will be the polarity of the voltage applied to the coil since this will determine the direction that the current flows through the coil. Another factor that will affect the direction is the way in which the coil is wound. If you wind the coil one way, then for a given voltage polarity the current will flow through the coil in one direction whereas if you wind it in the opposite way the current would flow through it the opposite way with the same polarity. Therefore basically the two factors that affect the direction of the magnetic lines of force are the polarity of the voltage applied to the coil which will determine the direction the current flows and the way in which the winding is put on the coil since this also will affect the direction of current flow.

23. What does coefficient of coupling mean?

The coefficient of coupling tells how closely two coils are coupled together. If the coils are placed in such a way that all of the flux produced by one coil cuts the turns of the other coil and all of the flux produced by the second coil cuts the turns of the first coil, the coefficient of coupling is one. If the coils are moved apart slightly so that some of the flux lines of one coil do not cut all the turns of the other, the coefficient of coupling is less than one. The maximum value that the coefficient of coupling can have is one. It can approach this value for coils wound on a high permeability coil form. In the case of air-core coils, the coefficient of coupling is usually quite low, it is usually some value less than .25.

24. How does the capacitance of a capacitor vary with the area of the plates; the spacing between the plates; the dielectric material between the plates?

The capacitance of the capacitor varies directly with the area of the plates. This means that if the two plates of a capacitor have a certain area and you double the area of the plates, you will double the capacity of the capacitor. If you cut the area of the two plates of the capacitor in half, the capacity will be cut in half.

The spacing between the plates of a capacitor has a very great effect on the capacity. We say that the capacity varies inversely as the spacing. By this we mean that if you double the spacing between the plates of the capacitor, the capacity will be reduced to one-half. On the other hand, if you reduce the spacing between them by one-half the original spacing, the capacity will be increased by a factor of two.

The capacity of a capacitor varies directly as the dielectric constant of the dielectric material between the plates. In the case of an air-type capacitor where the dielectric between the two plates is air, if you insert a material between the plates that has four times the dielectric constant of air, the capacity of the capacitor will be increased by a factor of four.

In summary, the capacity of a capacitor varies directly with the area of the plates and the dielectric material between the plates and it varies inversely with the spacing between the plates.

25. Assuming the voltage on a capacitor is at or below the maximum allowable value, does the value of the capacitor have any relationship to the amount of charge it can store? What relationship does this storage of charges have to the total capacitance of two or more capacitors in series; in parallel?

Yes; the amount of charge which can be stored by a capacitor is directly related to the value of the capacitor. The relationship between charge and capacitance can be expressed by the formula:

$$Q = CE$$

Where Q is the charge in Coulombs, C is the capacitance in Farads, and E is the applied voltage in volts.

The total charge stored in series connected capacitors will be less than could be stored in any of the individual capacitors. This is because the total equivalent capacitance of capacitors in series is less than the value of any of the series capacitors, and from the formula given above if C is small, Q will be small.

When two or more capacitors are connected in series to a voltage source, the same current will flow in the series circuit to charge the capacitors. Therefore each capacitor will have the same charge.

The voltage across each capacitor will be in accordance with the formula given above:

$$E_1 = \frac{Q}{C_1} \quad E_2 = \frac{Q}{C_2} \quad E_3 = \frac{Q}{C_3}, \text{ etc.}$$

This means that since each capacitor has the same charge, Q, that the voltage across each capacitor will be inversely proportional to the capacitance. That is, the smallest capacitor will have the largest voltage and the largest capacitor will have the smallest voltage.

The amount of charge which can be stored in parallel connected capacitors will be greater than the charge stored in any of the individual capacitors because the total capacity is the sum of the individual capacitors. Since each capacitor has the same applied voltage, each will have a charge determined by:

$$Q_1 = EC_1 \quad Q_2 = EC_2 \quad Q_3 = EC_3, \text{ etc.}$$

and the total charge is $Q_1 + Q_2 + Q_3$, etc.

26. How should electrolytic capacitors be connected in a circuit in relation to polarity? Which type of low-leakage capacitors are used most often in transmitters?

When connecting electrolytic capacitors in a circuit you must observe polarity. One side of an electrolytic is connected to the positive side of the circuit and the other side must be connected to the negative side of the circuit. If an electrolytic capacitor is connected in the circuit backwards, it will act almost as a short circuit and it will be destroyed.

Mica capacitors have the lowest leakage of any capacitor and are usually used in transmitters. They are not too often used in receiving equipment because there are other less expensive types that will work just as well, but since reliability is an extremely important factor in transmitters, the best quality components are usually used. Mica capacitors have a very low leakage and very seldom give trouble and therefore are widely used in transmitters.

27. What is the impedance of a parallel circuit which is composed of a pure inductance and a pure capacitance at resonance? Of a series circuit at resonance?

At resonance a parallel circuit acts like a high resistance. Therefore the impedance is pure resistance and very high value. In a series-resonant circuit, the circuit acts like a low resistance. Therefore, impedance is low and pure resistance. In a theoretical circuit consisting of pure inductance and pure capacitance, when the two are in parallel, the impedance would be infinite and in the case of a series circuit where there is no resistance in the circuit, if the inductance is pure inductance and the capacitance is pure capacitance, the impedance would be zero.

28. What is the "Q" of a circuit? How is it affected by the circuit resistance? How does the "Q" of a circuit affect bandwidth?

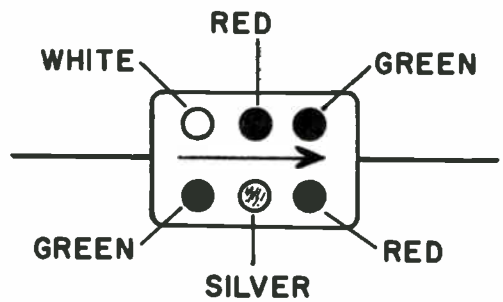
The Q of a circuit is the ratio of the reactance to the resistance. In the case of a coil, the Q of the coil is equal to the inductive reactance divided by the ac resistance. In most circuits, the Q of the coil is the factor that determines the Q of the circuit. Increasing the resistance in the circuit will reduce the Q since you have a larger number to divide into the reactance in order to get the Q. Reducing the resistance of the circuit increases the Q.

The bandwidth of a circuit is affected by the Q of the circuit. A high Q circuit has a narrow bandwidth and is used in applications where you want to be able to separate signals very close together. A low Q circuit has a wide bandwidth and is used in applications where you have to be able to pass a wide band of signal frequencies. You will go into the importance of high and low Q circuits later. You will find that the tuning circuits in communications-type receivers are high Q circuits because the receiver is designed to separate different stations operating closely together. On the other hand, the resonant circuits in television receiving and transmitting equipment are comparatively low Q circuits because the picture signals in television contain a wide band of frequencies and all these signals must be passed by the resonant circuit. You will understand this more clearly later when you study communications receivers and television. For the present simply remember that high Q circuits are very sharp and pass only a limited frequency band and low Q circuits are broad circuits that pass a wide number of signal frequencies.

29. Using the EIA Standard six dot color system what would be the value and tolerance of a capacitor whose first row of colors are (from left to right): WHITE, RED, GREEN: second row: GREEN, SILVER, RED? 0-2-5-00-10% - 500v TOL DC
2500pf - 10% - 500v

The EIA (Electronics' Industry Association) is an organization formed by companies manufacturing electronic equipment. They set up various standards as an aid to manufacturers so that there will be some standardization in the industry. The standard six-dot color system is frequently used to identify the capacity of mica capacitors. The dots are arranged in two rows with an arrow between the two rows as shown in the example.

With the values given, the capacitor would be set up, as shown below. The white dot simply indicates that the standard EIA color code is used. The dot beside the white dot is the first significant figure in the value of the capacitor. The green dot indicates the value of the second significant figure in the value of the capacitance. As you



work around the capacitor in a clockwise direction starting at the white and then the red to get the first significant figure and then the green dot to get the second significant figure, the next dot you come to is the red dot on the bottom right row. This tells you how many zeros there are after the second significant figure. The silver dot in the center of the lower row gives you the tolerance and the green dot tells you the voltage rating of the capacitor in hundreds of volts.

The color code is the same as used in identifying resistors. Since the first significant figure is represented by a red dot, the first figure is a 2. The second dot is a green dot; therefore the second significant figure is a 5 and since the dot indicating the number of zeros is red it means we have two zeros following the 5. Therefore the capacity of the capacitor is 2500 picofarads. (This may also be expressed as 2500 mmfd; micro microfarads was used for a long time in place of picofarads). The capacity could also be expressed in microfarads and 2500 picofarads is equal to .0025 mfd. The tolerance of the capacitor is 10% since silver indicates tolerance of 10% and the green dot indicates 5 which means the voltage rating of the capacitor is 500 volts.

You should memorize the EIA color code because you will have to use it to identify resistor and capacitor values. Also memorize the way the dots are used on the capacitor to enable you to identify any capacitor you might encounter. The chances are that you will have to determine the value of some capacitor on your FCC license test, but even if you should not be asked this question, you will have to deal with capacitors over and over again and be able to determine their capacity.

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NRI STUDY GUIDE FOR FCC LICENSE EXAMINATION

PART II

30. Why is impedance matching between electrical devices an important factor? Is it always to be desired? Can it always be attained in practice?

Impedance matching between electrical devices is important because it allows maximum power transfer. In other words, when load is matched to a generator, maximum power can be transferred from the generator to the load.

Impedance matching is not always desired. In some audio amplifications exact impedance matching would result in excessive distortion. There are other applications where we may be interested in maximum voltage or maximum current rather than maximum power. In addition, in some applications impedance matching may be undesirable because it may load the source too heavily.

Exact impedance matching cannot always be attained. For example, it's impossible to match the output impedance of a pentode or beam-power tube to a speaker because both pentode and beam-power tubes have a very high impedance. However, they are practical impedances into which we can have the pentode or tetrode tube work to obtain an efficient power transfer from the tube to the speaker.

31. A loudspeaker with an impedance of 4 ohms is working in a plate circuit which has an impedance of 4000 ohms. What is the impedance ratio of an output transformer used to match the plate circuit to the speaker? What is the turns ratio?

The impedance ratio is equal to the ratio of the plate impedance to the speaker impedance. Thus,

$$\text{impedance ratio} = \frac{4000}{4}$$
$$= 1000 \text{ to } 1$$

The turns ratio on a transformer needed to match the impedances will be equal to the square root of the impedance ratio. Thus if we let N_p equal the number of turns in the primary and N_s equal the number of turns on the secondary, the turns ratio will be:

$$\frac{N_p}{N_s} = \sqrt{\frac{1000}{1}}$$
$$= \frac{31.6}{1}$$

32. In an iron-core transformer, what is the relationship between the transformer turns ratio and primary-to-secondary current ratio; between turns ratio and primary secondary voltage ratio? (Assume no losses.)

In a transformer, if there are no losses, then the power taken from the secondary winding will be exactly equal to the power that the primary winding takes from the power line to which it is connected. The voltage ratio between the primary and secondary windings depends upon the turns ratio. If the secondary winding has more turns on it than the primary, then the secondary voltage will be higher than the primary voltage, and if the secondary winding has fewer turns on it than the primary winding then the secondary voltage will be lower than the primary voltage. The voltage is directly proportional to the turns ratio because the flux produced by the primary winding cuts the secondary winding inducing voltage in the various turns. The more turns you have on the secondary winding the more voltage will be induced in it. If the secondary winding of a transformer has twice as many turns as the primary, the secondary voltage will be twice the primary voltage, and the secondary current will be half the primary current. The product of the primary voltage times the primary current will give you the power input to the transformer. The secondary voltage times the secondary current will give you the power taken out of the transformer. If there no losses, these two products must be equal. If the secondary voltage is higher than the primary voltage, then the secondary current must be lower than the primary current in order for the products to be equal. Therefore, if the transformer has more turns on the secondary than on the primary, then the secondary current will be lower than the primary current and the secondary voltage will be higher than the primary voltage. If the transformer has fewer turns on the secondary winding than on the primary winding, the secondary current will be higher than the primary current and the secondary voltage will be lower than the primary voltage. We call a transformer where the voltage produced by the secondary winding is higher than the voltage applied to the primary winding "a step-up transformer". Where the secondary voltage is lower than the primary voltage, we call the transformer a "step-down transformer".

33. What prevents high currents from flowing in the primary of an unloaded power transformer?

A transformer is a self-regulating device. The transformer takes only enough power into the primary winding to supply the power taken from the secondary plus the losses in the transformer itself.

When there is no load connected across the secondary winding of a power transformer, the primary winding has a comparatively high impedance. This limits the current that can flow in the primary of the transformer. In fact, the only current that will flow will be the current required to supply the core losses in the transformer. As the transformer is loaded, the load on the secondary is reflected back into the primary, reducing the impedance of the primary and allowing a higher current to flow in the primary of the transformer.

34. An audio transformer has a resistive load connected across the secondary terminals. What is the relationship between this resistance, the turns ratio and the input impedance at the primary terminals? How is this principle useful in matching impedances?

The ratio of the primary impedance of a transformer is equal to the square of the turns ratio. We can represent this in a formula if we let Z_p be equal to the primary impedance Z_s the secondary impedance, N_p the number of turns on the primary and N_s the number of turns on the secondary. Then we will have

$$\frac{Z_p}{Z_s} = \left(\frac{N_p}{N_s}\right)^2$$

If the secondary impedance is a pure resistance, this formula tells us that the primary impedance will be equal to the secondary impedance times the turns ratio squared, or in other words:

$$Z_p = Z_s \left(\frac{N_p}{N_s}\right)^2$$

This principle is useful in impedance matching, because we now know that the square of the turns ratio of a matching transformer must be equal to the ratio of the primary to secondary impedances to be matched. For example, if we have a generator of one impedance and a load of another impedance, we can get the impedance ratio by dividing the load impedance into the generator impedance. The square of the turns ratio must equal this value if a matching transformer is to be used between the generator and the load to match the impedances.

35. How is power lost in an iron-core transformer? In an air-core transformer?

There are four principal losses in an iron-core transformer. These are hysteresis losses, eddy current losses, copper losses and flux leakage losses. In an air-core transformer there are two losses, copper losses and flux leakage losses.

Hysteresis losses in an iron-core transformer are losses encountered in magnetizing the iron core. When the alternating current builds up in amplitude the core of the transformer is magnetized. When the current drops back to zero, the magnetism of the core decreases, but there is some lag and all of the magnetism does not disappear from the core. Additional power is needed to reduce the magnetism back to zero. This is the hysteresis loss.

Eddy current losses are due to the fact that currents are induced in the core of the transformer itself. This current is induced because the metal core forms a complete circuit. Eddy current losses are kept at a minimum by using a laminated core rather than a solid core. The various laminations are insulated from each other to prevent complete circuit paths across the entire transformer core.

Copper losses are losses due to the resistance of the copper used in the transformer winding. These losses are equal to the product of the resistance of the wire times the current squared. They are the same in both the iron-core and air-core transformers.

Flux leakage losses are produced due to the fact that all of the flux lines produced by the primary winding do not cut the turns of the secondary winding. In an iron-core transformer where the coefficient of coupling between the primary and secondary windings is high, flux leakage losses are quite low. However, in an air-core transformer, where the coefficient of coupling between the two windings is low, flux leakage losses may be comparatively high.

36. Explain the operation of a break-contact relay; a make-contact relay.

In a break-contact relay, the relay contacts are normally closed. This means that the relay is set up so that when there is no current flowing through the relay winding, the contacts of the relay are closed. When the relay is energized the contacts are pulled apart, so we break the contact.

In a make-contact relay, when there is no current flowing through the coil the relay contacts are apart; in other words there is no circuit. When the relay coil is energized the contacts are pulled in such a way that the contacts close, and the circuit is completed through the contacts.

37. Discuss the physical characteristics and a common usage of each of the following electron-tube types:

(a) Diode
(b) Triode
(c) Tetrode
(d) Pentode
(e) Beam power
(f) Remote cut-off
(g) Duo-triode
(h) Cold-cathode
(i) Thyratron

(a) A diode is a tube with two elements called a cathode and an anode. The cathode is the source of electrons. It may be in the form of a cylinder which has a heater inside of it. Oxides of certain metals that emit electrons at a low temperature are used to coat the outer cathode surface. When the cathode is heated by the heater, an abundant supply of electrons will be available. This type of cathode is called an indirectly-heated cathode. The cathode may also be a filament-type cathode. The filament is simply a wire through which a current flows and the current flowing through it causes it to heat and emit electrons. This type of cathode is referred to as a directly-heated cathode.

In receiving-type tubes the filament is coated with oxides to give off electrons at low temperatures. In medium-power transmitting tubes a tungsten filament which has had a large amount of thorium dissolved in it is used. This is called a thoriated filament. In high-power tubes, pure tungsten filaments are used. Oxide-coated filaments or thoriated filaments would be literally torn apart by the high plate voltage used in high-power tubes.

The anode, which is often called the plate, is usually made in the form of a cylinder which is placed around the cathode. When a positive voltage is applied to this plate or the anode, the electrons emitted by the cathode will be attracted to the plate. If a negative voltage is applied to the plate, the electrons will be repelled from the plate and there will be no current flow through the tube.

Diodes are used primarily as detectors and rectifiers. Diode rectifiers may be vacuum types, in other words tubes from which all the air has been exhausted, and they may also be gas-filled types. In some applications a gas such as mercury vapor is put inside the tube. This type of rectifier is referred to as a mercury-vapor rectifier. Diodes and other tubes from which all the gas has been evacuated are called hard tubes, whereas tubes in which a gas has been introduced are called soft tubes.

(b) A triode is a three-element tube. In addition to the cathode and plate found in the diode tube a third element called a grid is used. The grid is placed between the cathode and plate. It is much closer to the cathode than it is to the plate and therefore a small grid voltage will have a much greater effect on the current flow through the tube than the same voltage applied to the plate will have. The grid is often made like a spiral-wound type coil, which is wound on two supports and placed around the cylindrical-type cathode. The closer the spacing of the grid wires and the closer the grid is to the cathode, the more effective it will be in controlling the flow of current through the tube. The grid is normally operated with a small negative voltage applied to it. Triode tubes may be used both as voltage and power amplifiers.

(c) A tetrode tube is a four-element tube. In addition to the three elements found in a triode, a tetrode has a fourth element placed between the grid and plate. This element is called the screen grid or simply the screen. It is usually a spiral-wound wire coil.

A positive voltage is applied to the screen. Because the screen is much closer to the cathode than the plate is, the voltage applied to the screen will have much more effect on the current flow through the tube than the plate voltage will have.

The primary purpose of placing the screen in the tube is to reduce the capacitance between the plate and grid of the tube. In a triode tube, a signal fed from the plate of the tube back to the grid of the tube can cause an amplifier to go into oscillation. The capacity between the plate and grid is comparatively large. However, in a tetrode tube the plate-to-grid capacity is greatly reduced. Instead of having a single capacitor between the plate and grid the tube acts as though there were two capacitors in series, a capacitor between the plate and screen and another between the screen and grid.

This effectively reduces the capacity because, as you will remember, when two capacitors are connected in series the total capacity is less than the capacity of the smallest capacitor. Also, by operating the screen at signal-ground potential by means of a suitable bypass capacitor, energy fed from the plate of the tube back to the screen can be effectively bypassed to ground.

Screen-grid tubes are particularly suited to use in rf amplifiers where the high plate-to-grid capacity of a triode tube could cause oscillation.

(d) A pentode tube is a five-element tube. In addition to the four elements found in the screen-grid tube, a fifth element called a suppressor grid or simply a suppressor is placed between the screen and plate. This element is also made in the form of a spiral wire coil and its purpose is to eliminate the undesirable effects of secondary emission which are encountered in some tetrode tubes. In a tetrode tube electrons reaching the plate of the tube may hit the plate at such a high speed that they will knock other electrons off the plate of the tube. These electrons drift off into space and are attracted by the positive voltage applied to the screen of the tube. By placing the suppressor grid in the tube between the plate and screen and operating the suppressor at ground potential or with a small negative potential, electrons that bounce off the plate of the tube are repelled by the suppressor grid back to the plate.

Pentodes are used both as voltage and power amplifiers in both audio and rf amplifications.

(e) A beam-power tube is usually referred to as a tetrode tube. The beam-power tube has a cathode, grid and screen grid and a plate. In addition, there are often beam-forming plates used in the tube. In most tubes the cathode is made in the form of a round cylinder. In the beam-power tube the cathode is flattened out so it has two flat sides and then rounded ends, something like an oval racetrack. The electron-emitting material is placed on the flat sides of the cathode so that the electrons are emitted in the form of two beams rather than in a circular pattern as in most tubes. Beam-forming plates placed at the rounded ends of the cathode further help to restrict the electron flow into the two beams.

Beam-power tubes are widely used both in af and rf power amplifier applications. They are particularly suited for this application because they have what we call a high-power sensitivity. This means that a relatively high power output can be obtained with only a small driving power applied to the grid. The power sensitivity of a beam-power tube, for example, is much higher than that of a triode tube.

(f) A remote cut-off tube is a tube which requires comparatively high negative grid voltage in order to cut off the flow of plate current. As the grid voltage is made negative the flow of plate current will decrease gradually rather than sharply as in the case of a sharp cut-off tube. A remote cut-off tube is made with a grid that has a variable spacing between the turns. For example, the various turns of the spiral-wound grid may be quite close together at the two ends of the grid but spaced comparatively far apart in the center of the grid. This means that if we apply a negative voltage to the grid of the tube the voltage applied to the ends will have comparatively high effect insofar as cutting off the flow of plate current is concerned, because the turns of the grid are spaced quite close together. However, in the center of the grid they will have very little effect because the turns are spaced far apart. As we increase the negative voltage applied to the grid, the part of the grid through which electrons still pass is gradually reduced until eventually when the negative voltage is made high enough the flow of plate current will be completely cut off.

In a tube where the spacing between the various turns of the grid is constant, the plate current can be cut off with comparatively low grid negative voltage. The flow of current from the cathode to the plate through the grid turns is reduced at a constant rate over the entire grid structure as a negative voltage is applied to the grid. This type of tube is called a sharp cut-off tube because it cuts off the flow of plate current with a comparatively low grid voltage. In the case of a remote cut-off tube, because of the variable spacing, the various parts of the grid have a different effect on the flow of plate current as the negative voltage is applied to the tube. Since it requires a much higher negative voltage to cut off the flow of plate current in this type of tube, we refer to the tube as a remote cut-off tube. Remote cut-off tubes are used in rf and i-f stages of receiving equipment where the gain of the stage is to be controlled either manually or by an automatic gain control.

(g) Duo-triode. This type of tube is simply a tube in which two triodes have been placed inside the same envelope. In most cases the two triode sections will have identical characteristics; however, this is not always the case. With modern compactron tubes, you might even run into tubes where there are three triodes inside the same envelope. The advantage of dual or triple tubes of this type is the saving in space and cost.

(h) A cold-cathode tube is a tube in which the cathode of the tube is not heated. This type of tube will usually have a cathode and plate and the tube will be filled with a gas rather than evacuated as in the case of most diodes. This type of tube is widely used as a voltage regulator. It is used in most cases in series with a resistor. When the voltage is applied and no current is flowing through the resistor, there will be no voltage drop across the resistor. This will enable the voltage between the plate and cathode of the tube to rise to the full applied value.

If the voltage is high enough the gas inside of the tube will ionize. When this happens current begins to flow from the cathode to the plate of the tube.

Once the gas is ionized the gas has a certain ionization potential depending upon the type gas in the tube. In other words, the voltage between the plate and cathode of the tube will remain at some fixed value. This value will be less than the voltage required originally to ionize the gas. The current flowing through the tube to the resistor will be such that the voltage drop across the resistor, when subtracted from the applied voltage, will maintain the voltage across the tube at the ionization potential.

If the applied voltage tends to go up, the current will increase so that the voltage drop across the resistor will increase and the voltage drop across the tube remains constant. If the opposite happens and the applied voltage decreases, the current flow through the tube will decrease so that the voltage drop across the resistor will decrease, and the voltage drop across the tube will remain constant. Cold-cathode tube are used primarily as voltage regulators.

(i) A thyratron is a three-element gas-filled tube. When a positive voltage is applied to the plate of the thyratron and a negative voltage to the grid, no current will flow through the tube. However, once the grid voltage is reduced to a low value or removed completely, electrons will begin to flow from the cathode to the plate of the tube. The electrons flowing through the tube will strike gas molecules knocking electrons from the molecules and ionizing the gas. This causes a comparatively low-resistance path to be formed between the cathode and the plate of the tube. The gas molecules, from which electrons have been removed, will be ionized and will all have a positive charge. If we apply a negative voltage to the grid to try to cut off the flow of plate current, these positively charged gas ions will simply be attracted to the grid and thus prevent it from cutting off the flow of current to the tube. Once current has begun to flow through the thyratron tube the only effective way of permitting the grid to regain control is by bringing the plate voltage down to zero.

Thyratrons are frequently used in control applications. They are usually used in applications where an ac voltage is applied to the plate so that the plate voltage will drop to zero each cycle and permit the grid to regain control of current flow through the tube.

38. What is the principal advantage of a tetrode tube over a triode tube in a radio-frequency amplifier?

The principal advantage of a tetrode over a triode as a radio-frequency amplifier is that the tetrode tube normally does not require neutralization. This is due to the fact that the plate-to-grid capacity of a tetrode is much lower than the plate-to-grid capacity of the triode. The screen grid of the tetrode tube effectively forms an electrostatic shield between the plate and grid so that little or no energy can be fed from the plate of the tube back to the grid. This means that the tube is not likely to go into oscillation when it is used as an amplifier. A triode tube, if it is to be used as an rf amplifier, must be neutralized or otherwise it will oscillate. Other advantages of a tetrode over a triode are higher amplification factor (μ), higher mutual conductance (g_m) and higher plate resistance (r_p). A pentode tube has the same advantage over a triode.

39. Compare tetrode and triode tubes in reference to high plate current and inter-electrode capacitance.

A tetrode tube has a much higher plate resistance than a triode tube. You will remember that in a tetrode tube the screen has a much greater effect on plate current than the plate does. This means that for a given change in plate voltage there is a very small change in plate current. This accounts for the high plate resistance of the tetrode tube. In a triode tube, however, a change in plate voltage will cause an appreciable change in plate current. Therefore the triode has a comparatively low plate resistance. Due to the higher plate resistance of the tetrode tube, it can work into a much higher load impedance. This means that for a comparatively small change in plate current, compared to that required from a triode, a relatively large power output can be obtained. We often refer to this fact by saying that a tetrode has a higher power sensitivity than a triode.

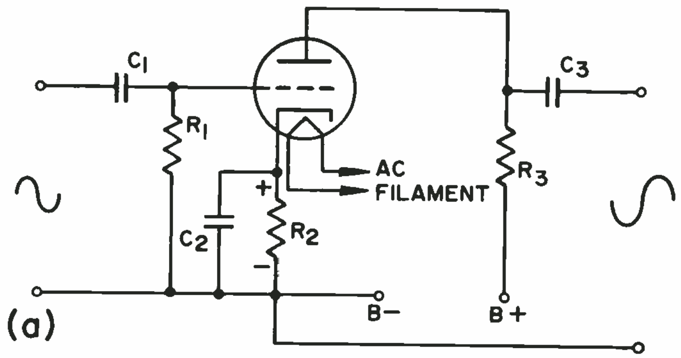
As far as inter-electrode capacities are concerned, the important capacity is the plate-to-grid capacity of the tube. In a tetrode tube, the screen grid effectively forms an electrostatic shield between the plate and grid so that the plate-to-grid capacity of the tetrode tube is comparatively small. The plate-to-grid capacity of a triode tube is large enough so that considerable energy can be fed from the plate back to the grid, and in most rf applications the triode will oscillate unless it is neutralized to eliminate the effects of this undesirable capacitance.

40. Are there any advantages or disadvantages of filament-type vacuum tubes when compared with the indirectly heated types?

The advantage of the filament-type vacuum tube is that it can be used in tubes where comparatively high plate voltages are used. If you try to use an indirectly heated cathode-type tube with a metal oxide on the cathode, the high plate voltage will pull the oxide right out of the cathode structure and ruin the tube. The disadvantage of filament-type vacuum tubes is that they are more subject to hum than indirectly heated types when an ac filament voltage is used. In addition, the efficiency of the filament-type tube is low and considerably more power is required to heat the filament to operating temperature than with the indirectly-heated type.

41. Draw a simple circuit diagram of each of the following and describe operation. Show a signal source, and include coupling and bypass capacitors, power supply connections and plate load.

(a) AF "grounded-cathode" triode amplifier with a cathode resistor biasing, as for class A operation.
(b) AF "grounded-cathode" pentode amplifier with battery biasing, for class A operation.
(c) RF "grounded-grid" triode amplifier with LC tank plate-load for class B operation.
(d) AF "cathode-follower" triode amplifier.
(e) AF "push-pull" pentode amplifier, operated class B with transformer coupling to a speaker.



The circuit shown above is an af grounded-cathode triode amplifier with cathode biasing. The ac voltage applied to the heater heats the cathode of the tube. Electrons will flow from B minus through the cathode bias resistor R2 to the cathode of the tube. They will flow from the cathode of the tube to the plate and through R3 back to B+.

When the electrons flow through R2 they will set up a voltage drop across the resistor having the polarity shown. The value of the voltage will depend upon the current flowing and the size of the resistor. The size of the resistor is selected so that when normal plate current flows through the resistor, the required bias will be developed.

The ac input signal causes the voltage between the grid and cathode of the tube to vary. When the signal swings the grid positive the bias between the grid and cathode will be reduced and this will cause the current flowing from the cathode to the

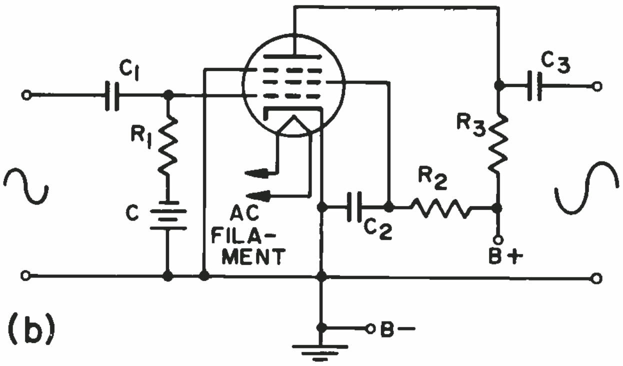
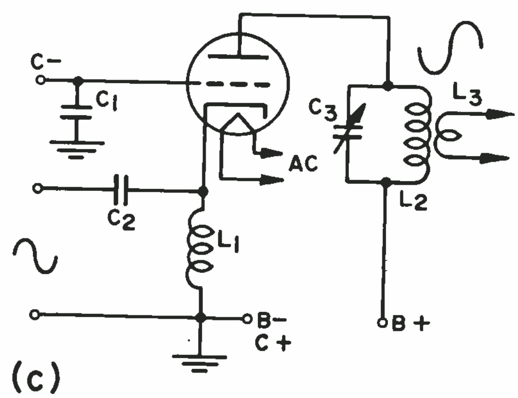


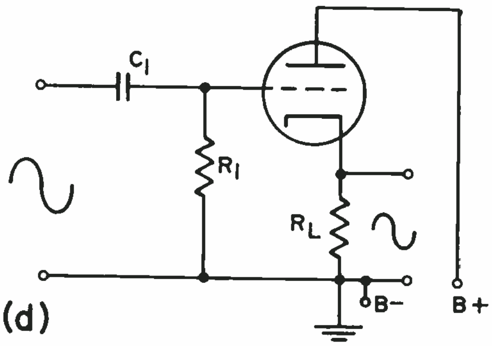
plate of the tube to increase. This increase in current will cause an increase in voltage drop across R3. Thus the potential between the plate of the tube and ground will swing in a negative direction as shown in the output signal.

When the input signal swings in a negative direction, it will add to the bias applied between the grid and cathode of the tube and reduce the flow of plate current through the tube. This will reduce the voltage drop across R3, causing the plate of the tube to swing in a positive direction, as shown in the output signal.

The circuit above is a pentode audio-frequency amplifier with battery biasing. The C battery connected between ground and the grid resistor R1 applies a negative voltage to the grid of the tube. The battery voltage will be sufficient to bias the tube on the mid-portion of the characteristic curve. The operation of the amplifier is similar to the operation of the triode amplifier discussed previously.

The circuit shown below is an rf grounded-grid amplifier. The C battery places sufficient bias on the grid of the tube to bias the tube to cut-off, which is the bias required for class B operation. The input signal is fed into the cathode circuit through the capacitor C2. L1 is an rf choke used to isolate the cathode from rf ground and at the



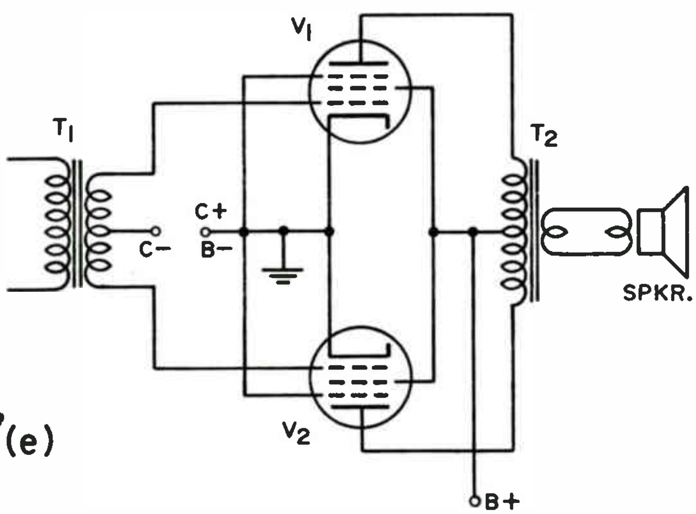


(d)

same time provide a dc path from B-2 to cathode. The tank circuit, consisting of L2 and C3 in the plate circuit of the tube, is a parallel-resonant circuit. It restores the plate current pulses to a sine wave as shown. L3 is inductively coupled to L2 and L3 can either feed to an antenna or to a following stage.

The circuit shown above is a cathode-follower amplifier. The input signal is fed to the grid of the tube through C1 and is applied between grid and ground across R1. The actual signal applied between grid and cathode of the tube will be the input signal less the signal voltage developed across the load resistor R2, which is in the cathode of the circuit. Notice that the signal in the cathode circuit is smaller in amplitude than the input signal; this is true of all cathode-followers. Also notice that the output signal is in phase with the input signal.

A push-pull audio frequency class B power amplifier is shown below. The two tubes are biased to cut-off by the C battery that connects between ground and the center tap of the secondary of the input transformer T1. Notice the input transformer is a step-down transformer used to match the low input impedance of the grid circuit. The input impedance will be low because the grids



(e)

will draw grid current. The output transformer T2 matches the plate impedance of the tubes to the low speaker voice coil impedance. Detailed operation of the push-pull amplifier is given in your lesson text.

42. What kind of vacuum tube responds to filament reactivation and how is reactivation accomplished?

Thoriated filaments respond to filament reactivation. When the emission from this type of filament falls off, it is due to the fact that the layer of thorium or the surface of the filament has been used up. By increasing the filament voltage of the tube to two or three times its normal value for a short period of time, additional thorium can be more or less boiled out of the interior of the filament up to the surface, and will once again be available to supply an abundance of electrons.

43. What is meant by "space charge"? By secondary emission"?

When electrons are emitted from the cathode of a vacuum tube they form a cloud around the cathode between the cathode and the grid of the tube. Since all the electrons are negative, they form a charge in the space around the cathode. This is referred to as the space charge.

When an electron strikes the plate of a tube, it is generally travelling at a fairly high speed. Often it will knock other electrons off the plate of the tube. This knocking loose of electrons is referred to as the secondary emission.

44. What is meant by the "amplification factor" (mu) of a triode vacuum tube (amplifier)? Under what conditions would the amplifier gain approach the value of mu?

The amplification factor of a tube is the ratio of the change in plate voltage to the change in grid voltage required to produce the same change in plate current. In other words, if we can change the plate current in a tube 2 milliamperes by changing the plate voltage 20 volts and can also change the plate current 2 milliamperes by changing the grid voltage 2 volts, the amplification factor of the tube would be $\frac{20}{2} = 10$.

A vacuum-tube amplifier acts much like a generator with an internal resistance. The internal resistance is the plate resistance of the tube. The total amplified signal produced by the tube is equal to the input signal times the amplification factor of the tube. This amplified signal is divided between the plate resistance of the tube and the plate load resistor. The larger the plate load resistor is, in comparison with the plate resistance of the tube, the closer the gain of the amplifier approaches the value of mu.

45. What is meant by "plate resistance" of a vacuum tube? Upon what does its value depend?

The plate resistance of a vacuum tube is the ratio of a small change in plate voltage to the change in plate current it produces. Plate resistance of a vacuum tube depends upon how effective the plate voltage is in controlling plate current. This in turn depends upon the tube type. For example, in a tetrode or pentode tube the screen voltage primarily controls plate current; the plate voltage has very little effect on plate current. Therefore a comparatively large change in plate voltage will cause very little change in plate current. This means that the tube will have a high plate resistance.

In a triode tube, however, a change in plate voltage has far more effect on the plate current. In triode tubes the relative spacing between the grid and cathode and plate and cathode will primarily determine the plate resistance of the tube.

46. What is meant by the voltage "gain" of a vacuum-tube amplifier? How does it achieve this gain?

The voltage gain of a vacuum-tube amplifier is the ratio of the output signal voltage to the input signal voltage. A vacuum-tube amplifier is capable of producing gain because the relatively small input signal voltage causes a change in plate current. This changing plate current flowing through the plate load resistor produces an amplified signal voltage across the load resistor that is greater than the input signal voltage.

47. What advantage may a bridge rectifier circuit have over a conventional full-wave rectifier?

With a bridge rectifier a secondary of the power transformer need not be center-tapped. Therefore for a given output voltage, fewer turns are required on the secondary winding with a bridge rectifier than with a full-wave rectifier.

48. What are "swinging chokes"? Where are they normally used?

A swinging choke is a choke that saturates rather easily. As the current through the choke increases the inductance decreases because of core saturation. This means that the opposition that the choke offers to the flow of current through it varies.

Swinging chokes are normally used in power supplies, particularly power supplies using mercury vapor rectifier tubes. They are used with the input choke between the filament of the tube and the first filter capacitor. The variable inductance helps provide for better voltage regulation.

49. What are the characteristics of a capacitor-input filter system as compared to a choke-input system? What is the effect upon a filter choke of a large value of direct current flow?

With a capacitor-input filter you would normally get a higher dc voltage than with a choke-input filter. However, the voltage regulation of a capacitor-input filter system is poorer than that of a choke-input system.

A large value of direct current flowing through a filter choke may tend to saturate the choke and reduce its inductance. It will most certainly reduce its effectiveness as a filter, particularly if there is a large ac current component superimposed on the direct current flow.

50. What is the purpose of a "bleeder" resistor as used in connection with power supplies?

A bleeder serves two purposes. It provides a minimum load on the power supply to keep the power supply voltage from rising to too high a value. A bleeder also serves as a safety device. Filter capacitors used in power supplies may hold a charge for a long time. When there is a bleeder used in the power supply, the capacitors can discharge through the bleeder and thus eliminate the possibility of the technician working on the power supplies receiving a dangerous shock even when the power is turned off.

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T-2:FCC

NRI Study Guide for FCC License Examination

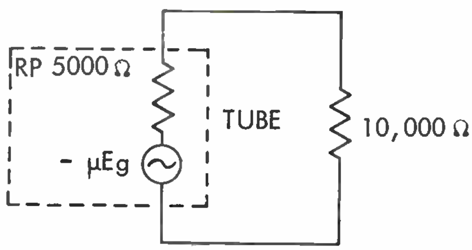
PART III

This part of the NRI FCC study guide deals with tubes and transistors. Study this part of the guide as soon as you receive it, and then answer the questions in the FCC-type examination TC-3 which you received along with the guide. Save the study guide when you are through so that eventually you will have a complete study guide you can use for review before taking your FCC license test.

51. A triode, "grounded cathode", audio amplifier has a "mu" (amplification factor) of 30, a plate impedance of 5000 ohms, load impedance of 10,000 ohms, plate voltage of 300 volts, plate current of 10 ma, and a cathode-bias resistor is used.

    (a) What is the stage gain of this amplifier?
    (b) What is the cut-off bias voltage eco?
    (c) Assuming the bias voltage is one-half the value of eco, what value cathode resistor would be used to produce the required bias?
    (d) What size capacitor should be used to sufficiently bypass the cathode resistor if the lowest approximate frequency desired is 500 cycles per second?

(a) In solving problems of this type we use what is called an equivalent circuit. In an equivalent circuit we represent the vacuum tube as a generator with an internal resistance in series with the generator. The internal resistance is the plate impedance of the tube; in the example given this is 5000 ohms. Connected across the generator with its internal resistance is the load impedance, which in this case is 10,000 ohms. The equivalent circuit would look like the following circuit.



The generator voltage is represented as $-\mu E_g$. The Greek letter μ (called mu) is used to represent the amplification factor of the tube. The total voltage of the generator will therefore be the input voltage times the amplification factor of the tube. Since no input voltage is given, we can assume that the input voltage is 1 volt, and this would give us a total generator voltage of $-30 \times 1 = -30$ volts. We indicate that this voltage is a negative voltage because the amplified voltage will be 180° out-of-phase with the grid voltage due to the reversal that occurs within the tube.

The total generator voltage of -30 volts is then divided across the internal resistance of the generator and the load impedance. In other words, the 30 volts will be divided across the total of 15,000 ohms. The amount of this voltage that will appear across the load resistance depends upon the ratio of the load resistance to the total resistance in the circuit. In other words, the amplified signal voltage will be equal to 30.

$$30 \times \frac{10{,}000}{10{,}000 + 5000} =$$

Triode gain $= \mu \times \frac{R_L}{R_L + R_p}$

$$\frac{30 \times 10{,}000}{15{,}000} = \frac{300{,}000}{15{,}000} = 20 \text{ volts}$$

This means that the output voltage is 20 volts for an input voltage of 1 volt. Therefore the gain of the stage is $\frac{20}{1} = 20$.

Cut-off bias $= E_p \div \mu$

(b) The cut-off bias voltage can be obtained from the knowledge that the amplification factor is 30 and the plate voltage is 300 volts. Since the amplification factor is 30, this means that the grid is 30 times more effective in controlling the plate current than the plate voltage. Therefore with a plate voltage of 300 volts, we need only to divide this voltage by the amplification factor of 30 to get the grid voltage that will be required to cut off the plate current. $300 \div 30 = 10$, therefore the cut-off bias voltage, eco, is 10 volts.

(c) Assuming the operating bias voltage is one-half the value of cut-off bias eco, the grid bias voltage should be 5 volts. Since the plate current is 10 ma, we can calculate the size cathode resistor required to get this voltage by using Ohm's Law.

$$R = \frac{E}{I}$$

Cathode R $= \frac{\frac{1}{2}\,eco}{I_p\,(\text{AMPS})}$

$$R = \frac{5}{.01}$$

$$= 500 \text{ ohms}$$

(d) When the reactance of the cathode bypass capacitor is 1/5 or less of the cathode resistance at the lowest desired operating frequency it will provide proper bypassing. Therefore, the reactance can be as high as 100 ohms.

Xc must be at least 1/5 of Cathode R or less

We know that the reactance of a capacitor is equal to $\frac{1}{6.28 \times f \times C}$ This is usually given in the formula

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

We can rearrange this formula by cross-multiplication, when we know the reactance and want to find the capacity, we simply change the position of Xc and C so that we will have:

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

and substituting 500 cycles for the frequency and 100 ohms for Xc we have:

$$C = \frac{1}{6.28 \times 500 \times 100}$$

$$C = \frac{1}{.314 \times 10^6}$$

$$C = 3.2 \times 10^{-6} \text{ farads}$$

$$= 3.2 \ \mu fd$$

Since the FCC-type examinations are multiple-choice, this value might be given as one of the choices or a somewhat larger capacitor value may be given. A larger capacitor will give even better response at the frequency of 500 cycles. However, 3.2 mfd is the smallest value that could be used and still provide proper bypassing.

52. Why is the efficiency of an amplifier operated Class C higher than one operated Class A or Class B?

The efficiency of an amplifier is the percentage power output compared to the power input. A Class A amplifier is essentially a linear amplifier. The tube is biased to operate on the mid-portion of its characteristic curve. Thus a fixed value of dc current flows at all times even when no signal is applied to the input. The ac voltage and ac current supplied to the load under these conditions are zero. When an input signal is applied to the tube, the plate current can be driven from its dc level to zero on one half-cycle and to twice the dc value on the other half-cycle. Similarly, the voltage across the load can be driven to zero on one half-cycle and a value equal to twice the dc level on the other half-cycle. Thus we have a situation where the signal voltage is superimposed on the dc voltage. The dc voltage at the output represents zero ac voltage. The peak value of the signal voltage, since it drives from the dc level to zero on one half-cycle and twice the value of the dc level on the other half-cycle, is equal to the dc voltage. Therefore the rms value of the voltage is .707 times the dc voltage. Similarly, the ac current is equal to .707 of the dc current. Therefore the power output is equal to $.707E \times .707I = .5EI$. This means that the output signal can have a maximum value equal to half the dc input; therefore the maximum possible theoretical efficiency of a Class A amplifier is only 50%.

In a Class B amplifier, the tube is biased to cutoff. Therefore when the grid is driven in a negative direction, no current flows. When the grid is driven in a positive direction, a high pulse of current flows, and a relatively high voltage will be developed across the plate load resistance. Since current flows only during one half-cycle, the average value of current flowing through the tube is much lower than in the case of a Class A amplifier. Therefore since its average value represents the dc input, the ratio of the power output to the power input is higher for a Class B amplifier than it is for a Class A amplifier.

A Class C amplifier is biased at two to four times cutoff. This means that plate current flows for even a smaller portion of a cycle in a Class C amplifier than it does in a Class B amplifier. Therefore the average or dc input through the stage will be even lower in a Class C amplifier than in a Class B amplifier. This means that for a given power output we will have a lower power input in a Class C amplifier than in either a Class B or a Class A amplifier. Since efficiency is the ratio of power output to power input, the efficiency of a Class C amplifier will be higher than the efficiency of either a Class A or Class B amplifier.

53. The following are excerpts from a tube manual rating of a beam pentode. Explain the significance of each item:

Control grid-to-plate capacitance.........1.1 uuf

The control grid-to-plate capacitance of the tube is the capacitance between the plate and grid of the tube. This represents the total capacitance formed between the elements in the tube, the leads connected to the elements and the tube pins.

Input capacitance...............................2.2 uuf

The input capacitance of the tube is the total capacity seen between the grid and cathode of the tube. It is made up mainly of the actual capacity between the grid and cathode of the tube plus any additional capacity that may be added by the grid-to-plate capacity and the output capacity of the tube.

Output capacitance.............................8.5 uuf

The output capacitance of the tube is the total capacity between the plate and cathode of the tube.

Heater voltage................................6.3 volts

The heater voltage is the nominal voltage which is to be applied to the heater of the tube. Most tube manufacturers recommend holding the heater voltage as close as possible, but they do rate the voltage with a tolerance usually of ±10%.

Maximum dc plate-supply voltage.........700 volts

This is the maximum power-supply voltage. Usually the actual dc voltage applied to the tube must be held less than this value.

Maximum peak positive pulse voltage...7,000 volts

This is the maximum value that a positive pulse reaching the plate of the tube may have. In certain applications, for example in the horizontal sweep in a television receiver, a large pulse may be applied to the plate of the horizontal sweep output tube. The maximum value that this pulse can reach safely without danger of damage to the tube is given as the maximum peak positive pulse voltage.

Maximum negative pulse plate voltage.......1,500 volts

This is the maximum peak value that any negative pulse reaching the plate of the tube may have without damaging the tube.

Maximum screen grid voltage.............175 volts

This is the maximum dc voltage which can be applied to the screen grid of the tube.

Maximum peak negative control grid voltage.... 200 volts

This is the maximum negative voltage that can be applied to the grid of the tube. This is usually made up of a combination of bias voltage plus the negative half-cycle of the input signal voltage. The sum of the two must not exceed 200 volts.

Maximum plate dissipation.................20 watts

This is the maximum power which the plate of the tube can dissipate. It represents the dc input to the tube plate circuit, less the output signal. Usually, operating the tube in excess of the plate dissipation will greatly reduce the life of the tube.

Maximum screen grid dissipation.........30 watts

This is the maximum power that the screen grid can dissipate. Since the screen grid is normally operated at ac ground potential, this usually represents the dc input to the screen.

Maximum dc cathode current.................200 ma

This is the maximum average or dc current that the cathode of the tube can supply. The current can exceed this value for short intervals providing it also drops below this value for an equal time and amount, so that the average or dc current does not exceed 200 ma.

Maximum peak cathode current................700 ma

This is the maximum peak current that the cathode current can reach during any part of the cycle. In other words, if the grid is driven highly positive so that a very high cathode current pulse flows, the cathode current cannot be allowed to exceed 700 ma.

Maximum control-grid circuit resistance...... .47 megohms

This is the maximum dc resistance that may be in the grid circuit of the tube. If the dc resistance exceeds this value, electrons accidentally striking the grid may set up excessive bias across the grid resistor, or gas within the tube may cause the grid voltage to become sufficiently positive to adversely affect the operation of the tube.

54. Name at least three abnormal conditions which would tend to shorten the life of a vacuum tube. Name also one or more probable causes of each condition.

Incorrect heater or filament voltage will shorten the life of a vacuum tube. If the heater or filament voltage is too high, there is a danger of the heater or filament simply burning out. Also there will be excessive emission and the active material, particularly in an oxide-coated tube, will be used at an excessive rate.

If the heater voltage is too low, particularly in a tube that uses a very high plate voltage, the high plate voltage would simply pull material directly off the filament and soon ruin it.

Excessive plate or screen voltage will cause excessive current to flow and exceed the plate or screen dissipation of the tube. Such a condition may be due to a higher than normal plate-supply voltage. The screen voltage may be excessive if the screen-dropping resistor is too small or if the screen voltage is obtained from a power supply having too high a voltage.

Excessive grid current will quickly damage a beam power tube. Excessive grid current is caused by excessive drive.

55. Would varying the value of a bleeder resistor in a power supply have an effect on the ripple voltage?

The ripple voltage in a power supply is affected by the load on the power supply. Therefore if the value of the bleeder resistor is changed appreciably so as to change the load on the power supply, it will have an effect on the ripple voltage. Changes in the bleeder resistor will be particularly noticeable when the power supply is heavily loaded, so that any change in current drawn from the power supply will be more effective in changing the ripple voltage.

56. What effect does the amount of current required by the load have upon the voltage regulation of a power supply? Why is voltage regulation an important factor?

The voltage regulation of a power supply is equal to the no-load voltage minus the full-load voltage over the full-load voltage. This figure is usually multiplied by 100 to present this figure as a percentage. The greater the load on the power supply, the lower the voltage will drop. Therefore if the full-load voltage is high, the voltage will drop more than it would with a light load. This means that the voltage regulation of the power supply would be poorer with a heavy load than it would be with a light load.

Voltage regulation is an important factor because changes in voltage from the power supply can affect the operation of the equipment. In the case of a transmitter, changes in voltage from the power supply can affect the oscillator frequency and the power output. In addition, if the voltage varies over a wide enough range, some of the components in the equipment can actually be damaged when the voltage reaches its peak value.

57. Discuss the relative merits and limitations as used in power supplies of the following types of rectifiers:

(a) Mercury-vapor diode
(b) High-vacuum diode
(c) Copper oxide
(d) Silicon
(e) Selenium

A mercury vapor rectifier tube has a fixed internal voltage drop of about 15 volts. This voltage drop remains essentially constant regardless of the current flowing through the tube. The tube is an efficient rectifier and can handle a comparatively high current. However, it is somewhat critical in operation.

Before applying a voltage to the plate of the tube you must bring the tube up to operating temperature. Also, sometimes mercury-vapor tubes are a source of rf interference.

A high-vacuum rectifier is not as critical in operation as a mercury-vapor tube. However, the voltage drop across a high-vacuum diode will vary with the current through the tube. The tube has the additional advantage that it usually will not generate rf interference and also will withstand a higher peak-inverse-voltage than a mercury vapor tube.

The copper oxide rectifier is primarily used in meters rather than in power supplies. It is used in low-voltage, low-current applications, where linear characteristics are required.

The silicon rectifier is capable of very high currents compared to its size. Modern silicon rectifiers can be made to operate at comparatively high voltages and currents. However, they are subject to damage by current surges and are quite critical as to peak-inverse-voltages.

The selenium rectifier was quite widely used before silicon rectifiers were developed. It is somewhat less susceptible to temperature changes than the silicon rectifier and can be made to withstand comparatively high voltages by stacking rectifier cells in series. However, the unit has a much higher resistance than the silicon diode, and therefore it dissipates far more heat, and large cooling fins must be used to get rid of its heat. Selenium rectifiers are much larger than silicon rectifiers rated at the same current. Both selenium and silicon diodes have the advantage that no heater power is required.

58. Explain the action of a voltage regulator (VR) tube.

A voltage regulator is a gas filled tube that has a cathode and plate. If a voltage is applied across the tube so that the cathode is negative and the plate positive, and the plate voltage reaches a high enough value, the gas inside the tube will ionize. This voltage is called the firing potential. Once the gas has ionized, the voltage across the tube will drop to a somewhat lower value called the ionization potential. The exact value will depend upon the gas used and the pressure of the gas inside the tube. Voltage regulator tubes are usually operated in series with a resistance so that if the voltage across the tube starts to increase, the current through the tube will increase. This will increase the voltage drop across the resistor so the voltage drop across the tube will remain almost constant.

59. If the plate, or plates of a rectifier tube suddenly become red hot, what might be the cause; and how could remedies be effected?

When the plates of a rectifier tube suddenly become red hot, it means that the rectifier tube is being overloaded. The most probable cause of the trouble is a shorted filter capacitor in the power supply. However, the trouble could also be due to a shorted bypass capacitor or some tube that is drawing excessive current. The obvious remedy is to check the filter capacitors and bypass capacitors for shorts, and if none are found, then the trouble should be traced to an individual stage.

60. If a high-vacuum, high-voltage rectifier tube should suddenly show severe internal sparking and then fail to operate, what elements of the rectifier-filter system should be checked for possible failure before installing a new rectifier tube?

The filter capacitor has probably shorted. This would place the rectifier tube directly across the high voltage supplied by the transformer and it will usually break down almost immediately. Before installing a new rectifier tube, the filter capacitors should be checked and the defective one replaced.

61. What does a blue haze in the space between the filament and plate of a high-vacuum rectifier tube indicate?

A blue haze between the plate and filament of a high-vacuum rectifier tube indicates that the tube is gassy. Such a condition demands that the tube be replaced immediately.

62. What is meant by the "peak-inverse-voltage" rating of a diode and how can it be computed for a full-wave power supply?

The peak-inverse-voltage rating of a diode is the peak or maximum reverse voltage that can be applied across the diode. Normally when the plate of a diode is positive, the tube conducts and there is a relatively low-voltage drop across the tube. However, when the plate is made negative and the cathode positive, the tube will not conduct. This is called an inverse voltage or a reverse voltage. There is a maximum or peak value that a tube can withstand before it will break down. If you exceed this peak-inverse-voltage rating, the chances are that the tube will arc internally.

In a full-wave power supply, the high-voltage secondary winding of the power transformer is normally center-tapped. Current flows through one-half during one half-cycle and the other half during the other half-cycle.

When current is not flowing in one half of the winding, the voltage has the wrong polarity to make the diode conduct. In other words, the voltage applied to the plate of the tube will be negative. If a transformer secondary has a total voltage of 1000 volts, then the voltage across one-half of the winding will be 500 volts. This is the rms value of the voltage. The peak value of the voltage will be 1.4 times this value or approximately 700 volts. The tube must be able to withstand at least this much voltage. PIV = RMS X 1.4
RMS = 1/2 voltage applied across center-tapped secondary

In addition to the voltage applied to the tube by the transformer winding, the filament of the tube is also connected to the power supply filter network. In the case of a capacitor input-type filter, at light loads, the capacitor may charge up to a value almost equal to the peak value of the ac voltage. We have just determined that this is 700 volts, therefore the input filter capacitor may charge up to a value of almost 700 volts. This means we have a voltage between ground and the cathode or filament of the rectifier tube of +700 volts. We have voltage between ground and the plate of the rectifier tube of -700 volts. Thus the total voltage applied between the plate and cathode of the tube is 1400 volts. Therefore the diode must have a peak-inverse-voltage rating of at least 1400 volts, or there is the danger it will arc across internally.

63. Describe the difference between positive (P-type) and negative (N-type) semiconductors with respect to:

(a) The direction of current flow when an external emf is applied.
(b) The internal resistance when an external emf is applied.

When a voltage is applied across an N-type semiconductor material, current in the form of electrons will flow from the negative potential to the positive potential. In other words, the current flow through the semiconductor is by means of electron carriers and is in the same direction as in any other circuit.

When a potential is applied across a P-type semiconductor, current is carried to the semiconductor by means of holes. These holes will be repelled by the positive potential connected to the semiconductor material and attracted to the negative terminal. Thus the holes will be moving through the semiconductor from positive to negative. In the complete circuit, electrons will leave the negative terminal of the voltage source and flow to the negative terminal of the P-type material. Here the electrons will fill holes. Meanwhile, the holes flowing across the semiconductor material reaching the negative terminal will be continually filled by electrons arriving at the negative terminal from the voltage source. At the positive terminal, the positive potential will be removing electrons from atoms near the terminal thus creating holes. The electrons removed from the atoms will flow from the terminal back to the positive source of the voltage source.

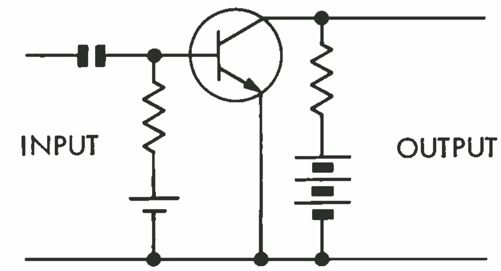
The internal resistance of a semiconductor material will depend upon how heavily the material is doped. The heavier the doping, the more carriers will be available to conduct the current through the material, and hence the lower the resistance will be. The resistance is also affected by the mobility of the carrier. The electrons in the N-type material have a higher mobility than the holes in the P-type material. Thus if equal sizes of N-type material and P-type material which were equally doped were compared, the N-type material would have the lower resistance.

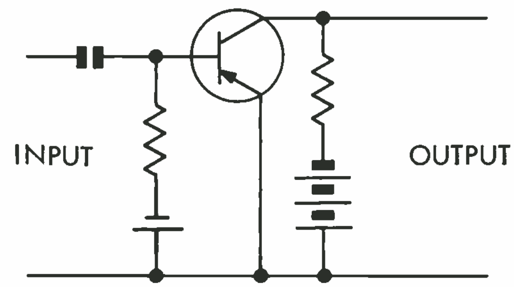
64. What is the difference between forward and reverse biasing of transistors?

When a junction is forward biased, a voltage is placed across it so that majority carriers can cross the junction. In other words, in the case of a P-N junction, the negative potential is applied to the N-side of the junction and the positive potential is applied to the P-side of the junction. The negative repels electrons from the N-type material across the junction; the electrons are then attracted by the positive voltage applied to the P-side of the junction. At the same time, the positive voltage applied to the P-side of the junction repels holes across the junction. The holes are attracted by the negative potential applied to the N-side of the junction.

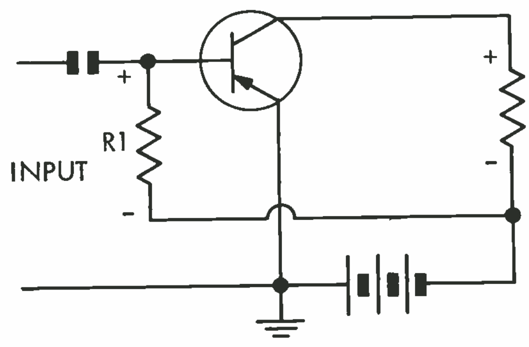
When a junction is reverse biased, it is biased so that majority carriers cannot cross the junction. In the case of a P-N junction, a positive voltage is applied to the N-side of the junction and a negative voltage to the P-side of the junction. A positive voltage applied to the N-side will pull electrons in the N-type material away from the junction and attract them towards the positive potential. At the same time it will repel holes, near the junction in the P-type material, away from the junction. Similarly, the negative voltage applied to the P-side of the junction will repel electrons, near the junction in the N-type material, away from the junction and attract holes in the P-side of the junction towards the negative potential. The reverse voltage has the effect of increasing the width of the depletion layer which forms around the junction and prevents the majority carriers from crossing the junction.

65. Show connections of external batteries, resistance load and signal source as they would appear in a properly (fixed) biased common-emitter transistor amplifier.





66. Draw a circuit diagram of a method of obtaining self-bias with one battery, without current feedback in a common-emitter amplifier. Explain voltage drops in the resistors.



In the circuit shown a PNP transistor is used. The base of the transistor is connected to the negative side of the voltage source through R1. This will place a forward bias across the emitter-base junction. A few electrons will flow through R1 to the base and combine with holes crossing the base. Even though the number of electrons combining with holes in the base is comparatively small, by using a very large resistor for R1, we can develop a voltage drop across it almost equal to the entire supply voltage. This way we can provide a forward bias voltage across the emitter-base junction of only a few tenths of a volt which is all that is needed for most transistors.

The voltage drop across R2 is produced by electrons flowing from the negative side of the battery to the collector. Here the electrons fill holes arriving at the collector. Meanwhile, holes flow from the emitter, across the emitter-base junction, across the base, and then across the base-collector junction to the collector where they are filled by electrons flowing through R2. The holes in the emitter region are produced by electrons flowing from the emitter to the positive side of the voltage source which is connected to ground.

The same circuit can be used with an NPN transistor, but in this case the carriers through the transistor will be electrons instead of holes, and the battery polarity must be reversed.

67. The value of the alpha cut-off frequency of a transistor is primarily dependent upon what one factor? Does the value of the alpha cut-off frequency normally have any relationship to the collector-to-base voltage?

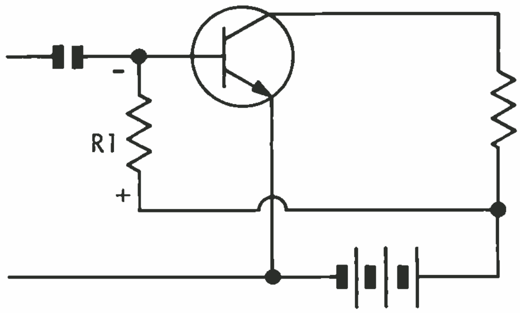
Alpha is the current gain of a transistor in a common-base circuit. The value of alpha is always less than 1 because some of the electrons that leave the emitter and cross the emitter-base junction combine with holes in the base. In a PNP transistor some of the holes crossing the emitter-base junction combine with electrons in the base. The result is that the number of majority carriers reaching the collector is always slightly less than the number of majority carriers leaving the emitter. Thus, the alpha of the transistor is primarily dependent upon the number of majority carriers that are lost in the base region. This to a great extent depends upon the thickness of the base. The alpha cut-off frequency is the frequency at which the value of α drops 3 db.

The collector-to-base voltage does have an effect on the value of the alpha cut-off frequency. If the collector-base reverse bias is made large, the depletion layer across this junction increases in size. This has the effect of reducing the thickness of the base which in turn increases the alpha cut-off frequency.

68. Why is stabilization of a transistor amplifier usually necessary? How would a "thermistor" be used in this respect?

Transistors are usually arranged in a circuit so they can be operated from a single battery. The forward bias required across the emitter-base junction can be obtained by placing a rather large resistor between the base and the collector-voltage source. A small base current flows, and this small current will produce a voltage drop across the base bias resistor so that the net bias across the emitter-base junction will be only a few tenths of a volt. This is usually all that is required to forward bias the transistor.

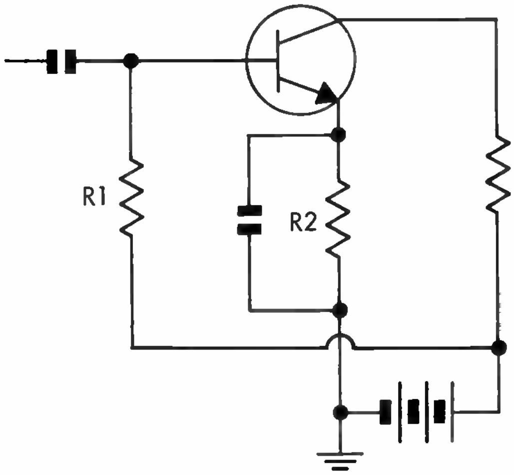
In the circuit shown, an NPN transistor is used.



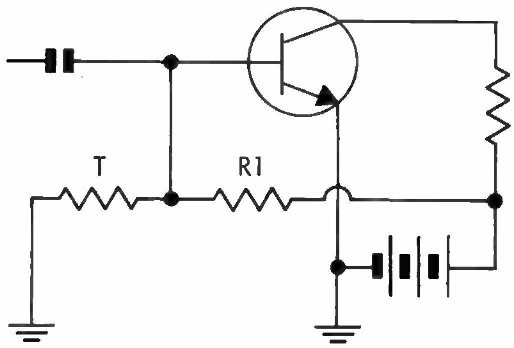
In this circuit, electrons will flow from the emitter across the emitter-base junction and most will flow through the base to the collector. However, a few will leave the base and flow through the resistor R1 developing the voltage drop shown. This voltage drop subtracts from the collector voltage source so that only a small voltage drop is left across the emitter-base junction.

In an NPN transistor, electrons are the majority carriers; holes are the minority carriers. Some holes will cross the base-collector junction and flow from the collector into the base. These holes will recombine with electrons in the base region and in doing so reduce the current flow through R1. This will cause the voltage drop across R1 to decrease, and hence the forward bias across the emitter-base junction will increase. The number of holes crossing from the collector to the base will depend upon the resistance of the junction, and this in turn will depend upon the temperature. The higher the temperature of the junction, the lower the resistance, and hence the more holes that will be crossing from the collector into the base. An increase in current due to an increase in forward bias across the emitter-base junction will cause more electrons to cross the base-collector junction and heat the junction still further. This will cause still more holes to cross from the collector to the base, reducing the voltage drop across R1 still further, causing the forward bias across the emitter-base junction to increase still more. This action will continue, and as a result the current will build up, and we will have what is known as thermal runaway. The current will be of such a high value the transistor will be destroyed.

One of the simplest means of overcoming this thermal runaway problem is to place a resistor in the emitter circuit between the emitter and ground as shown.



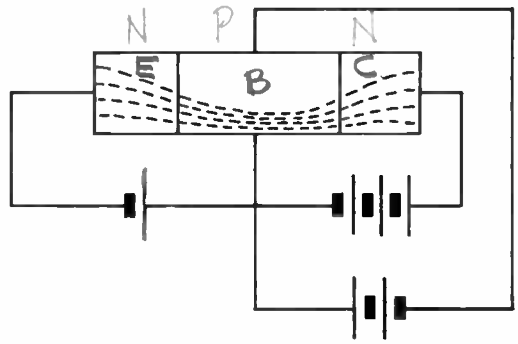
With this arrangement, if the forward bias across the emitter-base junction increases and current through the transistor increases, the voltage drop across R2 will also increase. This voltage across R2 will subtract from the forward bias across the emitter-base junction and tend to hold this bias constant so that the current increase will be limited.



Another circuit that can be used to prevent thermal runaway makes use of a thermistor. A thermistor is a resistor that has a negative temperature coefficient. In the circuit shown, if the current through R1 decreases due to a drop in electrons flowing from the base of the transistor through R1 to the positive voltage source, then the current through the thermistor will increase to make up for this change. If the current increases, the thermistor temperature will increase and this will cause the resistance of the thermistor to decrease. The decrease in resistance will lower the voltage drop across it, and hence tend to maintain the voltage at the base of the transistor essentially constant.

69. What is a junction-tetrode transistor? How do they differ from other transistors in base resistance and operating frequency?

The junction-tetrode transistor is a transistor with two base connections. In many respects it is operated in the same way as a typical transistor. A forward bias is placed across the emitter-base junction, and a reverse bias across the base-collector junction. However, the base has two connections; a negative potential is applied to one connection and a positive to the other, as shown in the illustration. The negative potential, applied to the second base connection forces the electrons flowing through the transistor to take a curved path as shown. The higher the negative potential applied to this terminal, the more the electrons are bent and the more restricted the channel through the base region becomes.



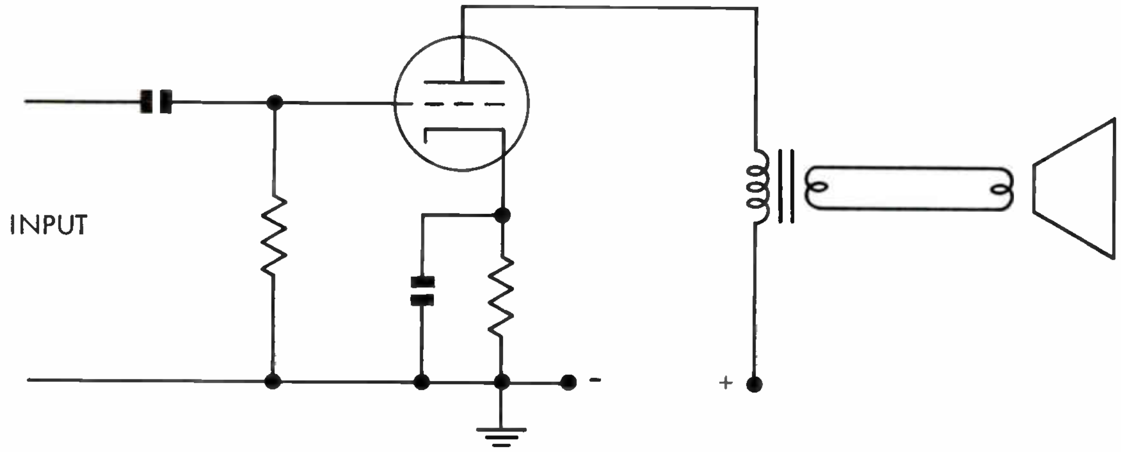
By placing a signal voltage between the voltage used to bias the base and the battery used to bias the base, the bias voltage can be varied, and hence the resistance of the base region can be made to vary; and this will cause a varying current to flow through the transistor.

Due to the fact that the active areas of the junction are quite small in a tetrode transistor, there is less capacitance across the junctions, and hence the transistor is usually capable of operating at a higher frequency than the equivalent triode transistor. The base region of the tetrode transistor has a higher resistance than that of a triode transistor because current flows through only a comparatively small part of the base.

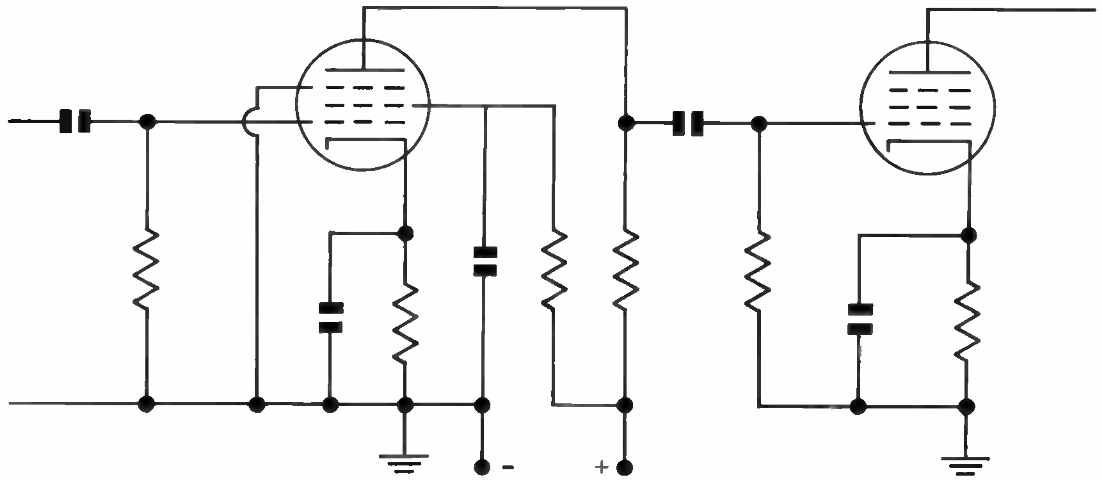
70. Draw a simple schematic diagram illustrating the following types of coupling between audio-amplifier stages and between a stage and a load:

(a) Triode vacuum tube inductively coupled to a loudspeaker.
(b) Resistance coupling between two pentode vacuum tubes.
(c) Impedance coupling between two tetrode vacuum tubes.
(d) A method of coupling a high-impedance loudspeaker to an audio-frequency amplifier tube without flow of plate current through the speaker windings, and without the use of a transformer.

(a)

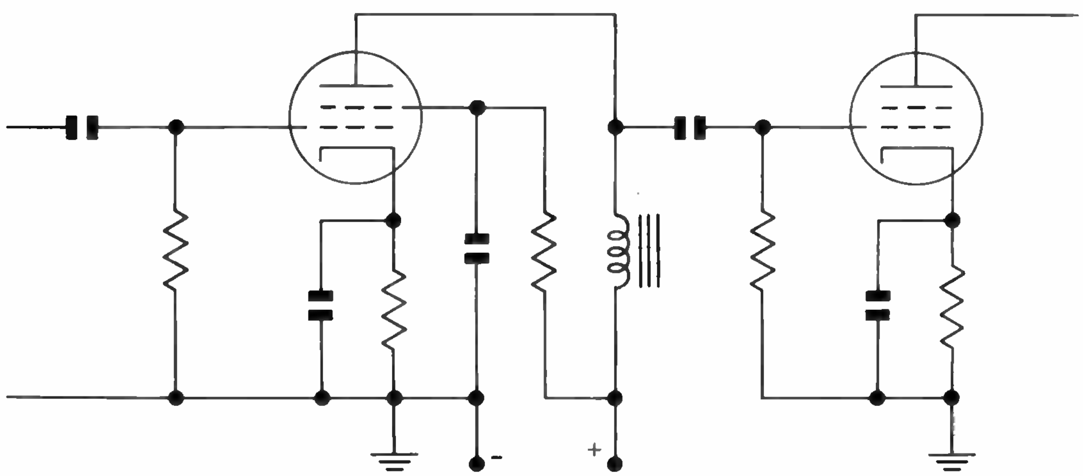


(b)

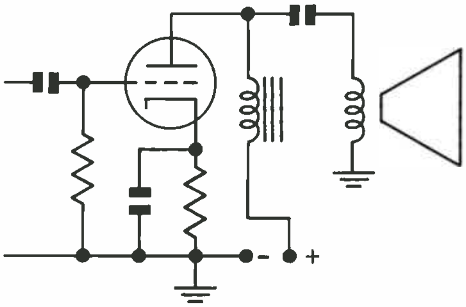


(c) Impedance coupling is essentially the same as resistance coupling except that a choke is used in the plate circuit of the first stage instead of a load resistor. Impedance coupling has the advantage that the choke offers a low resistance to dc and therefore the full power supply voltage is available to operate the plate of the tube. However, it has the disadvantage that the inductance of the choke will vary with frequency, and hence the load that the first stage sees will vary with frequency. At low frequencies, unless a very good quality choke with a high inductance is used, the load will be so low that the gain of the stage will be extremely low. Another disadvantage of impedance coupling is that the cost of the choke is far greater than the cost of a simple plate resistor. As a result the impedance coupling is seldom used.

(c)



(d) Most loudspeakers are low-impedance devices. However, a few high-impedance speakers have been made, but they are not particularly common. To connect a high-impedance speaker to the audio amplifier, you can use impedance coupling. You simply put a choke coil in the plate circuit of the output tube to provide a path for the dc to flow through, and then couple the plate of the tube to the loudspeaker through a suitable coupling capacitor. The circuit is shown after this discussion. As far as the circuit itself is concerned, it is not particularly important but since it is one of the questions that the FCC frequently asks on their license examinations, you should remember the circuit.



71. What would probably be the effect on the output amplitude of a waveform if the cathode-resistor bypass capacitor in an audio stage were removed?

If a bypass capacitor across a cathode resistor in an audio stage is removed, degeneration will be introduced in the stage. This is due to the fact that the varying plate current flowing through the cathode resistor will set up a varying voltage drop which will subtract from the voltage applied between the grid and cathode of the tube. This will result in a reduction of the output amplitude, but at the same time it will also cancel or reduce distortion introduced in the stage. Omitting the cathode bypass capacitor is one simple method often used to improve the amplifier fidelity.

72. Why do vacuum tubes produce random noise?

The emission from the cathode of a vacuum tube is not entirely uniform. In other words, you do not get exactly the same number of electrons coming off the cathode at all times. There is some variation, and this variation effectively modulates the electron stream producing a random noise.

Additional noise is produced in multi-element tubes because the current division between the elements does not remain exactly constant at all times. For example, in a pentode tube, the current from the cathode divides between the plate and screen. However, the division does not remain at exactly the same ratio at all times, there is some random variation which adds additional noise.

73. Why are decoupling resistors and capacitors used in stages having a common power supply?

Decoupling resistors and capacitors are used in stages having common power supplies to prevent feedback from one stage to another through the power supply. Feedback of this type could be degenerative, which might be undesirable because it would reduce the gain of the amplifier substantially. The feedback could also be regenerative and this would be undesirable because it would result in amplifier instability; in fact the amplifier might actually go into oscillation.

74. How would saturation of an output transformer create distortion?

If the dc current through the primary of an output transformer is excessive, the core may saturate. This means it contains all the magnetic lines of flux it can, and any further increase in current causes no further increase in flux.

When the output transformer saturates, its inductance goes down, and as a result, the load seen by the tube working in the primary of the transformer changes. This will introduce distortion because the tube will no longer be operating into the correct load impedance, it will also introduce distortion because the output will go down as the transformer is saturated.

75. Why is noise often produced when an audio signal is distorted?

When an audio signal is distorted, it is usually because a tube or transistor is operated on a nonlinear portion of its characteristic curve. This may result in a number of different types of distortion such as frequency distortion, amplitude distortion and harmonic distortion. In addition, we may run into another type of distortion known as inter-modulation distortion. This is the mixing of the different audio signals present along with the mixing of the signal and their harmonics. All this creates signals which were not present in the input which usually take the form of background noise.

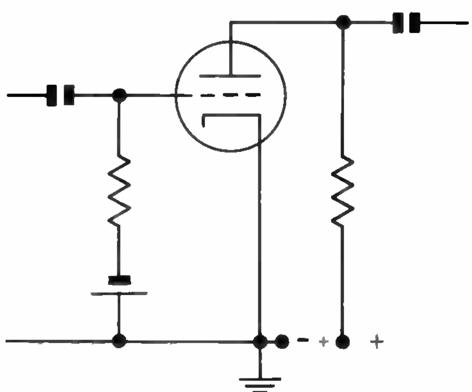
76. What are the factors which determine the correct bias voltage for the grid of a vacuum tube?

The correct bias to be applied to the grid of a vacuum tube depends upon the type of service in which the tube is to be used. For example, a Class A amplifier requires a bias approximately midway between zero bias and cut-off bias. Class B amplifier requires bias at approximately cut-off, and a Class C amplifier requires bias two to four times cut-off bias. The grid bias not only places the tube in the correct part of its characteristic curve, but it also limits the plate current in the tube to a safe value.

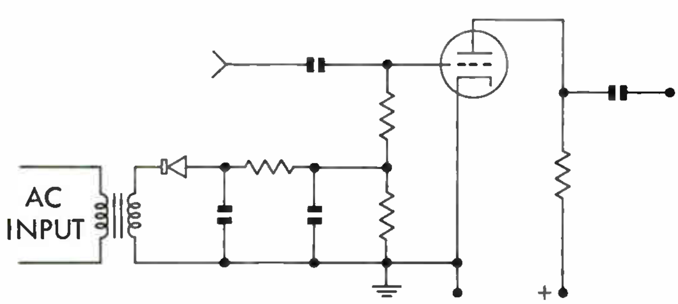
77. Draw schematic diagrams illustrating the following types of grid biasing and explain their operation:

(a) Battery (b) Power supply
(c) Voltage divider (d) Cathode resistor

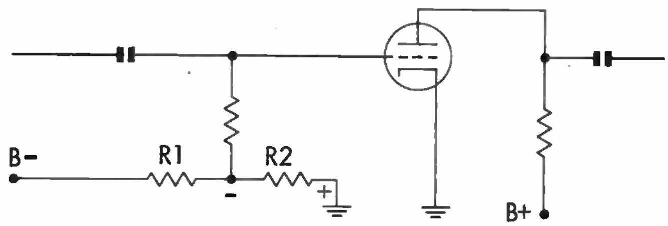
(a) The simplest type of bias is battery bias. Here a battery, usually referred to as a C battery, is connected in the grid circuit to place the required negative voltage between the grid and cathode of the tube.



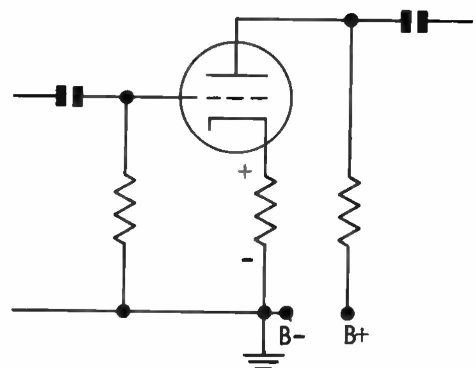
(b) In some of the more elaborate pieces of electronic equipment, bias is obtained by means of a separate winding on the power transformer or by a separate transformer in the bias supply. An arrangement of this type is shown. Here the power supply simply takes the place of the battery in the preceding circuit. Where a power supply is used for this purpose, the supply is usually loaded with a small value of resistor and the impedance of the supply kept low so that any current variation through the resistor will not appreciably vary the bias on the tube.



(c) A voltage divider network may be placed in the negative side of the power supply, as shown in the schematic, to provide bias for a tube. The power supply current flowing through the resistors R1 and R2 sets up a voltage drop having the polarity shown. The value of R2 is adjusted to provide the required bias for the tube.



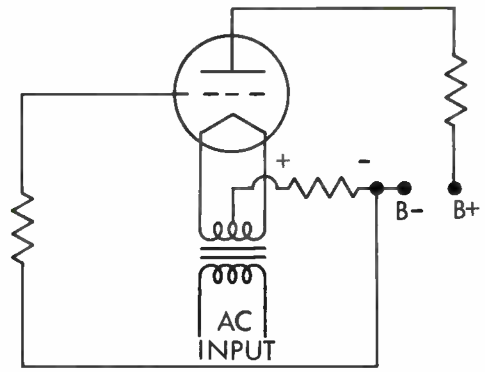
(d) Cathode-resistor bias is perhaps the most widely used type of bias in receiving-type equipment. The current flowing from B- through the resistor to the cathode of the tube develops a voltage drop across the resistor which places the cathode positive with respect to ground. The grid is returned directly to ground through a grid resistor, and since in a Class A stage there is no current flow, there will be no voltage drop across this resistor. This means that as far as dc is concerned the grid is at ground potential and the cathode is positive with respect to ground; in other words the grid is negative with respect to the cathode.



78. Is grid-leak biasing practical in audio-amplifier stages?

No: grid-leak biasing is not practical in audio-amplifier stages. Most audio-amplifier stages are Class A amplifiers. A Class A amplifier operates on the linear portion of the characteristic curve of the tube and does not draw grid current. Therefore there will be no bias developed in the grid circuit by means of a grid resistor.

79. Draw a diagram showing a method of obtaining grid bias for a filament-type vacuum tube by use of resistance in the plate circuit of the tube.



In the circuit shown, the power supply returns to ground through a resistance which develops a voltage drop across the resistance having the polarity shown. The grid of the tube is brought through the grid resistance back to this point so that the grid will be negative with respect to ground. The filament of the tube is grounded either through a center tap on the filament winding on a power transformer (as shown in the diagram) or else by means of a small center-tapped resistor.

80. Explain how you would determine the approximate value of cathode-bias resistance necessary to provide correct grid bias for any particular amplifier.

The required grid bias for a tube can be obtained from a tube manual. The manual will give the required bias for different values of plate voltage. The tube manual will also list the normal dc current flow through the tube under these circumstances.

Once you know the current that will flow through the tube and the bias voltage required, you can determine the size cathode resistor required to develop this bias simply by using Ohm's Law. If, for example, the bias required on the tube is 4 volts and the normal current is 20 milliamperes, we can find the bias as follows:

Using Ohm's Law $R = \frac{E}{I}$

and substituting 4 volts for E and .02 amps for I we have:

Grid bias Ω = Bias voltage / dc current through tube normally

$$R = \frac{4}{.02}$$

$$= 200 \text{ ohms}$$

After you have found the size of the resistor required, you need to determine the wattage rating that the resistor must have in order to safely dissipate power that must flow through it. Since you already know the voltage and current, you can find the wattage rating from the formula:

$$P = E \times I$$

$$P = 4 \times .02 = .08 \text{ watts}$$

Since 1/2-watt carbon resistors are readily available, a 200-ohm, 1/2-watt carbon resistor would make a suitable cathode-bias resistor.

Size of grid resistor = Bias voltage X normal current

National Radio Institute

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T-3:FCC

NRI Study Guide for FCC License Examination

PART IV

This part of your NRI FCC study guide deals primarily with oscillators and radio-frequency amplifiers. Study this part of the guide as soon as you receive it, and then answer the questions to the FCC-type examination TC-4 which you received along with the guide. Be sure to save this part of the study guide along with the parts you received previously so they will be available for review before taking your FCC license examination.

If you find that some of the questions covered in this guide are not clear to you, be sure to go over the explanation several times and also review your lesson texts. If you still have difficulties, make use of the NRI Consultation Service to write in requesting additional help. It is far better to take the time now to write for help than it is to wait. Small problems that you might be having now might cause you difficulty in later lessons. If you get these problems cleared up when they come up, you'll find that in most cases you will have very little difficulty with the later lessons.

81. What is an RFC? Why are they used?

    An RFC is a radio-frequency choke. A radio-frequency choke is a coil that has a high inductive reactance at the radio frequency in use in the circuit involved, but at the same time has a low dc resistance. Radio-frequency chokes are usually used to keep radio-frequency signals out of some circuits, usually power supply circuits, and at the same time allow dc or low-frequency ac signals to be applied through the choke to the circuit from which the choke is providing isolation.

    In some applications radio-frequency chokes are used as a radio-frequency load. The choke provides a high-impedance load for the radio-frequency signal, and at the same time has a low-resistance dc. A radio-frequency choke might be used in the plate circuit of a radio-frequency power amplifier. The choke acts as a high impedance and therefore the radio-frequency signal from the plate of the tube can be fed on to other circuits, usually through a capacitor. At the same time, the radio-frequency choke has a low dc resistance, which provides a convenient method of applying the plate voltage to the tube.

82. What are the advantages of using a resistor in series with the cathode of a class C radio-frequency amplifier tube to provide bias?

    Some class C amplifiers develop their entire operating bias by means of a grid resistor. When the grid of the tube is driven positive, the grid will draw grid current. A capacitor used in the grid circuit is charged by this current and then the capacitor discharges through the grid resistance. The resistance is selected so that the current flowing through it provides the correct operating bias for the tube. The disadvantage of this system is that if the tube should lose drive, in other words if a preceding stage fails, then there will be no rf signal applied to the grid of the tube and hence the grid will not be driven positive during the peak of the positive-driving signal. This means the tube will not draw any grid current, and hence there will be no grid bias developed by the tube. Without the grid bias the plate current may rise to a very high value and the tube may soon be destroyed.

    If part of the bias required for the tube is developed by means of a resistor in the cathode circuit of the tube, and if the grid drive should fail, the plate current flowing through the cathode resistor will still develop some bias across this resistor. As a matter of fact, if the plate current increases, the bias developed across the cathode resistor will also increase, and this more or less regulates the total plate current and should prevent the tube from drawing such a high plate current that it will destroy itself.

83. What is the difference between rf voltage amplifiers and rf power amplifiers in regards to applied bias? What type of tube is generally applied in rf voltage amplifiers?

    RF voltage amplifiers are used primarily in radio receivers. They are designed to take the small rf signal voltages picked up by the antenna and amplify them to a usable value. RF voltage amplifiers are class A amplifiers; therefore they are biased mid-way between zero grid voltage and cut-off voltage. They make use of small receiving-type tubes.

    RF power amplifiers are found primarily in transmitters. They are usually operated as class B or class C amplifiers. This means that the bias will be approximately cut-off in the case of a class B amplifier, and two to four times cut-off in the case of a class C amplifier. RF power amplifiers are used to develop power and usually large power-amplifier tubes are used in these applications. Of course, the size of the tube will depend upon the amount of power the stage must develop.

84. Explain the principle involved in neutralizing an rf stage.

In some rf stages, energy is fed from the plate of the tube back to the grid of the tube through the internal capacitance of the tube. When the energy fed from the plate back to the grid of the tube is in phase with the grid signal, it will reinforce the grid signal, and if the feedback is strong enough it will cause the stage to go into self-oscillation.

To prevent this type of oscillation we neutralize the stage. The idea behind neutralization is to feed a second signal back into the grid circuit, which is 180° out-of-phase with the signal fed to the grid through the tube. If the signal fed back into the grid circuit deliberately is equal to or slightly greater than the signal fed back to the grid through the tube, it will cancel the signal and prevent oscillation. If the signal is slightly larger than the signal fed back through the tube, a small amount of degeneration will be introduced, but this will add to the stability of the stage.

85. Draw a schematic diagram of a grounded-grid rf amplifier and explain its operation.

In the schematic diagram below, the rf input signal is coupled from the preceding stage to the cathode through C1. When the rf signal drives the cathode in a negative direction as shown during the first half-cycle, current flows from the cathode of the tube to the plate. The grid will be positive with respect to the cathode and therefore some current will flow from the cathode to the grid of the tube. This current will charge the grid capacitor C2 with the polarity shown. The majority of the current flows from the cathode through the tube to the plate of the tube. In the plate circuit we have a resonant circuit made up of C3 and L2. Other types of circuits, such as a pi network, could also be used in the plate circuit as well as shunt fed through a radio-frequency choke coil.

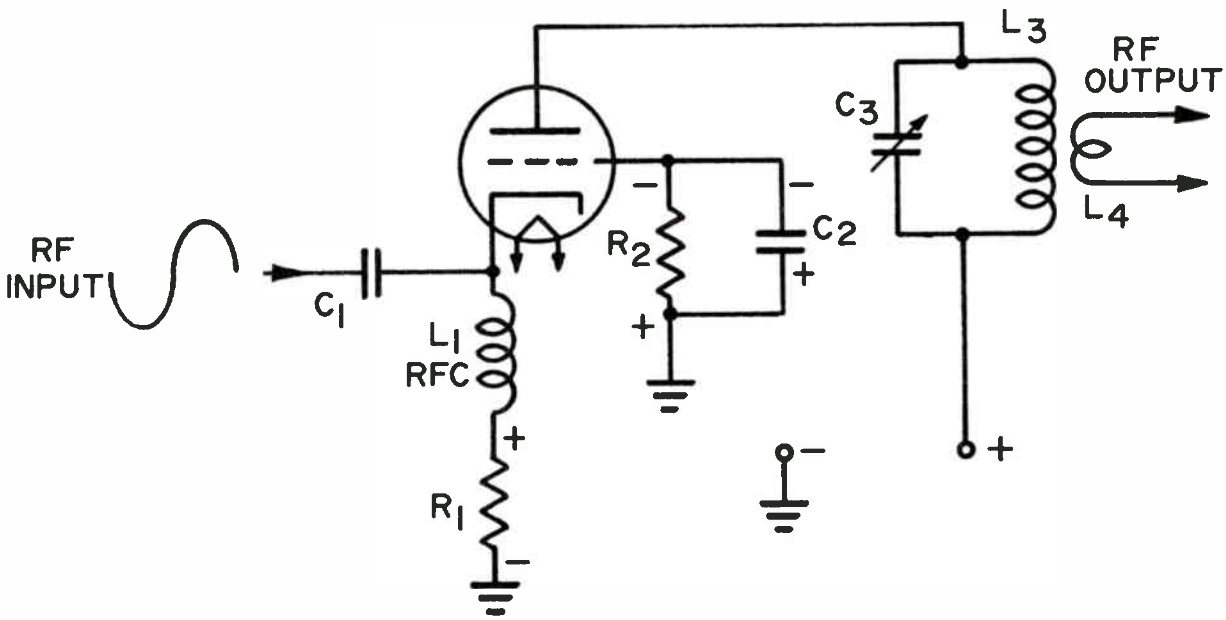
When the rf signal drives the cathode in a positive direction, the cathode will be positive with respect to the grid. The electrons that have charged C2 will flow through R2, developing a bias voltage across this resistor with the polarity shown. This bias voltage will be high enough to keep the tube cut off so that no plate current will flow through the tube.

Some bias is developed across R1 when the tube is conducting, because the cathode current flows through this resistor. In some circuits, the resistor may be omitted, but using a resistor of this type in this circuit provides some protection for the tube against a loss of drive.

The grounded-grid amplifier normally does not need neutralization. This is because the grid acts as a shield between the plate and cathode. The grid is held at rf ground potential by the grid capacitor, C2, and effectively isolates the plate from the cathode of the tube so that energy cannot be fed from the plate back to the cathode to cause oscillation.

86. State some indications of and methods of testing for the presence of parasitic oscillations in a transmitter.

Parasitic oscillations are oscillations that occur at a frequency other than the operating frequency,



or as a harmonic of the operating frequency. In some cases you will run into a low-frequency parasitic oscillation due to the resonant circuits set up by rf chokes and capacitances in the circuit. In other cases you will run into vhf parasitic oscillations which are caused by resonant circuits formed by the leads used to connect the tank circuit components to the vacuum tube.

In most cases these oscillations are self-sustaining; in other words they will occur even though there is no drive applied to the tube. They make it difficult or impossible to neutralize the stage. You can detect parasitic oscillation with a grid-dip meter. With the grid-dip meter you look for an indication that rf power is being developed at a frequency considerably removed from the operating frequency. You must look for both low-frequency and vhf parasitics. Parasitic oscillations can also be detected with a receiver providing it covers a wide enough frequency range to enable you to tune over the frequency range where the parasitic oscillations might be occurring.

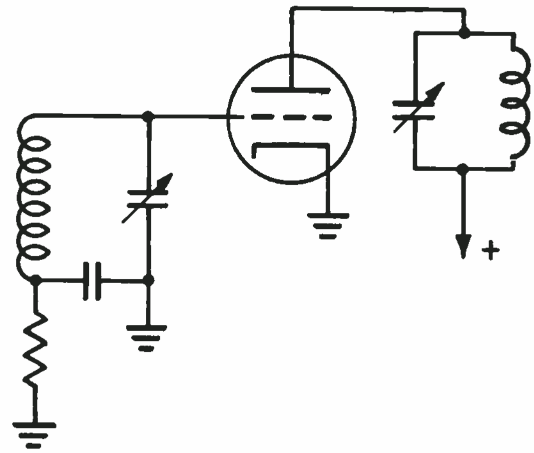
87. Explain, step-by-step, at least one procedure for neutralizing an rf amplifier stage.

RF amplifier stages are neutralized with the B+ removed from the plate of the tube, but with bias and drive applied to the tube. If the stage has a grid current meter, you can check the neutralization by tuning the plate circuit through resonance. If the stage is incorrectly neutralized, this will cause a drop in grid current. The neutralizing capacitor should then be adjusted until the grid-current meter remains constant as the plate circuit is tuned through resonance.

Another method of neutralizing a stage, also with the B+ supply removed from the plate circuit, is to place a small neon bulb close to the plate tank coil. With the excitation applied to the grid, if there is an appreciable amount of power getting through the tube into the tank circuit the neon bulb will light. The neutralizing capacitor should be adjusted until the bulb no longer lights.

A more sensitive indicator can be made by means of a diode, a pick-up coil, and a sensitive meter. The pick-up coil is placed near the plate-tank circuit. The B+ is removed from the tube as before and excitation applied to the grid. The neutralizing capacitor is then adjusted for minimum or no reading on the meter. This will indicate that the stage is neutralized inasmuch as little or no power is getting through the tube to the plate-tank circuit.

88. Draw a schematic diagram and explain the operation of a harmonic-generator stage.

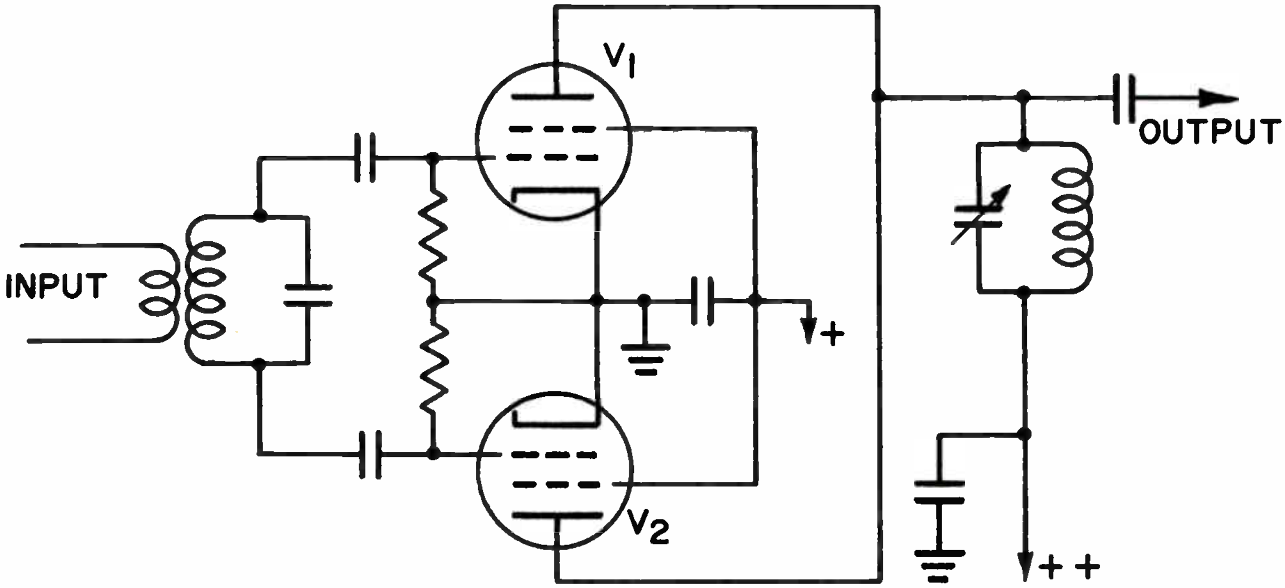


The circuit above looks very much like a class C power amplifier. However, the bias on the harmonic generator is much higher than in the case of the class C amplifier. This means that the plate current flows through the tube in much narrower or much sharper pulses. The resonant circuit in the tank circuit of the tube is adjusted to the harmonic desired. It is readily excited into oscillation by the sharp plate current pulse. By the use of a comparatively high Q tank circuit there is very little damping of the output wave between pulses from the tube.

89. Draw a schematic diagram of a "push-push" frequency multiplier and explain the principle of its operation.

In the circuit on page 4 note that the grids of the two tubes are driven from the opposite ends of the grid coil. In other words, during one half-cycle of the input signal, if the grid of V1 is driven positive and this tube conducts, the grid of V2 will be driven negative so that this tube will not conduct. A single pulse of energy will be supplied to the tank circuit in the output of the two tubes from V1. During the next half-cycle, the grid of V2 will be driven positive and this tube will conduct. Now this tube will supply a single pulse of energy to the tank circuit in the output. Notice that the plates of the two tubes are tied together and feed the tank circuit. The tank circuit is tuned to an even harmonic of the input signal. If it is tuned to a second harmonic, the tank circuit will get a pulse each cycle, one pulse from V1 during one cycle, and a pulse from V2 during the next cycle. If the tank circuit is tuned to the fourth harmonic, the tank circuit will get a pulse from V1 and go through the remainder of the cycle and the next cycle. Then it will receive a pulse from V2 and go through the remainder of the cycle and the next cycle.

Push-push frequency multipliers will produce only even harmonics of the input signal.



90. Push-pull frequency multipliers normally produce what order of harmonics: even or odd?

Push-pull frequency multipliers are normally used to produce odd-order harmonics such as the third, fifth, etc. You will remember that push-pull amplifiers will cancel out even-order harmonics generated within the stage. Therefore they are poor generators of even-order harmonics.

91. What class of amplifier is appropriate to use in a radio-frequency doubler stage?

A class C rf amplifier should be used in a radio-frequency doubler stage. A class C stage makes a particularly good doubler if the bias is somewhat higher than is usually used for normal class C operation. This will provide you with a high-amplitude, rather sharp pulse which will shock-excite the tank circuit in the output of the stage tuned to twice the input frequency.

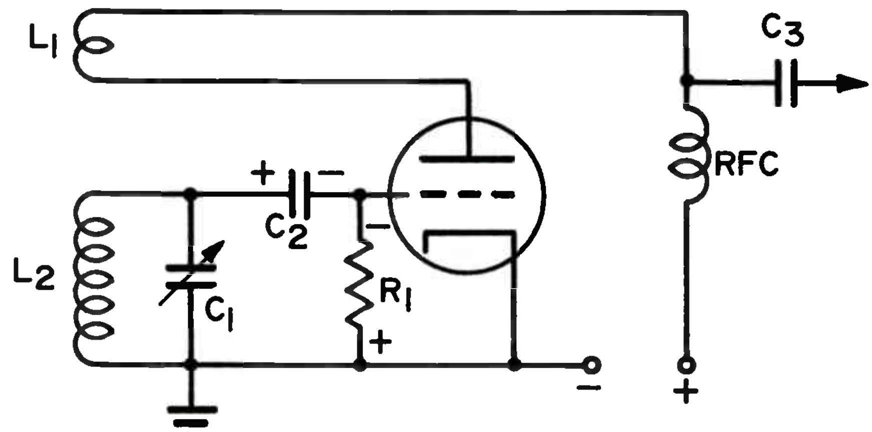
92. Draw circuit diagrams of each of the following types of oscillators (include any commonly associated components). Explain the principles of operation of each.

(a) Armstrong
(b) Tuned-plate tuned-grid, series-fed and shunt-fed (crystal and LC controlled)
(c) Hartley
(d) Colpitt's
(e) Electron-coupled
(f) Pierce (crystal-controlled)

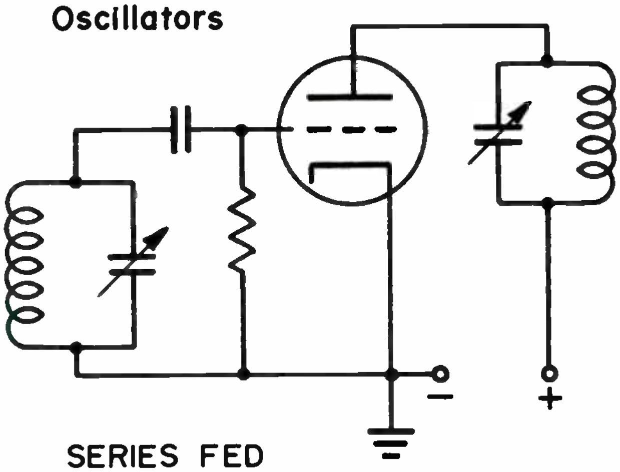
In the Armstrong oscillator, the output signal is fed through coil L1 which is inductively coupled to the coil L2. The coils are phased so that the energy coupled from L1 and L2 reinforces the signal in L2.

In operation, the signal across the resonant circuit consisting of L2 and C1 is applied between the grid and cathode of the tube. It is applied to the tube through the grid capacitor C2. When the input signal drives the grid positive, the electrons flow from the cathode to the grid of the tube and charge the grid capacitor with the polarity shown. This capacitor then discharges through

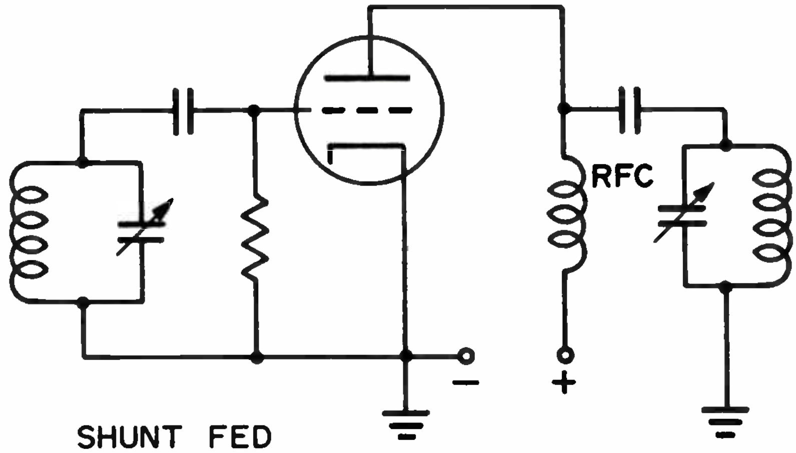
(a)
Armstrong
Oscillator



(b) LC-Controlled Oscillators



SERIES FED



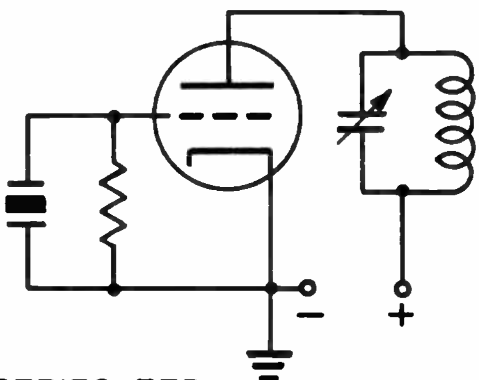
SHUNT FED

the grid resistor, R1, developing the bias shown for the tube. Current flows through the tube in a series of pulses, and these pulses are fed back from L1 to L2. The pulses are in phase with the voltage across L2 so that when a signal is inductively coupled from L1 into L2 it reinforces the signal in L2. Thus any losses in the tank circuit are supplied by pulses from the plate of the tube. The RFC in the plate circuit provides a load for the tube and the output can be fed through the capacitor C3 on to the next stage.

In the tuned plate-tuned grid oscillator, energy is fed through the plate to grid capacitance in the tube, from the plate circuit back into the grid circuit. The energy fed back into the grid circuit is in phase with the grid signal and reinforces it so that oscillation is sustained. Both shunt-fed and series-fed circuits are shown above.

The input tank circuit, consisting of L1 and C1 in both oscillators, can be replaced by a crystal. Then we have the circuits of the tuned plate-tuned grid crystal oscillators. The crystal in each case simply acts as a resonant circuit and energy is again fed back through the tube from the plate back into the grid circuit. This type is shown below.

(b) Crystal-Controlled Oscillators



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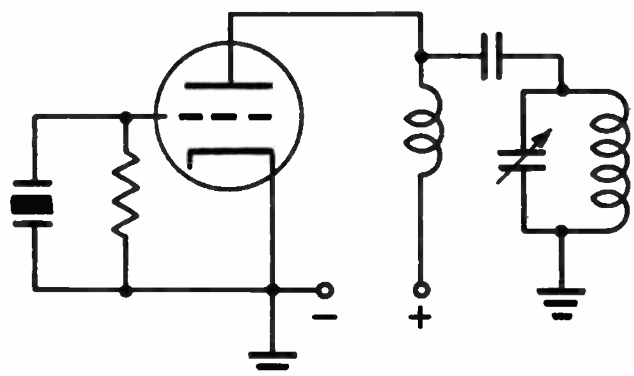
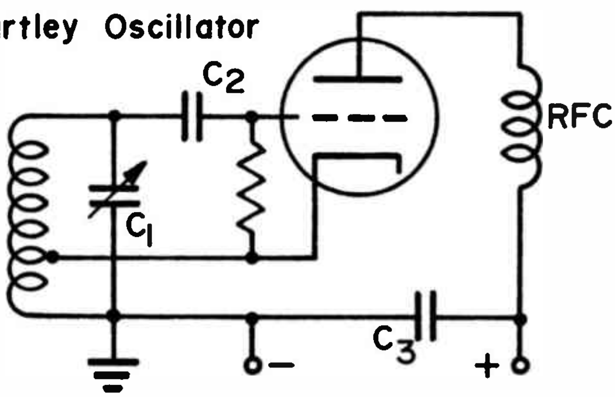


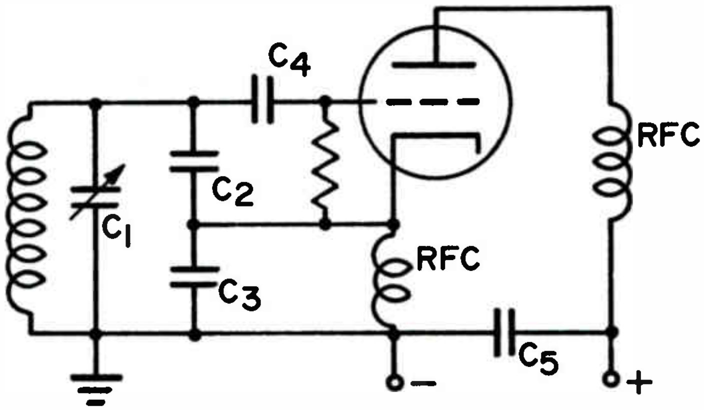
SHUNT FED

In the Hartley oscillator at the right, the energy is inductively fed from the plate circuit back into the tank coil. When the grid moves in a positive direction, this causes the plate current - and hence the cathode current - to increase. The increase in current flowing through the lower part of the coil sets up a magnetic field which cuts the entire coil and induces a voltage in it. This voltage has the right polarity to aid the grid voltage producing the increase in current. Hence energy is fed from the output circuit back into the tank circuit to sustain oscillation.

(c) Hartley Oscillator

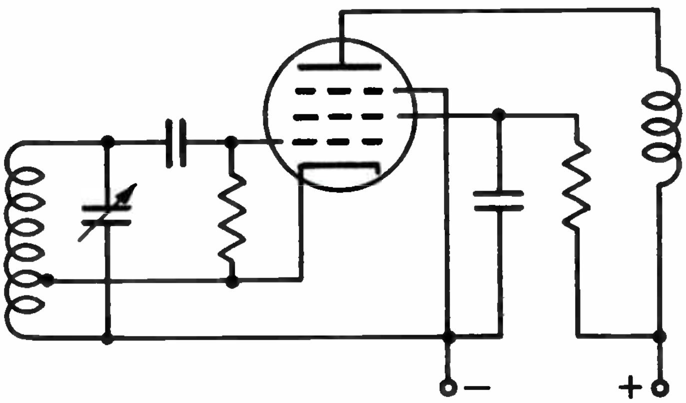


(d) Colpitt's Oscillator



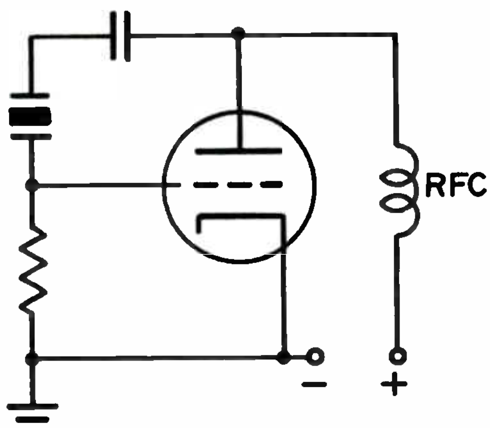
The Colpitt's oscillator is a circuit in which feedback is controlled by a capacity voltage divider made up of C2 and C3 in the circuit shown. The amount of energy fed back into the input circuit depends upon the reactance of the two capacitors. If the value of C2 is small compared to the value of C3 the reactance will be high and hence the feedback will be high. The Colpitt's oscillator is quite widely used.

(e) Electron-Coupled Oscillator



A schematic diagram of an electron-coupled oscillator is shown. Notice that the electron-coupled oscillator makes use of a pentode tube. In the circuit shown we have used the Hartley oscillator circuit, in which the screen of the tube acts as the plate. The operation of this part of the circuit is exactly the same as in any Hartley oscillator circuit. Energy is fed through the electron stream into the plate because most of the electrons travelling towards the screen pass right through it and flow to the plate of the tube. You can have other types of electron-coupled oscillators as well as the Hartley. The main feature of the electron-coupled oscillator is that coupling is by means of the electron stream.

(f) Pierce Oscillator



In the Pierce oscillator the crystal is placed between the plate and the grid of the tube. Feedback is from the plate through the capacitor and crystal back into the grid circuit. One of the characteristics of the Pierce oscillator is that its output is rich in harmonics.

All of these oscillators are described in detail in your lesson text. You should be able to draw the schematic diagram of each of these oscillators and explain how they work. If you are unable to do this now, stop and try drawing the diagrams from memory a number of times. If you get stuck, look at your textbook or at the study guide, refresh your memory, and then close the book and try to draw them from memory again. Also go over the explanation a number of times; you can be sure that detailed questions on one or more oscillators will be asked of you on your FCC examination.

93. What are the principal advantages of crystal control over tuned-circuit oscillators?

The principal advantage of crystal oscillators over tuned-circuit oscillators lies in their frequency stability. The crystal acts as a very high Q circuit element and it is capable of maintaining the oscillator frequency within very close tolerance, particularly if the temperature of the crystal is maintained constant.

94. Why should excessive feedback be avoided in a crystal oscillator?

Excessive feedback may cause a high crystal current which could crack the crystal.

95. Why is a separate source of plate power desirable for a crystal-oscillator stage in a radio transmitter?

The oscillator frequency may be affected by small changes in plate voltage. Therefore, if it is required to hold the oscillator frequency to a close tolerance, it is usually best to have a separate voltage source which is well filtered and regulated for the oscillator.

96. What might result if a high degree of coupling exists between the plate and grid circuits of a crystal-controlled oscillator?

A high degree of coupling between the plate and grid circuits of a crystal-controlled oscillator will result in excessive feedback, which may destroy the crystal.

97. Explain some methods of determining if oscillation is occurring in an oscillator circuit.

One method of checking for oscillation in an oscillator circuit is with a high-resistance dc voltmeter. You can measure the voltage across the grid resistor. If the stage is oscillating there will be grid current flowing, and this will result in there being voltage across the grid resistor. The polarity of this voltage will be such that the grid is negative.

Another method of detecting oscillation is by means of a small neon bulb. If you bring the bulb close to the oscillator-plate circuit it should light. In the case of a low-power oscillator it may be necessary to actually touch one of the bulb terminals to the plate circuit.

You can also often detect oscillation by tuning the plate circuit. As the plate circuit approaches resonance the plate current should drop. If you place too much capacitance in the plate circuit there will be a sudden increase in plate current, indicating that the oscillator has gone out of oscillation.

98. What determines the fundamental frequency of a quartz crystal?

The fundamental frequency of a quartz crystal is determined primarily by the thickness of the crystal. The type of cut also has some effect on the frequency.

99. What are the characteristics and possible uses of an "overtone" crystal? A "third mode" crystal?

Crystals to operate on very high frequencies are very thin and fragile. To overcome this problem overtone crystals are used. An overtone crystal is a crystal that is cut expecially to oscillate on the third, fifth or seventh overtone of the fundamental frequency of the crystal. The overtone is not an exact harmonic; for example the third overtone will be approximately three times the fundamental frequency of the crystal, but not exactly three times. Overtone crystals are used to generate a high frequency directly with a crystal oscillator.

A third mode crystal is a crystal designed to emphasize overtone.

100. Explain some of the factors involved in the stability of an oscillator (both crystal and LC-controlled).

In the case of a crystal oscillator, one of the major factors that affect the stability is the crystal cut. Some cuts are more susceptible to changes in temperature, and as a result the frequency may vary as the temperature changes. The most important thing in a crystal of this type is to keep the temperature constant.

In an LC-controlled oscillator the components in the frequency-controlling tank circuit will have an effect on the frequency. A high quality coil with low leakage and high Q will usually give the best frequency stability. However, changes in temperature will affect inductance and capacitance in the circuit and have an effect on the oscillator frequency unless some special temperature-compensating components are used in the circuit to counteract the temperature change.

Oscillators are often sensitive to changes in plate voltage. Therefore, for best stability the plate voltage, and the screen voltage in the case of a tetrode or pentode oscillator, should be regulated.

The loading on an oscillator may have an effect on the oscillator frequency. A heavily loaded oscillator will often change frequency particularly if the load is subject to variations.

101. What is meant by the temperature coefficient of a crystal?

The temperature coefficient of a crystal indicates how much the crystal will drift with a change in temperature. The temperature coefficient may be either positive or negative. A crystal that has a positive temperature coefficient will increase in frequency with increases in temperature; a crystal with a negative temperature coefficient will decrease in frequency with increases in temperature. The temperature coefficient is usually given as so much per degree centigrade. Sometimes it is given as a percentage and it also is sometimes given as so many parts per thousand kc.

102. Is it necessary or desirable that the surfaces of a quartz crystal be clean? If so, what cleaning agents can be used which will not adversely affect the operation of the crystal?

The surfaces of a quartz crystal must be kept clean or the crystal will not oscillate. Probably the best cleaning agent is plain soap and water. The crystal can be dried in a soft cleansing tissue. If the crystal has become excessively oily, it can be cleaned with carbon tetrachloride. If you use carbon tetrachloride avoid inhaling the fumes.

103. What is the purpose of a buffer amplifier in a transmitter?

A buffer amplifier is usually found between the oscillator stage and the first power amplifier. The buffer amplifier is operated as a class A amplifier. It provides a very light load on the oscillator stage and hence helps improve the oscillator frequency stability. It sometimes is called simply a buffer because it serves as a buffer or isolates the oscillator stage from the first power amplifier stage.

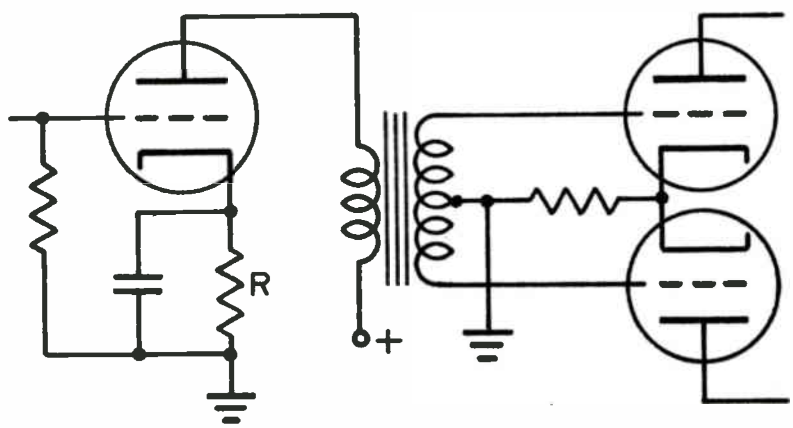
104. Why does a class B audio-frequency amplifier stage require considerably more driving power than a class A amplifier?

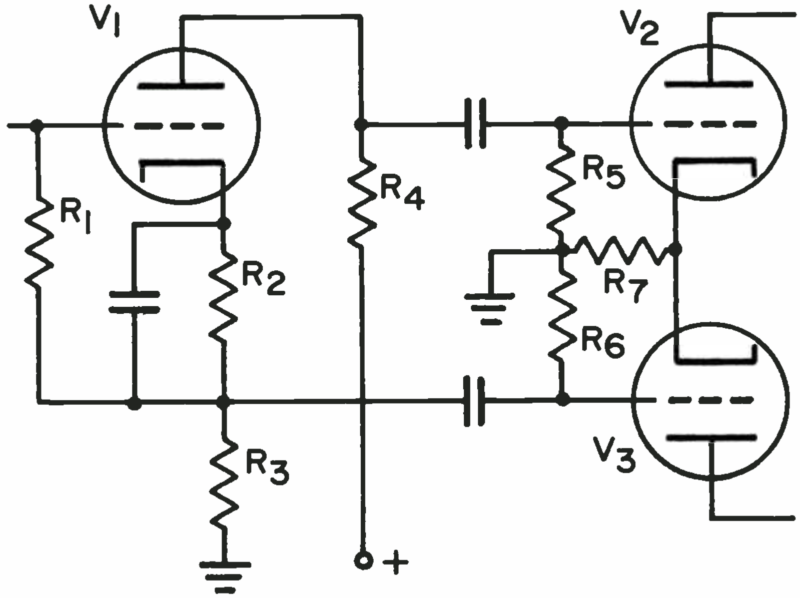
Actually, the grid of a class A power amplifier is never driven positive. As a result, there is no grid-current flow and so the net product of voltage times current in the grid circuit is zero. This means that the stage does not actually require driving power. You drive a class A amplifier simply by applying a varying signal voltage.

In a class B power amplifier, the grid is biased at cut-off. The driving signal has sufficient amplitude to drive the grid positive. When this happens, grid current flows and there is actually power consumed in the grid circuit. Thus a class B power amplifier requires driving power, whereas the class A power amplifier does not actually require any driving power.

105. Show by use of circuit diagrams two ways of using single-ended stages to drive a push-pull output stage.

The simplest method is shown in the diagram below and involves the use of an audio transformer between the single-ended stage and the push-pull output stage. The secondary winding of the audio transformer is center-tapped; the center tap is grounded; two voltages 180° out-of-phase with respect to each other are developed for application to the grids of the push-pull output stages.





The second method is by means of a stage called a phase splitter. Resistor R3 in the cathode circuit of V1 is equal to R4. The voltage developed across R3 is equal to but 180° out-of-phase with the voltage developed across R4. The voltage developed across R4 is fed to V2 and the voltage developed across R3 is fed to V3; thus the push-pull output stage is driven by equal voltages 180° out-of-phase. (Refer to the circuit diagram shown above.)

106. Name some causes of hum and self-oscillation in audio amplifiers and the methods of reducing them.

Hum may be due to inadequate filtering in the power supply. The obvious remedy in this situation is to increase the size of the filter capacitors. Also, a defective filter capacitor may lose capacity resulting in excessive hum. In this case simply replacing the capacitor should clear up the trouble.

Hum may also be due to cathode-to-heater leakage in a tube. This type of hum can be identified as 60-cycle hum and it can be eliminated by replacing the defective tube. In some amplifiers, particularly very high gain amplifiers, even a very small amount of leakage between the cathode and the heater of the tube can cause hum. This leakage may be so small it cannot be picked up by a tube tester.

Hum may also be caused by stray pick-up due to hum being fed from one wire into another. This type of hum is frequently encountered after major repair work where it has been necessary to move the wires in the equipment around considerably. Restoring the wires to their original position will usually correct this type of hum.

Self-oscillation is due to an in-phase signal being fed from the output of the stage back into the input of that stage or a preceding stage. Self-oscillation may be due to feedback loops established because the wiring routes the signal around the amplifier in such a way that it can feed from the plate circuit of one stage back to the grid circuit of the preceding stage. Self-oscillation may occur in very high gain stages, particularly when the plate circuit is inductively loaded. Tetrodes and pentodes are less susceptible to oscillation of this type because of the lower grid-to-plate capacitance.

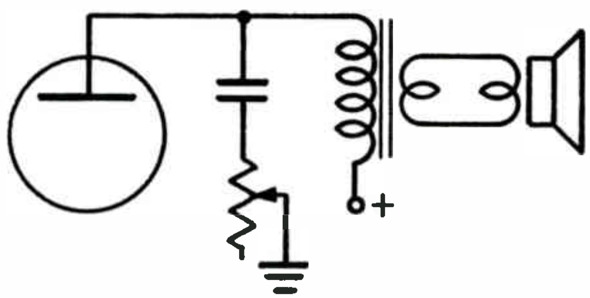
107. What factors should be taken into consideration when ordering a class A audio-output transformer? A class B audio-output transformer feeding a speaker of known value?

When ordering the audio-output transformer, the first thing to be taken into consideration is the power that the transformer must handle. If the transformer is underrated and not capable of handling the power that the output stage can put out, you are sure to run into transformer saturation which will cause considerable distortion.

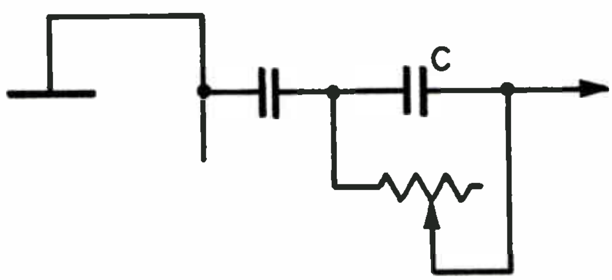
In the case of a single class A power output stage you need to get a transformer with the correct turns ratio so that the speaker impedance will be matched to the plate impedance of the tube. This way the tube will be working into the correct load impedance.

In selecting a transformer for class B output stages you must remember that the load impedance that the individual tube will see will be one quarter of the plate-to-plate impedance of the primary winding. In other words, if the tube is to work into a 2000-ohm load, then you want a transformer with a turns ratio that will match the speaker voice coil impedance to an 8000-ohm load. This way, each tube will see a load impedance of 2000 ohms.

108. Draw circuit diagrams and explain the operation of two commonly used tone-control circuits.



The most commonly used tone-control circuit is shown above. In this circuit a capacitor is used between the plate of the tube along with a potentiometer. The capacitor is selected to provide a low reactance to the high-frequency signals and a comparatively high reactance to low-frequency signals. As the resistance of the potentiometer is taken out of the circuit the higher frequency signals are bypassed to ground. This reduces the highs and gives an apparent increase in the base response.



The second circuit above is a treble circuit. Here the capacitor C is selected to provide a low reactance path to high-frequency signals and a high reactance to low-frequency signals. When the treble control is adjusted so that all the resistance is out of the circuit, the capacitor is bypassed and both high-frequency and low-frequency signals are passed. As the resistance in the circuit is increased the high-frequency signals are passed by the capacitor and the low-frequency signals attenuated by the tone control. This gives an apparent increase in the high-frequency gain of the amplifier.

109. Draw two cycles of a sine-wave on a graph of amplitude versus time. Assume a frequency of 5 mcs.

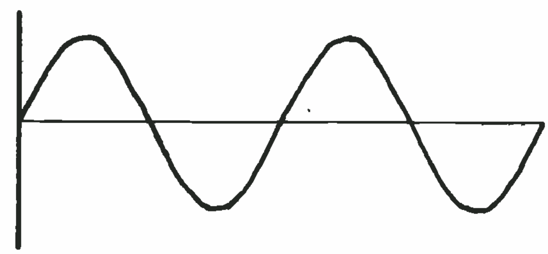
(a) What will be the wavelength of one cycle in meters? In centimeters?

(b) How many degrees does one cycle represent?

(c) How much time would it take for a wave to rotate 45°? 90°? 280°?

(d) If there were a second harmonic of this frequency, how many cycles would be represented on this graph?

(e) On the same graph draw two cycles of another sine wave leading the first by 90°.



(a) The wavelength of one cycle would be 60 meters. This is equal to 6000 centimeters. To find the wavelength in meters you use the formula:

$$\lambda = \frac{300}{f\ (\text{mcs})}$$

$$= \frac{300}{5} = 60 \text{ meters}$$

(b) One cycle represents 360°. When we use a vector or a phaser to represent a sine wave, in tracing out one cycle, the vector goes through one complete revolution or 360°.

(c) The period of a five-megacycle wave is

$$\frac{1}{5{,}000{,}000} = .2 \text{ microseconds}$$

This is the time it takes for the wave to go through one complete cycle. Therefore it would take one-eighth of this value to go through 45°, which is equal to .025 microseconds. To rotate 90° it would take one-quarter of the period, which is equal to

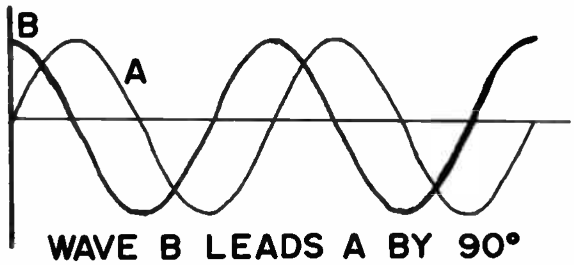
$$\frac{.2}{4} = .05 \text{ microseconds}$$

To rotate 280° we can determine the time by multiplying

$$\frac{280}{360} \times .2 = .156$$

(d) If there are second harmonics on this frequency four cycles could be represented on this graph.

(e)



WAVE B LEADS A BY 90°

110. What would be the velocity of the wave in the preceding question or any other electromagnetic wave in free space?

The velocity of the electromagnetic wave will be 186,000 miles per second or 300,000 kilometers per second.

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NRI Study Guide For FCC License Examination

PART V

The material in this part of the Study Guide supplements the material you have been studying in your lesson text. Be sure you study this part of the Study Guide carefully because it contains information on the type of question you will probably be required to answer when you go for your radiotelephone operator's license.

If you find that you need additional study on any of the material covered in this part of the Study Guide, be sure you go back to your lesson text and do whatever extra studying is necessary. Keeping up with the material in your course as you go along is the best way of making sure that you will be able to pass your examination and get your license.

Most of the questions in this section deal with modulation and measurements. There are also a number of problems involving calculations which you should be able to do without any difficulty if you have completed the mathematics reference text in your course.

Be sure to save this part of the Study Guide along with those you have received previously so you will be able to make a complete review before going for your license.

111. The output of an amplifier stage having a voltage gain of 30 db is 25 volts. What is the input voltage level?

To solve this problem we start with the formula used to give us the ratio of two voltages in decibels. The formula is:

$$db = 20 \log \frac{e_2}{e_1}$$

In this case we know that the voltage gain is 30 db so we can rewrite our formula, substituting this information. We will then have

$$30 = 20 \log \frac{e_2}{e_1}$$

Transposing the 20 from the right side of the equation to the left, or simply dividing both sides of the equation by 20, we get

$$\frac{30}{20} = \frac{3}{2} = 1.5 = \log \frac{e_2}{e_1}$$

This means that the value of the logarithm of $\frac{e_2}{e_1}$ is 1.5. We know that a logarithm is made up of two parts, the characteristic and mantissa. The characteristic tells us simply where to place the decimal point and the mantissa gives us the original number. Looking up .5 in the log table, we find that it is equal to 316. If we remember that there is one more position to the left of the decimal point than the value of the characteristic, this means that the number whose log is 1.5 is 31.6. In other words, log 31.6 equals 1.5.

Now we know that the value of

$$\frac{e_2}{e_1} = 31.6$$

Therefore $$e_1 = \frac{25}{31.6}$$

$$= .79 \text{ volts}$$

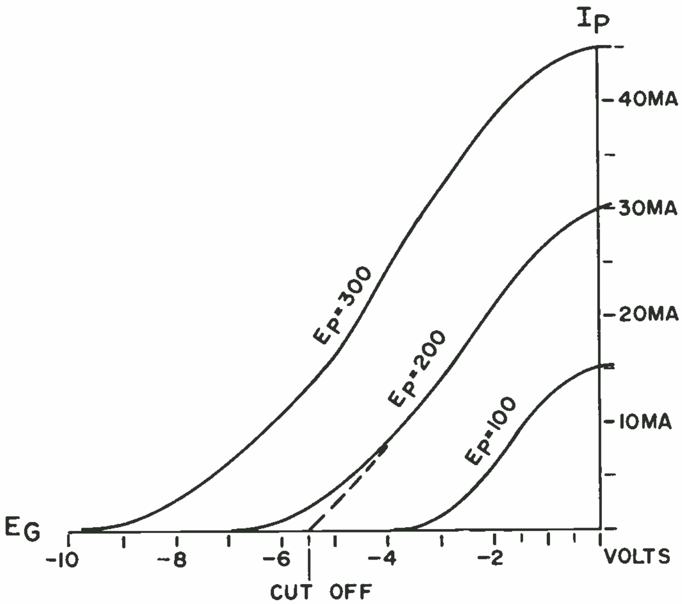
This means that the input voltage must be .79 volts.

This is a comparatively important problem because you will probably have a problem like this on your examination. Therefore, to get some extra practice, try working out the following problem: The gain of an amplifier is 27.6 db. If the output voltage is 48 volts, what is the input voltage?

Answer: 2 volts.

112. Draw a rough graph of plate-current versus grid-voltage (I_p vs E_g) for various plate voltages on a typical triode-vacuum tube.

(a) How would output current vary with input voltage in class A operation? Class B operation? Class AB operation? Class C operation?
(b) Does the amplitude of the input signal determine the class of operation?
(c) What is meant by "current cut-off" bias voltage?
(d) What is meant by plate-current "saturation"?
(e) What is the relationship between distortion in the output current waveform and
 (1) the class of operation?
 (2) the portion of the transfer characteristic over which the signal is operating?
 (3) the amplitude of the input signal?
(f) What occurs in the grid-circuit when the grid is "driven" positive? Would this have any effect on biasing?
(g) In what way is the output current related to the output voltage?



(a) In class A amplifier operation, the plate current will vary directly with the input voltage. As the input signal voltage increases in a positive direction the plate current will increase, and as the grid voltage swings in a negative direction the plate current will decrease. The output plate current will be a replica of the input voltage; in class A operation the input signal never drives the plate current to cut-off, nor does it drive the grid of the tube positive.

In class B operation the tube is biased to cut-off. The tube conducts only on the positive half-cycle. Thus the output current would be essentially a replica of the positive half of the input signal. There will be a series of pulses produced when the input signal swings positive, followed by no current flow through the tube during the negative half-cycle.

In class AB operation the tube is biased somewhere between class A operation and class B operation. In class A operation current flows through the entire 360° of the input cycle, but in class B operation it flows only during 180° (or the positive half) of the input cycle. In class AB operation it flows for more than 180°, but less than 360°. The output current would resemble the positive half of the input signal, but the negative half would be flattened out and drop to zero during the part of the negative half-cycle.

In class C operation the tube is biased several times cut-off bias; therefore, the plate current will flow in a series of sharp pulses during the period when the input signal is able to overcome the bias and cause the plate current to flow. The plate current pulses would exist for considerably less than 180° of the input cycle.

(b) The amplitude of the input signal does not determine the class of operation. The class of operation is determined by the bias applied to the tube.

(c) "Current cut-off" bias voltage refers to the grid bias required to cut-off the flow of plate current. You'll notice that the lower end of the three-plate current curves are not linear. Therefore it is difficult to say exactly at what voltage the true current will drop to zero. As a matter of fact, with tubes of the same type, it will vary somewhat. The exact cut-off voltage is the voltage at which the current will stop. However, since the lower end of the curve is not linear we often use a ruler to extend the linear or straight portion of the curve down to where it cuts the grid voltage axis and consider this as the cut-off bias. Actually, with this bias there will be some plate current flowing. On the curve representing a plate voltage of 200 volts we have extended the curve by means of a series of dashed lines; you can see that the cut-off bias is about 5-1/2 volts. Actually, there will be some small current still flowing in the tube with this bias applied. You can see from the curve that the bias will have to be almost 7 volts before the plate current is completely cut-off.

(d) You'll notice that the plate-current curves shown are straight diagonal lines over a reasonable portion of the curve. The top of the curve flattens out. This indicates plate-current "saturation". In other words, all of the electrons that will flow to the plate with the applied voltage are reaching the plate of the tube and increasing the bias further in a positive direction yet they will not result in any further increase in plate current. We say we've reached plate-current saturation -- the plate current is as high as it can be with the applied plate voltage.

(e) (1) The amount of distortion produced in a stage depends upon the type of operation. You can get less distortion with a class A amplifier than with any of the other types. Class C operation has the highest distortion.

(2) The portion of the curve over which the signal is operating has an effect on the

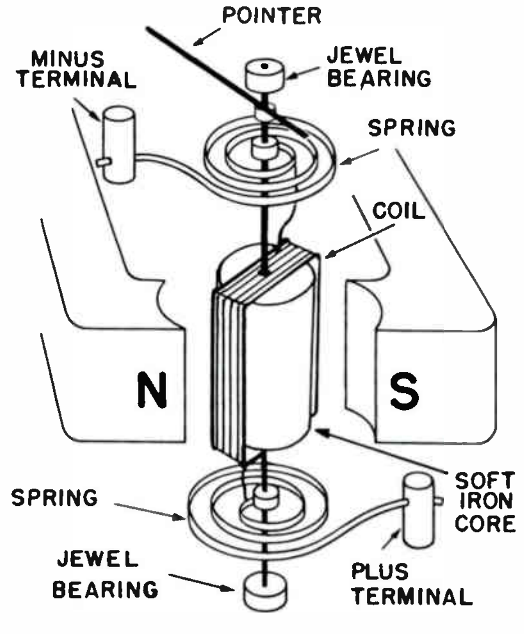
amount of distortion. In class A operation the tube should be operating over the center linear portion of the curve. If we try to drive the stage too hard and the input voltage extends the plate current on to either end of the curve where the curve is bent then the distortion will increase. Since we are operating over the curved portion of the curve in class B and class C operation, there will be more distortion than with class A operation.

(3) The amplitude of the input signal has an effect on the distortion. A small or low-level signal driving a class A stage will drive the tube over only a relatively small portion of the curve, thus making it possible to keep the stage operating in the linear portion of the curve. As the voltage increases there is more likelihood of driving the tube on to non-linear portions of the curve. In a class B amplifier we drive the tube hard enough to draw grid current. This usually causes the tube to operate not only over the lower bend in the curve but also up into the plate current saturation region. This could cause distortion. The same is true of class C operation.

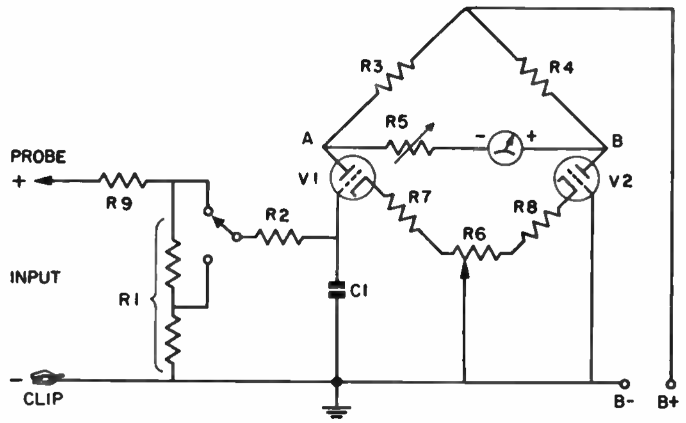
(f) When the grid of the tube is driven positive, grid current flows. In an amplifier stage where there is resistance in the grid circuit, this will develop voltage across the grid resistor and will have an effect on the bias in the stage. As a matter of fact, in a class C amplifier where the grid is intentionally driven positive we can make use of this grid current to develop all of the bias required by the stage.

(g) The output current is directly related to the output voltage. If the output voltage doubles, the output current will double. Notice that for given values for bias the current with 200 volts on the tube is approximately double the current with 100 volts on the plate of the tube. The relationship may not hold exactly, but since the load impedance in the stage is essentially constant, if the output signal voltage doubles, then the output signal current will double.

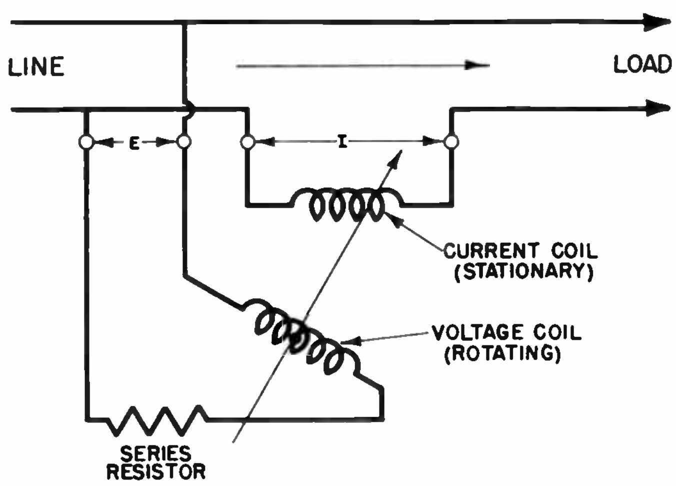
113. Make a sketch showing the construction of the D'Arsonval-type meter and label the various parts. Draw a circuit diagram of a vacuum-tube voltmeter and a wattmeter.



D'Arsonval Meter

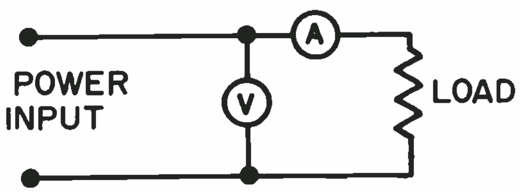


Vacuum-tube Voltmeter



Wattmeter

114. Show by a diagram how a voltmeter and an ammeter should be connected to measure power in a dc circuit.



115. A 0-1 dc milliammeter is to be converted into a voltmeter with a full-scale calibration of 100 volts. What value resistance should be connected in series with the milliammeter?

There are two ways of tackling a problem of this type. They are in reality essentially the same, but sometimes you can remember one system more easily than the other. The first method is to take the full-scale voltage (100 volts) and the maximum current which is to flow through the meter (1 milliampere). Now you use these two values in Ohm's Law to find out what the total resistance in the circuit must be. Substituting these values in Ohm's Law we have

$$R = \frac{100}{.001}$$

We can remove the decimal in the divisor by multiplying both the numerator and divisor of the fraction by 1000. We then have

$$R = \frac{100 \times 1000}{.001 \times 1000}$$

$$= \frac{100{,}000}{1}$$

The total resistance in the circuit must therefore be 100,000 ohms. This must consist of a resistor in series with the meter, plus the resistance of the meter. In this case, since the resistance of the meter is not given you assume that it is so low it doesn't matter; therefore, the resistor used would be a 100,000-ohm resistor.

The other method that you can use is to find what we call the ohms-per-volt rating of the meter. You can do this by dividing the meter current into 1 volt. Thus we have

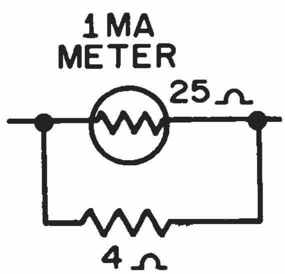
$$R = \frac{1}{.001}$$

$$R = 1000 \text{ ohms}$$

Therefore, the ohms-per-volt rating of the meter is 1000 ohms. This means that if you want the meter to read 1 volt full scale, the total resistance must be 1000 ohms; if you want it to read 2 volts full scale, then the resistance must be two times 1000, or 2000 ohms. If you want it to read 100 volts, then the total resistance must be 100 times 1000, or 100,000 ohms.

116. A 1-milliampere meter having a resistance of 25 ohms was used to measure an unknown current by shunting the meter with a 4-ohm resistor. It then read 0.4 milliamperes. What was the unknown current value?

With the shunt connected across the meter we have the circuit shown below. We have a resistance of 25 ohms and in parallel with it we have a resistance of 4 ohms. The 25-ohm resistance is the resistance of the meter.



You could work this problem out in proportions. The higher current will flow through the lower resistance. We know that the current flowing through the 25-ohm resistance is .4 milliampere; therefore, we know that the current flowing through the 4-ohm resistor will be higher, but it will bear the same ratio as 25 has to 4. We then set up our ratio in proportion as

$$\frac{25}{4} = \frac{I}{.4}$$

and when we cross multiply we get

$$4I = 10$$

$$I = 2.5 \text{ ma}$$

This means that the current flowing through the shunt is 2.5 milliamperes; thus the total current flowing in the circuit is 2.5 + .4 = 2.9 milliamperes.

Another method that can be used to solve this type of problem is to find the voltage across the known resistor. In this case, the known resistor is the 25-ohm meter resistance; you know the current flowing through it so you can get the voltage from the formula E = I × R.

$$E = .4 \times 25 = 10 \text{ millivolts}$$

Once you have the value of E you can get the current flowing through the 4-ohm resistor; then by adding these two currents you can determine the total current.

$$I = \frac{10}{4} = 2.5 \text{ ma}$$

$$I = 2.5 + .4$$

$$= 2.9 \text{ ma}$$

117. An rf vtvm is available to locate the resonance of a tunable primary tank circuit of an rf transformer. If a vtvm is measuring the voltage across the tuned secondary how would resonance of the primary be indicated?

When the primary is tuned to resonance there will be maximum circulating current in the primary coil. Since this coil is inductively coupled to the secondary, there will be maximum voltage induced in the secondary. Therefore, resonance of the primary would be indicated by a maximum reading on the vtvm. As the primary circuit is adjusted to resonance the voltage reading would increase until it reaches a peak; this is the point where resonance has been reached.

118. Define the following terms and describe a practical situation in which they might be used.

(a) RMS voltage

(b) Peak current

(c) Average current

(d) Power

(e) Energy

(a) RMS voltage means "root-mean-square" voltage. It is the effective value of an ac voltage and it is equal to .707 times the peak value of the ac sine wave. The rms voltage is the value we measure when we measure an ac voltage such as the power line voltage. If we state that the power line voltage is 120 volts we mean that the rms value (or the effective value) of the power line voltage is 120 volts.

(b) Peak current may be used in several different ways. For example, in a class C amplifier the current flows through the tube in a series of pulses. The peak current in this case is defined as the maximum value that the current reaches. This is an example of the use of the term in a case where the waveform is non-sinusoidal.

In discussing the current from an ac power line and referring to the peak current we mean the maximum value that the current reaches. Most current meters indicate in rms values and give the ac current as the equivalent of a dc value. For example, when we say that the ac current in a circuit is 1 amp we mean that the current flowing in the circuit would have the same heating effect that 1 amp of dc flowing in the circuit would have. Actually the dc current is a steady value and remains constant, whereas the ac current is continually varying. The peak value of the ac current is 1.41 times the rms or effective value of the current.

(c) In a class C amplifier in which the current flows through the tube in a series of pulses, the dc plate-current meter indicates the "average current". The meter can't respond to the series of pulses -- it simply indicates the average value of these pulses and gives a dc current reading.

Sometimes the term "average current" is also used in conjunction with the ac power line. This is the net current flowing when the value is averaged. The average value of a sine wave is .636 times the peak value. Thus if the peak current flowing in the circuit supplied by an ac power line is 100 amperes then the average current flowing would be 63.6 amperes.

(d) Power is the rate at which we use energy. Power is often used in reading the input to an amplifier stage - it is also used in reading output. When we refer to the input power we mean the rate at which the stage is taking energy from the power supply. When we refer to the output power we mean the rate at which the stage is supplying energy to an antenna or to a following stage.

(e) Energy is the ability to do work. In the case of an electrical device such as a battery, it is the battery's ability to supply electrical power. In other words, the battery has the ability to supply this energy at a given rate. When we take this energy out of the battery and use it we are using power.

119. Describe how horizontal and vertical deflection take place in a cathode-ray oscilloscope. Include a discussion of the waveforms involved.

Horizontal deflection of the electron beam in a cathode-ray oscilloscope is usually accomplished by feeding a sawtooth voltage to the horizontal deflection plates. The voltage waveform fed to the plate is arranged so that a positive voltage is applied to the horizontal deflection plate on the left of the tube, as you face the tube, and a negative voltage to the deflection plate on the right. This pushes and pulls the beam over to the left side of the tube. As the positive voltage on the left plate and the negative voltage on the right plate decrease in a linear or straight-line fashion, the electron beam moves over to the center of the screen until, when these voltages drop to zero, the beam is essentially in the center of the screen. Then the polarity of the voltage applied to the two plates reverses so that the voltage begins to build up in a negative direction on the left plate and in a positive direction on the right plate. The negative voltage on the left plate pushes the beam toward the right; the positive voltage on the right plate pulls the beam. If the change in voltage is made linear or in a straight line, then the movement of the beam across the face of the tube will be constant over the entire motion from the left to the right. When the electron beam reaches the right side of the screen, we suddenly reverse the voltages applied to the plate so that the beam will fly back to the left side of the screen very rapidly and then be ready to trace out the next line across the tube.

Vertical deflection is obtained by applying the signal to be viewed to the vertical deflection plates. As the waveform and voltage of the signal vary the beam will move up and down on the face of the cathode-ray tube. Thus with the beam moving across because of the sawtooth voltage applied to the horizontal deflection plate, and moving up and down because of the signal voltage applied to the vertical plates, we can trace out on the face of the crt a picture of the voltage waveform applied to the vertical deflection plate.

120. What is the meaning of the term "carrier frequency"?

The "carrier frequency" refers to the frequency of the unmodulated radio-frequency signal. With no modulation applied to the radio transmitter it will radiate an rf signal at the carrier frequency.

121. If a carrier is amplitude-modulated, what causes sideband frequencies?

When an rf carrier is amplitude-modulated, the rf signal is mixed with the modulating signal and two new signals are produced; one signal is equal to the carrier frequency plus the modulating signal, and the other is equal to the carrier frequency minus the modulating signal. These two new signals produced are the sideband frequencies. They are created in the modulation process and they are essentially due to the mixing of the carrier and modulating signals.

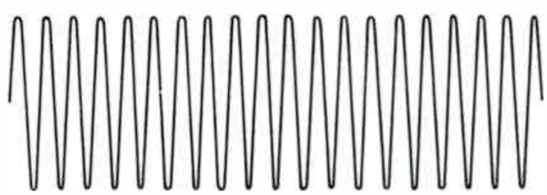
122. What determines the bandwidth of emission for an am transmission?

The bandwidth required for am transmission depends upon the frequency of the modulating signal. The highest frequency to be transmitted determines the bandwidth. For example, if the highest frequency signal to be transmitted is 2000 cycles, then the 2000-cycle signal will mix with the carrier and produce sidebands 2000 cycles or 2 kc above and below the carrier frequency. A bandwidth of 4 kc is required to accommodate the carrier and both sidebands. On the other hand, if the highest audio frequency to be transmitted is 10,000 cycles, and 10,000 cycles is equal to 10 kc, an upper sideband 10 kc above the carrier will be produced and a lower sideband 10 kc below the carrier will be produced. The bandwidth required for this transmission would be 20 kc.

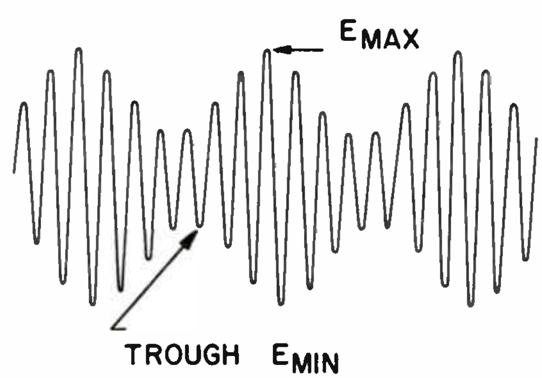
123. Why does exceeding 100% modulation in an am transmission cause excessive bandwidth of emission?

When a transmitter is modulated in excess of 100%, harmonics of the modulating frequency are produced. These harmonics in turn modulate the carrier and produce sidebands considerably outside the normal bandwidth required by the transmitter. Therefore, the am signal spreads out and radiates over a very wide bandwidth. As an example, if we are modulating a transmitter with a 5000-cycle signal, we would have a 5 kc bandwidth each side of the carrier or a total bandwidth of 10 kc. However, if we overmodulate the transmitter, a certain amount of third harmonic of the modulating signal may be produced. The third harmonic of 5 kc is 15 kc. This 15kc-signal would produce a sideband 15 kc above the carrier frequency and another sideband 15 kc below the carrier frequency. Therefore, the signal will spread out over a total bandwidth of 30 kc, instead of 10 kc as in the previous case where the transmitter was not overmodulated. Overmodulation can produce very high order harmonics and cause considerable spreading out of the carrier over an even wider bandwidth than the 30 kc in the example given.

124. What is the relationship between percentage modulation and the shape of the waveform "envelope" relative to carrier amplitude?



CARRIER E_C



With no modulation on the carrier the amplitude of the carrier is constant. But with a relatively small percentage of modulation (for example, 25% modulation) on the crest, the amplitude of the carrier will increase. At the same time, the troughs in the carrier decrease by an equal amount. When we have 100% modulation the crest or maximum amplitude of the carrier will be twice the amplitude of the carrier with no modulation, and the trough in the carrier will drop completely to zero. Thus the percentage modulation is an indication of the relationship between the amplitude of the crest and the troughs in the output signal.

The percentage modulation of the transmitter is given by the formula

$$\text{\% Modulation} = \frac{E_{MAX} - E_{MIN}}{2E_c}$$

E_{MAX} represents the amplitude of the crest.

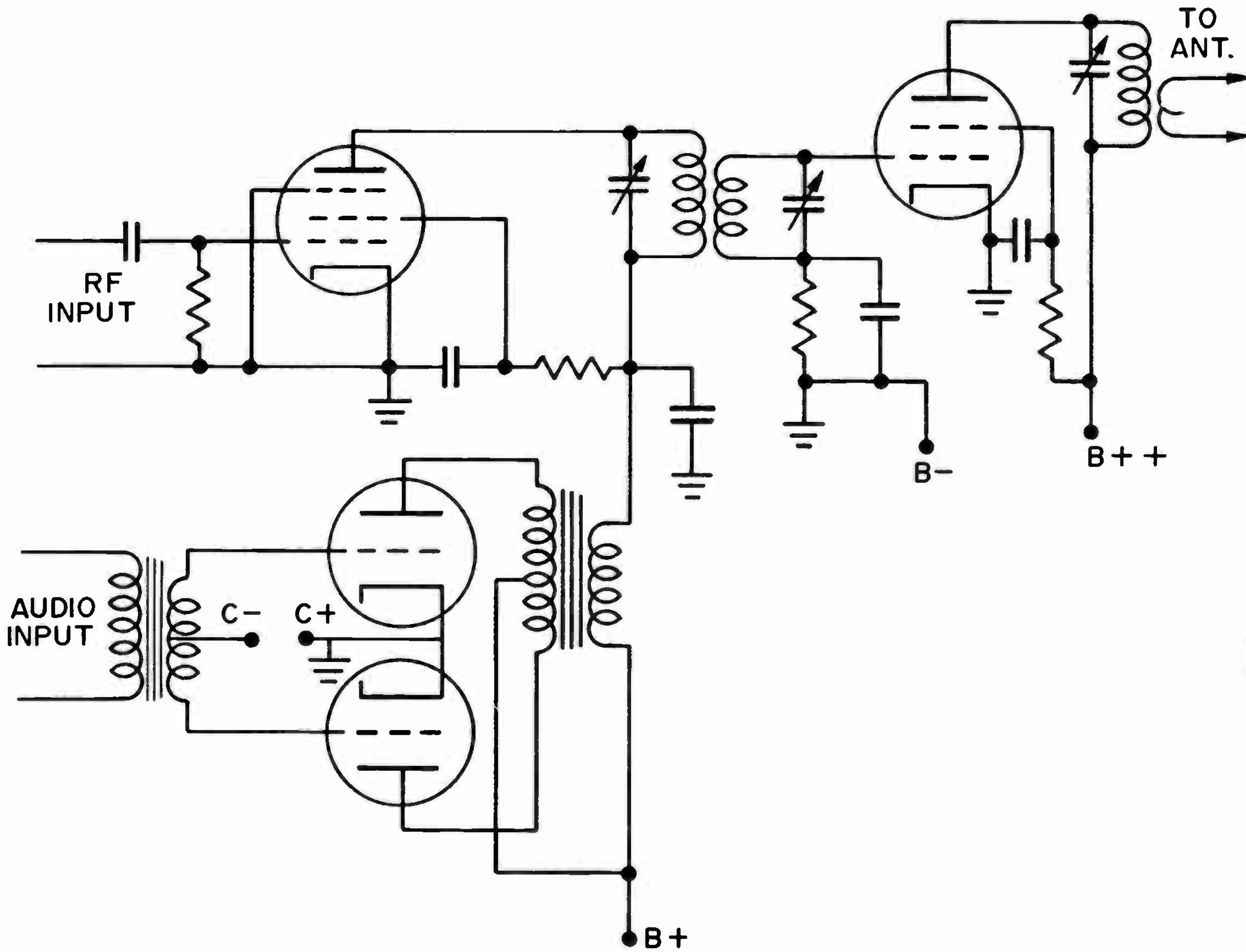
E_{MIN} represents the amplitude of the trough.

E_C represents the carrier amplitude with no modulation.

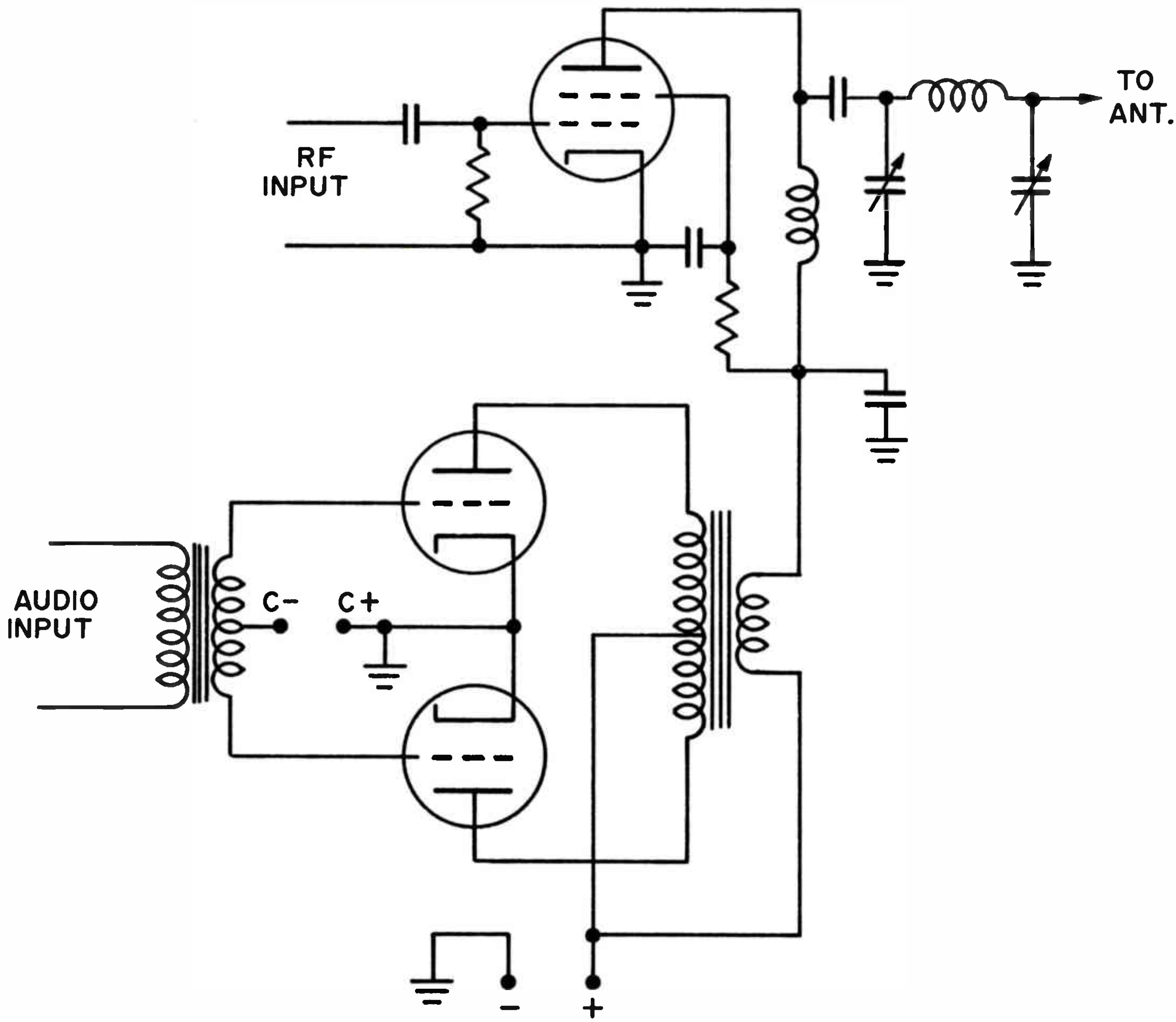
125. Draw a simplified circuit diagram of the final stages (modulator, modulated amplifier, etc.) of a type of low-level plate-modulated transmitter, utilizing a pentode tube in the modulated stage. Explain the principles of operation. Repeat, using a tetrode to provide high-level modulation.

In the transmitter using low-level modulation a low-power stage is modulated and then the modulated signal is amplified by linear amplifier stages before it is fed from the modulated stage to the antenna. In a high-level modulation, the final power-output stage is modulated and then the signal is fed directly to the antenna.

We are to show plate modulation in both examples required for this question. The diagram labeled A illustrates low-level modulation, in which the pentode tube is plate-modulated and followed by a linear amplifier. The diagram labeled B illustrates high-level modulation, using plate modulation on the tetrode tube.



(A) LOW-LEVEL PLATE MODULATION



(B) HIGH-LEVEL PLATE MODULATION

126. How does a linear power amplifier differ from other types?

A linear amplifier is biased to operate on the linear portion of its characteristic curve. A radio-frequency amplifier that is designed to amplify a modulated signal must be a linear amplifier. If a non-linear amplifier is used, considerable distortion will be introduced.

For example, a class C power amplifier is a non-linear power amplifier. It is fine for use in amplifying an unmodulated rf carrier because the amplitude of the unmodulated signal fed into the input of the stage remains constant. However, if we try to use a class C amplifier to amplify a modulated rf signal where the amplitude varies and the class C amplifier is operated correctly, the peaks in the modulated signal will be clipped and distortion will be introduced. When the amplitude of the input signal drops, it may be so low that it can't drive the grid of the class C tube into the current production region, so there will be no output from the stage. Considerable distortion will thus be introduced.

127. Draw a simple schematic diagram showing a method of coupling a modulator tube to a radio-frequency power amplifier tube to produce grid modulation of the amplified rf energy. Compare some advantages or disadvantages of this system of modulation with those of plate modulation.

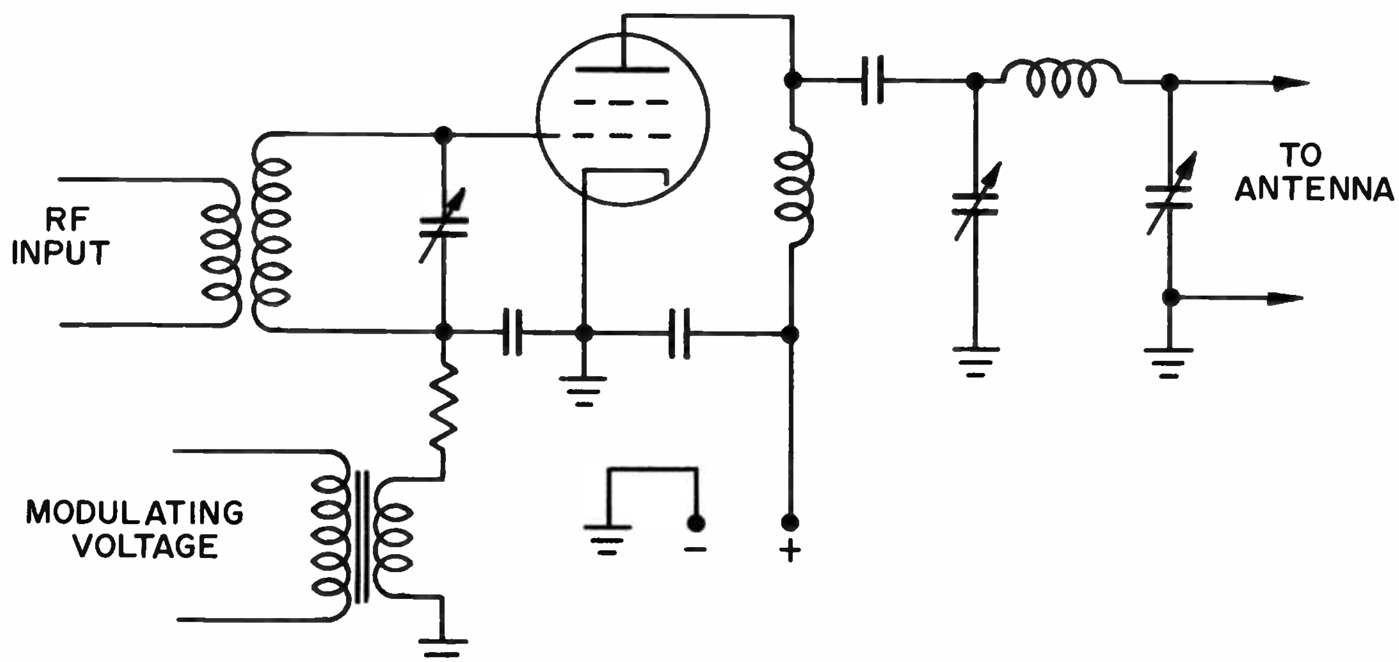
The advantage of grid-modulating a radio-frequency power amplifier is that comparatively little audio power is required to modulate the rf power amplifier compared to the audio power that would be required to plate-modulate the same stage. However, the disadvantage of grid modulation is the relatively poor efficiency of the system. An efficiency of about 30% to 35% is about all you can expect from a grid-modulated stage.

128. What is meant by "frequency shift" or "dynamic instability" with reference to modulated rf emission?

The terms frequency shift or dynamic instability mean that the carrier shifts frequency with modulation.

129. What would cause a dip in the antenna current when am is applied? What are the causes of carrier shift?

There are a number of things that can cause the antenna current to dip with am modulation. One of the most frequent causes is poor power supply regulation; another common cause is a defect in the modulator or the inability of the modulator to supply the power required on modulation peaks. Other problems that cause a dip in the antenna



GRID MODULATION

current are incorrect bias on the modulated stage or incorrect or insufficient rf drive or input to the modulated stage.

When the dc plate current in a plate-modulated stage fluctuates, we refer to this as carrier shift. This may be caused by poor power supply regulation and also a defect in the modulator. Poor power supply regulation, modulator defects and improper impedance match between the class C modulated stage and the modulator frequently cause a downward kick in the plate current meter with modulation. If the plate current meter kicks up with modulation, it is frequently a sign of overmodulation. It may also be due to incomplete or improper neutralization of the i-f power amplifier.

130. What is the relationship between the average power output of the modulator and the plate-circuit input of the modulated amplifier under 100% sinusoidal plate modulation? How does this differ when normal voice modulation is employed?

With 100% sinusoidal plate modulation, the modulator stage must supply an average power output equal to 50% of the dc power input to the modulated stage. However, with voice modulation, since there are many irregularities and peaks of very short duration that will cause this 100% modulation, the average power required by the modulator is much less than 50%.

131. What is the relationship between the amount of power in the sidebands and the intelligibility of the signal at the receiver?

The higher the power in the sidebands, the stronger the audio signal at the receiver. If the percentage modulation of the carrier is relatively low, even though a relatively strong signal might be received from the station, the actual audio output from the receiver could be comparatively low. On the other hand, if the modulation percentage is as close to 100% as possible without overmodulation, then you'll get maximum audio for the strength of the signal received.

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T-5:FCC

NRI Study Guide for FCC License Examination

PART VI

The material in this part of your study guide is meant to supplement what you have already studied in your lesson texts. Be sure to study this material carefully before answering the questions in the examination TC-6. The material covered in this part of the study guide contains information on the type of questions you are likely to be required to answer when you take your radiotelephone operator's license examination.

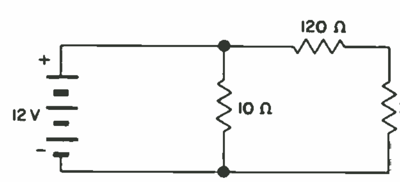
Be sure to spend any additional time you might need studying this part of the study guide or your lesson text if you find that some of the material is giving you difficulty. Keeping up with your course is the best way of being sure that you'll be able to pass the FCC examination and get your license. If you pass over small sections now that you do not understand completely, it is likely that these difficulties will cause you problems with later lessons.

Be sure to save this part of the study guide along with those you have already received so you will be able to make a complete review before going for your license.

Most of the questions in this section deal with antennas and transmission lines. There are also a few problems involving calculations which you should be able to do easily.

132. Draw a circuit composed of a 12-volt battery with three resistors (10, 120, and 300 ohms respectively) arranged in a "pi" network.

(a) What is the total current; the current through each resistor?
(b) What is the voltage across each resistor?
(c) What power is dissipated in each resistor?
(d) The total power dissipated in the circuit?



(a) From the circuit shown you can see that the 10-ohm resistor is connected directly across the battery. Therefore, considering this resistor first, the current through it can be found from Ohm's Law

$$I = \frac{E}{R}$$

and substituting 12 volts for E and 10 ohms for R we get

$$I = \frac{12}{10} = 1.2 \text{ amps}$$

In the second branch of the circuit we see that we have a 120-ohm resistor in series with a 300-ohm resistor across the 12-volt battery. Therefore the total resistance in this path is 420 ohms. Substituting these values in Ohm's Law we get

$$I = \frac{12}{420} = .0286$$

We can round this off to .029 amps. Therefore the current flowing through the 120-ohm resistor and the 300-ohm resistor will be approximately .029 amps. The total current flowing through the circuit will be 1.2 + .029 = 1.229 amps.

(b) Since the 10-ohm resistor is connected directly across the battery, the voltage across this resistor is 12 volts.

The 120-ohm and 300-ohm resistors are connected in series across the battery. The total voltage across the 420 ohms will be 12 volts. The voltage across the 120-ohm resistor will be

$$E = \frac{120}{420} \times 12$$

$$= 3.43 \text{ volts}$$

The remainder of the 12 volts will be dropped across the 300-ohm resistor. We can find this figure by subtracting 3.43 volts from 12 volts for a total of 8.57 volts.

(c) Since we know the voltage across each resistor, the current through each, and the resistance of each, we can find the power dissipated in each resistor by using any form of the power equation. Using the formula

$$P = \frac{E^2}{R}$$

and substituting 12 volts for E and 10 ohms for R we find that the power dissipated by the 10-ohm resistor is equal to

$$P = \frac{12 \times 12}{10} = 14.4 \text{ watts}$$

Substituting 3.43 for the voltage across the 120-ohm resistor and 120 ohms for R and using the formula again we get

$$P = \frac{3.43 \times 3.43}{120}$$

$$= \frac{11.76}{120} = .098 \text{ watts}$$

This is the power dissipated by the 120-ohm resistor. To find the power dissipated by the 300-ohm resistor, we substitute the voltage across it (8.57 volts) and the resistance (300 ohms) in the power formula. Now we have

$$P = \frac{8.57 \times 8.57}{300}$$

$$= \frac{73.44}{300} = .245 \text{ watts}$$

The total power dissipated in the circuit is equal to the sum of the powers dissipated by the individual resistors. Thus we have

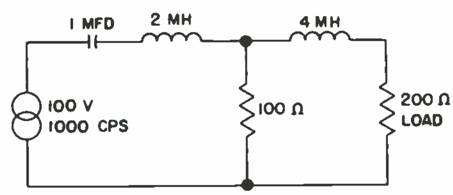
$$P_t = 14.4 + .098 + .245$$

$$= 14.783 \text{ watts}$$

We can round this off to 14.78 watts or we could even round it off to 14.8 watts and be close enough.

133. Draw a circuit composed of a voltage source of 100 volts, 1000 cps, a 1-microfarad capacitor in series with the source, followed by a "T" network composed of a 2-millihenry inductor, a 100-ohm resistor and a 4-millihenry inductor. The load resistor is 200 ohms.

(a) What is the total current; the current through each circuit element?
(b) What is the voltage across each circuit element?
(c) What "apparent" power is being consumed by the circuit?
(d) What real or actual power is being consumed by the circuit; by the 200-ohm resistor?



(a) The first step in solving a problem of this type is to find the value of the reactive components. To find the reactance of the capacitor we use the formula

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

Substituting 1000 cps, which is equal to 1×10^3, for f, and 1×10^{-6} farads, which is equal to 1 mfd, for C, we get:

$$X_c = \frac{1}{6.28 \times 1 \times 10^3 \times 1 \times 10^{-6}}$$

$$= \frac{.159}{10^{-3}} = 159 \text{ ohms}$$

We find the value of the inductive components by using the formula

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

Substituting 1000 cycles or 1×10^3 for f and 2×10^{-3} for L, we can find the reactance of the 2-mh inductor. We will get

$$X_L = 6.28 \times 1 \times 10^3 \times 2 \times 10^{-3}$$

$$= 12.56 \text{ ohms}$$

and we can round this value off to 12.6 ohms. The 4-mh inductor will have twice the reactance of the 2-mh inductor. Therefore its reactance will be 25.12 ohms; we can round this off to 25 ohms.

The next step in solving the problem is to find the impedance of the parallel branch consisting of the 100-ohm resistor in parallel with the 4-mh inductor and the 200-ohm resistor. The impedance of the circuit consisting of the 4-mh inductor and the 200-ohm resistor can be expressed as

$$Z = 200 + j25$$

Expressing this in polar form we will get the impedance as $202\angle 7.1^\circ$.

If we assume a voltage of 10 volts across the parallel combination of the 100-ohm resistor and the 4-mh choke in series with the 200-ohm resistor, we can calculate the current that would flow through each branch. With 10 volts across the circuit, the current through the 100-ohm resistor would be 0.1 amp. The current through the series branch would be

$$I = \frac{10}{202\angle 7.1^\circ} = .0495\angle -7.1^\circ \text{ amp}$$

$$= .049 - j.00605 \text{ amp}$$

We can get the total current now by adding the current through the two branches,

$$I = .1 + .049 - j.006$$

$$= .149 - j.06 = .15 \text{ amps}$$

Now we find the impedance of the circuit by dividing the current into the assumed voltage of 10 volts and we get

$$Z = \frac{10}{.15} = 66.666 \text{ ohms}$$

We can round this value off to 67 ohms.

Now we can find the total impedance of the circuit. The impedance of the parallel combination of the 100-ohm resistor in parallel with the 4-mh coil and the 200-ohm resistor is close to being a pure resistance; the phase angle is so small that we can assume it is a pure resistance, and will not appreciably affect the calculations. Then we will get as the impedance of the total circuit

$$Z = 67 - j159 + j12.5$$

$$= 67 - j146 \text{ (approximately)}$$

Converted to polar form, the total impedance is

$$Z = 161\angle -65.4^\circ$$

Since we know the voltage is 100 volts and we now have the impedance of the circuit, we can find the total current flowing. The total current will be

$$I = \frac{E}{Z}$$

$$= \frac{100}{161\underline{/-65.4^\circ}} = .62 \text{ amps}$$

Thus the current through the 1-mfd capacitor and the 2-mh choke coil will be .62 amps. Knowing the impedance of the parallel circuit and the total current flowing, we can calculate the voltage across this circuit.

$$E = IZ$$

Substituting .62 for I and 67 ohms for Z we get

$$E = .62 \times 67 = 41.5 \text{ volts}$$

Now that we know that the voltage across the parallel combination is 41.5 volts, we can use this information to find the current through the individual branches. The current through the 100-ohm resistor can be found from

$$I = \frac{41.5}{100} = .415 \text{ amps}$$

The current flowing through the branch consisting of the 4-mh coil and the 200-ohm resistor can be found by dividing the voltage by the impedance. This will give us

$$I = \frac{41.5}{202} = .206 \text{ amps}$$

Since we know the current flowing through the resistor, the voltage across the 200-ohm resistor will be

$$E = .206 \times 200 = 41.2 \text{ volts}$$

The voltage across the 4-mh coil can be found by substituting the values for the current flowing through it and the reactance of the coil. This will give us

$$E = .206 \times 25 = 5.15 \text{ volts}$$

We have already calculated the voltage across the 100-ohm resistor and found it to be 41.5 volts. The voltage across the 1-mfd capacitor can be determined since we know its reactance and the current flowing through it. Substituting in the ac form of Ohm's Law we get

$$E = .62 \times 159 = 98.7 \text{ volts}$$

and the voltage across the 2-mh coil is

$$E = .62 \times 12.6 = 7.8 \text{ volts}$$

(c) The apparent power being consumed by the circuit is equal to the voltage times the current. Thus the apparent power is

$$P = 100 \times .62 = 62 \text{ volt amps}$$

(d) The real and actual power being consumed by this circuit is equal to the voltage times the current times the power factor. The power factor is equal to the cosine of the phase angle. Since we found the phase angle is 65.6°, the real power is

$$P = 100 \times .62 \times \cos 65.4^\circ$$

$$= 25.8 \text{ watts}$$

The power being consumed by the 200-ohm resistor can be found from the formula

$$P = \frac{E^2}{R}$$

$$= \frac{41.2 \times 41.2}{200} = 8.48 \text{ watts}$$

The figures in this problem have been rounded off -- most calculations were performed using a slide rule. However, if you work these problems out by multiplying and dividing each individual formula you should come close to the values given. If you decide to do this, be sure to round your numbers off rather than carry long decimals which are really of no significant value.

134. What is meant by the "characteristic" (surge) impedance of a transmission line; to what physical characteristics is it proportional?

A transmission line is made up of a pair of spaced conductors. It can be considered as being divided into small segments or parts composed of inductance, capacitance and resistance. The current flowing to each segment or part produces a voltage drop across each section which causes a current to flow in the next section and so on down the line. The impedance measured across each section or part of a line is the same and constant. This impedance is called the characteristic or surge impedance of the line.

The characteristic impedance of a transmission line varies directly as the square root of the inductance and inversely as the square root of the capacitance. The characteristic impedance is given by the formula

$$Z_0 = \sqrt{\frac{L}{C}}$$

135. Why is the impedance of a transmission line an important factor with respect to matching "out of a transmitter" into an antenna?

For efficient operation of a transmitter, you want to get power from a transmitter into the antenna. The power is transported from the transmitter to the antenna by the transmission line. If the transmission line is terminated in its characteristic impedance, all of the energy going down the line will be absorbed by the impedance on the end of the line. To accomplish this, the antenna must be matched to the transmission line. Therefore the impedance of the transmission line is important; the line must be connected to the antenna at a point where the impedance of the antenna is equal to the characteristic impedance of the transmission line. Then the losses on the transmission line will be at a minimum, there will be little or no radiation from the transmission line, and the overall radiation pattern will be that of the antenna.

136. What is meant by "standing waves", "standing-wave ratio (SWR)," and "characteristic impedance" as referred to in transmission lines? How can standing waves be minimized?

We have already defined the characteristic impedance of a transmission line. It might also be considered as the impedance the generator would see if it were connected to a transmission line that was infinitely long.

When energy is sent down a transmission line, if the transmission line is terminated in its characteristic impedance, all the energy will be absorbed or dissipated in the load. However, if the transmission line is not terminated in its characteristic impedance, then only part of the energy is absorbed by the load. The remainder of the energy is reflected back along the transmission line toward the source. The energy moving back from the load toward the source interferes with the energy coming from the source toward the load. The reflected voltage will add to the transmitted voltage algebraically so that at points along the line the voltage reaches a maximum value and at other points it reaches a minimum value. A wave that is standing on the line can actually be traced out along the transmission line and is called a standing wave.

The standing-wave ratio along the transmission line is equal to the maximum voltage divided by the minimum voltage standing along the line. When the line is terminated in its characteristic impedance, the maximum voltage will be equal to the minimum voltage so that the standing-wave ratio will be 1. This is the ideal situation. If any other value of load is connected across the line, E max. will be greater than E min. and the value of the standing-wave ratio will increase above 1. The higher the value of the standing-wave ratio, the more poorly the load terminates the transmission line. Therefore the standing-wave ratio can be minimized by terminating the transmission line with a load having a resistance as close as possible to the characteristic impedance of the transmission line.

137. If standing waves are desirable on a transmitting antenna, why are they undesirable on a transmission line?

Standing waves are desirable on a transmitting antenna because the antenna should radiate power into space. However, in a transmission line, we want to transport the power from the transmitter to the antenna. We do not want the transmission line to radiate power. Therefore standing-wave ratios on the line are undesirable.

It might at first appear inconsequential whether power is radiated by the antenna or by the transmission line. However, by the right design, the antenna can be constructed to radiate power in a given pattern. In the case of a broadcast station located north of a city to which you want to direct your programs, the antenna should radiate maximum power toward the south to cover the city. Therefore the antenna would be designed for this purpose. However, if the transmission line is radiating power, there is no predicting in which direction the power will be radiated. It may be radiated to the north where the population is comparatively sparse. This is power that is lost and can't be radiated to the south, where you want the power radiated.

138. What is meant by "stub tuning"?

Stub tuning is making use of a section of transmission line to match the antenna to the transmission line. An example of stub tuning is a quarter-wave matching stub. Here a length of transmission line that is one quarter wavelength long is used to match the antenna to the transmission line. The impedance of the line used to make the quarter-wave matching stub must be the geometric mean between the antenna impedance and the line impedance. You can find the impedance of the required quarter-wave stub by using the formula:

$$Z = \sqrt{Z_1 \times Z_2}$$

where Z is the impedance of the quarter-wave stub, Z_1 is the antenna impedance, and Z_2 is the impedance of the transmission line.

139. What would be the considerations in choosing a solid-dielectric cable over a hollow pressurized cable for use as a transmission line?

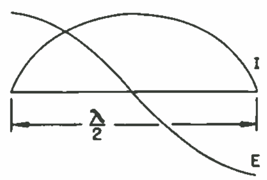
A solid-dielectric cable is flexible and can usually be routed in any required direction. On the other hand, a hollow pressurized cable is rigid and much more difficult to route. The solid-dielectric cable also is much less expensive.

The upkeep of a solid-dielectric cable is practically zero. Installed, this type of cable has a reasonably long life requiring no maintenance. On the other hand, most hollow pressurized cables continually lose gas, making it necessary to keep a gas cylinder continuously connected to the cable to regulate the pressure. This can be fairly costly over a length of time.

140. Explain the voltage and current relationships in a one-wavelength antenna; a half-wavelength (dipole) antenna; one quarter-wavelength "grounded" antenna.

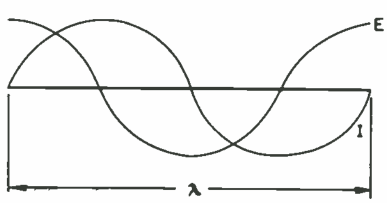
The basic antenna that we use in explaining antenna phenomena is the half-wavelength antenna. An antenna acts actually like an open transmission line. The electrical energy travels down the antenna to the open end and is reflected back, setting up a standing wave along the antenna. In the case of a half-wave antenna the voltage and current distribution along the antenna is shown in Fig. A. The voltage reaches its maximum at the open end of the antenna and is zero at the center of the antenna. The current standing wave along the antenna falls to zero at the ends of the antenna and reaches its maximum in the center. A dipole is simply a center-fed half-wave antenna; the voltage and current distribution along it are the same.

(A)



A one-wavelength antenna acts simply like two half-wave antennas connected in series. The voltage distribution along a one-wave antenna is shown at (B). Notice that at the end of the antenna the voltage standing wave is at a maximum. A quarter-wavelength along the antenna, which would represent the center of a half-wave antenna, the voltage has dropped to zero. Another quarter-wave along the antenna, which would be the end of a half-wave antenna, the voltage wave is at a maximum. The wave continues through the cycle so that we have a maximum at the two ends and a maximum voltage wave in the center. A quarter-wavelength in from each end of the antenna we have a zero voltage node.

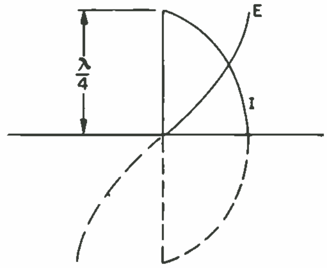
(B)



The current distribution on the antenna is the same as it would be on two half-wave antennas connected in a series. At one end the current is zero; a quarter-wavelength in the current has reached its maximum value. This is the equivalent to the center of the first half-wave antenna. Another quarter-wave along the antenna the current has dropped back to zero again. Thus, as far as the current standing wave is concerned, we have zero current at the two ends and at the center, and a maximum current one quarter-wavelength in from each end.

A quarter-wave grounded antenna acts like half of a half-wave antenna. The open end of the antenna has the voltage maximum and current minimum. At the grounded end of the antenna we have zero voltage and maximum current. The earth more or less acts like a mirror, causing a mirror reflection of the other quarter-wavelength of the antenna. Thus in many respects the antenna acts like a half-wave antenna. The voltage and current distribution along a ground quarter-wavelength antenna are shown at (C).

(C)



141. What effect does the magnitude of the voltage and current at a point on a half-wavelength antenna in "free space" (a dipole) have on the impedance at that point?

The impedance of an antenna at a particular point is given by the formula

$$Z = \frac{E}{I}$$

The higher the voltage at the particular point and the lower the current, the higher the impedance will be. A dipole has an average impedance of about 72 ohms at the center. Thus, a dipole is frequently fed at the center by means of a 72-ohm transmission line because the transmission line will be terminated in its characteristic impedance. However, the actual impedance at the center point or at any other point along the antenna will be influenced by nearby objects that affect the voltage and current standing waves on the antenna.

142. How is the operating power of an AM transmitter determined using antenna resistance and antenna current?

The operating power of an AM transmitter can be obtained using the power formula,

$$P = I^2 R$$

where I is the transmission line current and R the characteristic impedance of the transmission line. The antenna current meter in the transmitter actually reads the transmission line current, not the antenna current. Of course, this system works only when the antenna matches the transmission line so that there are no standing waves on the line. If there are standing waves on the line, then the current reading you obtained by putting an rf ammeter in the transmission line would be meaningless for determining the power fed to the antenna. If you happened to be at a spot where you had a maximum in your current standing wave, you would get a high current reading; if you were where there was a minimum, you would get a low current reading. You would have no way of knowing what type of reading you are actually getting.

However, if the transmission line is terminated in its correct impedance by the antenna, the power being fed to the antenna from the transmission line will be equal to the square of the current on the line times the characteristic impedance of the line.

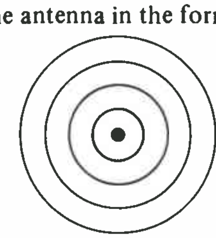
To find the power that the transmitter is delivering to the antenna with no modulation, measure the antenna current with zero modulation. If you want to find the maximum modulated power, then you should bring the modulation up to 100% (or as close to 100% as you intend operating the transmitter). Then use the current reading obtained under these conditions in calculating transmitter power output.

143. What kinds of fields emanate from a transmitting antenna and what relationships do they have to each other?
There are three fields radiated from a transmitting antenna. One field, called an induction field, exists only very close to the antenna and is of no importance insofar as communications are concerned. The other two fields are radiated by the antenna and propagated through space at right angles to each other. These are the electric field and the magnetic field. A vertical antenna will radiate an electric field that is polarized vertically. We call this vertical polarization. A horizontal antenna will radiate an electric field that is horizontal; this is called horizontal polarization.
When the electric field is vertical, the magnetic field will be horizontal; when the magnetic field is vertical the electric field will be horizontal. These two fields will always be at right angles to each other.

144. Can either of the two fields that emanate from an antenna produce an emf in the receiving antenna? If so, how?
In a radio wave both the electric field and the magnetic field will be present at all times. One cannot exist without the other. The presence of the two causes a current to flow in the receiving antenna. However, the electric field has a more significant influence on the signal picked up in a straight-wire antenna than the magnetic field will have. On the other hand, the magnetic field will have more effect on the current picked up in a loop-type of antenna.

145. Draw a sketch and discuss the horizontal and vertical radiation patterns of a one-quarter wave vertical antenna. Would this also apply to a similar type of receiving antenna?
A vertical antenna radiates a horizontal radiation pattern equally well in all directions. Looking at the antenna from the top, as shown at A, we see the radiation spreading out around the antenna essentially in the form of circles. In actual practice, the circles might be distorted somewhat due to the presence of buildings, trees, hills, etc. However, unless the radiation pattern is modified by some external object, the energy would be radiated from the antenna in the form of circles.

A



The vertical radiation from the vertical antenna is low-angle radiation, as shown at B. This means that the angle between it and the surface of the earth is small. The rf signal tends to skim along close to the earth rather than shoot on up into the ionosphere at a very sharp angle.

B

The characteristics of the vertical receiving antenna are essentially the same as those of the vertical transmitting antenna. You will receive signals equally well from all directions. The vertical receiving antenna will receive a signal travelling parallel to the surface of the earth better than it will pick up a signal that is coming at it in a sharp angle from the ionosphere.

As far as transmitting and receiving are concerned the similarities between the characteristics of the vertical antenna are true of all antennas. An antenna that transmits energy in a comparatively narrow beam will also receive energy only from a comparatively narrow beam. On the other hand, an antenna that transmits equally well in all directions will receive equally well in all directions.

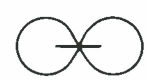
146. Describe the directional characteristics of horizontal and vertical loop antennas.

A vertical loop, as shown at (A), radiates a horizontal pattern that looks like a figure 8, as shown at (B). The radiation is best in the plane of the loop. The vertical radiation pattern is essentially in the form of a circle and is primarily high-angle radiation.

A

SIDE VIEW OF LOOP ANTENNA

B



TOP VIEW SHOWING HORIZONTAL RADIATION

If the loop is tipped over so that it becomes horizontal, then the radiation patterns are turned 90°. The loops become omnidirectional in the horizontal direction and transmit equally well in all directions. The vertical radiation pattern is primarily high-angle radiation.

147. In speaking of radio transmissions, what bearing does the angle of radiation, density of the ionosphere and frequency of emission have on the length of the skip zone?

In general, the lower the angle of radiation the further the signal travels from earth to the ionosphere and back to earth again. A high angle of radiation favors a relatively short skip zone.

The density of the ionosphere will determine the maximum frequency signal that will be reflected back to the earth by the ionosphere. Generally speaking, the more dense the ionization of the ionosphere the higher the frequency of the signal that will be reflected back to the earth. The higher the frequency of the signal, the longer the skip zone will be.

148. Why is it possible for a sky-wave to "meet" a ground wave 180° out-of-phase?

The phase that two signals radiated by a single antenna and travelling over different paths have in relationship to each other when they meet at a distant point depends upon the distance the two signals have travelled. If one signal has travelled an odd number of wavelengths more than the other signal, the two signals will be 180° out-of-phase where they meet and will tend to cancel each other. On the other hand, if the two signals have travelled an even number of half-wavelengths, they will be in phase when they meet and will tend to aid each other. Often fading in and out of stations, particularly when this occurs rapidly and causes a garbled signal, is due to the fact that the path the sky-wave is travelling is changing and hence distance is changing. It ultimately aids and then opposes the sound wave, causing flutter or rapid fading in and out of the signal.

149. What is the relationship between operating frequency and ground-wave coverage?

Generally speaking, the higher the frequency at which a station is operating, the more rapidly the ground wave is attenuated. Thus the ground wave from very low frequency stations will travel great distances, while the ground wave from very high frequency stations will travel only a relatively short distance.

150. Explain the following terms with respect to antennas (transmission or reception):

(a) field strength; (b) power gain; (c) physical length; (d) electrical length; (e) polarization; (f) diversity reception; (g) corona discharge.

(a) The field strength is the strength of the radio wave at a certain distance from the antenna. We usually express the field strength as a certain voltage per meter. This is the amount of voltage that would be induced in an antenna by the rf wave for every meter of the antenna length. For example, if an antenna is exactly one meter long and a passing rf signal induced a signal of 10 microvolts in the antenna, we say that the field strength is 10 microvolts per meter. The field strength can be expressed in millivolts or volts. Since it is usually quite small, it is more often expressed in microvolts per meter.

(b) The power gain of an antenna is an indication of how effective an antenna is in comparison to a dipole. If we say that the power gain of a certain antenna is 10 db, we mean that at a given point the field intensity would be 10 db higher than it would be if the same power were transmitted by a dipole antenna.

(c) The physical length of an antenna is the actual length of the antenna measured in meters or some other measurement such as feet.

(d) The electrical length of an antenna is a measurement that tells you what part of a wavelength the antenna is. For example, due to the end effect of an antenna, the antenna acts like it is longer than it really is. Therefore a half-wave antenna designed for operation on 50 meters would not be cut to a length of 25 meters but to some physical length slightly less than 25 meters. The electrical length of the antenna will be 25 meters because you'll have a complete half-wave of voltage and a half-wave of current standing on the antenna.

(e) Polarization may refer to the polarization of the radio wave or the polarization of an antenna. When we say that a radio wave is horizontally polarized we mean that the electrical signal is polarized horizontally. The magnetic signal will be at right angles to it, and therefore it will be vertically polarized. If we say that the signal is vertically polarized, then the electric signal is vertical and the magnetic signal will be horizontal.

A dipole signal radiates an electric signal having the same polarity as the dipole itself. Therefore if the dipole is placed in a horizontal plane that is parallel to the earth, we say it has horizontal polarization and it radiates a horizontally polarized wave. If the dipole is placed in a vertical position, we say that it is vertically polarized and it will radiate a vertically polarized wave.

(f) Diversity reception refers to reception from two antennas. In this type of reception the two antennas are located some distance from each other. Thus if the signal at one antenna begins to fade, chances are that the signal at the other antenna will be comparatively strong and giving satisfactory reception. Provisions are made for automatically switching to the antenna with the strongest signal.

(g) A corona discharge is an electrical discharge from the surface of a conductor, occurring when the voltage becomes excessive. For example, you may get a corona discharge off the end of a sharp point because of the stresses set up in the electric field around that point. It will take a much higher voltage to produce a similar discharge from a ball. Here the electrical stresses built up would not be as high for a given voltage as they would be from a sharp object. Corona discharges from a whip-type antenna such as used on mobile equipment can produce static or noise in the receiver. You are much more likely to get a discharge from a whip antenna if the antenna comes to a sharp point. To cut down on these discharges, you will often find a ball or cap on the end of the antenna.

151. What would constitute the ground plane if a quarter-wave "grounded whip" antenna, 1 meter in length, were mounted on the metal roof of an automobile; mounted near the rear bumper of an automobile?

With the antenna mounted on the roof of the car, the car roof would serve as a reasonably good ground plane. With a quarter-wave antenna, the diameter of the ground-plane radials should be a half-wavelength; in this case it would be 2 meters. The car roof would come pretty close to meeting this requirement.

With the antenna mounted on the bumper, the bumper itself would to some extent act as a radial as would the rear of the car. However, in this case the effectiveness of the ground plane would not be as high as with the antenna mounted in the center of the roof.

152. Explain why a "loading coil" is sometimes associated with an antenna. Under this condition, would absence of the coil mean a capacitive antenna impedance?

A loading coil is sometimes used with an antenna to increase the antenna's electrical length. In the case of a half-wave antenna that is slightly shorter than a half-wavelength, a loading coil may be used with the antenna to increase the electrical length to a half-wavelength. If the loading coil is removed, the antenna would act like an inductance. A loading coil may also be used with a quarter-wavelength antenna to permit the use of an antenna that is actually shorter than a quarter-wavelength. The loading coil adds to the length so that the electrical length of the antenna plus the loading coil becomes a quarter-wavelength. In this circumstance, if the loading coil is removed, the antenna would act like a capacitive reactance. An antenna that is electrically less than a quarter-wavelength in length has a capacitive impedance. At exactly quarter-wavelength it acts as a pure resistance. Between a quarter-wavelength and a half-wavelength it acts as an inductance. At a half--wavelength once again it acts as a pure resistance. Between a half-wavelength and a three quarter-wavelength it acts like an impedance; at three quarter-wavelength it will act as a pure resistance again. Between three quarter-wavelength and one-wavelength it will act as an inductance; at a full-wavelength, a pure resistance. As the length of the antenna is increased, the reactance and resistance impedances repeat themselves.

153. What radio frequencies are useful for long-distance communications requiring continuous operation?

Low frequencies (frequencies below 500 kHz) are used with large antennas and high-powered transmitters for continuous long-distance communications. Sometimes long-distance communications are possible with comparatively low power using the short wavelengths, but propagation of the high-frequency short-wavelength signal varies from season to season and also varies during the day. Short-wave transmission is also subject to variation as the sun spot cycle varies. Therefore this type of communication is not nearly as reliable as communications on the low frequencies.

154. What type of modulation is largely contained in "static" and "lightning" radio waves?

Amplitude modulation. Because the static and lightning cover a wide frequency range, the amplitude varies rapidly. Therefore no matter what frequency your receiver is tuned to, you'll pick up some of the signal. Since its amplitude is varying, it will act like an amplitude-modulated signal.

155. Will the velocity of signal propagation differ in different materials? What effect, if any, will this have on the wavelength of a frequency?

The velocity of a signal travelling through different materials may vary. For example, a signal moving down a transmission line does not move down the line as rapidly as it moves through free space. Therefore the material affects the wavelength. If the signal is moving slower along a transmission line, the wavelength is shorter than it would be in free space. The velocity of propagation, however, has no effect on the frequency. The frequency is determined by the generating source.

156. Discuss series and shunt feeding of quarter-wave antennas with respect to impedance matching.

Shunt feed is used with a grounded quarter-wave antenna. At the grounded end of the antenna, the antenna impedance is zero. As you move up the antenna from the grounded end toward the open end, the antenna impedance increases. In shunt feed you connect the transmission line to the point on the antenna where the antenna impedance is equal to the line impedance. In the case of a 72-ohm coaxial cable, for example, you connect the ground on the cable to the ground point on the antenna and the inner conductor from the cable at a point up on the antenna where the antenna impedance is 72 ohms.

Series feed is used with an insulated quarter-wavelength antenna. Here, in the case of a vertical antenna, the lower end is insulated from ground. The impedance at this point will be approximately 36 ohms. This is the theoretical impedance; it will actually be influenced somewhat by surrounding objects. However, the usual procedure is to use a matching network of some type to match the transmission line or the transmitter output to the 36-ohm impedance of the antenna. In the case of a transmitter that is being fed to a quarter-wave insulated antenna by means of a 72-ohm transmission line you could use a quarter-wave matching section between the transmission line and the antenna. The matching section should be a quarter-wavelength long and should have an impedance equal to the geometric mean of the impedance of the antenna (36 ohms) and the impedance of the transmission line (72 ohms.) In this case, the geometric mean would be

$$\sqrt{36 \times 72} = 51 \text{ ohms (approx.)}$$

NRI Study Guide for FCC License Examination

PART VII

The material in this part of the study guide supplements the material you have been studying in your lesson text. Be sure you study this part of the study guide carefully because it contains information on the type of questions you will probably be required to answer when you go for your radiotelephone operator's license.

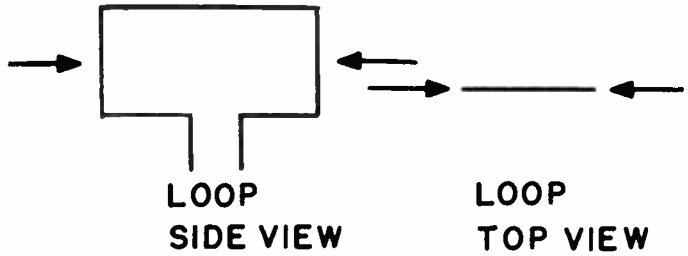
If you find that you need additional study on any of the material covered in this part of the study guide, be sure you go back to the lesson text and do whatever extra studying is necessary. Keeping up with the material in your course as you go along is the best way of making sure that you will be able to pass your examination and get your license.

The questions covered in this section are quite varied. There are also several questions involving calculations. If you have trouble with calculations go back and review lessons 5X, 9X, 15X and 19X. The calculations that you will have to do in the examination accompanying this part of the study guide are similar to those that you will have to be able to perform in order to get your FCC radiotelephone operator's license.

157. Discuss the directivity and physical characteristics of the following types of antennas:

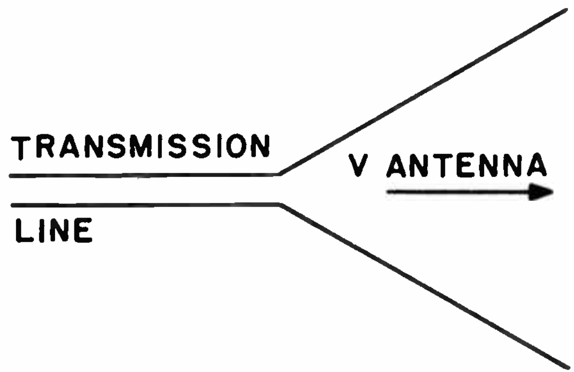
(a) Single loop
(b) V-beam
(c) Corner-reflector
(d) Parasitic array
(e) Stacked array

(a) The single loop-type antenna is generally made of rather heavy conductor so that it can be self-supporting. The direction in which the loop receives best is in the plane of the loop. The loop is bi-directional inasmuch as it will receive as well from one side of the loop as it will from the other. Transmitting characteristics of a loop antenna are the same. Loop antennas are often used in direction-finding equipment. In this type of equipment the loop is placed in the vertical direction and rotated. Once you find where you have maximum signal pickup, you know that the signal is coming from a direction in the plane of the loop. Of course, it may be coming from either side, but other means are used to determine from which side of the loop the signal is coming.



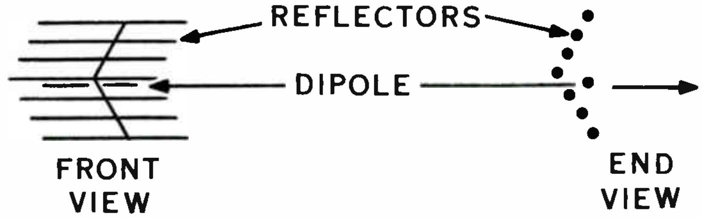
ARROWS SHOW DIRECTION OF MAXIMUM SIGNAL PICK-UP

(b) The V-beam antenna is so called because it is arranged so that it looks like the letter V. A top view of a V-beam antenna is shown. Each



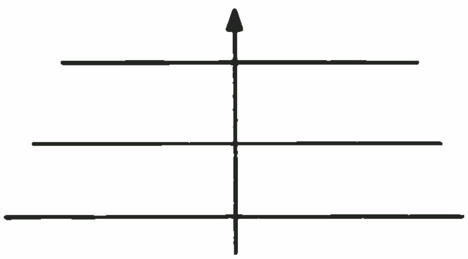
leg of the V-beam antenna is usually several wavelengths long. The longer the legs are, the more directional the antenna is and the higher the gain we can obtain from the antenna. In transmitting, the maximum signal is transmitted in the direction shown, in receiving, the antenna receives best from the direction it transmits best. Except at vhf, V-beam antennas are quite large and usually take up considerable space.

(c) The corner-reflector is usually used for vhf and uhf reception. A drawing of a corner-reflector is shown. The corner-reflector an-



tenna is a unidirectional antenna, when used as a transmitting antenna it transmits in the direction shown and when used as a receiving antenna, it receives best from the direction in which it transmits best. It is a high-gain antenna and can be mounted either horizontally or vertically. In the horizontal position it transmits and receives a horizontally polarized signal whereas in the vertical position it transmits and receives a vertically polarized signal.

(d) A parasitic array may consist of a comparatively simple antenna made up of a dipole and a single parasitic element which may be used either as a reflector or a director depending on its length or it may be a comparatively complex antenna system consisting of a dipole plus a number of parasitic elements. A top view showing a parasitic array consisting of a director, a dipole and a reflector is shown. All parasitic



arrays are somewhat similar to the one shown. The dipole is 1/2 λ long, directors are about 10% shorter than the dipole and reflectors about 10% longer. If additional directors or reflectors are added, the directivity of the antenna increases and at the same time the gain increases. However, the increase in gain, once you get past the so-called three-element parasitic array consisting of a reflector, dipole and director increases slowly as you add elements. In order to increase the gain 3 db you must double the number of parasitic elements. Thus the more elements you add, the less additional elements affect the gain and directivity of the array.

Parasitic arrays are quite widely used at frequencies of about 10 mc and up. Below this frequency the parasitic array becomes quite large and cumbersome.

(e) Stacked array. A stacked array can be made by taking two identical parasitic arrays and stacking one above the other. The arrays may be used to increase the low-angle radiation of the antenna. In addition, adding a second stacked array effectively doubles the elements in the antenna assembly so a gain of approximately 3 db can be expected. Stacked arrays are usually used only at the higher frequencies, usually in the vhf range and above because at lower frequencies they become extremely large, heavy and difficult to manage.

158. Why are insulators sometimes placed in the antenna guy wires?

Frequently the guy wires used to support an antenna are long enough to become an appreciable part of a half-wavelength in length. When this happens, the guy wire may act like a parasitic element and distort the antenna pattern. This can be avoided by breaking the guy wire into a number of sections and using insulators between the various sections. Insulators called egg-type insulators are frequently used. These are porcelain insulators and the two pieces of guy wires are looped through the insulator in such a way that if the insulator should break, the guy wires themselves will still be looped together so that the antenna cannot fall.

159. Compare some properties of electrostatic and electromagnetic fields.

An electromagnetic field is a field set up by an electromagnet. The electromagnet may consist of a number of turns of wire wound in the form of a coil and designed to set up this electromagnetic field, or it may be a single conductor with a current flowing through it. A single conductor will set up an electromagnetic field around it and the strength of the field will depend upon the current flowing through the conductor.

Electromagnetic fields are identical to magnetic fields. Just as the magnetic fields produced by two north poles of two permanent magnets will repel each other, so will two identical electromagnetic fields. On the other hand, if the electromagnetic fields have the opposite polarities then they will attract each other just as two dissimilar permanent magnets will.

An electrostatic field is similar to an electromagnetic field except that it is an electric field set up between two charged objects. For example, there is an electrostatic field existing between two plates on a charged capacitor. The strength of the electrostatic field depends upon the potential applied to the two charged objects. It also varies inversely with the distance separating the two charged objects. Electrostatic fields set up flux lines just like electromagnetic fields do.

160. In what way are electrical properties of a common-circuit element affected by electromagnetic fields? Are interstage connecting leads susceptible to these fields?

Some circuit elements are affected by fixed electromagnetic fields, others are influenced only by varying magnetic fields.

For example, a cathode-ray tube that is in a strong fixed electromagnetic field will be affected by the field. The field will have an influence on the direction that the electron beam moves down the tube. Thus usually provisions are taken to shield the tube both from fixed and varying electromagnetic fields.

Varying electromagnetic fields can induce a voltage in a conductor. If the field happens to be from a lead carrying 60-cycle power and the lead is in the first stage of the audio amplifier in a radio or television receiver, sufficient hum voltage may be picked up or induced in that lead and then amplified by the following stages in the receiver so that an objectionable hum will be produced. In the case of i-f transformers in a radio or television receiver, energy could be fed from one i-f transformer to the other through the electromagnetic field and this could induce a voltage in the second transformer and cause feedback in the stage so that the stage would be unstable or may break into oscillation and be of no use whatsoever. Once again, shielding is usually used in order to prevent this from happening.

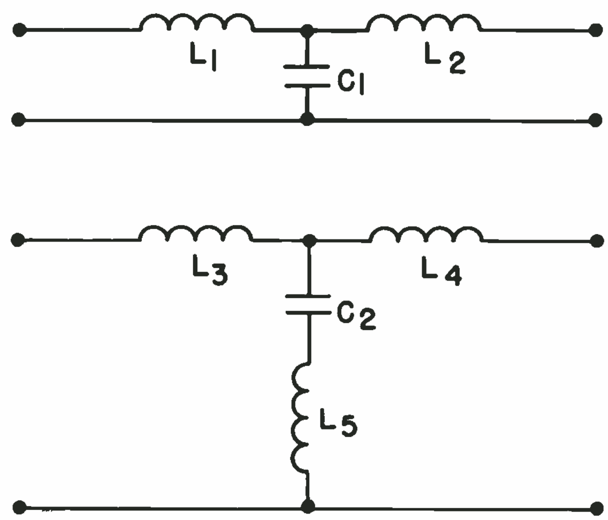
161. What factors determine the amplitude of the emf induced in the conductor which is cutting magnetic lines of force?

The emf induced in a conductor cutting magnetic lines of force will depend upon the number of lines of force the conductor is cutting and the rate at which the conductor cuts them. The more lines of force the conductor is cutting, the higher the voltage that will be induced in it. The faster the field is cutting the lines of force, the higher the emf that will be induced in it.

162. Explain the theory of molecular alignment as it affects magnetic properties of materials.

The atoms of materials are influenced by strong external magnetic fields. When placed in a strong field, the atom of a material itself may become magnetized. Groups of atoms are magnetized and aligned to form small areas called domains which are themselves magnetized. If in the material these domains are close enough together so that the small magnets produced by the domains add together the material is classified as a magnetic material. If the domains are spaced so that the small magnetic fields produced by each do not add, then the materials are non-magnetic materials. Some magnetic materials retain their magnetism for a long time, others will lose it as soon as or shortly after the external magnetic field is removed. Material such as alnico which is used in the manufacture of loudspeakers, is an example of a material that retains its magnetic alignment for a long time after the external field that magnetized it is removed.

163. Draw a circuit diagram of a low-pass filter composed of a "constant-K" and an "M-derived" section.



The filter made up of L_1, L_2 and C_1 forms what is known as a T type, constant-K filter. This type of filter is called a constant-K filter because the product of X_L and X_C is constant for all frequencies. If the inductance and capacitance values are modified so an additional inductance can be added we have the second filter consisting of L_3, L_4, C_2 and L_5. The second filter is a derived filter. It is derived from the basic T-type constant-K filter.

The characteristic of the M-derived filter is that it will have a very high attenuation at a specific frequency. At some frequency C_2 and L_5 will form a series-resonant circuit. At this frequency the attenuation will be the highest. The sharpness of the attenuation curve for this type of filter will depend on a factor called the M factor which can be determined by a formula. The formula can be obtained from any handbook; it is not necessary for you to remember it. However, the fact that the filter is derived from the constant-K T-type filter and the M factor affects the sharpness of the curve is where we get the name M-derived filter.

The constant-K section of the filter offers increasing attenuation as the frequency increases. The reactance of the two coils L_1 and L_2 increases and the capacitive reactance of the capacitor decreases as the frequency increases, therefore this type of filter will pass low-frequency signals but attenuate high-frequency signals. Hence it is called a low-pass filter. In the M-derived filter, the reactance of L_3 and L_4 increases as the frequency increases, hence it also is a low-pass filter, with a very high attenuation at the frequency at which L_5 and C_2 form a series resonant circuit.

164. In general, why are filters used? Why are "band-stop", "high-pass", and "low-pass" filters used? Draw a schematic diagram of the most commonly used filters.

Filters are used to block signals of certain frequencies and permit signals of other frequencies to pass through. For example, a filter may be used in the output of a transmitter to help eliminate second-harmonic radiation. The filter would be designed to pass the transmitter-fundamental frequency, but offer a high attenuation to the second harmonic.

Filters may also be used in receiving equipment. For example, a receiver designed to operate on very high frequencies might be located near a number of stations operating at lower frequencies. A filter designed to eliminate interference from the low-frequency signals might be used in the antenna circuit to prevent the strong low-frequency stations from causing cross modulation by overloading the i-f stage. The low-pass filters would still permit the higher-frequency signals to pass through to the rf stage.

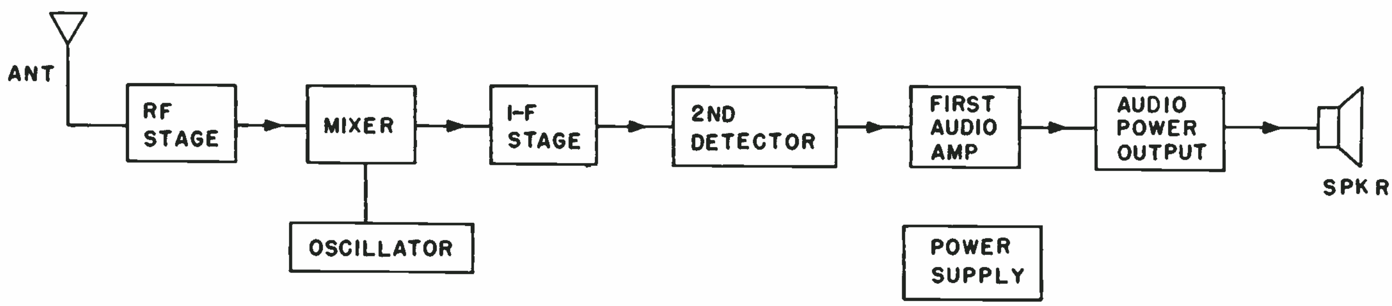
A band-stop filter is a filter designed to stop a certain band of frequencies. It will permit frequencies above the band-stop and below the band-stop to pass through with little or no attenuation. A band-stop filter designed to block signals near the i-f frequency of a receiver might be used in the antenna circuit to prevent i-f interference.

A high-pass filter is a filter that is designed to permit signals above a certain frequency to pass with little or no attenuation, but to offer a high attenuation to signals below this frequency.

A low-pass filter is a filter designed to permit signals below a certain frequency to pass through with little or no attenuation and to offer a high attenuation to signals above this frequency.

Review the section on filters from page 29 through page 44 of lesson 29CC. You should be able to draw from memory all the filters shown in Fig. 50 on page 42.

165. Draw a block diagram of a single-conversion superheterodyne AM receiver. Assume an incident signal and explain briefly what occurs in each stage.



The incoming signal strikes the antenna and causes a current to flow in the antenna. This signal is coupled to the primary winding of the rf coil and fed to the rf stage where it is amplified. The rf amplifier will have a resonant circuit in the input which will be tuned to resonance at the frequency of the incoming signal.

The amplified signal from the rf stage is fed to the mixer where it is mixed with a signal from the local oscillator. The local oscillator operates at a fixed frequency either above or below the frequency of the incoming signal. In most AM receivers it operates at a frequency above the incoming signal. In the mixer, the incoming signal beating with the signal from the local oscillator generates two new signal frequencies, one equal to the sum of these two frequencies and the other equal to the difference. In the output of the mixer circuit there is a transformer called the intermediate-frequency transformer which is tuned to the difference frequency. This new difference-frequency signal contains the original AM modulation on the incident signal.

The intermediate-frequency signal is amplified by the i-f stage and then fed to the second detector. In the second detector the modulation on the carrier is removed from the carrier and then the modulation signal, which will be the audio signal, is fed to the first-audio amplifier. This is a voltage amplifier that builds up the strength of the signal. From the first audio amplifier the signal is fed to the power amplifier stage and from there to the loudspeaker.

In addition to the circuit shown there will be a power supply in the receiver which supplies power to the various stages in the set.

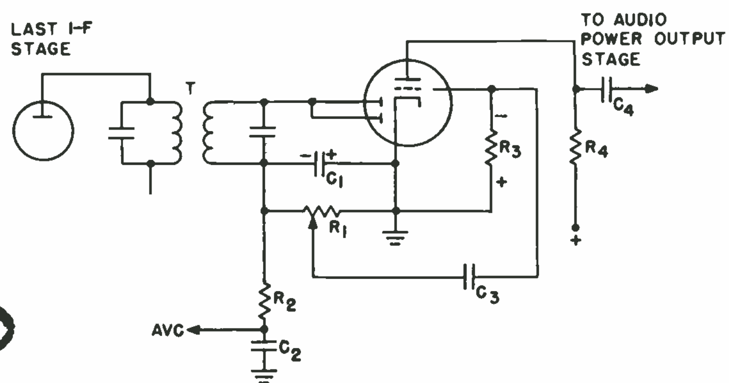
166. Explain the relation between the signal frequency, the oscillator frequency and the image frequency in the superheterodyne receiver.

In a superheterodyne receiver the oscillator is usually operated at a frequency above the frequency of the incoming signal. It beats with the incoming signal to produce a difference frequency which is the i-f frequency. As an example, suppose the superheterodyne is tuned to receive an incoming signal of 1000 kc. If the receiver has an i-f frequency of 500 kc then the local oscillator would be tuned to 1500 kc. The difference between the incoming signal of 1000 kc and the local oscillator signal of 1500 kc is 500 kc which is the i-f frequency.

However, if there also was a station operating on 2000 kc this would beat with the signal from the oscillator and the difference between these two signals would also be 500 kc. This signal would be amplified by the i-f amplifier as well as the desired signal. The 2000 kc signal is called an image signal. The image signal is a signal on a frequency equal to the i-f frequency plus the oscillator frequency. The desired signal is a signal equal to the oscillator signal minus the i-f signal.

167. Draw a circuit diagram of an AM second detector and af amplifier (in one envelope), showing avc circuitry. Also show coupling to and identifications of all adjacent stages.

(a) Explain the principles of operation.
(b) State some conditions under which readings of avc voltage could be helpful in troubleshooting a receiver.
(c) Show how this circuit would be modified to give davc.



(a) In the circuit shown the transformer T is the output i-f transformer. This transformer couples the signal from the last i-f stage to the diode detector. The tube used in the detector circuit is a duplex diode-triode type tube. This tube takes two diodes and a triode in the one envelope.

The two diode plates are connected together and connected to one side of the i-f transformer. When the signal polarity is such that the diode plates are positive, the diodes will conduct; current will flow through the cathode of the tube to the diode plates through the transformer and charge the capacitor C_1 with the polarity shown. The amplitude of the charge on C_1 will vary with the strength of the signal and of course in amplitude modulation the variation in strength of the signal is due to the amplitude modulation. Therefore we'll have an af voltage appearing across C_1 caused by the original modulation placed in the rf carrier.

R_1 provides a method for C_1 to discharge when the amplitude of the audio signal goes down. However, the time constant of R_1 and C_1 is selected so it is long compared to the individual rf cycles and therefore C_1 cannot discharge appreciably between the positive rf pulses to the diode. However, the time constant of the resistor-capacitor combination is short enough to be able to follow changes in charge across C_1 due to modulation changes.

R_1 also serves as a volume control. Part of the audio signal voltage developed across C_1 is fed through C_3 to the grid of the triode section of the tube. The triode section will amplify the signal. The audio signal will cause the current flowing through the triode to vary and this will cause a varying voltage to appear across R_4. This voltage is fed through C_4 to the following stage.

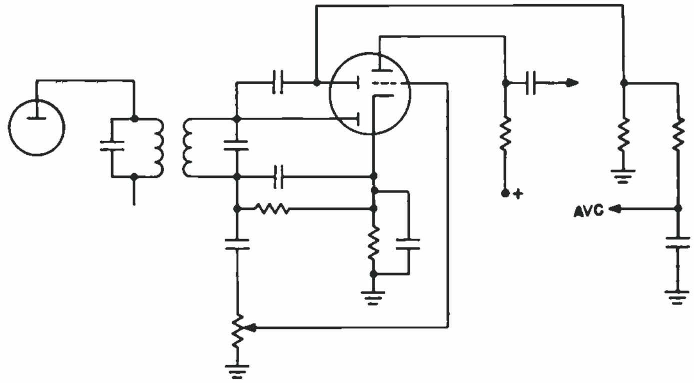
Bias for the triode section of the tube is developed across the resistor R_3. This type of bias is often referred to as convection bias. R_3 is a large value resistor and some of the electrons travelling from the cathode over towards the plate will accidentally strike the grid. These few electrons flowing through R_3 will develop a voltage across it having the polarity shown on the diagram and this voltage will bias the tube.

The varying avc voltage across C_1 will depend primarily upon the amplitude of the carrier. This voltage is filtered by the R-C network consisting of R_2 and C_2 and used as the automatic volume control (avc) to control the gain of the receiver. If the overall strength of the signal being received increases, the average voltage across C_1 will increase. This voltage when filtered by R_2 and C_2 will cause the voltage across C_2 to increase. The increased negative voltage across C_2 is used to reduce the gain of the mixer and/or the i-f stages.

(b) AVC voltage readings can often be helpful in isolating the trouble in a receiver. For example, if there is no avc voltage developed then the chances are there is no signal reaching the second detector. On the other hand, if as you tune across the bank, you notice that the avc voltage varies as you tune through what you would expect to be stations, then the chances are that the receiver is working up to the second detector and a defect in the receiver is in the audio section.

If the receiver is intermittent, checking the avc voltage may help you to isolate where the intermittent defect is. For example, you tune in a station and then connect a voltmeter across the avc. If when the output from the receiver drops or disappears, the avc voltage remains constant, then the defect is at some point after the point at which the avc voltage is developed. On the other hand, if the avc voltage drops when the signal drops then you should look for trouble in the rf common mixer or i-f stages or in the avc circuit itself.

(c) DAVC is the abbreviation for delayed avc. Delayed avc is automatic volume control that does not begin to work until the strength of the received signal exceeds a certain amplitude. The circuit shown takes advantage of the voltage developed in the cathode circuit to delay the production of avc. Notice that the two diodes are separated by means of a capacitor. The one diode serves as the second detector. The other diode that fed the i-f signal through the capacitor is the avc diode. In the circuit shown notice that the avc diode resistor returns to ground. Therefore as far as the diode plate is concerned, the cathode will be several volts positive. The signal voltage fed to the diode must overcome this positive voltage on the cathode before the avc



diode will conduct and produce any avc voltage.

168. Discuss the following items with respect to their harmonic attenuation properties, as possibly used in a transmitter or receiver.

(a) Link coupling
(b) Tuned circuits
(c) Degree of coupling
(d) Bias voltage
(e) Decoupling circuits
(f) Shielding

(a) Link coupling is frequently used between the stages of a transmitter. Coupling by means of a link between two resonant circuits effectively isolates the circuits and keeps the harmonic radiation at a minimum. Very little harmonic energy is induced into the link because the link provides a low-impedance path only at the frequency at which the two resonant circuits are tuned. Similarly, there is very little harmonic energy induced from the link into the second resonant circuit. Thus harmonic suppression is obtained both in inducing very little harmonic energy from the first harmonic circuit into the link and from the link into the second resonant circuit. Link coupling is used in transmitters rather than in receivers.

(b) Harmonic attenuation is a direct function of the number of tuned circuits. The more tuned circuits there are tuned to the fundamental frequency, the greater the harmonic attenuation will be. If a fundamental signal and a harmonic signal of equal amplitude are fed to a resonant circuit where the attenuation of the harmonic signal is such that it is reduced to one quarter of that of the original harmonic signal, and then the two signals are fed to a second resonant circuit and the harmonic signal again reduced by one quarter, the resultant harmonic amplitude would be reduced to 1/16 the strength of the original at the output of the second resonant circuit. If a third resonant circuit followed, and the same reduction obtained, the harmonic would be reduced to 1/64 the strength of the fundamental signal at the output of the third tuned circuit.

(c) The degree of coupling between resonant circuits has an effect on the harmonic transfer. Generally speaking, the looser the coupling, the greater the harmonic attenuation. Where there is tight coupling between circuits and where the circuits are overcoupled, there is not nearly as much harmonic attenuation as in the case of loosely coupled circuits.

(d) Bias voltage. The bias at which a Class C stage is operated has a direct bearing on the amount of harmonics generated by the stage. In general, if you want to operate a Class C stage as a frequency doubler or tripler, you operate it with a very high bias and drive the stage with a strong driving signal. This results in short sharp pulses in the Class C stage which are very conducive to harmonic generation. Therefore to attenuate harmonic generation in a Class C stage you should operate it at as low a bias as possible. Generally speaking, the bias voltage should be two to three times cut-off for normal Class C operation. Operating a Class C stage with a higher bias will result in an increase in harmonic generation.

In receiver circuits the bias on the rf and i-f stages should be controlled by an automatic volume control system. Also the tubes or transistors in the receiver should be operated in the center portion of the characteristic curve so that linear operation may be obtained. Strong signals that drive the input stages or the i-f stages into the nonlinear region will produce harmonics. In some cases, of course, where the receiver is close to a strong transmitter a certain amount of overloading and harmonic generation is difficult to avoid. In such instances the use of a wave trap tuned to the frequency of these stations will reduce the amount of signal reaching the antenna terminals of the receiver and cut down on harmonic generation.

(e) Decoupling circuits are used both in transmitters and receivers to prevent rf signals from being fed through the power supply from one stage to another. Decoupling circuits are particularly important in receivers where strong signals from the i-f stages may be fed back to the rf stages resulting in harmonics from the i-f stages reaching the rf stages. Decoupling circuits are also used in transmitters to prevent high energy signals from the output stages getting back into the low level stages of the output transmitter.

(f) Shielding is widely used both in receivers and transmitters to prevent harmonics from one stage from travelling to another stage and also to prevent feedback of the fundamental signal from one stage to another. The i-f transformers and receivers are shielded, often the various rf and mixer coils are shielded or shielded from each other.

In transmitters, the various rf power generating stages are shielded to prevent harmonic radiation directly from the resonant circuit in the transmitter. By shielding the stages the harmonic radiation, along with the fundamental radiation can be kept at a minimum. Then the desired signal is taken off through appropriate coupling circuits and any filters that may be needed to reduce harmonic radiation can be incorporated into the coupling circuits.

National Radio Institute

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NRI Study Guide for FCC License Examination

PART VIII

The material in this part of the study guide is meant to supplement the material you have already studied in your test. Be sure you study this part of the study guide carefully as soon as you receive it. After you have studied the study guide, answer the questions in the examination TC-8. The material covered in this part of the study guide along with the lessons you have studied should enable you to answer all of the questions in TC-8. The questions asked in this examination are similar to those you are likely to be required to answer when you take your Radiotelephone Operator's License Examination.

Be sure to spend any additional time you might need studying this part of the study guide or your lesson text if you find that some of the material is giving you difficulty. Keeping up with your course is the best way of being sure that you will be able to pass the FCC Examination and get your license. If you pass over small sections now that you do not understand completely, it is likely that these difficulties will cause you problems in later lessons.

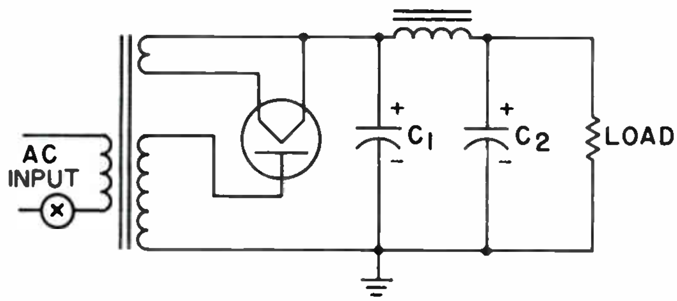
Be sure to save this part of the study guide along with those you have already received, so that you will be able to make a complete review before going for your license.

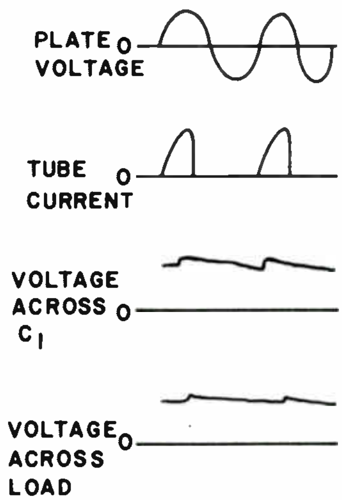
169. Draw a diagram of each of the following power supply circuits. Explain the operation of each, including the relative input and output voltage amplitudes, waveshapes, and current waveforms.

    (a) Vacuum-tube diode, half-wave rectifier with a capacitive-input "pi-section" filter.
    (b) Vacuum-tube diode, full-wave rectifier with choke input filter.
    (c) Silicon diode, doubler-circuit rectifier with a resistive load.
    (d) Nonsynchronous-vibrator power supply, with silicon diode, bridge-circuit rectifier and capacitive input "pi-section" filter.
    (e) Synchronous-vibrator power supply with capacitive input "pi-section" filter.

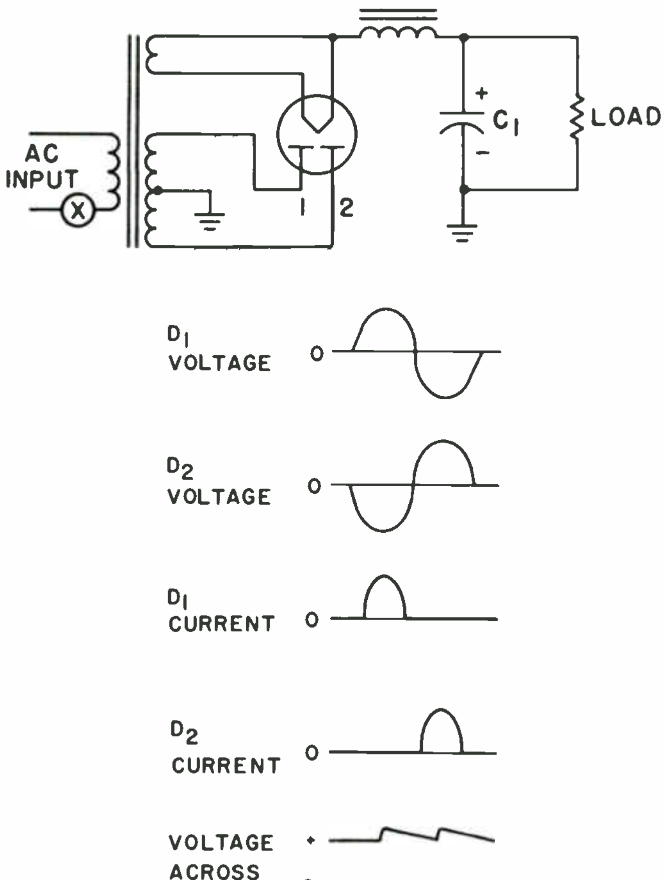
The schematic diagram of a half-wave rectifier with a capacitive-input pi-section is shown. In a rectifier circuit of this type, the diode conducts for less than half a cycle. It conducts when its plate is positive and the ac voltage is greater than the dc voltage built up across the input filter capacitor. The ac voltage applied to the plate of the diode is a sine wave. The current through the diode is shown, and as you can see current flows in a series of pulses during the positive half of the ac cycle.

When current flows through the diode, capacitor C_1 is charged to a value almost equal to the peak of the ac voltage. When the diode stops conducting, the capacitor discharges through the filter choke and through the load. The dc voltage across the output filter capacitor C_2 is much more constant than the voltage across C_1. This is due to the additional filtering action produced by C_2 and the filter choke.





169(A)



169(B)

The schematic diagram of a full-wave vacuum-tube rectifier using a choke input filter is shown. In this circuit we have used a single dual diode as the rectifier. The rectifier tube has two separate plates which act as two separate diodes. When the ac from the transformer secondary makes diode plate 1 positive, current will flow through this diode and not through the other diode.

During the next half-cycle, when the polarity is reversed, current will flow through diode 2 and not through diode 1. Actually, two separate diode tubes could be used in this power supply. In high-voltage power supplies found in transmitters you will usually find two separate diode tubes used rather than a dual diode as shown.

Since the current flows through first one diode and then the other, we have a series of 120 p.p.s. from the output of the rectifier tube. The peak amplitude of these pulses is limited by the filter choke so that the peak current flow through the diodes is not as high as in the case of the capacitor input filter. The filter choke and the capacitor C_1 filter the pulsating dc from the rectifier tube so that we have essentially a constant dc voltage across the load.

One of the characteristics of a choke input-type filter is that the output voltage is somewhat lower than could be obtained with a capacitive input filter. However, the voltage regulation obtained with a choke-input filter is usually better than that which can be obtained with a capacitive input filter.

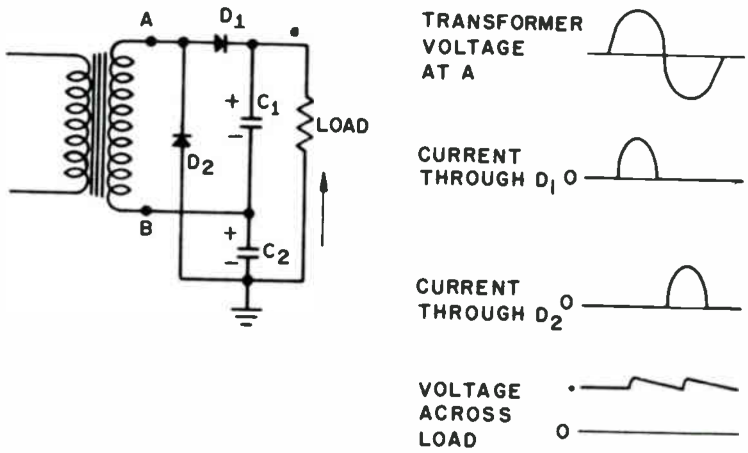
Two schematic diagrams showing voltage-doubler circuits using silicon diodes are shown. The circuit shown at A is a full-wave voltage-doubler, whereas the one shown at B is a half-wave doubler. In the full-wave voltage-doubler circuit there will be 120 current pulses per second fed to the load, whereas in the half-wave doubler circuit there will be only 60 pulses fed to the load.

Needless to say, the half-wave doubler output is harder to filter, but the circuit is somewhat more flexible inasmuch as one side of the transformer secondary can be connected directly to B-.

This type of power supply can also be used in transformerless equipment. In other words, the power line can be connected directly to the voltage-doubler rectifier circuit. In a full-wave circuit, such as that shown in A, one side of the power line cannot be connected to ground and this type of power supply is usually used only in circuits where a transformer is used. The waveforms showing the current through the diodes in both supplies plus the voltage waveform across the load for each power supply is shown.

In the full-wave rectifier circuit, when terminal A of the transformer secondary is positive and terminal B negative, electrons flow from terminal B into the negative plate of C_1, out of the positive plate and through D_1 back to terminal A of the secondary. Thus, C_1 is charged to a voltage approximately equal to the peak transformer secondary voltage. During the next half-cycle when terminal B is positive and terminal A is negative, electrons leave terminal A and flow through the diode D_2 into the negative plate of C_2. Electrons flow out of the positive plate and back to terminal B of the transformer.

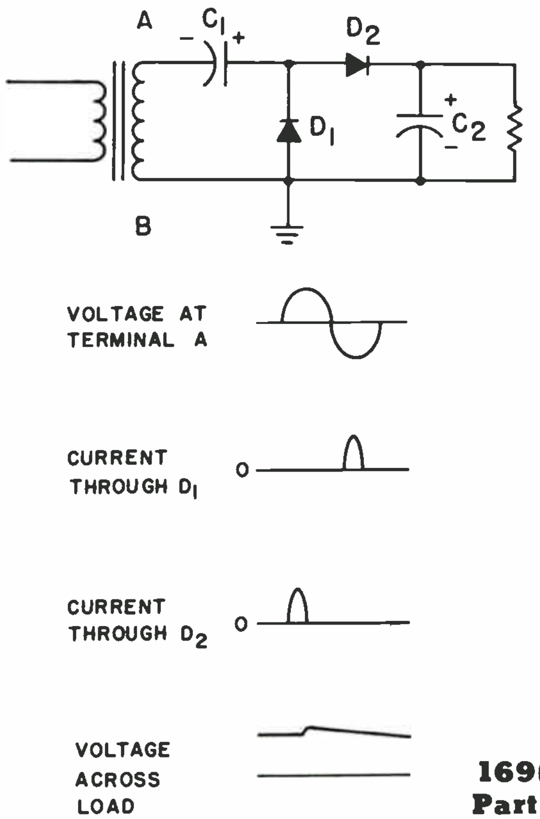
Thus, in this half-cycle the capacitor C_2 is charged to a voltage almost equal to the peak line voltage. Since capacitors C_1 and C_2 are connected in series, the total voltage across them is almost twice the peak transformer secondary voltage. Thus, the output from the transformer has been doubled. The exact value of the output voltage will depend on the capacity of C_1



169(C) Part A

and C_2 and on how much current is drawn from the power supply by the load. In supplying current to the load, electrons leave the negative terminal of C_2 and flow through the load into the positive terminal of C_1. This causes some electrons to leave the negative terminal of C_1 and flow into the positive terminal of C_2.

In the half-wave rectifier circuit, when terminal A of the transformer is negative and terminal B positive, electrons leave terminal A of the transformer and flow into the negative terminal of C_1. Electrons flow out of the positive terminal of C_1 and through D_1 back to terminal B of the transformer. On this half-cycle, C_1 is charged to a value equal to the peak line voltage with the polarity indicated. During the next half-cycle, terminal A of the transformer secondary is positive and terminal B is negative.



169(C) Part B

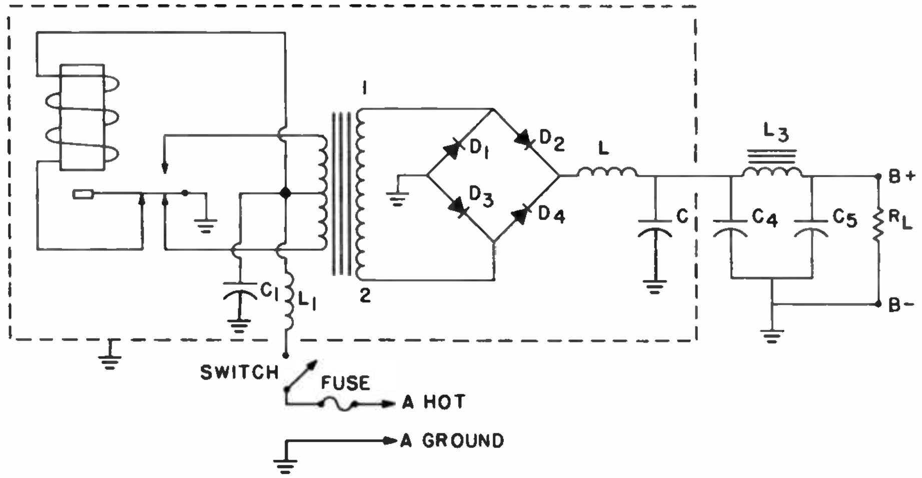
The transformer secondary voltage will be placed in series with the charge on C_1. Electrons will leave terminal B of the transformer and flow into the negative plate of C_2. Electrons will also leave the positive plate of C_2 and flow through D_2 to the positive plate of C_1. Since the total charging voltage is equal to the transformer secondary voltage plus the charge on C_1, C_2 will be charged to this total voltage, which will be approximately twice the peak value of the transformer secondary voltage.

Once again, the exact output voltage from the doubler will depend on the size of C_1 and C_2 and on the current taken from the supply by the load.

A schematic diagram of a nonsynchronous-vibrator power supply that has silicon diodes used in the bridge-rectifier circuit and capacitor input pi-section filter is shown on the next page. You will remember that a nonsynchronous-vibrator is a vibrator which is used with a separate rectifier. The vibrator is not self-rectifying.

In the power supply circuit shown, when terminal A of the secondary of the transformer is positive and terminal B is negative, electrons flow from terminal B through the diode D_3 to ground and then to the load. Current then flows through the load, through the filter choke L_3 and through the rf choke L, back through D_2 to terminal A of the transformer secondary.

During the other half-cycle, when terminal A of the secondary is negative and terminal B is positive, electrons will leave terminal A of the transformer and flow through D_1 to the negative side



169(D)

of the load. Electrons will then flow through the load and chokes and back through D_4 to terminal B of the transformer. The rf choke L and the capacitor C are not considered part of the pi-section filter. These parts are used to eliminate rf interference which is generated by the vibrator.

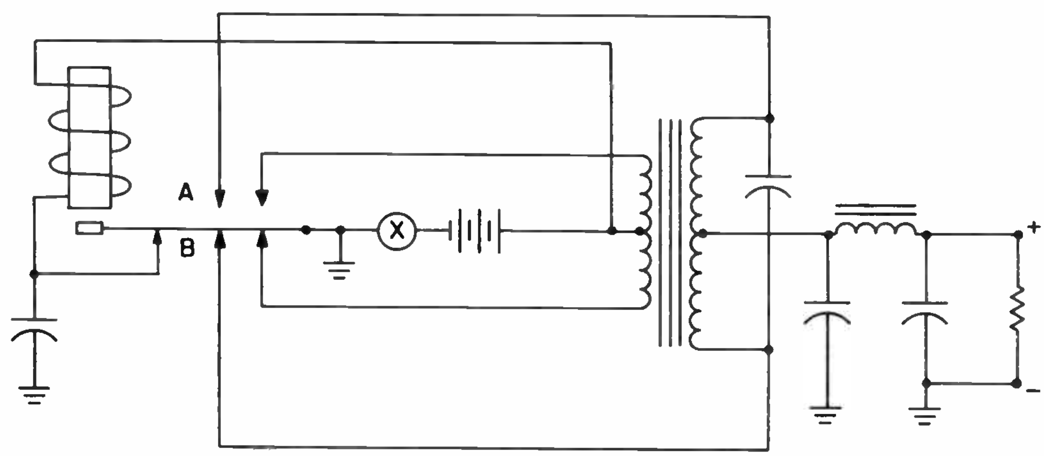
A synchronous-vibrator power supply with a capacitive pi-section filter is shown. The synchronous-vibrator is a self-rectifying vibrator. One set of contacts on the vibrator is used to operate the vibrator and the other set is used to rectify the ac produced in the secondary of the transformer.

When the vibrator is turned on, the coil is energized through the contact on the reed. This causes the reed to be attracted over towards the coil, making contact at terminal A, and the current to flow through one-half of the transformer primary, inducing a high voltage in the secondary. The voltage induced in the secondary will cause the current to flow from the transformer through the vibrator contact to ground.

Electrons then flow from ground through the load and through the filter choke back to the center tap of the power transformer secondary. During the next half-cycle when the contact through the vibrator coil is broken, the reed vibrates over to side B, closing the primary contact on this side, and current flows through the other half of the transformer primary.

Since it flows through the primary in the opposite direction, it induces a voltage of the opposite polarity in the secondary of the transformer. This causes a current to flow from the transformer through the other high-voltage rectifier contact to ground and then through the load and the choke back to the center tap of the high-voltage secondary winding.

Vibrator power supplies were widely used at one time in mobile equipment, but these supplies have been for the most part replaced in modern equipment by solid-state inverter-type power supplies. However, even though this type of power supply is obsolete, since the FCC may ask questions on it on your Radiotelephone Operator's



169(E)

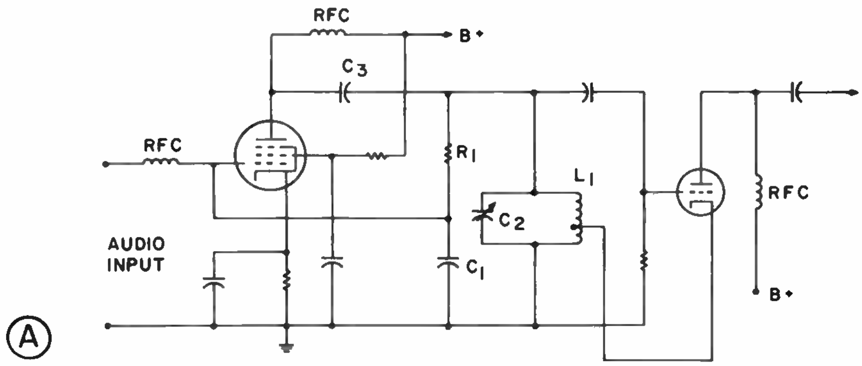
License Examination, it is important that you remember the diagrams of the synchronous and the nonsynchronous types of vibrator power supplies and how they work.

170. Draw a schematic diagram of a frequency-modulated oscillator using a reactance-tube modulator. Explain its principle of operation.

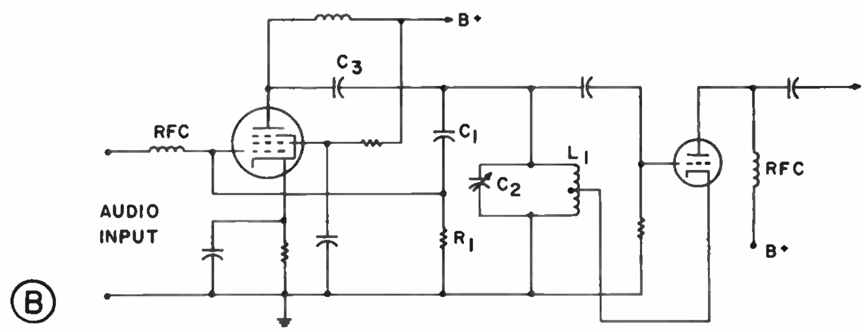
Two reactance-tube modulators are shown. The circuit shown at A adds inductive current to the oscillator circuit, whereas the one shown at B adds capacitive current to the oscillator circuit.

Decreasing the inductance in the circuit will increase the frequency; increasing the inductance will reduce the frequency.

Similarly, if we change the capacitor current, we can change the effective capacity across the tank circuit. If we increase the capacitive current, we will have the effect of increasing the capacity in the circuit in order to make the capacitive reactance lower. Increasing the capacity in the circuit will reduce the oscillator frequency. Similarly, if we reduce the capacitor current, this will have the effect of increasing the capacitive reactance, reducing the capacity in the circuit.



170(A),(B)



In an oscillator circuit, the frequency is determined by the inductance of the oscillator coil and the capacitance of the capacitor that is connected across this coil. In both circuits the coil is labelled L_1 and the capacitor C_2. In this oscillator circuit a certain current will flow through the coil and through the capacitor.

If by an external means, such as a reactance-tube modulator, we can make the current through either the coil or the capacitor change, this has the effect of changing the net value of the inductance or the capacitance in the circuit. For example, if the coil current increases, this has the effect of reducing the inductance of the coil; whereas if the coil current decreases, it has the effect of increasing the inductance in the circuit.

Reducing the capacity in the tank circuit will cause the oscillator frequency to increase.

In the circuit shown at A, we have a reactance-tube which produces a signal current in the plate circuit that will be in phase with the current in the tank coil L_1 and varies the inductive current. The oscillator tank voltage is fed to the network consisting of R_1 and C_1. The resistance of R_1 is made large in comparison to the reactance of C_1 so that the series circuit made up of R_1 and C_1 will act like a resistance, and the net current flowing through the network consisting of R_1 and C_1 will be in phase with the oscillator tank voltage. The current flowing through C_1 will produce a voltage across it which lags it by 90°. Therefore, we have a lagging voltage applied to the grid of the reactance-tube. This will cause a current through the reactance-tube which will lag the oscillator tank voltage by 90°. This current is fed back to the oscillator tank circuit through C_3, and since it lags the oscillator tank voltage it will act like an inductive current.

In the reactance-tube modulator, a fixed lagging current is fed from the reactance-tube back to the tank circuit. The amount of current fed by the reactance-tube back to the tank circuit will depend upon the dc bias on the reactance-tube. The oscillator circuit and the reactance-tube current are adjusted to operate the oscillator on the desired frequency. This is called the center frequency or resting frequency.

The oscillator can then be modulated by feeding an audio signal into the reactance-tube grid circuit. This signal is superimposed on the bias applied between the grid and cathode of the reactance-tube and causes the current through the tube to increase or decrease, depending on the polarity of the signal fed to the grid of the tube. If the reactance-tube current increases, then we have an increase in the coil current through the tank circuit. This has the effect of reducing the inductance in the tank circuit, and causes the oscillator to operate on a higher frequency. On the other hand, if the audio signal causes the current from the reactance-tube to decrease, then the current fed back to the tank circuit decreases and the coil current goes down. This has the effect of increasing the inductance in the tank circuit, causing the oscillator to operate on a lower frequency.

In the circuit shown at B, the reactance of C_1 is much higher than the resistance of R_1. Therefore, in a series circuit consisting of R_1 and C_1 we have, in effect, a capacitive circuit. This means that the current flowing through this circuit will lead the voltage by 90°. The oscillator tank voltage is applied across the combination of R_1 and C_1; therefore, we have a current flowing through this network that leads the oscillator tank voltage by 90°.

The current flowing through R_1 will produce a voltage across this resistor that is in phase with the current and, therefore, the voltage across R_1 will lead the oscillator voltage by 90°. This causes a current flow through the reactance-tube which will lead the oscillator voltage by 90°. This leading current is fed through C_3 back into the oscillator tank circuit. Since it is a leading current, it will add to the capacitor current.

If the signal volt ge applied to the grid of the reactance-tube causes the current through the reactance-tube to increase, the current through the oscillator tank capacitor will increase. The increased current has the effect of reducing the capacitive reactance. The capacitive reactance can be reduced by increasing the capacity. Therefore, we have the effect of the capacity increasing, which causes the oscillator frequency to go down. If the signal voltage applied to the grid of the reactance-tube causes the reactance current to decrease, then the reactance current fed back to the tank circuit goes down. This causes the current through the oscillator tank capacitor to decrease. This has the effect of increasing the capacitive reactance, which is the same as reducing the capacity across the oscillator tank capacitor. Reducing the capacity of the capacitor causes the oscillator frequency to increase.

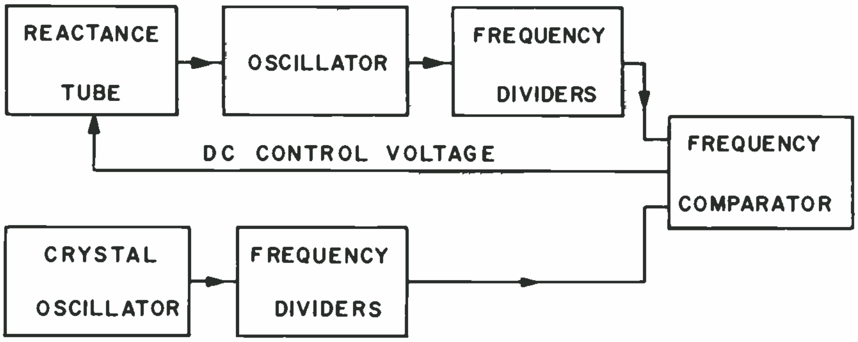
171. How is good stability of a reactance-tube modulator achieved?

The stability of a reactance-tube modulator is usually controlled by an automatic frequency control system. The usual procedure is to feed the output of the oscillator to a divider chain which reduces the signal frequency. The output from the divider chain is fed to a phase comparator. At the same time a highly stable crystal oscillator is operated in the transmitter. The output from this oscillator is fed to a divider and from there to the frequency comparator. The frequency comparator detects the difference in frequency between the two signals fed to it, and in the event of a frequency difference, develops a dc correction voltage which is fed back to the reactance modulator stage. This causes the current fed by the reactance-tube to the oscillator either to increase or decrease as necessary to correct the oscillator frequency.

A block diagram of the system used to obtain reactance-tube modulator stability is shown.

172. Discuss the following in reference to frequency modulation.

(a) The production of sidebands.
(b) The relationship between the number of sidebands and the modulating frequency.
(c) The relationship between the number of sidebands and the amplitude of the modulating voltage.
(d) The relationship between the percent of modulation and the number of sidebands.
(e) The relationship between modulation index or deviation ratio and the number of sidebands.
(f) The relationship between the spacing of the



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sidebands and the modulating frequency.

(g) The relationship between the number of sidebands and the bandwidth of emission.

(h) The criteria for determining bandwidth of emission.

(i) Reasons for pre-emphasis.

(a) The number of sidebands produced in frequency modulation depends upon the deviation and on the modulating frequency. Any modulating signal produces at least two sidebands, one above the carrier and one below the carrier. However, if the deviation is insufficient, additional sideband pairs are produced. The actual number of sideband pairs that will be produced depends upon the modulation index. The modulation index, M, depends on the deviation in frequency and the modulating frequency. The modulation index is given by the formula:

$$M = \frac{\text{deviation}}{\text{modulating frequency}}$$

A modulation index of 1 will mean that one pair of sidebands is produced. A modulation index of 2 indicates there will be two sideband pairs, two above the resting frequency and two below the resting frequency.

(b) The number of sidebands produced is influenced by the modulating frequency, as you can see from the formula given for the modulation index. The lower the modulating frequency for a given deviation in frequency, the greater the number of sidebands that will be produced. Conversely, the higher the modulating frequency for a given frequency deviation, the fewer number of sidebands that will be produced.

(c) The frequency deviation of an FM transmitter is determined by the amplitude of the modulating frequency. Therefore, the higher the amplitude of the modulating frequency, the greater the deviation in frequency.

By referring to the formula for the modulation index, you will see that the greater the deviation in frequency, the higher the modulation index and the greater the number of sideband pairs produced. Thus, a high amplitude audio signal will produce more sideband pairs than a low amplitude audio signal of the same frequency.

(d) The percentage modulation in an FM system indicates the amount of deviation of the oscillator carrier. For example, in FM broadcasting, the maximum deviation permitted by the FCC is 75 kc. Therefore, at 100% modulation the carrier is swinging 75 kc above and 75 kc below the resting frequency. In FM sound used with television, the maximum deviation permitted is 25 kc. Therefore, in this system when we say that the percentage modulation is 100%, we mean that the signal is deviating 25 kc above and 25 kc below the resting frequency.

Since the modulation index depends on the deviation in frequency, the higher the percentage modulation, the greater the number of sideband pairs that will be produced.

(e) The modulation index is a direct indication of the number of sidebands produced. The higher the modulation index, the greater the number of sideband pairs produced.

(f) The spacing between the sidebands is a direct function of the modulating frequency. The higher the modulating frequency, the greater the spacing between the sidebands. For example, with a 500-cycle modulating frequency, there will be a sideband 500 cycles above the resting frequency and one 500 cycles below the resting frequency. Other sidebands will appear at harmonics of 500 cycles. In other words, second harmonics will produce sidebands 1000 cycles above the resting frequency and 1000 cycles below the resting frequency. Thus, the spacing between the sidebands will be 500 cycles.

On the other hand, if the modulating signal was a 1000-cycle signal, then the first sidebands would appear 1000 cycles above and 1000 cycles below the resting frequency. The second sidebands would appear 2000 cycles above and 2000 cycles below the resting frequency. Thus, in this case, the spacing between the sidebands would be 1000 cycles.

(g) The greater the number of sideband pairs, the wider the bandwidth of emission. For example, if the modulating signal is 1000 cycles and three sideband pairs are transmitted, then we'll have sidebands 1000 cycles, 2000 cycles, and 3000 cycles above and below the resting frequency. Thus, a total bandwidth of 6000 cycles would be required. If four sideband pairs are produced, then we would have sidebands at 1000 cycles, 2000 cycles, 3000 cycles, and 4000 cycles above and below the resting frequency. In this case, a total bandwidth of 8000 cycles would be required.

(h) The criteria for determining the bandwidth of emission is the maximum permissible deviation and the highest modulating frequency to be transmitted. Obviously, the greater the deviation that is permitted, the wider the bandwidth will be. In addition, the higher frequencies produce sidebands further from the resting frequency than the lower frequencies modulation provides. Hence, high modulating frequencies require a wider bandwidth. Obviously, FM broadcast stations, which are permitted a maximum deviation of 75 kc and also transmit comparatively high modulating signal frequencies in order to rebroadcast high-fidelity music, require a much wider bandwidth than mobile FM communication systems that are limited to a deviation of 5 kc. Furthermore, the FM communication systems which are used to transmit only voice frequencies, transmit comparatively low modulating frequencies, and this, along with the narrow deviation permitted, contain the FM signal in a comparatively narrow bandwidth.

(i) Pre-emphasis is used to get the best possible signal-to-noise ratio from the FM system. An FM system is much less susceptible to noise interference than an AM system. However, the noise rejection characteristics of an FM system are best at low modulating frequencies. By pre-emphasis, the higher modulating frequencies are permitted to produce a greater deviation than equal amplitude low modulating signals produce. This pre-emphasis enables us to overcome a less favorable signal-to-noise ratio than would be obtained at the higher modulating frequencies without pre-emphasis. In a receiver, a de-emphasis network which restores the different frequency components of the modulating signal to their original amplitude compensates for the pre-emphasis used at the transmitter.

173. What might be the effect on the transmitter frequency if a tripler stage in an otherwise perfectly aligned FM transmitter was slightly detuned?

If the tripler stage is slightly detuned, the output from the stage may be slightly below normal. This could result in a somewhat inadequate drive to the final amplifier in the output of the transmitter, but if the stage is only slightly detuned, it should have no effect on the transmitter frequency. The frequency of the transmitter is controlled primarily by the oscillator stage, and as long as it is operating on the correct frequency, the slightly detuned tripler should have no effect on the transmitter frequency.

The only way that the tripler stage could affect transmitter frequency would be for it to be detuned substantially so that instead of operating as a tripler, it operated either as a doubler or a quadrupler. In this case, the final amplifier stage would be operated on the wrong frequency. However, this would require a substantial detuning of the tripler to accomplish this.

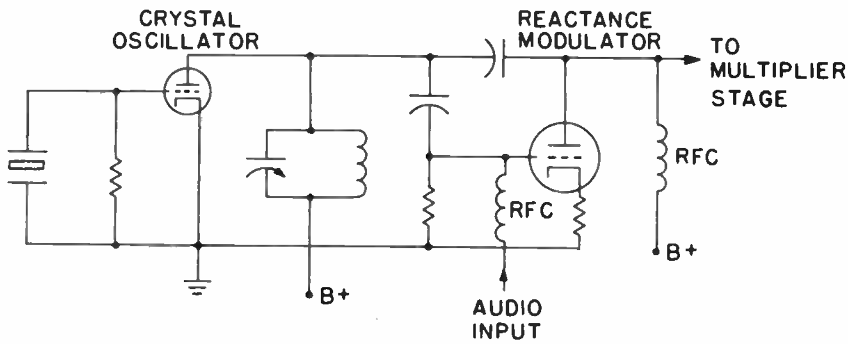
174. Discuss wide-band and narrow-band reception in FM voice communication systems with respect to frequency deviation and bandwidth.

A wide-band FM system is a system in which a wide frequency deviation is permitted and a narrow-band system is a system in which the frequency deviation is limited to a comparatively narrow deviation. Obviously, the wide-band system with the wider frequency deviation requires a wider bandwidth than the narrow-band system.

The advantage of the wide-band system is that it is more noise-free than the narrow-band system. However, with the tremendous demand for communication services, most voice systems use the narrow-band systems in order to conserve space in the frequency spectrum and permit the operation of more stations. The bandwidth required can also be reduced by limiting the modulating frequencies. In voice communications, only comparatively low-frequency audio signals need be transmitted. Hence, the modulating frequency is limited, which further conserves space in the frequency spectrum by keeping the total bandwidth required by each station at a minimum.

175. Draw a circuit diagram of a phase modulator. Explain its operation. Label adjacent stages.

Phase modulation is produced by means of a reactance-tube connected across the tank circuit of a crystal oscillator. In the circuit shown, the signal fed to the multiplier stage following the



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oscillator is actually made up of two components, one signal being fed directly from the crystal oscillator and the other from the reactance-tube.

The relationship between the two signals, E_O from the oscillator and E_R from the reactance tube, can best be shown by vectors. As you can see, in the vector diagram at A, the two signals add, producing the signal E, which is fed to the following multiplier stage.

You can see that the phase relationship between the signal from the oscillator and the signal actually fed to the multiplier stage depends upon the amplitude of the signal from the reactance modulator. When the audio signal swings the grid of the reactance tube in a positive direction so that the current through the stage increases, the current through the reactance stage will increase. This will advance the vector representing the output signal shown at B so that there will be an increase in the number of cycles occurring in a given period of time. The result is a small increase in output frequency.

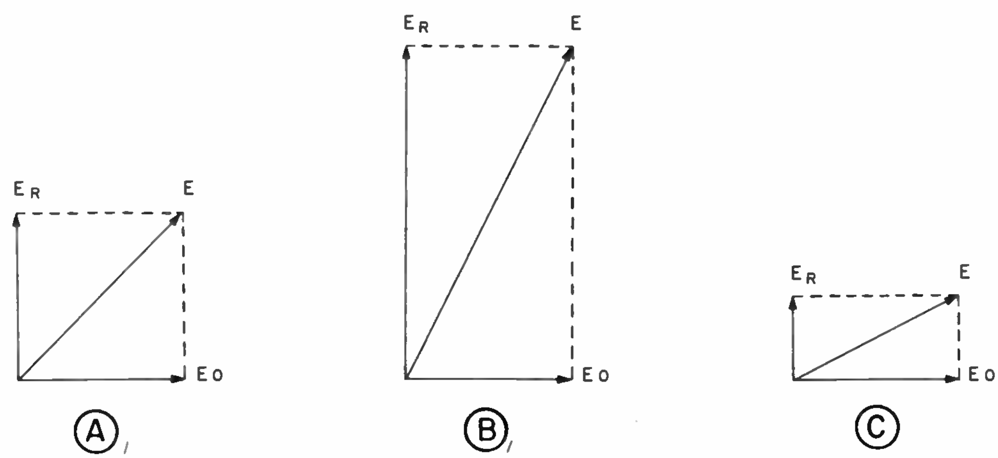
In the vector diagram at C, when the audio modulating signal swings the grid of the reactance tube in a negative direction and the current through this tube goes down, the output voltage vector E swings in a clockwise direction, representing a longer period of time. Thus, in a given interval, fewer cycles would occur and, in effect, the frequency of the oscillator will decrease.

The amount of frequency shift that can be obtained with a phase modulator is quite limited. However, by feeding the signal to a number of frequency multiplier stages an appreciable frequency deviation can be obtained. This type of frequency modulation is referred to as the indirect method.

176. Explain, briefly, what occurs in a waveform if it is phase-modulated.

When a waveform is phase-modulated, the frequency varies because the phase of the signal is alternately advanced and retarded. In the illustration shown on the next page you can see that when the grid of the reactance modulator swings in a positive direction and the current through the reactance-tube increases, the number of cycles occurring in a given period of time increases slightly, and the frequency of the wave increases. Conversely, when the audio signal drives the modulator grid in a negative direction, the signal slows down so that we'll have fewer complete cycles in a given period, and the frequency of the rf signal decreases.

177. Explain, in a general way, why an FM deviation



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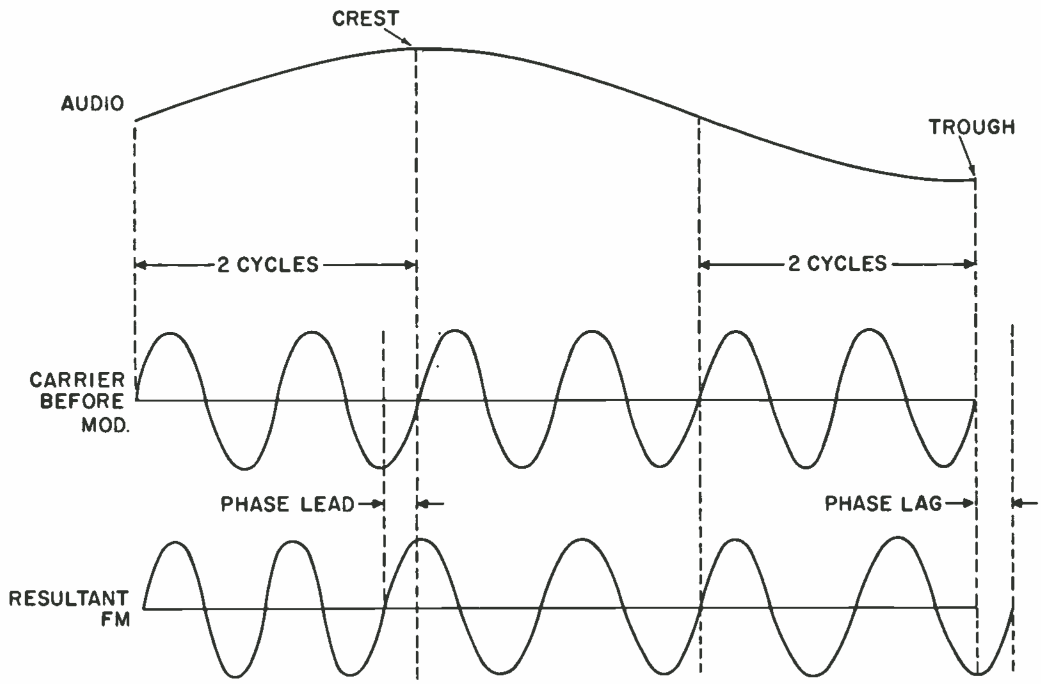


176

meter (modulation meter) would show an indication if coupled to the output of a transmitter which is phase-modulated by a constant amplitude, constant audio frequency. To what would this deviation be proportional?

When the oscillator in a transmitter is phase-modulated, a certain amount of frequency deviation is produced. By multiplying the oscillator frequency many times, this deviation is increased so that at the transmitter output we have an appreciable frequency variation. Thus, we have an FM signal and the frequency deviation meter will respond to this frequency modulation. In the case of an FM transmitter, the amount of deviation will be proportional to the amplitude of the modulating signal. Therefore, the higher the amplitude of the signal, the greater the deviation.

178. Draw a circuit diagram of each of the following stages of a phase-modulated FM transmitter. Explain their operation. Label adjacent stages.

(a) Frequency multiplier (doubler) with capacitive coupling in input and output.
(b) Power amplifier with variable link coupling to the antenna. Include circuit for metering the grid and plate currents.
(c) Speech amplifier with associated pre-emphasis circuit.

(a) A schematic diagram of a frequency doubler with capacitive coupling in the input and output is shown at A. As you can see, the doubler is no different from doublers you studied earlier. The grid resistor is selected so that the grid bias, developed by the class C stage, will be 4 to 10 times the cut-off bias. This will result in ample harmonics being produced in the output circuit. The fundamental signal is fed into the grid circuit. The tank circuit, in the plate circuit of the multiplier, is tuned to twice the incoming signal frequency.

(b) The power amplifier used in the FM transmitter is a typical class C power amplifier. A variable link coupling is shown connected to the output tank circuit. By adjusting this coupling, the loading on the final amplifier stage can be varied. Meters are placed in the grid and plate circuits to monitor grid and plate current. Both meters are bypassed by suitable capacitors to keep rf signals out of the meters.

(c) The first two stages of the speech amplifier are shown. The pre-emphasis circuit is made up of C_1 and R_1. The capacitor C_1 is chosen so that it has a relatively high reactance at the lower audio frequencies. Thus, lower frequency signals will be attenuated considerably. On the other hand, the capacitor will have a low reactance to the higher signals, so that they will be fed from the plate of V_1 to the grid of V_2 with little or no attenuation. Thus, the higher frequency audio signals will receive more amplification from the speech amplifier than the lower audio frequencies. As mentioned previously, in the FM receiver and in the detector output, a de-emphasis circuit is used to compensate for pre-emphasis applied in the transmitter.

179. Could the harmonic of an FM transmission contain intelligible modulation?

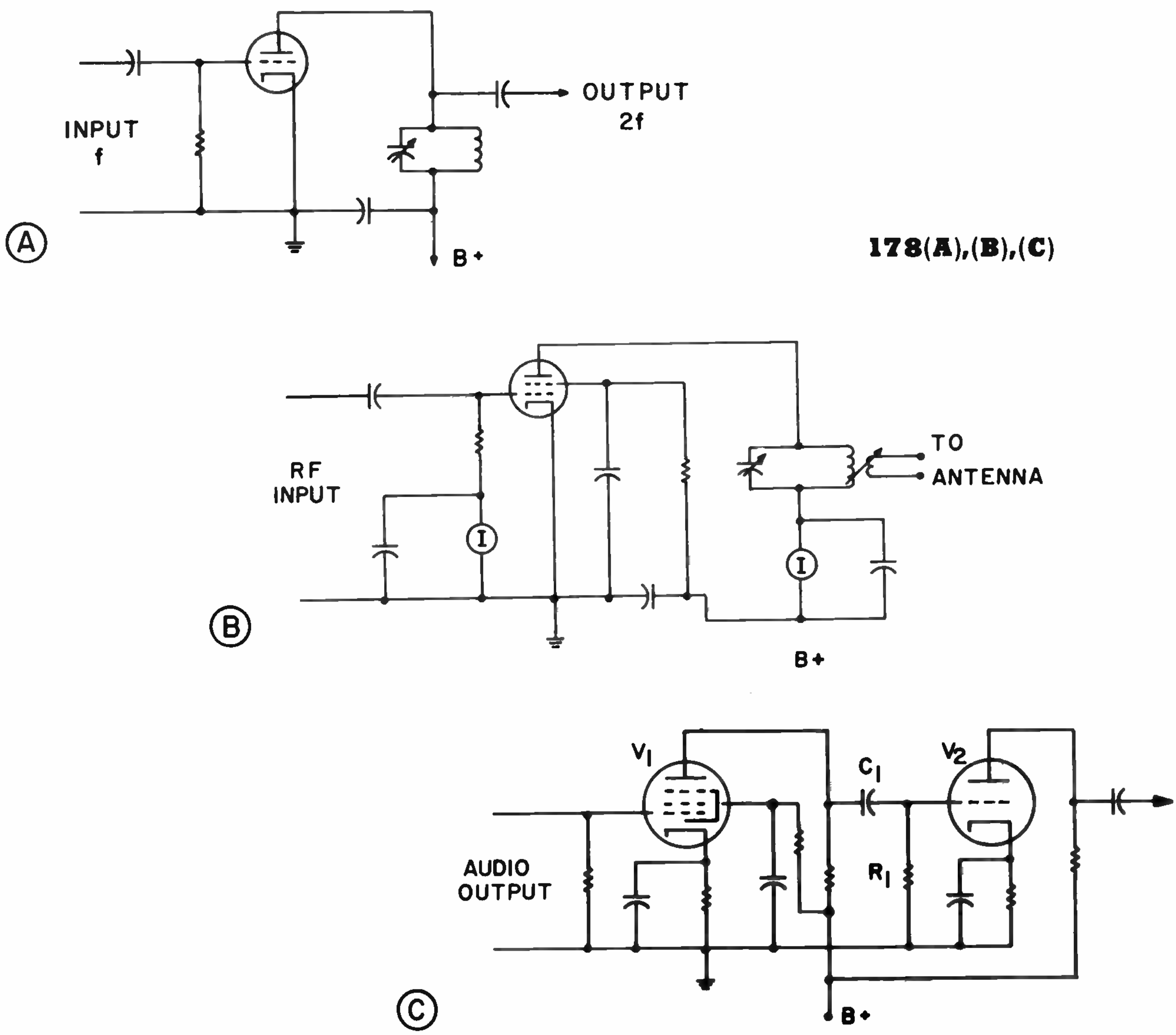
Yes. The modulating signal varies the rate at which the modulating signal varies above and below the resting frequencies. The amount that the signal is deviated depends on the amplitude of the signal. Therefore, the harmonic of an FM signal would simply have a greater deviation than the fundamental because the frequency has been doubled or tripled. However, the rate at which the signal varies above and below the resting frequency does not change, and hence, the modulation on the harmonic will be intelligible.

180. Under what usual conditions of maintenance and/or repair should a transmitter be re-tuned?

When a tube or transistor in the oscillator stage, one of the frequency multipliers or final amplifier stages is replaced, the chances are that the transmitter will require retuning. Retuning will also be required if a part in one of the frequency determining circuits is replaced. Furthermore, the transmitter should be retuned when measurements indicate that it is not operating according to FCC technical standards or that the efficiency of the transmitter is not as high as it should be.

181. If an indirect FM transmitter without modulation was within the carrier frequency tolerance, but with modulation out of tolerance, what might be some of the possible causes?

In the indirect FM system, a crystal oscillator is usually used, and the output phase-modulated by means of a reactance-tube modulator. Crystal



178(A),(B),(C)

oscillators are inherently very suitable so that it is unlikely that the defect is in the crystal. However, there are some circuits associated with the crystal oscillator that might be causing the trouble.

The trouble could be in the reactance modulator. A defect in the modulator may be loading the crystal oscillator circuit, causing the crystal frequency to vary. In addition, a defect in the speech amplifier could result in excessive audio signal being fed to the reactance modulator.

To get the stability required by FCC regulations, the power supply used in association with the crystal oscillator is usually a regulated supply. There may be some defect in the regulator. The audio signal may be causing the regulated voltage to vary in some way and this could cause the crystal oscillator to shift frequency. Feedback between the various stages in the transmitter might also be causing power supply variations. Feedback could be due to defective filter capacitors in the power supply and also due to a defective decoupling capacitor.

182. In an FM transmitter, what would be the effect on antenna current if the grid bias on the final power amplifier were varied?

If the rf excitation to the final power remains constant and the grid bias on the stage is increased, current will flow for a smaller part of each cycle, so the actual power delivered by the final output stage to the load will go down. In this case the antenna current will decrease. On the other hand, if the grid bias is reduced, the plate current will flow for a longer part of each cycle and the net power delivered to the load will increase. This will cause the antenna current to increase.

183. Assume you have the following instruments:

AC-DC VTVM
AMMETER
HETERODYNE FREQUENCY METER. (.0002% accuracy)
ABSORPTION WAVEMETER
FM MODULATION METER

Draw and label a block diagram of a voice-modulated (press-to-talk microphone), indirect (phase-modulated) FM transmitter having a crystal multiplication of 12.

(a) If the desired output frequency were 155.460 mc, what would be the proper crystal frequency?

(b) Consider the transmitter strip completely detuned; there are ammeter jacks in the control grid circuits of the multipliers and the control grid and cathode circuit of the final circuits of the final amplifier. Explain in detail, step-by-step, a proper procedure for tuning and aligning all the stages except the plate circuit of the final power amplifier (p.a.).

(c) Assume a tunable antenna with adjustable coupling to the plate circuit of the final p.a. With the ammeter in the cathode circuit of the p.a. and with the aid of a tube manual, describe a step-by-step method of obtaining maximum output power, without damaging the tube.

(d) If the p.a. in (c) above were a pentode, how would you determine the power input to the stage?

(e) In (c) above, how would you determine if the p.a. stage is self-oscillating; if so, what adjustments could be made?

(f) Assume the transmitter's assigned frequency is 155.460 mc, with a total tolerance of ± .0005%. What should be the minimum and maximum frequencies, as read on the frequency meter, which would assure the transmitter being within tolerance?

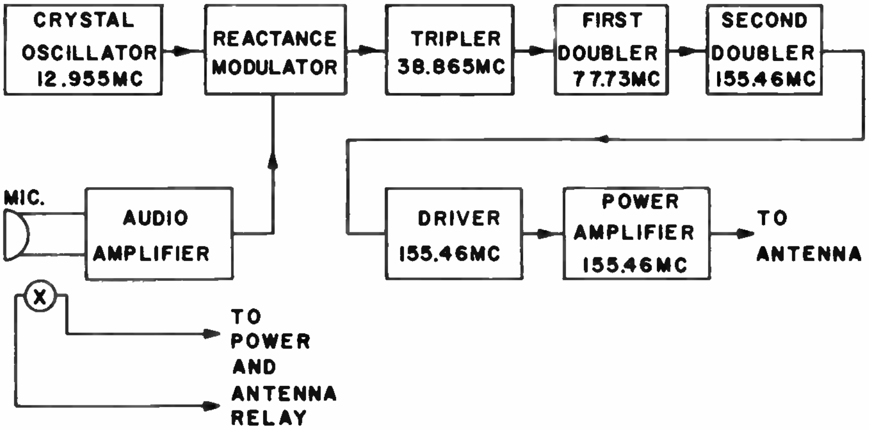
(g) Assume the 1 mc crystal oscillator of the frequency meter has been calibrated with WWV and that the meter is tunable to any frequency between each 1 mc interval over a range of 20-40 mc with usable harmonics up to 640 mc. Explain in detail what connections and adjustments would be made to measure the signal directly from the transmitter; also by means of a receiver.

(h) In checking the frequency deviation with the modulation meter, would you expect the greatest deviation by whistling or by speaking into the microphone with a low voice?

(i) If the transmitter contained a means for limiting and were overmodulating, what measurements and adjustments could be made to determine and remedy the fault?

(a) Since the crystal output must be multiplied twelve times to reach the desired operating frequency of 155.460 mc we can find the proper crystal frequency by dividing the output frequency by 12. 155.460 ÷ 12 = 12.955 mc.

(b) In some transmitters, provisions are made for reducing the plate (and screen) voltages on the power amplifier tube when the transmitter is being retuned. If the transmitter has such provisions, the voltages on the output tube should be reduced to prevent the current from rising to an excessive value during the tuning operation.



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To begin the actual tuning, the ammeter is inserted in the grid circuit of the tripler stage. Then any adjustments in the oscillator and/or reactance stage should be made. The adjustments are made to provide maximum current indication on the meter. Since the meter is in the grid circuit we want maximum reading; this indicates maximum drive to the tripler stage.

After the oscillator has been adjusted the ammeter should be moved to the grid circuit of the first doubler stage. When this is accomplished the tripler is adjusted for maximum indication on the meter. When you adjust the tripler, it may be possible to get more than one peak on the meter. If you can get more than one peak it indicates the tripler is either operating as a doubler or as a quadrupler.

Then you must use the absorption wavemeter to adjust the tripler to the correct frequency, in this case 38.865 mc. Once you have brought the adjustment to approximately the correct frequency with the absorption wavemeter, you can then go ahead and carefully complete the adjustment for maximum grid current in the doubler stage.

The next step is to move the ammeter to the grid circuit of the second doubler. Now the first doubler is adjusted for maximum output, which will be indicated by maximum reading on the second doubler current meter. Again, if you can get more than one peak as you adjust the first doubler, it indicates that the range of adjustment is sufficient to operate the doubler as a tripler or as a straight through amplifier. Again, you must use the absorption wavemeter to identify the various peaks. The meter should be brought near the output of the first doubler circuit, and the peak, which occurs at about 77 mc, identified. Then the first doubler adjustment can be carefully made for maximum indication on the meter at this frequency.

The next step of the adjustment procedure is to move the meter to the grid circuit of the driver. The output of the second doubler should now be adjusted for maximum grid current at approximately 155 mc. Again, if you can get more than one peak, you will have to use the absorption wavemeter to identify the correct peak.

The final step in the adjustment of the strip is to move the meter to the grid circuit of the p.a. Now the adjustments in the output of the driver circuit and the input of the pa should be made for maximum reading on the grid current meter. Here you do not have to worry about tuning the circuit through a harmonic because the second harmonic will be over 300 mc, and this is so far from the operating frequency that the transmitter tuning adjustments would not have sufficient range to permit you to tune the transmitter to the second harmonic.

(c) In the preceding section we went through a procedure for aligning the multiplier and driver stages in the transmitter. With the transmitter plate voltage still reduced, the grid current in the final stage should be checked and compared with the value listed in the transmitter instruction manual or in a tube chart. If provisions are made in the driver stage for adjusting the driver power, the power should be adjusted to give the correct value of final amplifier grid current.

Once the grid current has been adjusted, the meter can be moved to the cathode circuit of the final amplifier. The amplifier should be lightly coupled to the load and the plate voltage still reduced. Now the tuning in the plate circuit of the final amplifier is adjusted for minimum cathode current. As you tune the plate circuit, you will notice the cathode current decreases until eventually the minimum point in the dip is reached. This is the correct setting for the plate tuning control.

The next step is to apply the rated plate voltage to the final amplifier tube and then carefully retune the plate circuit. Using the tube manual to determine what the maximum cathode current should be, you increase the loading on the transmitter slightly and redip the plate circuit. You continue increasing the loading and dipping the plate circuit until you have the loading adjusted, so that when the plate is dipped, the cathode current is equal to the rated value for the tube. Remember that the cathode current will include not only the plate current, but also the screen and grid currents. If these currents are given separately from the plate current they must be added to the plate current. Often when you increase the plate voltage and load the transmitter the grid current will go down slightly. You should go back and recheck the grid current, and if it is low, readjust the driver slightly to bring the grid current up to the rated value. Once again go back and check the cathode current of the final amplifier. If it is now slightly above normal, you will have to reduce the loading and retune the plate circuit of the final amplifier stage.

(d) When we speak of the power input to a stage we are usually referring to dc plate power input. The power input is equal to the plate voltage times the plate current. Since the meter in the cathode circuit is measuring the plate, screen and grid currents, you need to subtract the screen and grid currents from the cathode current in order to get the plate current. You can get the value of the screen and grid currents by measuring them or by referring to a tube manual.

Most transmitters have a meter which indicates the plate voltage, but if the transmitter does not have one you will have to measure the plate voltage. To do this you first turn off the transmitter and make sure that all the filter capacitors are discharged. Then connect the ground lead of your voltmeter to ground and the positive lead to the output of the power supply.

If the meter is a multirange meter, it should be set on a high enough range so that you will be sure that the voltage is not too high for the meter.

Now turn the transmitter on and read the meter. Do not touch the meter, the meter leads, or try to switch the range on the meter because the voltages in many transmitters are dangerously high and you could easily be shocked. Once you have the plate voltage, you can use it with the plate current to get the plate power input. To disconnect the voltmeter turn off the transmitter, discharge the filter capacitor, and then disconnect the meter leads.

(e) If the final amplifier stage is self-oscillating, you will get power out from the stage even with the excitation removed. Therefore, you can simply remove the excitation and check to see if there is power being generated by the stage. You can use the absorption wavemeter as a check. If the transmitter has a separate bias supply and the tube is biased beyond cut-off by this supply, the plate current should drop to zero when the excitation is removed. If the current doesn't drop, this also is an indication that the stage is oscillating. If the stage is oscillating near the operating frequency, the stage requires neutralization. If the stage is operating at a very high frequency, you have parasitic oscillations and these must be eliminated by suitable parasitic chokes.

(f) Since the frequency meter has an accuracy of .0002%, we must allow for any error in the frequency meter when checking the transmitter frequency. If we want to keep the transmitter within a tolerance of .0005%, then we must hold the reading on the meter within .0003% to allow for the frequency meter error. Therefore, the total deviation that can be permitted will be .000003 times the operating frequency. This will give us a tolerance of 466 cycles. Therefore, the transmitter frequency must be maintained at 155.460 mc ± 466 cycles.

(g) Since the range of the frequency meter is 20-40 mc, you know that the meter's fourth harmonic will be needed to check the transmitter calibration. You can find the exact frequency you should read on the meter by dividing the operating frequency of 155.460 mc by 4, giving you 38.865 mc.

Now that you know the frequency that you should read on the meter, assuming the transmitter is exactly on frequency, the next step is to use the 1 mc crystal in the frequency meter to check the accuracy of the meter at 38 mc and at 39 mc. Thus, the compensation that may be necessary due to slight inaccuracies in the meter can be allowed for.

The next step is to feed a small signal from the transmitter to the heterodyne frequency meter and adjust the signal generator in the meter until you get a zero indication. There may be a meter on the heterodyne frequency meter that indicates zero when the two frequencies are the same, or you may get a zero beat in a pair of headphones.

As you bring the frequency of the frequency meter close to the transmitter frequency you will hear an audible tone. The frequency of this tone will decrease as the generator is brought closer to the transmitter frequency and it will drop to zero when the two are set on exactly the same frequency.

In using a receiver in conjunction with the frequency meter, the receiver is tuned to the transmitter frequency. Then the signal from the frequency meter is also fed to the receiver. As you approach the station frequency with the frequency meter, you get a beat tone. Adjust for the zero beat. Then you have the generator on the same frequency as the transmitter.

(h) Whistling into the microphone will produce a greater deviation than speaking into the microphone. Even if the actual amplitude of the whistling was no greater than the voice level, voice frequencies are comparatively low, whereas the whistle is a much higher frequency. In the higher frequency range the pre-emphasis circuit will take over and produce a stronger audio signal for the modulator, and hence, a greater deviation of the transmitter.

(i) An FM deviation meter can be used to check the transmitter deviation. If the deviation is excessive, then the limiter should be reset to prevent excessive deviation, regardless of how loud a signal is picked up by the microphone.

After this adjustment is made it is possible that excessive clipping is taking place. This could occur if the gain control in the speech amplifier is turned up too high. Reducing the setting of the gain control will prevent excessive clipping. You can use an audio generator and oscilloscope to check the setting of the gain control in the speech amplifier. The manufacturer of a transmitter usually gives information on how strong a signal should be fed into the speech amplifier in order to set the gain and limiting controls correctly.

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T-8: FCC

NRI Study Guide for FCC License Examination

PART IX

Be sure you study this part of the study guide carefully as soon as you receive it. After you study it, answer the questions in the examination TC-9. The material covered in this part of the study guide along with the lessons you have studied should enable you to answer all of the questions in TC-9. As we have mentioned previously, the questions asked in this examination are similar to those you are likely to be required to answer when you take your Radio-Telephone Operator's License Examination.

Be sure to spend any additional time you might need studying this part of the study guide carefully or review your lesson text if you find some of the material is giving you difficulty. You can be sure that you will get some questions on your FCC Examination on material covered in this section of the guide. If you understand the material covered, and are able to draw from memory all the diagrams asked for, you should have no difficulty answering the questions.

Be sure to save this part of the study guide along with those you have already received so you will be able to make a complete review before going for your license.

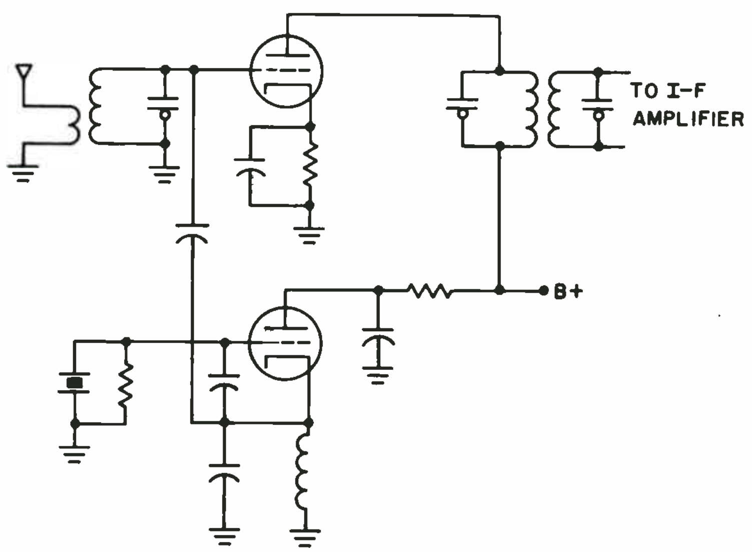
184. Draw a schematic diagram of each of the following stages of a superheterodyne receiver. Explain the principles of operation. Label adjacent stages.

(a) Mixer with injected oscillator frequency.
(b) I-F amplifier.
(c) Limiter.
(d) Discriminator.
(e) Differential squelch circuit.

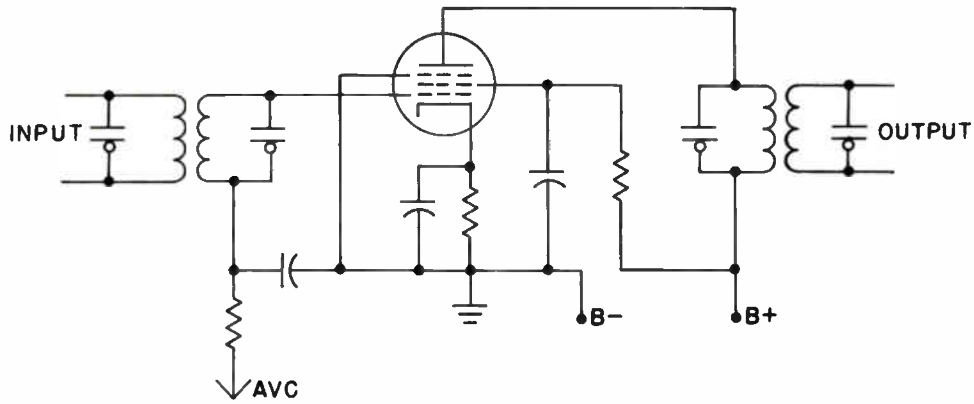
(a) A mixer and oscillator using separate triodes is shown. Actually, the two triodes could be in the same envelope of a single tube or in some cases a pentode may be used as the mixer and a triode as the oscillator.

We have shown a crystal oscillator rather than a variable tuned oscillator. In FM communications work, the service is generally assigned a particular channel frequency. At the most two or three channels might be available and in this case a switch would be provided for switching different crystals into the circuit to provide the correct oscillator frequency. FM communication receivers are often double-conversion superheterodynes. By that we mean that the incoming signal is converted to a high i-f frequency by means of a mixer-oscillator combination. Then the signal may be amplified and then fed to another mixer-oscillator combination which converts the signal to a lower frequency. The mixer-oscillator arrangement in each case may be the same, the only difference will be that the resonant circuit in the mixer input and output will be tuned to different frequencies and different frequency crystals will be used. Where a receiver must operate on several channels, the various crystals used to switch channels will be found in the first oscillator. The second mixer-oscillator circuit always works on a fixed input frequency converting the high i-f to a lower i-f.

In broadcast FM receivers the crystal is replaced by a resonant circuit whose frequency can be varied so that the various stations in the FM broadcast band can be tuned in.

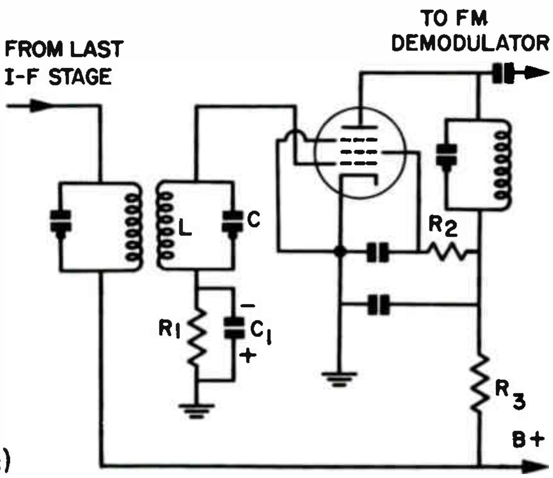


(a)



(b)

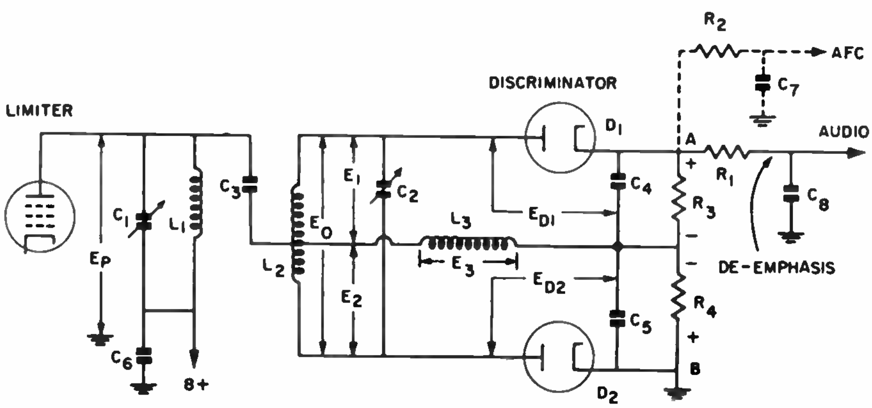
(b) The schematic diagram shown is that of a pentode i-f amplifier stage. Notice that as far as the schematic is concerned the circuit is identical to that found in an AM broadcast receiver. The only actual difference will be the i-f frequency. FM receivers designed for broadcast band operation use a 10.7 mc i-f. FM receivers designed for communications work use a lower frequency i-f, the exact value used will depend upon the particular receiver.



(c)

(c) The schematic diagram of a pentode limiter is shown. Notice the similarity between the limiter and the i-f amplifier. The main difference is the use of the RC network made up of R_1 and C_1 in the grid circuit of the limiter. The RC network is placed in the grid circuit so that when the grid of the limiter is driven positive a bias will be developed across this network. In addition, the plate and screen voltages on the limiter stage are usually quite low. Resistor R_3 drops the B supply voltage to a value considerably lower than that used on the i-f stages. The idea in back of a limiter is to operate it so that it is easily driven to cutoff and plate-current saturation. This results in clipping of the FM signal so that the amplitude of the output signal from the limiter is constant. Since the FM modulation does not change the amplitude of the signal, this limiting is possible; it does not cause any distortion. The modulation intelligence is all present in the form of frequency modulation rather than amplitude variations. Amplitude variations of the signal are usually caused by noise and therefore converting the signal to a constant-amplitude signal will reduce or almost eliminate the effect of noise pulses.

(d) In the discriminator circuit shown, the coil L_1 in the output of the limiter circuit is inductively coupled to the coil L_2. As a result a voltage is induced in series with L_2. This voltage, E_O, will be 180° out-of-phase with the voltage, E_P, across L_1. The voltage across L_1 is coupled through C_3 to the coil L_3 so the voltage E_3 will will be in phase with the voltage E_P across L_1.



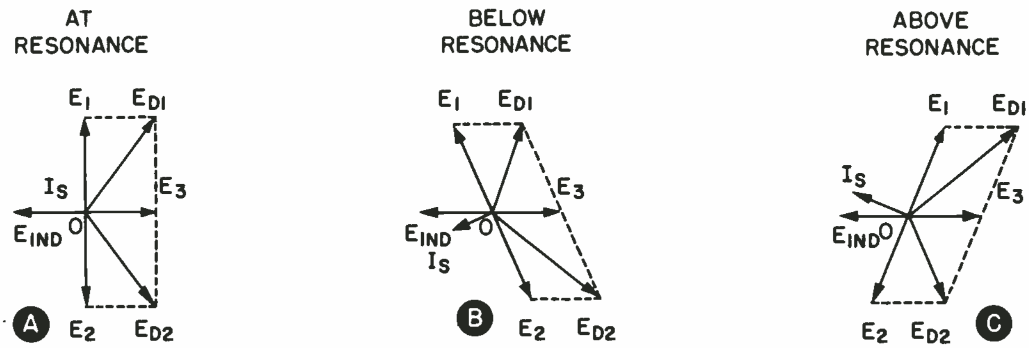
(d)

The voltage induced in L_2 will cause a current to flow in the series-resonant circuit consisting of L_2 and C_2. This current I_S will be in phase with the induced voltage E_O. The induced current flowing through L_2 will develop a voltage across L_2 which leads the current by 90°. However, this will be the total voltage across L_2. If instead of the total voltage we take the voltage from the center terminal of the coil to one end and label this E_1 we can say that this voltage will lead the current by 90° and then the voltage measured from the center terminal of the coil to the lower end of L_2 which we have labeled E_2 will lag the induced voltage by 90°. Thus we get the vector diagram shown at A. The voltage applied to the diode D_1 will be the vector sum of E_1 and E_3. The voltage applied to the diode D_2 will be the vector sum of E_2 and E_3. The diagram at A represents the conditions at the resting frequency; at this frequency the voltage E_{D1} is equal to the voltage E_{D2} and hence the voltages developed across R_3 and R_4 will be equal and opposite and therefore cancel. As a result the net output from the discriminator is zero.

in the vector diagram of C. Now the vector sum of E_1 plus E_3 will be greater than the vector sum of E_2 plus E_3 and therefore the voltage applied to D_1 will be greater than the voltage applied to D_2. This means that the voltage developed across R_3 will be greater than the voltage developed across R_4 and therefore will have a positive voltage developed at terminal A.

The rate at which the voltage at A changes polarity depends on the frequency at which the FM signal swings below and above the resting frequency. This of course depends on the frequency of the modulating signal. Therefore at A, we will have an audio signal corresponding in frequency and amplitude to the original modulation on the FM carrier.

In many FM discriminators a filter network such as R_2 and C_7 is used to develop a dc voltage if the i-f frequency is above or below the correct value. This voltage is then fed to a reactance tube which is used to control the oscillator frequency and produce the correct i-f frequency.

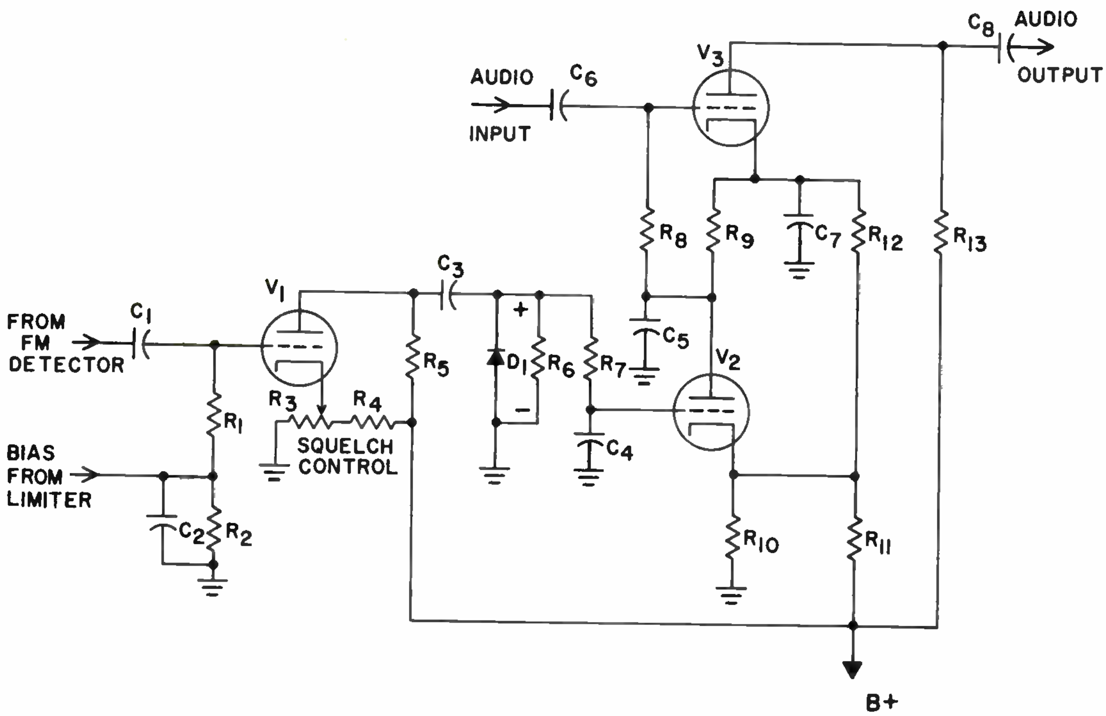


When the signal deviates below the resonant frequency then the reactance of the capacitor C_2 becomes greater than the reactance of L_2. As a result the circuit acts like a capacitor and the induced current I_S will lead the induced voltage E_O as shown in the vector diagram at B. Since the voltages E_1 and E_2 will still be 90° out-of-phase with the induced current then the vector diagram shown at B represents the conditions in the circuit. Now E_{D1}, the vector sum of E_1 and E_3 is less than E_{D2}, the vector sum of E_2 plus E_3. This means that the voltage applied to D_2 will be greater than the voltage applied to D_1 and therefore the voltage across R_4 will be greater than the voltage across R_3. This will result in a negative voltage appearing at point A at the discriminator output.

When the signal deviates above the resting frequency the reactance of L_2 will be greater than the reactance of C_2. Under these conditions the circuit acts like a coil and now the induced current I_S will lag the induced voltage E_O as shown

(e) The purpose of a squelch circuit is to turn off the receiver audio when there is no signal being received. For example, in mobile communications work, the transmitter is turned on only when messages are to be transmitted. On the other hand, the receivers in the mobile unit must be on at all times and when the transmitter is not on the air there is considerable atmospheric noise picked up and amplified by the receiver. This could be very annoying particularly over a period of time, so a squelch circuit is used to turn the receiver audio section off when the transmitter is not transmitting.

The operation of the squelch circuit is relatively simple. The signal from the FM detector output, which will contain considerable noise is fed to V_1 which is a noise amplifier. The bias on the cathode of this stage is set by means of a squelch control to give the desired amount of amplification. The signal is fed through the coupling capacitor to the diode D_1. When the cathode of this diode is driven negative, the diode will conduct



(e)

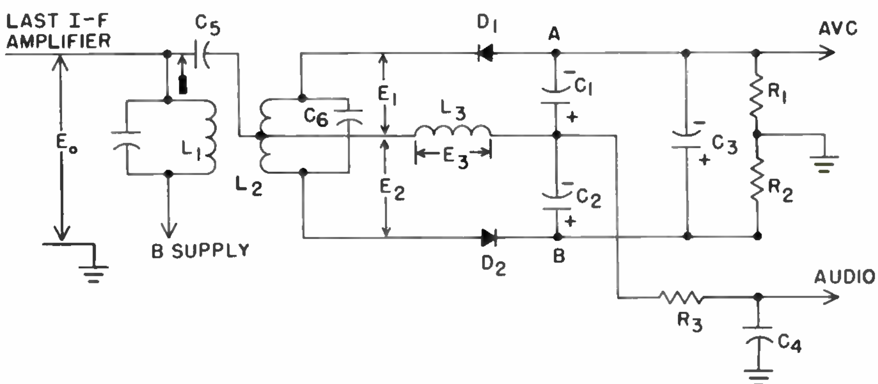
and current will flow from the cathode to the anode and then through R_6 back to the cathode of the diode. In flowing through the resistor R_6 a voltage is developed having the polarity shown.

The voltage developed across R_6 is a pulsating voltage and this is filtered by R_7 and C_4 and applied to the grid of the squelch tube V_2. The positive voltage fed to the grid of the squelch tube causes the tube to conduct heavily and the plate current for this tube flows through R_9, the cathode resistor of the first audio stage, V_3. The positive voltage developed across the cathode resistor biases the audio stage beyond cutoff so that the audio signal fed to it from the detector is not amplified.

When the transmitter is turned on, there will be a bias developed across the limiter grid resistor and this bias is fed to the grid of the noise amplifier, V_1. The bias is sufficient to cut off the noise amplifier hence the noise signal will no longer reach the noise detector and there will be no positive voltage fed to the grid of the squelch tube. The squelch is biased by means of R_{10} and R_{11} so that a positive voltage will appear in the cathode and unless a positive voltage is also present in the grid of the tube, the current through the squelch tube will be cut off. When current flow through the squelch tube is cut off, the audio amplifier stage will no longer be biased beyond cutoff and then the audio signal from the FM detector will be amplified by this stage and fed on to the next stage.

185. Draw a schematic diagram of a ratio detector and explain its operation.

The ratio detector is quite similar to the discriminator circuit. C_1 is equal to C_2 and R_1 is equal



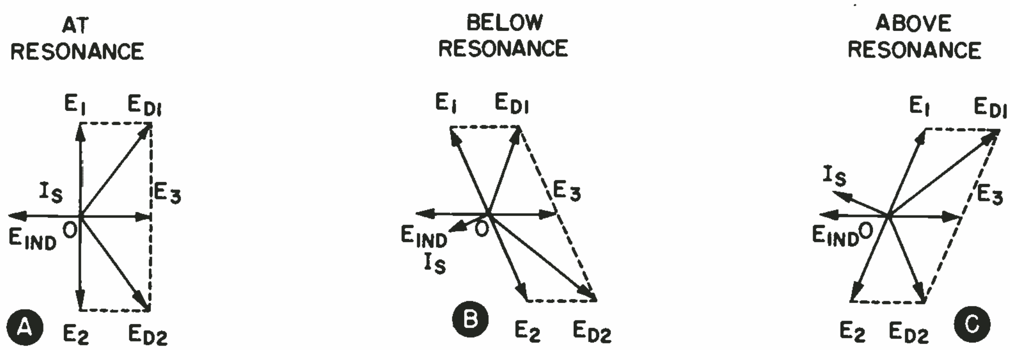
to R_2. However, notice that in the ratio detector the diodes are connected with the opposite polarity. In other words, D_1 has its cathode connected to a secondary of the i-f transformer and D_2 has its anode connected to the other end of the secondary.

In operation, a voltage is induced in the series-resonant circuit consisting of L_2 and C_6. This voltage is 180° out-of-phase with the voltage fed through the capacitor C_5 to the coil L_3.

Since L_2 and C_6 form a series-resonant circuit, the induced voltage will cause a current to flow around that circuit and this current will be in phase with the induced voltage.

capacitors. The total voltage across C_1 and C_2 will remain constant at all times as long as the amplitude of the signal remains constant. Therefore since we are interested only in FM modulation and not in AM modulation we connect the large electrolytic C3 across the series combination of C_1 and C_2. This electrolytic capacitor, which is called the stabilizing capacitor, holds the voltage across the two capacitors constant so that if a noise pulse should arrive and tend to change the total voltage, the electrolytic simply absorbs the extra current in the circuit and prevents any rapid voltage change.

When the FM signal frequency swings below resonance, the series-resonant circuit of L_2 and C_6



At the i-f center frequency, the current flowing through L_2 will produce the voltages E_1 and E_2, shown at A in the vector diagram, that will lead and lag the induced current I_S by 90°. Thus at resonance the vector diagram shown at A, which is identical to the vector diagram for the discriminator circuit, also applies to the ratio detector. Notice that E_{D1}, the voltage applied to D_1, is equal to E_{D2}, the voltage applied to D_2.

However, since the two diodes are connected in series, when the cathode of D_1 is negative, the anode of D_2 is positive. Under these conditions we will have a current flow from the center tap of L_2 through the upper part of the transformer winding to the cathode of D_1, through D_1 to the plate and into the negative terminal of C_1. Electrons will flow out of the positive terminal and through L_3 back to the center tap of L_2. At the same time, electrons will leave the center terminal of L_2 and flow through L_3 to the negative side of C_2. Electrons will leave the positive side of C_2 and flow through the diode D_2 back to the center tap of L_2. Thus we have equal voltages across C_1 and C_2. Therefore the voltage between terminals A and B will be the sum of the voltages across C_1 and C_2. Notice R_1 and R_2. These are equal value resistors; the junction is connected to ground and therefore the junction is ground potential. Similarly the junction of C_1 and C_2 will be at ground potential since these are equal value

begins to act like a capacitance. Under these circumstances, the current I_S which is the secondary induced current leads the induced voltage. Thus we have the condition represented in the vector diagram at B. Under these circumstances, the voltage applied to the diode D_2 is greater than the voltage applied to the diode D_1. This means that the voltage drop across C_2 will be greater than the voltage drop across C_1. Therefore the junction of C_1 and C_2 will swing in a negative direction.

When the FM signal swings above resonance, the series-resonant secondary circuit will act like an inductance. Under these circumstances the induced voltage will lead the induced secondary current and we have a situation represented by the vector diagram shown at C. Now the voltage applied to D_1 will exceed the voltage applied to D_2. The voltage across C_1 will be greater than the voltage across C_2 and the junction of the two capacitors will swing in a positive direction.

Thus at the junction of C_1 and C_2 we have a voltage that is moving positive and negative as the incoming FM signal swings above and below resonance. Since the signal swings at the audio ratio of modulation, therefore we will have the audio signal picked up at the junction of C_1 and C_2.

The filter consisting of R_3 and R_4 eliminates any rf that may be present at the junction of C_1 and C_2. We can also pick up an automatic volume control or automatic gain control voltage at point A since the amplitude of the total voltage across C_1 and C_2 will depend upon the amplitude of the FM signal. Therefore a very strong signal will produce a higher negative voltage at terminal A than a weaker signal. This negative voltage can be used to control the gain of the rf and i-f stages in the receiver.

186. Explain how spurious signals can be received or created in a receiver. How could this be reduced in sets having sealed untunable filters?

Spurious signals can be received in a receiver simply because a strong interfering station is operating close to the frequency of the station you are trying to receive. The selectivity of the receiver may simply be insufficient to reject the undesired signal.

Sometimes a very strong local station will cause cross-modulation either in the rf or mixer stage of the receiver. The strong interfering station may be so strong that it simply blocks the first stage of the receiver. It causes this stage to operate on the non-linear portion of its characteristic curve. The modulation on the signal simply over-rides the modulation on the carrier on the signal you are trying to receive, creating a new signal which can get through the i-f passband of the receiver.

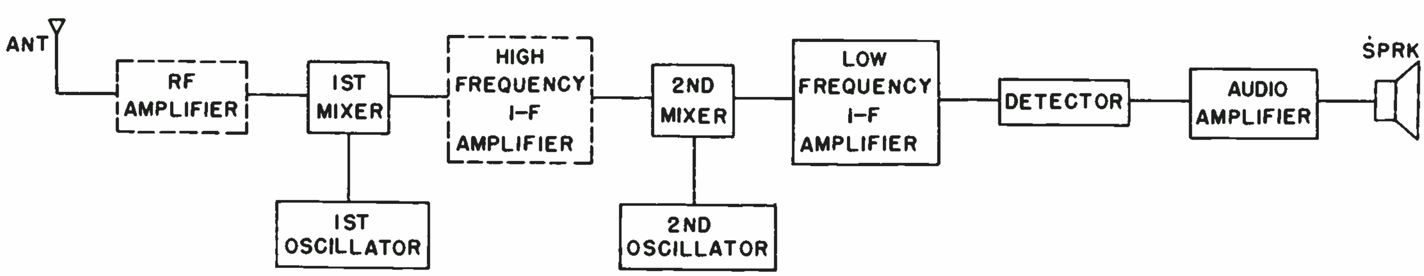
Sometimes two strong local stations can beat together producing spurious signals that may interfere with the signal from each station or they may interfere or completely block out the signal from a third station you are trying to receive. Again, this is usually the result of one or more of the strong signals driving the first stages in the receiver into the non-linear portion of the characteristic curve. Interference of this type can sometimes be reduced by means of directional antennas or by means of traps inserted in the input circuit of the receiver and tuned to the frequency of the interfering signal. A parallel-resonant trap can be used in series with the antenna lead-in between the antenna and the receiver. A series-resonant trap can be used across the input between the antenna lead-in and ground.

Spurious signals that can be created in the receiver itself, for example, in the mixing process, in the output of the mixer circuit you will have at least four signals; a signal equal to the frequency of the incoming signal, a signal equal to the frequency of the oscillator signal, a signal equal in frequency to the difference between the frequencies of the two signals, and a signal equal in frequency to the sum of the frequencies of the two signals. The only signal you are actually interested in is the signal equal in frequency to the difference between the frequencies of the incoming signal and the oscillator signal. However, either this signal or one of the other three signals may beat with another incoming signal and produce the correct i-f signal frequency and this signal can then get on through the i-f passband in the receiver. In interference of this type, the actual cause may not be too obvious. The incoming signal might be picked up by the antenna and fed to the mixer along with the desired signal. In this case a trap in the input circuit should be helpful. However, in the case of a very strong local signal, the signal might be picked up in the wiring in the mixer circuit and in this case better shielding in the receiver is the only effective remedy.

187. Describe, step-by-step, a proper procedure for aligning an FM double-conversion superheterodyne receiver.

As we mentioned previously, a double-conversion superheterodyne receiver is simply a superheterodyne where the incoming signal is first converted to a fairly high frequency intermediate frequency and then fed to a second mixer where it is converted to a lower frequency intermediate frequency. Between the first and second mixer stages there may or may not be i-f amplifiers depending on the quality of the receiver. A block diagram of a double-conversion FM receiver is shown. The stages shown in dashed lines may be omitted in some receivers.

In FM receivers the two most widely used detectors are the discriminator and ratio detector. The procedure for aligning the two is essentially the same. In the case of both the detectors you simply connect a vacuum-tube voltmeter or a high-impedance volt ohmmeter across



the entire output of the detector and then adjust the primary winding on the detector i-f transformer for maximum output. Then move the meter so it is connected across half the output, and adjust for the secondary of the detector transformer for zero output. For example, in the ratio detector you would connect the meter between the junction of C_1 and C_2 and ground.

If the receiver uses a ratio detector then you can connect the voltmeter across the entire detector output for the remainder of the alignment procedure. If the set uses a discriminator it will have a limiter and so you can connect the voltmeter across the limiter grid resistor and leave it there for the balance of the alignment. With the signal from the signal generator tuned to the low i-f frequency and fed into the second mixer circuit you now proceed to align the low-frequency i-f transformers working from the detector towards the mixer. The first step is to align the secondary of the last i-f transformer (not the detector transformer) and then the primary and repeat this procedure until you reach the primary of the first low-frequency i-f transformer.

After you have completed the alignment of the low-frequency i-f amplifier, you move the signal generator to the first mixer. In the case of a vacuum-tube type mixer you can usually feed the signal into the grid of the mixer. Set the signal generator to the high i-f frequency. If the crystal used in the oscillator circuit is of the plug-in type, it is usually a good idea to unplug it to prevent any interference with the i-f alignment. Now, align the i-f transformer in the grid circuit of the second-mixer and then the primary winding of the same i-f transformer. You simply proceed from the grid circuit of the second mixer and work toward the plate circuit of the first mixer.

After you have the two i-f amplifiers aligned, the next step is to connect the signal generator to the antenna terminals and set it to the frequency at which the receiver operates. Now put the oscillator crystal back in its socket, if you removed it during the i-f alignment, and then adjust the mixer input circuit. Next, if the receiver uses an rf stage, adjust the output circuit of the rf stage and then finally adjust the circuit in the input of the rf stage.

During the alignment procedure as you work from the low i-f amplifier towards the rf input, the output as indicated on the meter should be continually increasing. You must avoid letting the output voltage get too high, otherwise you will overload some stage in the receiver and it will be difficult to peak to various adjustments. As the output voltage as indicated by the meter increases, you should reduce the output from the signal generator. Some manufacturers indicate in their alignment instructions what the alignment voltage should be during the alignment procedure. Be sure not to exceed this value or you will have difficulty getting the best possible alignment job.

188. Name four materials that make good insulators at low frequencies, but not at uhf or above.

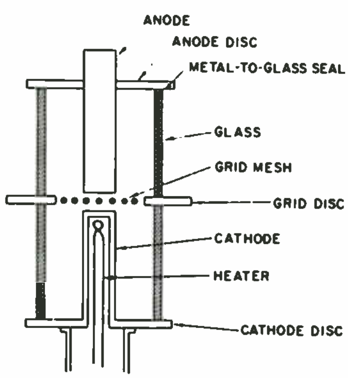
Glass, paper, wood, bakelite, porcelain and rubber are often used as insulators at low frequencies, but they are unsatisfactory as insulators at uhf or above.

189. Name at least three circuit factors (not including tube types and component values) in a one-stage amplifier circuit, that should be considered at vhf which would not be of particular concern at vlf.

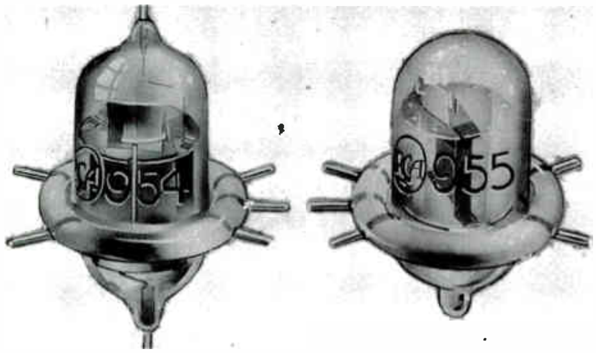
Proper grounding is of particular importance in vhf circuits. The leads should run in short direct circuits to a common ground. Lead lengths are important at vhf. In addition to the ground leads, all leads in the circuit should be kept as short and direct as possible. Shielding is important at vhf; the output of the circuit must be effectively shielded from the input. Stray capacities are also important in the vhf circuits. Even though the capacities are kept low, because of the very high frequencies involved, the reactance may be quite low. Therefore all stray capacities should be kept as low as possible.

190. What is a "lighthouse" triode? An "acorn" tube? These tubes were designed for operation in what frequency range?

A lighthouse triode or a lighthouse tube as they are often called is a special tube designed for uhf operation. The tube gets its name from the fact that it looks something like a lighthouse.



As shown in the illustration the cathode, grid and plate are arranged parallel to each other. In effect they are formed something like discs, and the entire tube is built so that it can be installed in a resonant cavity.



An acorn tube is also designed for use in the uhf region up to about 600 mc. This type of tube is similar to the conventional tube in that in the center of the tube is the cathode, the grid is wrapped around the cathode and the plate is around the grid cathode structure. However, the elements inside of the tube are made extremely small in order to cut down capacity in the tube. The elements are spaced very closely to reduce transit time to a minimum. Lead inductance is kept at a minimum by bringing the element leads directly out through the sides of the tube through the glass seal rather than bringing them to a plug-in type base on the bottom of the tube. The acorn tube gets its name from the fact that it looks something like an acorn.

191. Why are special tubes sometimes required at uhf and above?

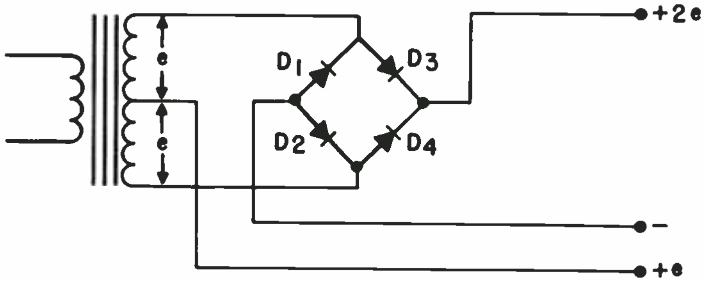
At uhf the transit time of a tube becomes important. The transit time is the length of time it takes the electrons to travel from the cathode of the tube to the plate. Even though this time may be extremely short, at uhf it may represent an appreciable portion of a cycle. As a result, inefficient operation of the tube results.

Another important problem encountered at uhf is stray capacity in the tube itself. At lower frequencies, the capacity between the elements of the tube do not upset the performance of the tube, however, at uhf these capacities may be high enough to prevent the tube from operating properly. Inductance in the leads is another important factor at uhf. The inductance may be high enough so that it along with the stray capacity between the tube elements forms a resonant circuit at a frequency below the required operating frequency. When such a situation is encountered, the tube is entirely unsatisfactory for use at the desired frequency. Special tube designs are employed to overcome these problems.

192. Show a method of obtaining two voltages from one power supply.

The simplest method of obtaining two or more voltages from a single power supply is by means of a voltage-divider network and bleeder connected across the output of the power supply. Taps can be arranged in the divider network to obtain the required voltages.

Another method of getting two voltages from a single power supply is by means of the conventional tapped secondary on the power transformer and by the use of four diode rectifiers. In the circuit shown all four diodes function as a bridge rectifier. The output voltage available will be determined by the entire secondary output voltage. Diodes D_1 and D_2 also function as a full-wave rectifier using first one half of the transformer secondary winding and then the other half.



The output from this circuit will be determined by the voltage across each half of the secondary winding. In other words, the power supply output voltage will be approximately half the voltage obtained from the rectifier circuit using the bridge rectifier.

193. Describe some factors in connection with the following items, which should be considered at vhf and above but would not be of particular concern at mf or below.

(a) Wire diameter and lead lengths.
(b) Wire configuration (placement and bending).
(c) Coaxial cables and transmission lines.
(d) Capacitor types.

(a) At vhf, there is a pronounced skin effect in any current-carrying conductor. By skin effect we mean that there is a tendency for the current to flow on the outer surface of the wire rather than through the entire cross-section of the wire. This will result in an increase in the ac resistance of the wire. This can be overcome by using a larger diameter wire or by using hollow tubing for connections. Lead length is also important at vhf, whereas it is not too critical at medium frequencies. At vhf, even a comparatively short piece of wire that has a relatively low inductance may have a comparatively high inductance reactance because inductive reactance goes up with the frequency. Therefore leads should be kept as short and direct as possible. In wiring at vhf, wire size and lead lengths are both important. If you use too large a wire size you increase the stray capacity; if you use too small a wire size the ac resistance becomes too high. As mentioned previously, too long a lead may result in excessive inductance reactance in the circuit.

(b) The placement of the wiring at vhf is extremely critical. Sharp bends in the wire should be avoided in order to prevent excessive losses. Also while the wiring should be kept as short and direct as possible, leads should be kept clear of the chassis in order to prevent excessive capacity and leads should be kept away from other components in the circuit in order to prevent feedback and excessive losses.

(c) As the frequency increases, the losses in coaxial cables and transmission lines increase. Therefore it is important to route such lines so that they are as short and direct as possible.

Some types of coaxial cable have such a high loss that even with a relatively short length of cable the power reaching the output will be attenuated so that it is substanially below the power fed into the input. Generally speaking, open-wire transmission lines and flat lines such as the twin lead used in TV have a lower loss at vhf than coaxial lines or cables.

(d) Special capacitor types are used at vhf. The capacitors are generally smaller than those found at mf, because lower capacity values are required. Also the capacitors are made with a better grade of insulating material in order to keep losses as low as possible. The capacitors are also arranged so that they have as low an inductance as possible. All capacitors have a certain amount of inductance and at some frequency the inductance and the capacity in the circuit will resonate. Generally, this is a disadvantage; however, in some special bypass applications this can be an advantage. For example, if you are bypassing the screen grid of a tetrode or pentode tube, if the capacitor happens to be resonant at the operating frequency, then you will have a series-resonant circuit connected between the screen grid of the tube and ground. As you know this will offer a very low resistance to the vhf signal and therefore the grounding of the screen will be more effective than it would be if it were bypassed by a non-resonant capacitor.

194. Define "transmitter intermodulation", a possible cause (or causes), its effects and steps that could be taken to reduce this.

One form of transmitter intermodulation may result in a location where two transmitters are operated close together so that there is some coupling between the two antennas. Under these circumstances, energy from one transmitter can get back into the other transmitter and produce intermodulation. For example, let's consider the transmitters as transmitter A, operating on one frequency, and transmitter B operating under a somewhat higher frequency. If some of the energy from transmitter B gets back into transmitter A, a signal equal to the difference of B - A may be produced. This signal in turn can produce an intermodulation product so that we have signals of A + (B - A) and A - (B - A). Additional harmonics of the B - A product may help produce additional intermodulation sidebands. Similarly, the transmitter at B can generate spurious outputs at B ± (B - A). This type of interference can be kept to a minimum by using filters in the transmission line or tuned stubs to attenuate or eliminate the the interfering signal.

Other sum and difference components can be generated in the transmitter itself. This is due to the fact that the Class C amplifiers used in a transmitter are all non-linear devices. Coupling between stages in the transmitter due to insufficient isolation and filtering in the power supply circuits can be a source of intermodulation. Again, if the intermodulation components are developed, suitable filters in the transmission line between the transmitter and the antenna should prevent their radiation.

195. State a probable cause of and a method of reducing transmitter spurious emissions (other than harmonics).

Transmitter spurious radiations occurring near the operating frequency are often the result of self-oscillation. This is caused by an improperly neutralized transmitter in most cases. Retuning the transmitter, particularly making sure that it is neutralized properly should eliminate radiation of this type.

Two other types of spurious radiation that may be encountered are very high frequency and very low frequency parasitic oscillations. VHF parasitics are caused by resonant circuits being produced by the lead inductances and the stray capacities of component parts. The proper selection of component parts will reduce the possiblity of low frequency parasitics. High frequency parasitics are eliminated by means of suitable parasitic chokes in the plate leads of the stage where the trouble is encountered. These chokes are loaded by low-value resistors so that sufficient losses are introduced in the circuit to prevent oscillation.

196. List several frequently used methods of attenuating harmonics in transmitters and explain how each works.

One of the most effective methods of eliminating harmonics from a transmitter is the low-pass filter. The low-pass filter permits the transmitter signal to travel through the filter and through the transmission to the antenna. However, the filter is designed so that it has a cut-off frequency slightly above the transmitter operating frequency.

The low-pass filter offers a very high impedance to any frequency above the transmitting frequency and thus effectively reduces any harmonic radiation produced in the transmitter.

Another method of keeping the harmonics at a minimum is by use of low-impedance link coupling between the stages of the transmitter. Link coupling effectively couples the desired signal frequency, but offers a high impedance to higher frequency signals.

Often a great deal of harmonic radiation is due to capacitive coupling between stages rather than inductive coupling. Therefore the use of a faraday shield between inductively coupled circuits will minimize the transfer of harmonics between inductively coupled circuits. A faraday shield offers no opposition to the magnetic lines of force coupling the two circuits together inductively, but it effectively acts as a capacitive shield between the two circuits thereby reducing harmonic coupling from one circuit to the other due to capacity between the two circuits.

197. What might cause FM in an AM radio-telephone transmitter?

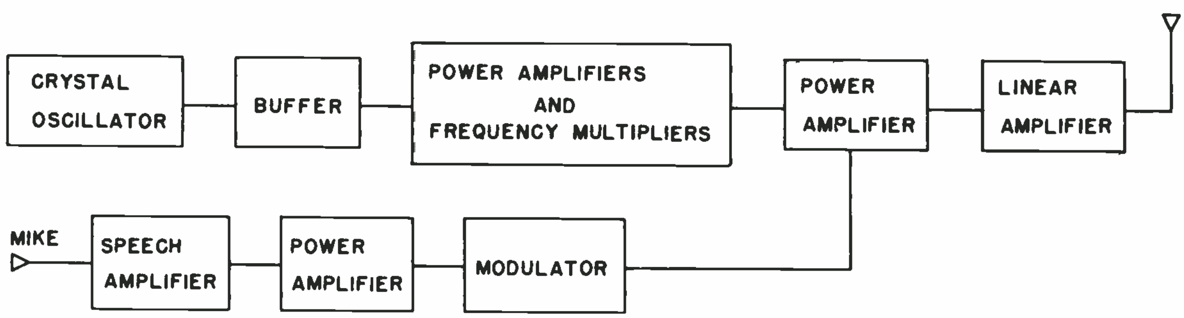
If the frequency of an AM radio-telephone transmitter deviates with the modulation, something must be causing the oscillator frequency to shift. The most likely cause of trouble of this type is poor regulation in the oscillator power supply. This may be due to a defect in the power supply itself so that as the transmitter is modulated, the dc output voltage from the oscillator power supply might vary causing the oscillator frequency to shift. Another possible cause of the trouble may be that the oscillator operating voltages are obtained from the same power supply that supplies the voltages to other parts of the transmitter. Insufficient decoupling between the various sections of the power supply might be the cause of the difficulty.

The oscillator frequency may also shift if the loading on the oscillator changes with modulation. The use of a Class A buffer stage at the output of the oscillator circuit will usually prevent trouble of this type. The Class A buffer stage provides a constant load on the oscillator so that the load cannot change with modulation.

198. Draw a block diagram of an AM transmitter.

The crystal oscillator in the transmitter is held at a very close tolerance by enclosing the crystal in an oven so that its temperature is kept constant. The oven is thermostatically controlled to maintain the crystal temperature to a very close tolerance. The output from the oscillator is fed to a buffer stage which provides a constant load on the oscillator. In broadcast band transmitters where the crystal frequency is the operating frequency, Class C power amplifiers are then used to build up the power to a sufficient level to drive the Class C power amplifier. In high frequency transmitters, frequency multipliers are used to increase the frequency to the desired value and then power amplifiers are used to develop sufficient power to drive the last Class C power amplifier. The final Class C power amplifier is modulated by the modulator stage.

In some small transmitters the output from the Class C amplifier may be fed directly to suitable filters and to the transmission line and from there to the antenna. However, in a high power AM broadcast station, the modulated output from the



last Class C power amplifier is fed to a linear amplifier where the power is increased still further. This is referred to as low-level modulation, where a stage before the final power amplifier stage is modulated. This requires less audio power than would be required to modulate the final amplifier in the case of a high power station.

In the audio section of the transmitter the audio signal is amplified by a series of Class A voltage amplifiers and finally fed to a Class A power amplifier which provides the driving power needed to drive the modulator which is used to modulate the Class C rf power amplifier.

Separate power supplies are used for the oscillator and low-level audio stages. The oscillators usually operate from their own regulated supply to prevent any frequency modulation of the oscillator as well as any frequency shift due to voltage variations. It is much easier to keep this supply voltage constant when the only stage operated from it is the oscillator. A separate high-voltage power supply is usually used as power for the final linear amplifier.

199. Explain the principles involved in a single-sideband suppressed-carrier (SSSC) emission. How does its bandwidth of emission and required power compare with that of full carrier and sidebands?

In single-sideband suppressed-carrier emission, the carrier is suppressed during the modulation process so that only the sidebands are produced. Then one of the sidebands is suppressed either by a filter system or by a phasing network in the audio circuits. Modulation in a suppressed carrier system is usually accomplished in a balanced type modulator so that the carrier is cancelled out in the modulator and the only time an rf output is produced is when a modulating signal is supplied to the modulator.

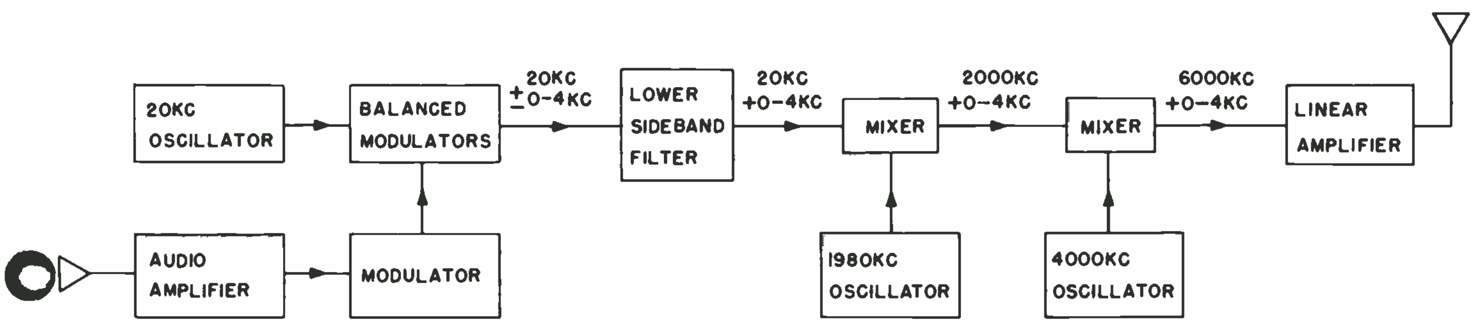
In the conventional amplitude modulation system, we have a carrier and two sidebands. In the single sideband system we have only one sideband so the bandwidth required with SSSC emission is one half that required in conventional AM type of modulation.

In a standard AM broadcast where we have a carrier and two sidebands, the power in the sidebands at 100% modulation will be equal to one half of the carrier power. Thus we must generate the carrier plus the two sidebands. Actually, only the sidebands are carrying the intelligence signal, the carrier itself contributes nothing insofar as the transmission of intelligence is concerned. Therefore only one third of the power actually being radiated by the transmitter contains the intelligence we are transmitting. Therefore in a situation where a 100-watt transmitter is 100% modulated the actual average power being radiated is 150 watts. The power in the sidebands will only be 50 watts. Therefore a single sideband transmitter with a power output of 50 watts would have roughly the same amount of power in the intelligence signal.

200. Draw a block diagram of an SSSC transmitter (filter type) with a 20 kc oscillator and emission frequency in the range of 6 mc. Explain the function of each stage.

In the block diagram shown the audio signal from the microphone is fed to the audio amplifier where it is built up to sufficient amplitude to feed it to the modulator. The output from the modulator is fed to the balanced modulator. At the same time, the 20 kc oscillator output is fed into the balanced modulator.

The balanced modulator is adjusted so that with no audio input, the output from the modulator is zero. When an audio signal is applied to the modulator, upper and lower sidebands are produced around the 20 kc carrier frequency. The upper and lower sidebands are fed from the modulator to a sideband filter. The sideband filter may be designed to eliminate either the upper sideband or the lower sideband. In the example we have shown, the sideband filter is designed to eliminate those sideband signals below 20 kc. Thus at the output of the filter we will have the upper sideband consisting of frequencies from roughly 20 kc up to the upper audio limit, that we



choose, pass through the audio amplifier. Usually since single-sideband transmissions are used for voice frequencies the response of the audio amplifier is limited to 3000 or 4000 cycles, so the sidebands would extend from 20 kc up to no higher than 24 kc.

The signal from the output of the sideband filter is fed to a mixer. A signal from a local oscillator is also fed to the mixer and in the mixer stage the two signals are mixed together producing upper and lower sidebands. By selecting a local oscillator frequency of 1980 kc, we can produce a sideband equal to the sum of the local oscillator plus the sideband signal and another sideband equal to the difference between the local oscillator and the sideband signal. By tuning the output of the mixer so that only the upper sideband is passed, we have a signal equal to the sum of the two which will be 2000 kc plus the upper sideband.

The output from the first mixer is then fed to a second mixer where it is mixed with the signal from the second local oscillator operating at 4 mc. This signal will beat with the signal from the first mixer again producing signals equal to the sum and difference of the two signals. The sum of the two signals will give us a signal equal to 6000 kc (6 mc) plus the upper sideband signal. The output from the second mixer is then fed to a linear rf power amplifier to increase the amplitude of the signal. A second linear rf power amplifier may be used, if required, to build up the amplitude of the signal still further.

201. Explain briefly how an SSSC emission is detected.

To detect a single-sideband suppressed-carrier sideband, we must reinsert the missing carrier. This is done by a local oscillator at the output of the i-f amplifier in the receiver. Thus if the receiver is designed with a 455 kc i-f amplifier, a 455 kc signal is generated locally and this signal along with the signal from the receiver i-f is fed to the detector. In the detector the two signals beat together so that a carrier and a sideband are produced and the original audio signal can then be recovered as in the conventional AM type of detector.

202. Explain, step-by-step, how to align an AM receiver, using the following instruments. Also explain briefly what is occurring during each step.

(a) Signal generator and speaker.
(b) Signal generator and oscilloscope.
(c) Signal generator and vtvm.

In each of the three preceding questions we are asked to align an AM receiver using the signal generator as the signal source. However, we are to use a different output indicator in each case. In the case of the speaker, you use a modulated signal from the signal generator, and you simply keep adjusting the receiver while listening to the speaker and listening for maximum output. All alignment adjustments are performed essentially with the purpose of increasing the speaker output. You must keep the gain of the receiver down reasonably low so that you can notice changes in sound level. This is not a particularly good way of aligning a receiver because the ear is relatively insensitive to small changes in sound level and therefore it is difficult to get each of the adjustments peaked exactly. The net result is that the gain of the receiver will usually be somewhat below what it could be if each individual adjustment was peaked more accurately.

In the second example we are to use an oscilloscope as the output indicator. Here a modulated signal from the signal generator is used and the oscilloscope vertical amplifier is connected to some convenient point in the audio section of the receiver. Usually between the plate of the audio output tube and ground is the best place to connect the oscilloscope vertical input. Now you can see the audio signal on the oscilloscope and notice any change in amplitude as you perform the various alignment adjustments. Again, you are to align the receiver for maximum amplitude on the oscilloscope. If the signal on the oscilloscope gets too large you can reduce its amplitude by reducing the signal from the signal generator. If the signal generator output is already low so the signal is down in the noise level, you can turn down the receiver volume control or the oscilloscope gain.

In the third example you are to use a vtvm as the output indicator. Here you can use a modulated signal, and connect the vtvm as an ac voltmeter to measure the ac signal voltage between the plate of the output tube and ground. If the vtvm does not have a built-in blocking capacitor, you should connect a blocking capacitor between the voltmeter input and the plate of the output tube. You may also use the vtvm as a dc voltmeter and connect the voltmeter across the avc line. Here, as you adjust the alignment adjustments the amplitude of the avc will increase as the best alignment is obtained.

After connecting the output indicator in the correct position, the first step in aligning the receiver is to set the signal generator to the i-f frequency. If you are using a speaker, an oscilloscope or an ac-connected vtvm as an output indicator you should use a modulated output from the signal generator. If you are using the vtvm as a dc voltmeter and have it connected across the avc line, you can use an unmodulated output from the signal generator. Next, feed the output from the signal generator into the input circuit of the last i-f stage and then adjust the last i-f transformer; that is, the transformer between the last stage and the second detector for maximum output. Once you have performed this adjustment, move the signal generator to the input of the preceding i-f stage and then adjust the interstage i-f transformer to give you maximum indication on the output indicator.

The next step is to move the signal generator to the input of the mixer. The signal generator should still be set at the i-f frequency and if possible the local oscillator should be stopped. You can do this by temporarily shorting together the oscillator plates of the tuning capacitor. Now the input i-f transformer is adjusted for maximum indication on the meter.

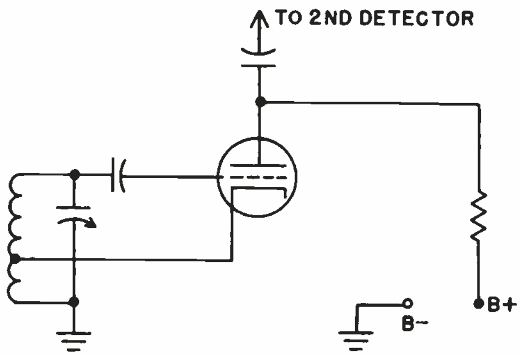
During the alignment procedure, as you move from the last i-f stage toward the input of the mixer, as the output indication on the output indicating device increases, you should reduce the setting of the signal generator output to use as small a signal as possible from the signal generator. If you use too strong a signal, some overloading may occur, and it may be impossible to accurately align the receiver.

After you have the i-f amplifier of the receiver aligned, with the signal generator still connected to the input of the mixer circuit, you should let the oscillator start operating and then tune the receiver to the high end of the band. Set the signal generator to approximately 1500 kc, in the case of an AM broadcast receiver, and tune the dial to 1500.

Now adjust the oscillator trimmer to make the signal come in at the correct point on the dial. Next, move the signal generator to the antenna input terminal and adjust the mixer trimmer for maximum output on the output indicating device. If the receiver has an rf stage you should now adjust the rf trimmer for maximum output.

Modern superheterodyne receivers have specially shaped plates in the oscillator section of the tuning capacitor, and therefore, if you have the i-f amplifier set to the correct frequency, the receiver will track across the broadcast band. However, it is a good idea to tune to the low end of the band and set the signal generator to some frequency around 600 kc. Check to see how closely the signal comes in at the correct spot on the band. If the signal is off substantially, you will usually find that one of the outside plates in the oscillator section of the tuning capacitor has slots in it. This is done to enable you to bend part of this plate slightly to try to correct the oscillator frequency so that it will track with the dial correctly.

203. Draw a BFO circuit diagram and explain its use in detection.



In the reception of cw signals, we need to provide a tone so the operator can recognize the various code characters being transmitted. A code transmitter simply interrupts the carrier so that the only signal being transmitted is the carrier. You know that the carrier is transmitted in a series of dots and dashes arranged to represent the various letters. The carrier alone may produce some hiss in the receiver but will not produce an audible tone. Therefore we use a BFO usch as the one shown, which is designed to operate at approximately the i-f frequency of the receiver. Some means is provided for varying the frequency of the BFO slightly. In the circuit shown, the capacitor in the resonant circuit is variable. The BFO frequency is tuned slightly off the i-f frequency so that when the signal from the BFO beats with the incoming cw signal a tone is produced. The tone is equal to the difference between the frequency of the two signals. Thus the dots and dashes being received by the receiver are turned into an audible tone that can be copied by the operator.

204. What would be the advantages and disadvantages of utilizing a bandpass switch on a receiver?

A bandwidth switch on a receiver permits the operator to adjust the bandwidth to the value required to receive the type of emission being received. For example, if you are listening to a standard AM voice-type transmission, you need a fairly wide bandwidth to pass the carrier and the two sidebands. However, if you are copying a cw signal, you need a bandwidth of only a few hundred cycles. Therefore by reducing the bandwidth of the receiver you can often eliminate another interfering signal. In addition, the narrower bandwidth usually cuts down on the noise, which is an advantage when copying a weak signal.

The narrow bandwidth has two disadvantages. If the receiver has any tendency to drift, you might lose the signal you are trying to copy. The second disadvantage is that in tuning the receiver to find a particular station, you must tune the receiver carefully, otherwise you could pass right over the station you are trying to locate. It is usually best to tune the receiver with the bandwidth somewhat wider than normal, and then after the station is tuned in correctly, the bandwidth can be reduced if this is desirable.

205. Explain: sensitivity of a receiver; selectivity of a receiver. Why are these important quantities? In what typical units are they usually expressed?

The sensitivity of a receiver is an indication of the ability of the receiver to produce a satisfactory output from a weak signal. The sensitivity of a receiver is usually measured in microvolts required to produce a standard output. In most cases a standard output is considered to be 50 milliwatts, and the sensitivity of the receiver, given in microvolts, indicates how weak a signal can produce this standard output of 50 milliwatts. For example, if the sensitivity of the receiver is 5 microvolts, this means that a signal of at least 5 microvolts is required to produce a standard output of 50 milliwatts. On the other hand, if the sensitivity of the receiver is 1 microvolt, this means that a signal of only 1 microvolt is needed to produce the standard output of 50 milliwatts.

The receiver's selectivity is a measure of the ability of the receiver to select the desired signal and reject undesired signals. For this purpose, the more selective the receiver the better it is. However, there is a limit to how selective a receiver can be, particularly for the reception of modulated signals. If the selectivity is too high, part of the sidebands will be lost and the intelligence of the signal may be destroyed. Of course, greater selectivity can be used for the reception of cw where only a relatively narrow bandpass is required. The selectivity is usually expressed in terms that indicate the attenuation of a signal a certain frequency from the desired frequency. For example, we might say that a signal 3 kc above or below the desired signal is so many db down. A signal that is 10 kc above or below the desired signal may be even further down in response. As an example, the response of a receiver may be stated as 3 db down, ±3 kc, 20 db down, ±10 kc. This means that a signal equal in amplitude to the desired signal that is 3 kc either above or below the desired signal will receive 3 db less amplification in the receiver. If the signal is 10 kc above or below the desired signal, but equal in amplitude, it will receive 20 db less amplification.

206. Explain briefly the principles involved in frequency-shift keying (FSK). How is this signal detected?

In frequency-shift keying instead of interrupting the carrier and turning the transmitter on and off to form a series of dots and dashes, the carrier is left on at all times. When we wish to send a dot or a dash, the carrier frequency is shifted so that the dots and dashes are formed by shifting the carrier of the transmitter from the fixed frequency to a new frequency. This is often done by means of a reactance tube connected across an oscillator. As the key is closed, the current through the reactance tube changes shifting the oscillator frequency. The output from the variable frequency oscillator is mixed with the output from a higher frequency crystal oscillator. The output from the mixer is then either mixed further to produce a higher output signal or simply amplified if the signal frequency is the desired transmitting frequency.

In the receiving FSK, any conventional superheterodyne receiver is used. The i-f signal is beat with a signal generated by a local BFO and fed to the second detector. Two audio tones are produced, one representing the dots and dashes, and the other simply representing the transmitter carrier. By tuning the BFO to exactly zero beat the transmitter carrier, the tone produced when there is no transmission and in the spaces between the dots and dashes will simply disappear and the only resulting tone will be the dots and dashes used for the various characters. Where two tones are produced by the mixing of the incoming signal and the BFO signal, a filter can be used to eliminate the tone produced by the uninterrupted carrier.

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NRI Study Guide For FCC License Examination

PART X

The material in this part of the study guide supplements the material that you have been studying in your lesson texts. Some of the material is a review of what you studied some time ago. It is reviewed here because of its importance and the likelihood that you will be asked questions on this material.

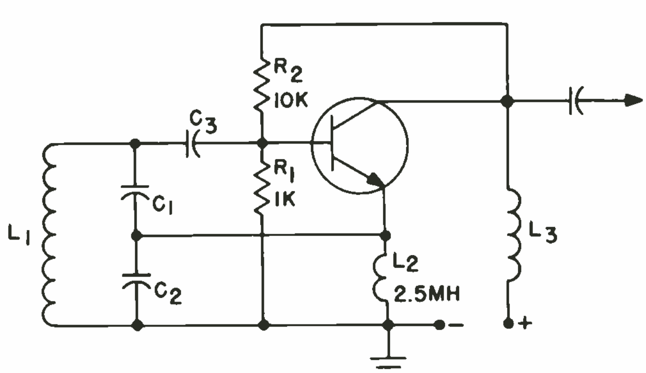
If you find that you need additional study in any of the material covered in this part of the study guide, be sure you go back to your lesson texts and do whatever studying is necessary. A little extra study now could mean the difference between success and failure in your efforts to get your FCC License. Do not ease off now, after you have gone this far.

Be sure to save this part of the study guide along with those you have received previously so you will be able to make a complete review before going for your license.

207. Draw simple schematic diagrams of the following transistor circuits and explain their principle of operation. Use only one voltage source; state typical component values for low power -10mc operation:

(a) Colpitt's-type oscillator
(b) Class B push-pull amplifier
(c) Common-emitter amplifier
(d) A PNP transistor directly coupled to an NPN type

(a) In the Colpitt's oscillator circuit the amount of feedback applied is divided by means of a capacitor-divider network. This is the distinguishing characteristic of the Colpitt's oscillator. In the circuit shown, capacitors C_1 and C_2 form the capacity-divider network and determine the amount of feedback. The total value of C_1 and C_2 connected in series are effective along with L_1 in determining the frequency at which the oscillator will operate.



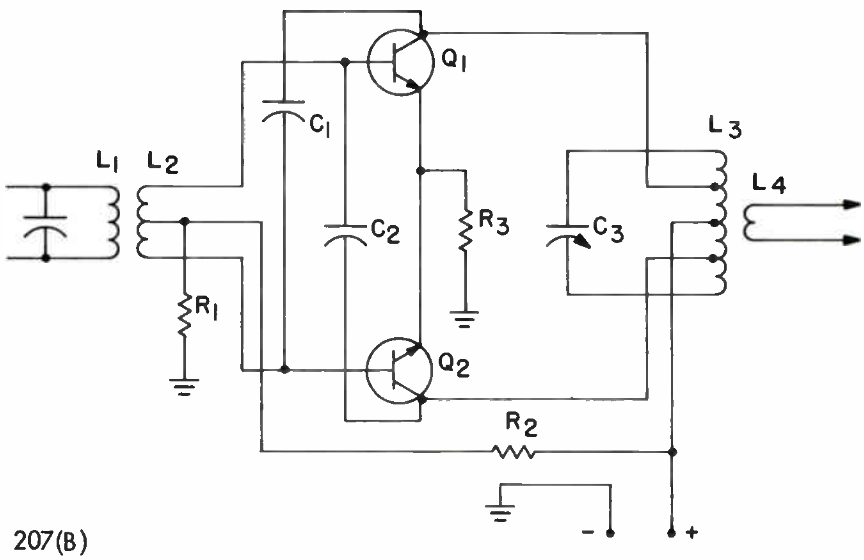
207(A)

In the operation of the equipment, when the equipment is first turned on, a small current will flow through the transistor due to the forward bias placed across the emitter-base junction by the divider network consisting of R_1 and R_2. This will cause a current to flow through L_2, the small rf choke that is used to complete the emitter circuit. Current flowing through this choke produces a voltage drop across it and this charges the capacitor C_2. The charge on capacitor C_2 will start an oscillation in the tank circuit which consists of coil L_1 and the two capacitors C_1 and C_2. The voltage developed across C_1 is applied to the emitter and the base; this is the feedback voltage. Energy is applied to the tank circuit by the transistor during the part of the cycle when the base of the transistor is driven in a positive direction. This causes a heavy burst of current through the choke L_2 which in turn supplies energy to C_2 which supplied energy to the entire resonant circuit. In the case of a Colpitt's oscillator using a PNP transistor, the energy would be supplied to the resonant circuit when the base is driven in a negative direction.

The resistor R_2 is used to supply a small starting forward bias across the emitter-base junction. Once the oscillator starts operating, the tank circuit voltage will drive the base in a positive direction and cause conduction through the transistor. However, if R_2 was omitted from the circuit, it is unlikely that the oscillator would be able to start oscillating.

(b) In the schematic diagram shown on the next page, the transistors Q_1 and Q_2 are connected in a push-pull rf power amplifier circuit. A small forward bias for the two transistors is provided by means of the divider network consisting of R_1 and R_2. The transistors used in this type of circuit may be either NPN or PNP transistors. We have shown NPN transistors. If PNP transistors were used, the polarity of the battery would be reversed.

The input transformer made up of L_1 and L_2 is a step-down transformer. This type of transformer is used in order to match the low input impedance of the transistor to the somewhat higher output impedance of the driver stage. Capacitors C_1 and C_2 are used to provide neutralization. Without these capacitors, it is quite likely that the amplifier would go into oscillation due to the internal capacities of the transistors.



207(B)

In some of the newer transistors the internal capacity has been reduced due to new design techniques and you may find these neutralizing capacitors are omitted.

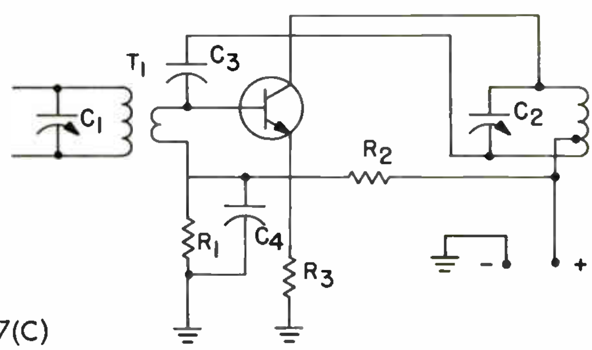
The resistor R_3 connected between the emitters of the transistors and ground is used to prevent thermal runaway. As the base-collector junction of the transistors heat, there is a tendency for a reverse current to flow across this junction due to the flow of minority carriers. This will have the effect of increasing the forward bias across the emitter-base junction and tend to cause the current through the transistors to increase. Any increase in current through the transistors will result in an increase in the voltage drop across R_3 which in turn will subtract from the forward bias across the emitter-base junction and hence prevent thermal runaway.

Notice that the collectors of the transistors are connected to taps on the output tank coil L_3. This is done in order to avoid excessive loading of the tank circuit. You will remember that the output impedance of a transistor is comparatively low when compared to that of a vacuum tube. A vacuum tube arranged in a circuit, of this type can be connected across the entire tank circuit without excessively loading the circuit. However, this is not the case with a transistor, therefore we tap down on the tank circuit coil to provide a better impedance match.

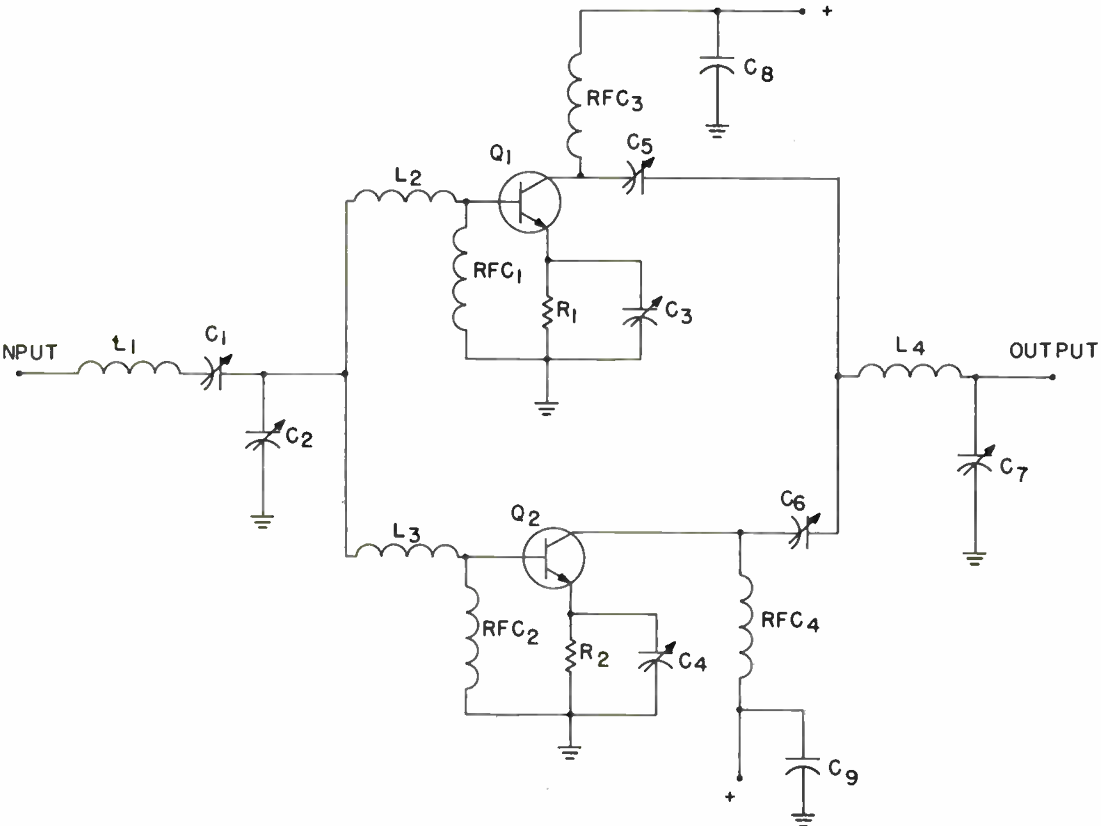
While you may find a Class B power amplifier operating in the 10mc range, it is doubtful that you will find this type of rf power amplifier used at much higher frequencies. Difficulty building suitable transformers to provide equal drive to the transistors and matching the transistors to the transformers generally makes this type of circuit impractical at the higher frequencies. A more practical arrangement is to connect the two transistors in parallel as in the second circuit shown. In this circuit on the opposite page, we have shown NPN transistors, once again PNP transistors could be employed, but the polarity is reversed.

(c) The common-emitter rf amplifier shown is typical of the type of amplifier you may find in the rf section of a receiver or in the i-f section. Forward bias for the emitter-base junction is provided by the voltage-divider network consisting of R_1 and R_2. R_3 in the emitter circuit is used to prevent thermal runaway. Notice that the input transformer, T_1, is a step-down transformer. Once again this is done in order to match the low input impedance of the transistor to the output impedance of the preceding stage. Neutralization is accomplished by means of capacitor C_3 which feeds a small part of the energy from the output circuit back into the base circuit to cancel the energy fed through the collector-base capacity of the transistor.

(d) A two-stage direct-coupled rf amplifier using a PNP transistor followed by an NPN transistor is shown. It is comparatively simple to connect a PNP transistor directly to an NPN transistor because of the opposite voltage requirements of

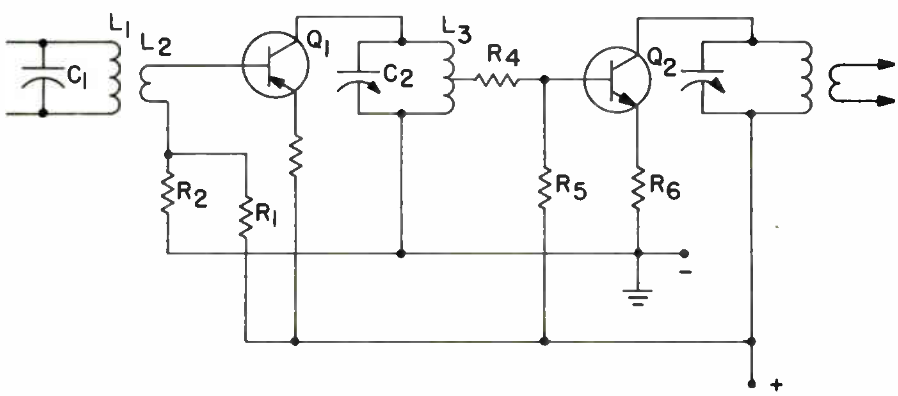


207(C)



the two transistor types. In the case of Q_1, the PNP transistor, forward bias is provided by the voltage-divider network consisting of R_1 and R_2. This network will provide a negative voltage on the base of the transistor with respect to the emitter voltage, and hence the requirement for forward bias across the emitter-base junction is fulfilled. The collector of the transistor connects through the tank coil, L_3, to the negative side of the power supply and hence we have a reverse bias across the base-collector junction.

The amplified signal is tapped off the coil L_3 and fed to the base of the transistor Q_2 through R_4. The voltage-divider network consisting of R_4 and R_5 will provide a voltage drop across R_4 that is greater than the voltage drop across R_6 so that the base of the transistor is slightly above (positive) the negative potential of the power supply. This will place the base of Q_2 positive with respect to the emitter, which is the condition for forward bias across the emitter-base junction of an NPN transistor. The collector of Q_2 comes back to the positive side of the power supply. The signal fed from the collector of Q_1 is fed directly to the base of Q_2 and amplified by the transistor and fed to the tuned circuit in the collector circuit of Q_2. In some circuits of this type you may find R_4 bypassed by a capacitor. The capacitor provides a low-impedance path for the signal, but does not affect the dc coupling between the two stages.

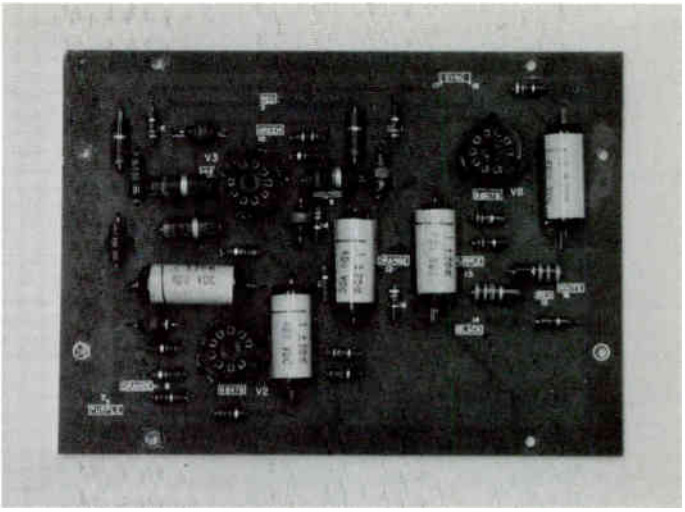


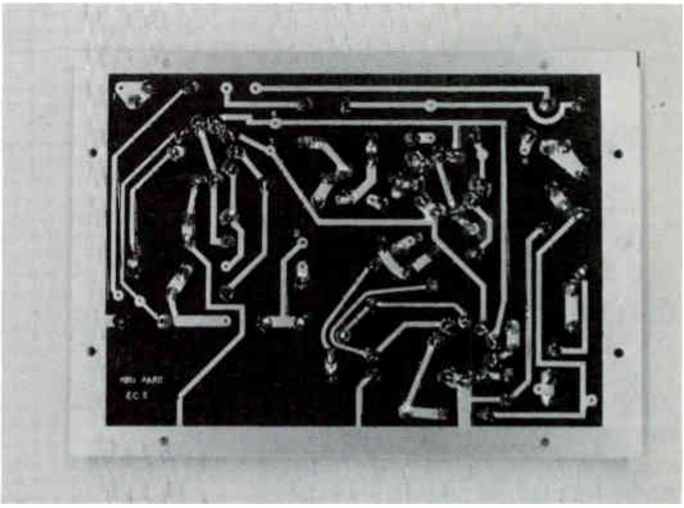
207(D)

208. Discuss etched-wiring printed circuits with respect to the following:

(a) Determination of wire breaks.
(b) Excessive heating.
(c) Removal and installation of components.

An etched-wiring printed circuit is a technique that is being widely used in modern equipment. In this type of wiring we usually start with what is called a circuit board. This board may be made of a phenolic type of material or it could be some glass type of epoxy material. The material is coated with a solid layer of copper. This layer of copper may be on one side, or it may be placed on both sides. The circuit desired on the board is then put on the board photographically. The circuit board is then placed in a chemical bath which etches away the undesired copper leaving only the copper required to make the various connections on the board. The board is then either drilled or punched to provide the means of inserting the resistors, capacitors, transistors or tube socket leads. Wiring is accomplished by inserting the parts through the holes and then soldering the parts to the copper remaining on the board. Most boards found in communications type equipment have the copper on only one side and the parts are mounted on the opposite side. In the photographs on the right, we have shown the top and bottom side of an etched circuit board so that you can see the components mounted on the one side and the etched wiring on the other.





Photos showing top (above) and bottom (below) of printed circuit board.

(a) Sometimes a circuit board will break. If part of the board simply breaks off or there is a large crack in the board, it is easy to see. However, sometimes a board will develop a hairline crack that is almost invisible. This type of defect is particularly annoying in mobile equipment because when the equipment is on the service bench and there is no vibration, there may be a good contact across the hairline crack and the equipment will work satisfactorily. However, when it is placed in the equipment in which it is used, the vibration will cause an intermittent opening and closing of the circuit and hence a malfunction of the equipment. This type of break can usually be detected by slightly flexing the circuit board. Sometimes you can see where the crack is and on other occasions by taking suitable voltage measurements as you flex the board you can determine which circuit is opening. In cases where you can't determine where the circuit is open the simplest method of affecting repair is to simply flow solder over the copper wiring on the board. The solder will simply bridge the crack in the wiring and establish a good contact across the crack.

Where the break in the copper is substantial and you can see it, sometimes you can repair the circuit simply by flowing solder across the break. However, when you can't do this, you can use a short piece of hookup wire across the break in the circuit. You simply solder one end of the hookup to the copper on one side of the break and then after it has cooled, solder the wire to the copper on the other side of the break.

(b) One of the problems in repairing etched circuit boards is that the copper may come up from the circuit board if you apply too much heat to the board. The copper is in effect glued to the board and excessive heat will break the bond between the board and the copper. Therefore it is important that you avoid applying too much heat when changing parts or repairing breaks in the copper wiring. Repairs should be made quickly and they should be made with an iron that has a comparatively low wattage rating. If the copper should come up off the board then you can usually repair it by using a piece of hookup wire to bridge across the connection from which the copper was removed.

(c) The removal and installation of components from a printed circuit board is not particularly difficult provided you use a reasonable amount of care. If you have to remove a resistor or a capacitor or any other part that has two leads the best method is to simply cut the leads to get the part out first and then remove the leads one at a time. You should remove the leads by heating the copper and then pulling the lead out from the phenolic or glass side of the board. This has a tendency to pull the copper down on the board. If you push the leads through from the phenolic side, there is a tendency for the lead to catch the copper and pull it up off the board. You should get the lead out as quickly as possible and as soon as you remove the lead, insert a round wood toothpick in the hole from which the lead was removed, quickly remove the soldering iron and let the board cool. After the copper and board cool, you can remove the toothpick and the hole will be free from solder.

When you have to remove a component having a large number of connections, such as a tube socket, often special heating type devices are available, enabling you to heat all the terminals of the socket at once so that these sockets can be removed. If you have to remove the socket by heating the terminals one at a time, by the time you have heated one terminal and moved on to the next the first terminal will have cooled off. This makes it an almost impossible task to get the socket out except by using your diagonals or cutters or some other device to cut the socket up so that you can remove the connections one at a time.

In installing the replacement component, the most important thing is to make sure that the holes through which the leads must pass are clean. Also be sure that the leads on the replacement component are clean. A good way to install a resistor or capacitor in the circuit is to insert the part in the circuit board in the position occupied by the original component. Make sure that the part is down against the board so that it will be held firmly in place. Then touch the soldering iron to the lead of the replacement component, apply a little solder to the lead and slowly move the iron down on to the board so that it will come in contact with both the lead and the copper on the board. As soon as the iron touches the board you can see that the solder will flow smoothly over the copper and the lead. Remove the soldering iron and allow the joint to cool. Do not hold the iron in place on the copper any longer than is necessary to make the solder flow smoothly over the board, otherwise you may get the board too hot and cause the copper to break loose.

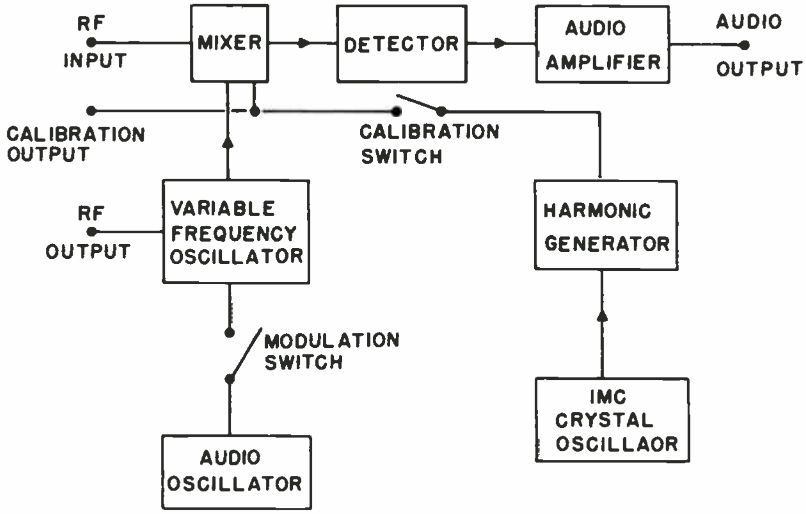
209. Draw a block diagram of a heterodyne frequency meter which would include the following stages:

Crystal Oscillator
Crystal-Oscillator Harmonic Amplifier
Variable Frequency Oscillator
Mixer
Detector and AF Amplifier
AF Modulator

Show rf-input and rf, af, and calibration-outputs. Assume a bandswitching arrangement and a dial having arbitrary units, employing a vernier scale.

(a) Describe the operation of the meter.
(b) Describe, step-by-step, how the crystal could be checked against WWV using a suitable receiver.
(c) Under what conditions would the af modulator be used?
(d) Describe, step-by-step, how the unknown frequency of a transmitter could be determined by the use of headphones; by use of a suitable receiver.
(e) What could be meant by calibration check points; when should they be used?
(f) If in measuring a frequency, the tuning dial should show an indication between two-dial frequency relationships in the calibration book, how could such a frequency value be determined?
(g) How could this meter be used as an rf generator?
(h) Under what conditions would it be necessary to recalibrate the crystal oscillator?

(a) In the block diagram shown, you can see that the frequency meter has a variable frequency oscillator. The signal from this oscillator is fed to a mixer and in the mixer is mixed with the incoming rf signal whose frequency you wish to check. In the mixer these two signals beating together produce new signals equal to the sum and difference of the two signal frequencies. The signals are fed to a detector which in effect eliminates the two original signals and the sum of the two frequencies leaving only the difference signal in the output. If the two signals are close enough together in frequency, the difference in frequency between the two will be an audio signal which is fed to an af amplifier and can be heard in the output by means of headphones. To check



209(A)

the frequency of the incoming signal the frequency of the variable frequency oscillator in the frequency meter is varied until this audio tone is heard. Then it is carefully adjusted until the frequency of the tone slowly drops down to a lower and lower frequency note and finally reaches such a low value that you can't hear it. In other words, we zero beat the variable frequency oscillator with the incoming signal. When we reach a zero beat, the variable frequency oscillator in the meter is tuned to the same frequency as the incoming signal. To determine the frequency of the incoming signal, you read the dial setting on the variable frequency oscillator, and then look that setting up in the calibration book supplied with the frequency meter. This will give the frequency of the incoming signal.

The accuracy of the variable frequency oscillator is checked by means of an accurate 1 mc crystal oscillator built into the heterodyne-frequency meter. The output from the crystal oscillator feeds to a harmonic generator so that there will be signals generated at every 1 mc. To calibrate the meter you simply close the calibrating switch and then tune the variable frequency oscillator across the dial. You should get a zero beat at every 1 mc calibration point on the variable frequency oscillator scale. If you do not, some means is generally provided either mechanically or electrically of adjusting the dial to give you the correct reading at every 1 mc check point.

When measuring an incoming signal, if you have no idea what the frequency is, the first thing to do is to find the signal and see where it comes in on the dial of the variable frequency oscillator in the meter. Then you should check the 1 mc check point above and below this frequency to make sure they are accurately calibrated. If not, make whatever compensation is necessary to get these two check points accurate and then go back and zero beat the unknown incoming signal and then read the variable frequency oscillator scale. Finally, look up the scale reading in the calibration book to get the frequency of the incoming signal.

(b) You'll notice that in the block diagram there is an output terminal provided from the harmonic generator. To check the calibration of the crystal oscillator the usual procedure is to tune the receiver to WWV and then feed the output from the harmonic generator to the receiver. You should find that the output gives you a zero beat on WWV. If you hear a very low frequency tone, it indicates that there is some slight discrepancy between the two signals. In other words, the 1 mc crystal oscillator is not operating on exactly 1 mc. It would only have to be off slightly to produce an audible tone if you checked it against WWV on 10 mc or higher. Provisions are made for correcting the crystal frequency; the frequency of a crystal can be shifted a small amount.

(c) The af modulator is useful when very accurate frequency measurements are required. For example, the response of the human ear falls off at low frequencies; some people can hear 25 or 30 cycles, some as low as 20 cycles, but frequencies below this value simply cannot be heard. When you have a zero beat with the incoming signal, you may actually be 20 or 30 cycles off frequency. To get a more accurate reading, turn on the audio oscillator. The tone from the audio oscillator will have a tendency to flutter; in other words it will grow louder and weaker at a rate equal to the difference between the incoming signal and the variable oscillator frequency. Therefore by carefully tuning the variable oscillator until the tone remains constant you will have the variable oscillator on exactly the frequency of the incoming signal. This may be important in measuring signals in the broadcast band, or very low frequencies where transmitters must be operated at tolerances of a few cycles, but at higher frequencies it is seldom possible to obtain a more accurate frequency indication by this method.

(d) We have already described the general procedure for using the frequency meter with headphones to determine the frequency of the transmitter. In using a receiver with the frequency meter, the station is tuned in on the receiver and then the output from the variable frequency oscillator is fed into the receiver antenna. You simply adjust the variable frequency oscillator to get a zero beat on the receiver and this will permit you to read the transmitter frequency from the variable frequency oscillator dial and the calibration book. Of course, you should check the dial accuracy with the crystal oscillator to insure maximum possible accuracy.

(e) Check points in the case of the meter shown in the block diagram are every 1 mc on the variable frequency oscillator dial. These are the points at which you can check the accuracy of the variable frequency oscillator using the crystal calibrator in the meter.

(f) Heterodyne frequency meters are supplied with a calibration book. The dial on the variable frequency oscillator may be calibrated from 0 to 500 or some other convenient figure. Usually a vernier is supplied along with the dial so that you can read the dial to four places.

These meters are supplied with calibration books to tell you what frequency each dial division indicates. For example, the meter might be used for measuring the frequency of a station operating in the 40-50 mc region. Suppose you tune in the station and you get a reading of 281.0 on the dial. You look in the calibration book and you see that they indicate that 42,246.1 kc should come in at 280.6 on the dial and 42,246.2 kc, the next frequency given, comes in at 282.2 on the dial. Then you know that the station frequency is somewhere between these two values. If you want to get a more accurate indication of the station frequency you'll notice that there are 1.6 dial divisions between the two stations for .1 kc (100 cycles). This means that each tenth of a dial division is equal to 100 ÷ 16 cycles. Therefore you can take 100 divided by 16 and multiply it by 4 and you get 25 cycles as your answer. Therefore the frequency of the station would be 42,246.125 cycles.

It is seldom that the dial reading you'll get on the frequency meter will fall exactly on one of these readings given in the calibration book and therefore it is generally necessary to interpolate as in the example given. If you are not interested in getting the frequency as accurate as possible and you may notice the dial reading is approximately halfway between the two check points, you can simply make a quick approximation of the frequency. However, if you want to get as close as possible then you have to go through the steps outlined.

(g) The meter can be used as an rf signal generator just as any other rf signal generator is used. The output from the variable frequency oscillator can be used as an accurate signal source in receiver alignment. You simply connect the output from the variable frequency oscillator to the receiver under alignment.

(h) The crystal oscillator frequency should always be checked whenever any repairs have been made in the crystal oscillator circuit. Such simple repairs as replacing a tube may affect the accuracy of the crystal oscillator.

The oscillator should also be checked if the unit is shipped or handled roughly. The accuracy should be checked on each measurement if there is any tendency for the line voltage to fluctuate. However, assuming a constant line voltage and careful handling of the equipment, the accuracy of the crystal oscillator need be checked only periodically. Some crystals have more of a tendency to drift with climatic changes than others. Usually it is a good idea in the case of a crystal oscillator to check it quite frequently until you become familiar with its characteristics and its tendency to drift. Then you can set up a regular routine for checking its accuracy depending on the needs of the particular crystal in the unit.

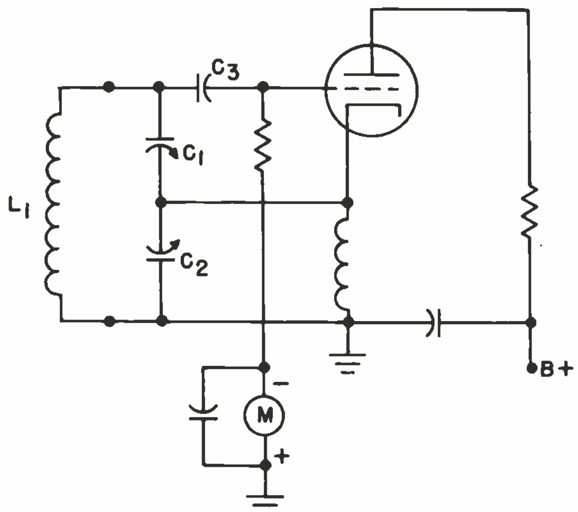
210. Draw a simplified circuit diagram of a grid-dip meter; explain its operation and some possible applications.

The grid-dip meter is simply an oscillator in which a sensitive meter is placed in the grid circuit. You know that when an oscillator is operating, the grid of the tube is driven positive and this causes grid current to flow. The amount of grid current that will flow depends upon the amount of feedback; the higher the feedback, the higher the oscillator grid current. In a grid-dip meter the coil in the oscillator tank circuit, L_1, is arranged on a coil form that can be plugged into the grid-dip meter. The coil is exposed so that it can be brought next to the resonant circuit you wish to check. Plug-in coils are used so that the grid-dip meter can be used over a wide frequency range. Some means of adjusting the frequency at which the grid meter oscillates is provided; usually the capacitor connected across the oscillator coil, C_1 - C_2 in the diagram, is a variable capacitor.

In use, the coil of the grid-dip meter is brought near the resonant circuit you wish to check. Then the frequency at the oscillator is varied by varying the capacity of the tuning capacitor. When you approach the resonant frequency of the circuit under test, the circuit will begin to take power from the grid-dip oscillator. When this happens, the amount of feedback in the grid-dip oscillator circuit will go down and hence the reading on the grid-dip meter connected in the oscillator circuit will go down. Usually as you tune the frequency of the meter toward the resonant frequency of the circuit under test, the grid-current meter takes a sharp dip, reaching minimum at the resonant frequency of the circuit.

Grid-dip meters are usually calibrated so you can determine the resonant frequency of the circuit. Of course, the frequency calibration is not extremely accurate, in addition, the resonant circuit has some tendency to pull the frequency of the grid-dip oscillator. However, a good indication of the approximate frequency can be obtained.

One particularly useful application of the grid-dip meter is in setting up the multiplier stages of a transmitter. Some multiplier stages might have such a wide range that they could be set either to double or to triple. You know approximately the frequency at which the stage should operate and you simply set the grid-dip meter to that frequency, and with the power on the transmitter off, tune the resonant circuit until the grid-dip meter shows a dip. Then you know that the circuit is tuned close to the correct frequency. In the case of the multiplier you could easily tell whether you

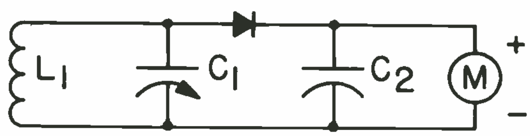


had the stages set to double or triple. In fact, you can go through the entire transmitter with the grid-dip meter and before you even apply the power to the transmitter, adjust all the resonant circuits in the transmitter close to the correct frequency.

Another useful application of the grid-dip meter is in locating circuits that may cause parasitics. For example, in the case of a transmitter operating in the 40 to 50 mc range, it is possible that there may be parasitic circuits which could cause oscillation above 100 mc. By using a grid-dip meter you can check for resonant circuits up above 100 mc. If you should encounter any indication of a resonant circuit then you can install suitable traps that will eliminate this problem.

Grid-dip meters can be used in receivers as well as in transmitters. You can check the approximate frequency of a resonant circuit in a receiver in the same way as you do in a transmitter. You can also locate the approximate resonant frequency of an antenna with a grid-dip meter simply by coupling the coil close to the antenna; at the resonant frequency of the antenna it will take power from the oscillator circuit and a noticeable dip will occur in the grid-current reading.

211. Draw a simplified circuit diagram of an absorption wavemeter (with a galvonometer indicator); explain its operation and some possible applications.



The absorption type wavemeter simply consists of a resonant circuit made up of L_1 and C_1. In some wavemeters a small incandescent bulb is connected across the resonant circuit. The meter is placed near an energized rf circuit and tuned to resonance. As resonance is approached, the bulb will light, reaching its brightest when the circuit is at exact resonance. Of course, there is a scale on the wavemeter that is calibrated in frequency and you can determine the frequency to which the meter is tuned from the scale.

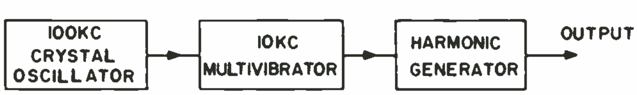
In the meter shown, instead of a bulb as an indicator we have a meter. Since a galvonometer is a dc device we have to use a rectifier to change the rf to dc. The rf is filtered by the small capacitor C_2 across the meter. In using an absorption wavemeter of this type, once again the meter is placed close to the resonant circuit and tuned to resonance. Resonance is indicated by a maximum reading on the meter. You have to be careful in the case of high power circuits to keep the meter from going off scale; in a situation of this type you simply move further away from the resonant circuit.

There are a number of applications for an absorption wavemeter. The most obvious is in checking the approximate frequency of the rf output from a resonant circuit. This is often helpful in the case of doublers or triplers; you can be sure that they are on the correct frequency. Another application is tuning for maximum output; the higher the reading on the meter the more the output from the resonant circuit. Absorption wavemeters can also be used when looking for parasitic oscillations. You tune the wavemeter for the frequency range where you expect the parasitic might exist. If there is a parasitic you will get an indication on the meter.

212. Draw a block diagram, showing only those stages which would illustrate the principle of operation of a secondary frequency standard. Explain the function of each stage.

The primary frequency standard in the United States is station WWV. This station operates on 2.5 mc, 5 mc, 10 mc, 15 mc, 20 mc, 25 mc and 30 mc. With WWV operating on so many frequencies it is possible to pick it up on at least one frequency, generally on two or three frequencies, in any part of the country, at any time of the day or night. Thus WWV is used as the primary frequency check. It is usually used mainly to check the accuracy of secondary frequency standards. A secondary frequency standard is shown in the block diagram. The oscillator is a 100 kc crystal-controlled oscillator. The crystal is maintained in a temperature-controlled oven so that it will remain at exactly the same temperature. The output from the crystal is fed to a frequency divider, usually consisting of a multivibrator or similar type of oscillator. The multivibrator is designed to operate at 10 kc. It is set up so that every tenth pulse from the 100 kc crystal oscillator synchronizes the multivibrator and keeps it operating at exactly 10 kc. Provisions are usually made so the 10 kc oscillator can be turned off to make it easier to identify the signal from the 100 kc crystal oscillator.

The output from the oscillators are fed to a harmonic generator so that check points are produced by the 100 kc oscillator at every 100 kc across any given frequency band. The 10 kc multivibrator provides check points at every 10 kc. By



using the frequency standard in conjunction with a communications-type receiver, or a frequency meter, the calibration of the receiver or meter can be checked every 10 kc and thus an accurate determination of the frequency of a signal may be made. Where necessary, another divider that divides the frequency down to 1 kc can be used. However, sometimes when using the frequency standards with a communications-type receiver, 1 kc markers are too close together and it is difficult to measure the frequency accurately.

In a frequency meter that is designed to cover only a limited range, a very accurate dial calibration can be made over the limited range that the meter is designed to cover. Then a secondary standard with a series of dividers such as shown can be used to calibrate the frequency meter at very close intervals. With such an arrangement, frequency measurements can be made accurate down to even a few cycles at comparatively high frequencies.

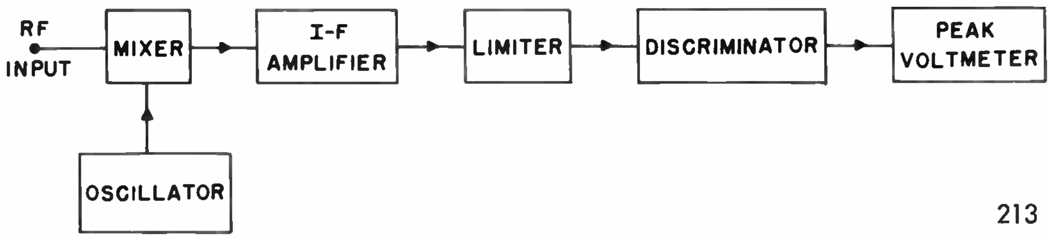
213. Draw a block diagram of an FM deviation (modulation) meter which would include the following stages:

Mixer
I-F Amplifier
Limiter
Discriminator
Peak Reading Voltmeter

(a) Explain the operation of the instrument.
(b) Draw a circuit diagram and explain how the discriminator would be sensitive to frequency changes rather than amplitude changes.

(a) As you can see from the block diagram, on the next page, of the FM deviation meter, the meter is quite similar in many respects to an FM receiver. However, instead of feeding the discriminator output to an audio amplifier and then to a loudspeaker, the output is fed to a peak reading voltmeter. The actual voltage that the meter indicates on peaks is of no importance; however, it can be calibrated to read directly in the frequency so that the deviation can be determined by reading the meter.

In operation, a signal from the FM transmitter is fed to the mixer in the frequency deviation meter, where it is mixed with a locally generated signal from the local oscillator. The frequency of the local oscillator is equal to the incoming signal frequency plus the i-f signal frequency so that a difference signal equal in frequency to the i-f frequency is produced. This signal is then amplified by the i-f amplifier.



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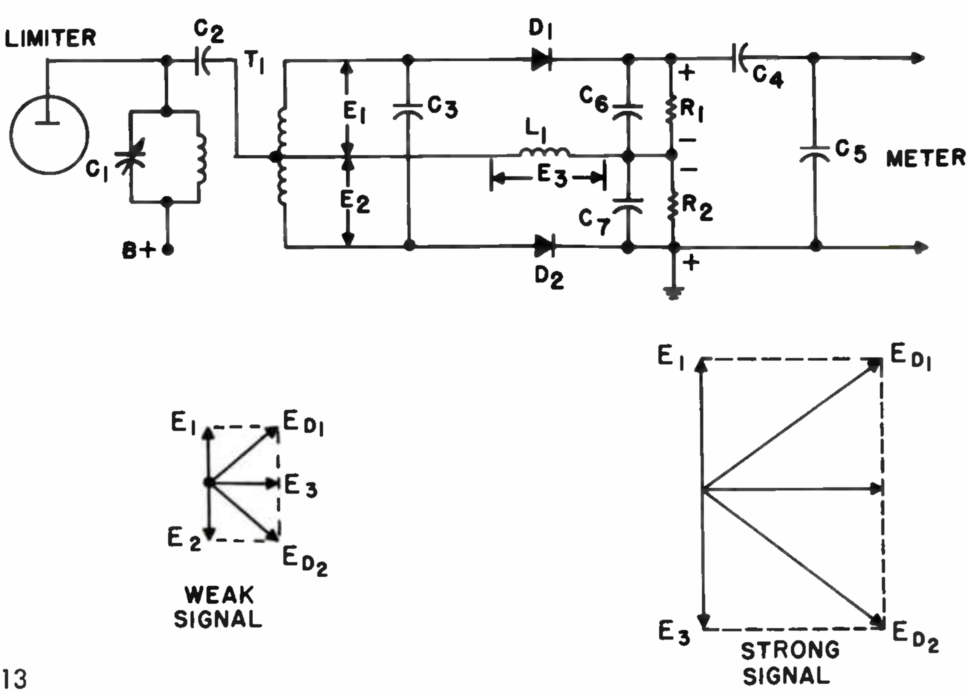
The output from the i-f amplifier is fed to a limiter so that a constant amplitude signal will be fed to the discriminator, regardless of variations in signal strength. The discriminator is a conventional FM discriminator that produces an output voltage as the signal frequency deviates. The greater the deviation, the greater the voltage produced by the FM discriminator.

The deviation voltage is measured by means of a peak reading voltmeter. The greater the deviation, the greater the voltage produced and hence the greater the indication on the meter. By calibrating the meter to read directly in frequency deviation rather than in voltage, the frequency deviation can be read directly from the meter.

(b) The schematic diagram is shown. The output produced by this discriminator will depend upon the frequency changes rather than amplitude changes. For example, the voltage applied to D_1 is equal to $E_1 + E_3$. The voltage applied to D_2 is equal to $E_2 + E_3$. If the amplitude of the signal increases then the amplitude of all three voltages will increase. The strength of E_1 and E_2 are dependent upon the current flowing through the primary winding of T_1. An increased signal current through this winding will induce a higher voltage in the secondary winding of T_1 and hence the amplitude of E_1 and E_2 will increase. At the same time, increased signal current will produce a higher signal voltage at the plate of the limiter, providing the limiting action is not complete and this would simply increase the amplitude of E_3. Assuming the signal remains on frequency, the net vector diagrams are shown as the signal amplitudes increase. Notice that while the voltage applied to D_1 increases with an increase in signal strength to the discriminator, the voltage supplied to D_2 also increases. Thus the voltage produced across R_1 increases as well as the voltage across R_2. Since these two voltages are 180° out-of-phase the net output from the discriminator remains zero.

However, when the frequency changes, the phase relationship between the voltages E_1, E_2 and E_3 changes, so that we have the results shown in the vector diagrams on the opposite page.

The amount of deviation will determine the phase shift of E_1 and E_2 and hence the ratio of unbalance between the voltage across R_1 and R_2. The ac voltage produced by the signal deviating back



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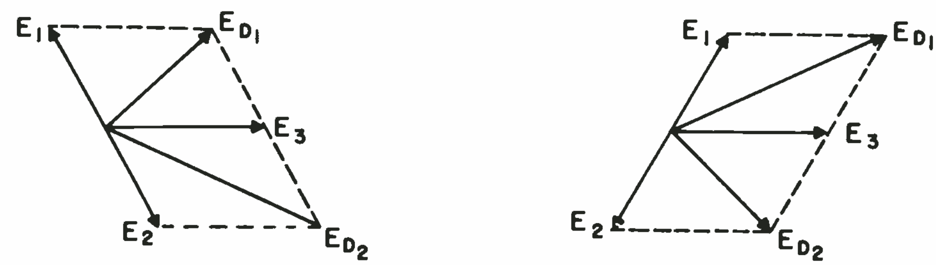

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and forth is fed through the coupling capacitor C4 to the deviation meter, which is a peak reading device. It indicates the peak voltage produced at maximum deviation.

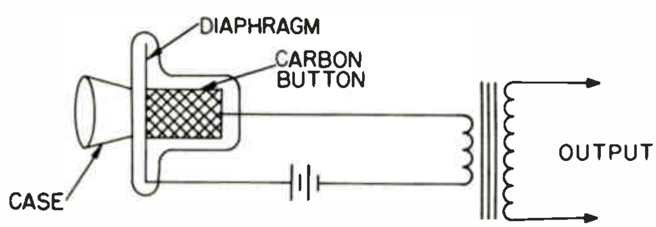
214. Describe a usual method (and equipment used) for measuring the harmonic attenuation of a transmitter.

The simplest method to measure the harmonic attenuation of a transmitter is by means of a field strength meter. A field strength meter is a simple receiver with a short antenna and a meter connected across the output in place of the speaker. The meter is calibrated to read in db.

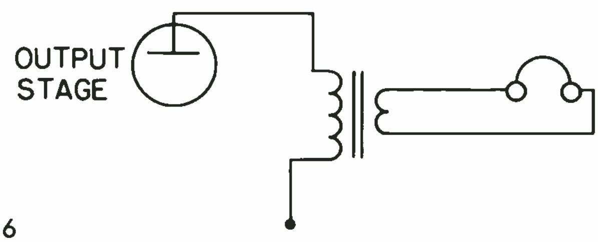
The procedure in measuring the harmonic attenuation is to check the signal strength of the transmitter on its fundamental frequency at a location some distance from the transmitter. The reading on the field strength meter is noted. The meter is then tuned to the second harmonic and the reading is noted. The difference between the two readings is the attenuation of the second harmonic. You can then tune to the third or fourth or higher harmonics and check the attenuation of these harmonics.

215. Draw a diagram of a single-button carbon-microphone circuit, including the microphone transformer and the source of power.

A carbon microphone consists of a button which is made of carbon granules which are tightly packed into a container. The diaphragm is placed against one side of the carbon crystals. An electrical circuit is established by means of a battery connected to the diaphragm. This causes a current to flow through the carbon granules and through the primary of the microphone transformer as shown. When we speak, the sound waves set up by our voices cause the pressure on the diaphragm to vary. This alternating increases and decreases the pressure on the carbon granules causing the resistance of the carbon button to change. This causes the current through the primary of the microphone transformer to change. The microphone transformer is a step-up transformer and the varying current flowing through the primary winding produces a stepped-up audio voltage across the secondary. The output from the secondary of the transformer is then applied to the first amplifier stage.



216. If low impedance head telephones of the order of 75 ohms are to be connected to the output of a vacuum-tube amplifier, how may this be done to permit most satisfactory operation?



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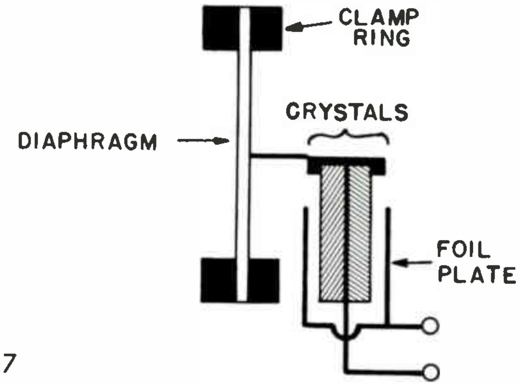
A 75-ohm headphone would present a very poor load to the plate circuit of a vacuum tube. Therefore to match the low impedance set to the plate of the vacuum tube a step-down audio transformer is used as shown in the schematic. The turns ratio of the transformer is selected so that the 75-ohm headset when reflected into the plate circuit of the vacuum tube presents the proper load impedance for the tube.

217. Describe the construction and explain the operation of a "crystal" type microphone. A "carbon button" microphone.

A crystal microphone is a piezoelectric device. This type of device operates on the principle that when stress is placed on a certain type of crystal an electrical voltage is developed. In the crystal microphone a slab of crystal is used and it is mechanically coupled to a metal diaphragm. When sound waves strike the diaphragm they cause it to



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vibrate to and fro and this in turn stresses the crystal which places a strain on it and causes it to induce a small voltage by the piezoelectric principle.

There are two types of carbon-button microphones: the single-button and the double-button microphone. In the single-button microphone carbon granules are packed into a single container and a diaphragm is placed against the open end of this container so that the diaphragm and the container seal the carbon granules inside of the unit called the button. The microphone operates on the principle that the sound waves striking the diaphragm cause the diaphragm to vibrate alternately increasing the pressure on the granules in the carbon button. This causes the resistance of the button to change and hence when we connect the carbon button in series with a battery in the primary winding of a microphone transformer, a varying current will be produced and this in turn will induce a voltage in the secondary winding of the microphone transformer.

In the double-button carbon microphone two carbon buttons are used and the diaphragm is placed between the two. The microphone is connected to a microphone transformer with a center tap as shown. In this type of microphone when the sound wave causes the diaphragm to move in one direction and compress the carbon granules hence reducing the resistance, the current through that button and one half of the primary winding of the transformer increases. Meanwhile the reduced pressure on the carbon granules on the other button causes the resistance of that button to increase so that current through the other half of the primary winding of the microphone transformer decreases.

Since the two currents are flowing through the transformer primary in opposite directions the two changes aid so that a somewhat higher output voltage may be obtained across the secondary of the microphone transformer.

The disadvantage of carbon microphones is that the battery current flowing through the carbon button does not remain exactly constant and as a result a hissing sound is produced by the microphone even when there is no sound signal present. The advantage of the carbon microphone is that it is rugged and has a relatively high output. By means of the microphone transformer an even higher output voltage can be made available to drive the first audio amplifier stage.

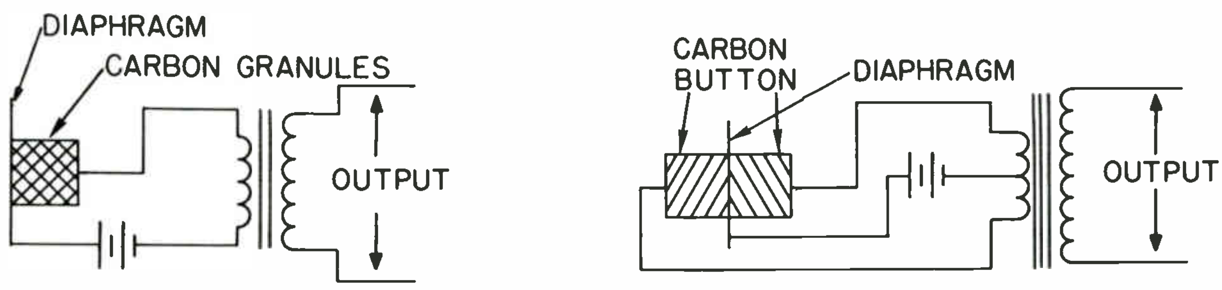
218. What precaution should be observed when using and storing crystal microphones?

Crystal microphones are easily damaged by excessive heat and/or high humidity. Therefore they should be stored in a cool dry place. In use, a crystal microphone should not be placed in direct sunlight or in any other hot or damp location. In addition, crystal microphones are quite fragile and you should avoid dropping them or striking them against any hard object. A sharp blow could easily crack the crystal.

219. Why is it important that transmitters remain on frequency and that harmonics be attenuated?

Transmitters are all assigned a certain frequency. For example, on the standard broadcast band, transmitters are assigned frequencies at every 10 kc. If the transmitter drifts off frequency, it may cause interference to another station operating on a nearby frequency.

The attenuation of harmonics is important because the second or third harmonic of a transmitter, or a strong higher order of harmonic for that matter, could cause interference to another station either in the same service or in another



Single-button microphone (left) and double-button microphone (right).

type of service. For example, a broadcast station operating on 600 kc could cause interference on another broadcast station operating on 1200 kc if it is radiating a strong second harmonic. Another broadcast station operating on 1500 kc could radiate a strong harmonic on 3000 kc or on 4500 kc and interfere with the services operating on these frequencies. Therefore with the radio spectrum as crowded as it is and with the tremendous demand for frequencies for the various types of services, it is essential that all transmitters remain on their correct operating frequency and that the harmonics from these stations be attenuated at least as much as specified by FCC Regulations.

220. How does a primary cell differ from a secondary cell?

A primary cell is a cell that provides electrical energy by using up the materials in the cell itself. The primary cell cannot be recharged. The ordinary flashlight cell where the acid in the cell eats up the zinc as electrical energy is produced, is an example of the primary cell.

A secondary cell is a cell where the chemical composition of the elements changes to produce electricity but the chemical action is the reversible action. This means that by putting electrical energy back into the secondary cell, the plates can be changed back to their original chemical composition and the cell used again to supply electricity. Thus the materials in the cell are not used up. The action of charging and discharging is reversible so that a secondary cell may be recharged.

221. What is the chemical composition of the electrolyte of a lead-acid storage cell?

The electrolyte of a lead-acid storage cell is a solution of sulphuric acid in water. In a fully charged storage cell the electrolyte will have a specific gravity of between 1.20 and 1.30 depending upon the temperature of the electrolyte. Normally the specific gravity of the electrolyte is checked by means of a hydrometer which indicates the charge on the battery. If the battery temperature is approximately 80°F, the specific gravity will be close to 1.3, but if the battery electrolyte is cold, a lower reading near 1.20 can be expected.

222. Describe the care which should be given a group of storage cells to maintain them in good condition.

In the lead-acid storage cell, one plate is made of sponge lead and the other plate of lead peroxide. If the cell is discharged to too low a value and both plates turn to lead sulphate, we refer to this as sulphation. If we permit this chemical action to go too far, you cannot reverse the action and recharge the battery. Thus the first step in caring for a storage battery is to avoid discharging the battery to too low a level.

In use, a certain amount of water will be lost from the battery due to the chemical action that occurs during the charging and discharging of the battery, and due to evaporation. Therefore it is important that the electrolyte be brought up to the proper level at regular intervals by adding water.

Storage cells should never be stored in a discharged condition. They should be fully charged when they are stored, and if they are to be stored for a long interval they should be taken out of storage and recharged at regular intervals. The conditions under which they are stored should be controlled, and they shouldn't be kept in either too hot or too cold a location.

223. What may cause "sulphation" of a lead-acid storage cell?

As pointed out previously, when a lead-acid cell is discharged, the sulphuric acid combines with the lead on the one plate and the lead peroxide on the other plate to form lead sulphate. If the cell is discharged to too low a level, excessive amounts of lead sulphate will be built up on the two plates so that the battery cannot be recharged. Therefore the primary cause of sulphation is discharging to too low a level.

224. What will be the result of discharging a lead-acid storage battery at an excessively high current rate?

When a battery is discharging a certain amount of heat is produced in the battery. If the battery is discharged at too high a rate, the battery will become too hot. This may cause the plates in the battery to warp and touch. Once this happens you have an internal short in the battery, and the battery is of no further use.

225. If the charging current through a storage battery is maintained at the normal rate, but its polarity is reversed, what will result?

If the charger is connected to the battery with the wrong polarity, the current flowing through the battery will discharge the battery instead of charging it. If this is permitted to continue too long, sulphation at the plates will result so that the battery will be discharged to such a low level that it will be impossible to reverse the action and recharge the battery.

226. What is the approximate fully charged voltage of of a lead-acid cell?

The voltage of a fully charged lead-acid cell under no load will be close to 2.1 volts. Thus the actual voltage of the so-called 6-volt storage battery will be approximately 6.3 volts and the voltage of a fully charged 12-volt storage battery under no load will be 12.6 volts.

227. How is the capacity of a battery rated?

Batteries are rated in ampere hours. This rating tells you how long a battery can be discharged at a given current. The rate at which a battery is being discharged can be determined by taking the current and multiplying it by the hours this current is drawn from the battery. For example, a 100-ampere hour battery can be discharged at a rate of 10 amperes for 10 hours or it could be discharged at a rate of 5 amperes for 20 hours before recharging would be required.

228. What steps may be taken to prevent corrosion of lead-acid storage cell terminals?

The terminals should be kept clean and covered with a light coating of a non-corrosive grease. You should avoid over-filling the cell with water to prevent spilling the electrolyte on the cell terminals.

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NRI Study Guide For FCC License Examination

PART XI

After you have completed your study of this part of the study guide and submitted answers to the accompanying group of questions, you will be ready to take your examination for your Second Class Radiotelephone Operator's License. If you have covered all of the rules and regulations in your regular studies, then the review of the technical questions that has been given you in the various study guides you have received should enable you to pass the Second Class Operator's License without any difficulty. Therefore, if it is convenient to do so, we suggest at this time that you go ahead and get your Second Class Radiotelephone Operator's License. This will mean that you will have to take Parts I, II and III of the FCC License Examination. After you have passed the examination on these three parts you will be given your Second Class Radiotelephone Operator's License and then very shortly, after you have completed the next three parts of the study guide, you will be ready to take your First Class Radiotelephone Operator's License test. If you already have the Second Class Radiotelephone Operator's License when you go for the First Class Examination, you do not have to repeat the examination given for Parts I, II and III. Thus the exam for the First Class License will be considerably shorter and will require less reviewing on your part. Therefore if you live in a large city where the examinations are given regularly, it will be easier to get your Second Class License now and then take the shorter test for your First Class License later. However, if you live some distance from the examining point, or if it is inconvenient to take the Second Class License examination now, there is no reason why you cannot go ahead and take the entire examination for a First Class Radiotelephone Operator's License at one time.

In either case, if you decide to go ahead and get your Second Class License now, or even if you decide to wait until later, this would be a good time to review the material covered in this and the preceding sections of the study guide. The time spent in review will be very worthwhile. You can't expect to retain all the material you have covered in your lessons and in the study guides without reviewing from time to time. Even after you have your license, you will find that you will want to review periodically.

Be sure to save this part of the study guide along with the parts you received previously so they will be available for review before taking your FCC license examination.

229. What determines the speed of a synchronous motor? An induction motor? A dc series motor?

The speed of a synchronous motor will be determined by the power-line frequency and the number of pairs of poles in the motor. For example, on a 60-cycle power line there will be $60 \times 60 = 3600$ cycles per minute. Therefore a synchronous motor with two poles has one pair of poles and its speed would be $3600 \div 1 = 3600$ rpm. If the motor has four poles then it has two pairs of poles and its synchronous speed would be $3600 \div 2 = 1800$ rpm.

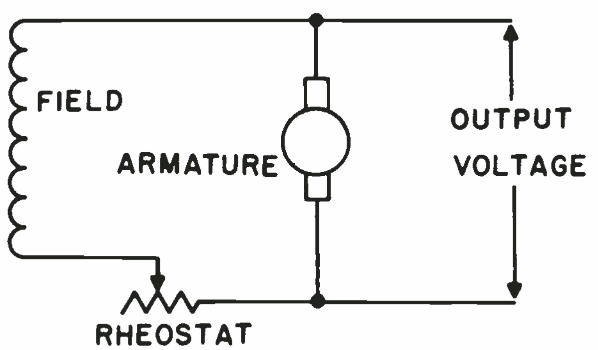
An induction motor runs at slightly below the synchronous speed. The slip between the rotating electrical field produced by the ac current in the field poles, and the speed at which the armature is rotating, produces the torque that makes the motor operate. The synchronous speed of an induction motor is the same as the speed of a synchronous motor. However, its actual operating speed will be slightly less than the synchronous speed, the greater the load the greater the difference between the synchronous speed and the operating speed.

The speed of a dc series motor varies inversely as the load. A dc series motor will operate at maximum speed with minimum load. As the load on the motor increases, the speed of the motor will go down.

230. Describe the action and list the main characteristics of a shunt dc generator.

As you can see from the schematic diagram in a shunt dc generator, the field is in parallel with the armature. The generator gets its name from the fact that the field is in parallel or in shunt with the armature.

In this type of generator there will be a small amount of residual magnetism maintained in the



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pole pieces of the generator. Thus as we start rotating the generator, a voltage is induced in the armature due to the residual magnetism of the field. This voltage causes a current to flow through the shunt field increasing the strength of the field and hence inducing a greater voltage in the armature. Thus the voltage builds up until the rated armature voltage is reached. The current flowing through the field is only a small portion of the total current produced by the armature because the shunt field has comparatively high resistance. The armature voltage may be regulated by controlling the current through the field by means of the rheostat that is in series with the shunt field.

The chief characteristic of this type of generator is reasonably good voltage regulation under varying load conditions.

231. Name four causes of excessive sparking at the brushes of a dc motor or generator.

Worn or dirty brushes will cause excessive sparking in both a dc generator and motor. Also, if the spring pressure on the brushes is too low so that they do not maintain proper contact with the commutator, excessive sparking will occur.

In both the motor and the generator the brushes must be set at the so-called neutral point so that when commutation occurs there is little or no current flow. Thus brushes that are not properly set at the neutral point will result in excessive sparking. Sparking may also be caused by either an overloaded motor or an overloaded generator.

232. How may radio frequency interference, often caused by sparking at the brushes of a high-voltage generator, be minimized?

Whenever a spark is produced at the brushes of a high-voltage generator, we actually generate a signal containing many different frequencies. This signal has a tendency to shock excite any resonant circuit. There are many resonant circuits formed by various circuits in the generator. This results in radiation which can cause radio-frequency interference.

Interference of this type can be kept at a minimum by the usual steps to obtain the best possible generator performance. By this we mean the brushes must be kept clean and with the proper pressure applied to them. The commutator must be kept clean and the brushes must be adjusted so that the commutation occurs at the neutral point. In addition, suitable bypass capacitors may be used across the brushes to keep the radiation at a minimum. In some cases, where these steps fail to eliminate the problem, a filter network in the output of the generator may be employed to keep the interference from getting out of the generator.

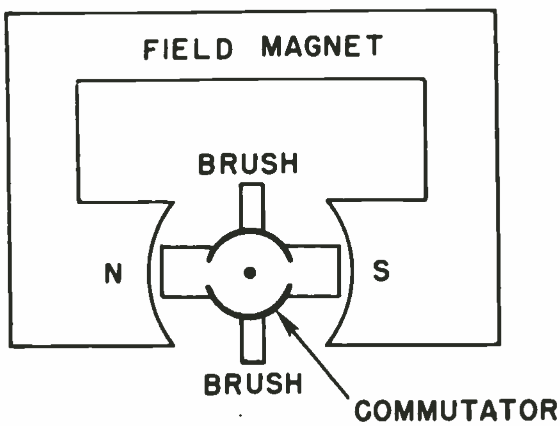
233. How may the output voltage of a separately excited ac generator, at constant output frequency, be varied?

In an ac generator we have a field that is excited by a dc source. The ac is produced by the armature rotating in the fixed polarity magnetic field. Instead of bringing the current out of the armature through the commutator and brushes, the current is brought out through slip-rings so we have the direction of current flow reversing each half cycle.

The voltage produced in the armature depends upon the strength of the field. Therefore the simplest method of controlling the voltage is to simply control the field voltage. Usually the field voltage is produced by a separate dc generator, thus by controlling the output of the dc generator we can control the current through the ac generator field and thus the output voltage of the ac generator.

234. What is the purpose of a commutator on a dc motor? On a dc generator?

In the diagram of the generator shown, the field polarity remains constant at all times. In other words, the one pole of the field magnet is always a north pole and the other is a south pole. As the armature rotates in this field, during one half revolution, current will flow through the armature winding in one direction. During the next half revolution, current will flow through the armature winding in the opposite direction. A commutator is used on the dc generator so that the current can be taken from the armature first with one polarity and then with the other. Thus in the external circuit, the current will always flow in one direction. Without the commutator



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and the brushes, the output from the generator would reverse and we would have ac instead of dc. The purpose of the commutator and brushes is therefore to enable us to generate dc.

In the dc motor, current must flow through the armature in one direction during one half revolution to cause the motor magnetic field to repel the stationary magnetic field. It must flow through the armature winding in the opposite direction, to cause the armature field to continue to repel the stationary field during the next half revolution. Therefore a commutator is needed to keep the current flowing through the armature winding in the correct direction in order to keep the armature rotating continuously. If the dc was applied to the armature through the sliprings, so that the direction of current did not change, the armature would rotate only part of a revolution. It would then reach a point where the two fields were lined up so that the north pole of the permanent field magnet would be lined up opposite the south pole in the armature and the south pole field magnet would be lined up with the north pole of the armature magnet. There would be no further rotation of the armature and hence the motor would not rotate. However, with the commutator, the current through the armature is switched so that when the armature reaches this position, where it would stop rotating, the direction of current through the armature is reversed causing the armature field to change and hence the field is repelled by the permanent field and rotation continues.

235. What may cause a motor-generator bearing to overheat?

The most frequent cause of a bearing overheating is insufficient or incorrect lubrication of the bearing. Overheating might also be caused by dirt in the bearings. If the motor-generator has been recently disassembled for repair and/or cleaning and the bearing overheats after if has been re-assembled, the overheating may be due to misalignment due to incorrect re-assembly of the motor-generator, assuming that the bearing has been lubricated properly.

236. What materials should be used to clean the commutator of a motor or generator?

The commutator may be cleaned with very fine sandpaper. Never use emery paper or emery cloth to clean a commutator.

Some technicians will clean the commutator of a motor or generator while the equipment is operating. They use a long piece of sandpaper about the width of the commutator and hold it against the commutator as the machine rotates. This is a very dangerous practice and is not recommended except in the case of low voltage equipment. It is much safer to clean the commutator when the equipment is off. This may take a little longer, but is certainly the preferable method, particularly in the case of a high-voltage generator or a motor that operates from a 115-volt or higher voltage power line.

237. If the field of a shunt-wound dc motor were opened while the machine was running under no load, what would be the probable result(s)?

In a shunt-wound dc motor, the armature and the field are connected in parallel and directly across the power line. The strength of the field will be determined by the power-line voltage, the resistance of the wire used to wind the field and the number of turns on the field. The armature, as it rotates, builds up a counter electromotive force (cemf) which opposes the line voltage. The difference between the line voltage and the counter electromotive force is the actual voltage which causes current to flow through the armature and produce the torque which causes the armature to rotate. Under no load conditions the cemf build-up is quite high so that the armature current will be low. The armature speed will be maintained within a safe value. However, if the shunt field is opened, there will be a certain amount of residual magnetism left in the field. However, the strength of the field will drop substantially. This means that the cemf in the armature will drop and thus the armature current will increase. This will cause the armature speed to increase. The armature speed may increase to such a high value, that the centrifugal forces produced, may cause the winding to fly right off the armature. If this doesn't happen, the high armature current may be sufficient to burn out the armature winding. Suitable protective devices should be incorporated to remove the voltage from the armature in the event that the field opens, to prevent the motor from destroying itself by operating at an excessively high speed or from drawing an excessive armature current.

238. The following are excerpts from a transistor handbook describing the characteristics of a PNP alloy-type transistor as used in a common-emitter circuit configuration. Explain the significance of each item.

Maximum and minimum ratings.

Collector-to-base voltage
(emitter open) .... -40 max volts.
Collector-to-emitter voltage
(base-to-emitter volts = 0.5) .... -40 max volts.
Emitter-to-base voltage .... -5.0 max volts.
Collector current .... 10 max ma.
Transistor dissipation at
ambient temperature of 25° C
For operation in free air .... 120 max mw.
At case temperature of 25° C
For operation with heat sink .... 140 max mw.
Ambient temperature range:
operating in storage .... -65 to +100° C

Collector-to-base voltage (emitter open). This is the maximum voltage which can be applied between the collector and the base without danger of breakdown of the collector-to-base junction. This voltage may be applied with the emitter open, in other words the emitter not connected to anything in the circuit.

Collector-to-emitter voltage (base-to-emitter voltage = .5). This is the maximum voltage that can be applied between the collector and the emitter of the transistor with a reverse bias applied across the emitter-base junction of .5 volts. If the voltage between the collector and the emitter exceeds this value, there is a danger of a breakdown occuring between the collector and the emitter.

Emitter-to-base voltage. This is the maximum forward-bias that can be applied across the emitter-base junction without the danger of exceeding the emitter-to-base or the emitter-to-collector current ratings of the transistor.

Collector current. This is the maximum safe collector current at which the transistor may be operated without the danger of overloading and hence destroying the transistor.

Transistor dissipation (25° C in free air). This is the maximum power which the transistor may dissipate without the danger of overloading and causing it to become so hot that it will break down where the surrounding air temperature is 25° C and no special means are provided for removing the heat from the transistor.

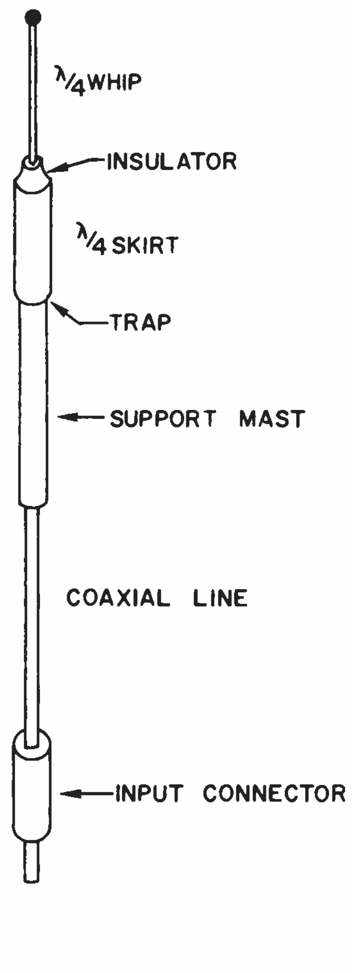
Transistor dissipation (at 25° C with heat sink). This is the maximum power that the transistor may dissipate without the danger of overloading and hence overheating the transistor where the transistor is used with a heat sink, and the surrounding air temperature is 25° C.

Ambient-temperature range (operating and storage). These are the design limits for both storage and operation of the transistor. The transistor must not be stored in any location where its temperature drops below -65° C nor must it be stored in a location where its temperature will exceed 100° C. Since these limits also apply to operating conditions, the temperature must not exceed 100° C when the transistor is operating. It is unlikely that the low temperature is of much concern under operating conditions because the normal heat dissipated by the transistor will keep it warm enough to prevent the low temperature from becoming of any great concern.

239. Draw a sketch of a coaxial (whip) antenna; identify the positions and discuss the purposes of the following components:

(a) Whip
(b) Insulator
(c) Skirt
(d) Trap
(e) Support mast
(f) Coaxial line
(g) Input connector

A sketch of the coaxial antenna is shown. The antenna is basically a half-wave antenna where the whip is one quarter wavelength long and the



skirt is one quarter wavelength long. The whip and the skirt are insulated from each other. The two elements both form part of the radiating system.

(a) The whip is connected directly to the center conductor of the coaxial cable. Thus it is simply an extension of the cable. The whip is made of a relatively thin piece of flexible material and it generally has a small bead on the end to minimize corona discharge. The whip is the top quarter wavelength of the antenna and forms part of the radiating system.

(b) The insulator is between the whip and the skirt. It prevents contact between the skirt and the whip and at the same time positions the whip in the center of the end of the skirt.

(c) The skirt is connected to the other conductor or ground conductor of the coaxial cable. The skirt is one quarter wavelength long and it functions as the other half of the half-wave radiating system. The overall length of the skirt and the whip will be slightly less than a half wavelength long to allow for end effect encountered in all antennas of this type.

(d) The skirt, as we pointed out is a quarter wavelength long. It is connected at the upper end to the outside of the connector on the coaxial cable. The outside conductor is at ground potential, thus in effect the upper end of the skirt is shorted. The quarter wavelength section of the skirt transposes the impedance at the opposite end. We have a low impedance at the end connected to the outside conductor of the coaxial cable and thus we have a high impedance at the open end. This high impedance essentially functions as a parallel-resonant trap and prevents any interaction between the skirt and the coaxial cable which is fed up through the center of the skirt.

(e) The support mast may be made of any suitable high strength material, and it must be insulated from the skirt. In the case of a mobile insulation in an automobile, the mast will be secured to some part of the automobile or truck so that it is grounded.

(f) The coaxial cable provides a means of transferring the rf power from the transmitter to the antenna. A 72-ohm transmission line is particularly suitable in applications of this type since the center impedance of a half-wave dipole is 72 ohms. Thus a good match is provided between the transmission line and the antenna and problems of standing waves on the transmission line can be avoided.

(g) The type of input connector used on the coaxial cable will depend upon the type of output connector on the transmitter. Usually a screw-type connector is used so that the cable can be inserted in the transmitter and then the coaxial cable connector is screwed on to the transmitter output terminal. This provides a good solid connection which won't work loose from vibration and at the same time provides a convenient means of disconnecting the transmission line from the antenna if repairs are needed either on the transmitter or on the antenna system.

240. What is "power factor"? Give an example of how it is calculated. Discuss the construction and operation of dynamotors.

Power factor is the cosine of the phase angle between the voltage and current in an ac circuit. When the voltage and current are exactly in phase, the phase angle between the two is zero. The cosine of the angle zero is 1, thus the power factor is 1. When the voltage and current in the circuit are 90° out-of-phase, the phase angle is 90°. The cosine of an angle of 90° is zero thus the power factor is zero. The power factor will vary somewhere between 1 and 0 depending on the phase angle.

The power factor gives us an indication of the ture power in the circuit versus the apparent power. For example, if a wattmeter, which measures the true power in the circuit, indicates that the power is 1000 watts whereas a voltmeter indicates that the voltage is 1000 volts and an ammeter indicates that the current is 1.5 amps we know that the power factor is somewhat less than 1. The product of the voltage times the current gives us a volt amp product of 1500. Therefore the power factor in the circuit will be

$$\frac{1000}{1500} = .667$$

Most ac circuits operate at a power factor somewhat less than 1 because of inductive and capacitive effects in the circuit. However, we usually try to keep the power factor as close to 1 as possible for most efficient operation of the equipment.

A dynamotor is a conbination motor and generator. The dynamotor has a single field and the motor and generator windings are placed on a common armature that rotates in the single field. The motor winding is connected to a dc source, usually a low voltage source. The generator winding usually has many more turns than the motor winding so that a much higher voltage is generated in the generator winding. Dynamotors are used in mobile equipment to provide the high dc voltage required to operate vacuum tubes from the comparatively low voltage available from the storage battery in the equipment. The motor and generator windings connect to separate brushes, thus we have a common field but separate motor and generator windings and brushes on a common armature.

241. List the comparative advantages and disadvantages of a motor-generator and transformer-rectifier power supplies.

The main advantage of a motor-generator type power supply is that it can be conveniently operated from either a dc or an ac source. In addition, comparatively good regulation can be obtained from a power supply of this type and it is usually comparatively easy to filter the dc output from the generator. The disadvantage of this type of supply is that it requires considerable maintenance. The rotating equipment must be kept properly lubricated, the brushes must be replaced at regular intervals and the commutator must be kept clean or excessive arcing will occur.

The transformer-rectifier type power supply must operate from an ac source. Of course, this can be overcome by means of a vibrator so that it can be operated from a dc source, but vibrators always prove to be relatively troublesome devices, particularly if the current requirements of the power supply are more than moderate. The transformer-rectifier supply is quite flexible inasmuch as it is easy to get any required voltage. In addition, if more than one voltage is required and it is not convenient to use a bleeder, separate taps can be provided on the power transformer and used with an additional rectifier to conveniently obtain the required voltages. As long as the design of the filter network is adequate the regulation from this type of power supply is satisfactory in most cases.

Transformer-rectifier power supplies require a minimum of maintenance although the filter capacitors required are relatively large if pure filtered dc is required from the output of the supply.

242. Discuss the cause and prevention of interference to radio receivers installed in motor vehicles.

The greatest source of interference to radio receivers in motor vehicles is the automobile ignition system. Each time a cylinder is fired, there is a spark generated by the ignition system and this causes considerable static radiation. Spark suppressors can be used on the spark plugs to keep this interference to a minimum. The spark suppressors are simply a resistance-type device that prevent the production of a sharp, hot, short-duration spark and tend to produce a longer somewhat steadier spark. They actually improve combustion in the cylinders and also reduce interference.

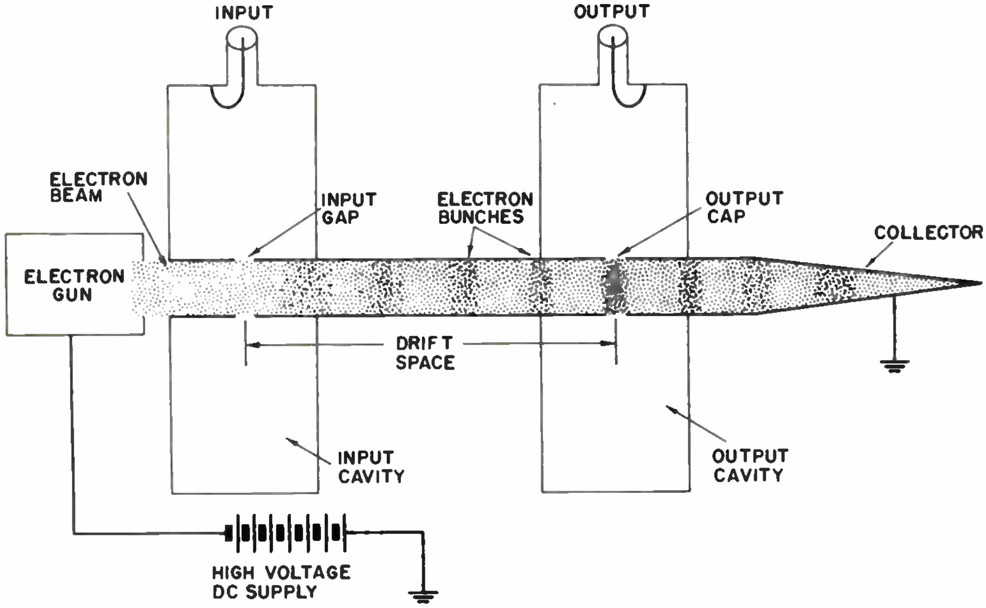
Generators and voltage regulators in the automobile can also cause interference. This type of interference can usually be eliminated by the use of suitable filter capacitors. In most automobiles one side of the generator is grounded. Connecting a capacitor such as a .5 mfd capacitor from the hot side of the generator to the car frame, which is grounded, should eliminate any generator noise. Similarly bypassing the voltage regulator by means of a suitable filter capacitor (.1 mfd to .5 mfd) should eliminate regulator noise.

In some of the older cars where the fenders and different body parts were rivetted to the frame, scraping and motion of these parts could cause static. However in the case of most modern vehicles the different body parts are welded into a unit type of construction so this type of interference is not so prevalent. However if you should run into it in an older type vehicle, a suitable bonding strap to bond the parts of the car to the main frame to prevent the generation of static electricity will eliminate this type of interference.

Of course, the antenna lead-in should be shielded and the shield grounded to the car frame. Any pickup on the lead-in is thus avoided and since the antenna itself is outside of the metal shell of the car it is not likely to pick up any additional interference. The metal shell and frame of the car in most cases provides adequate shielding between the antenna and the parts in the automobile ignition system.

243. Describe the physical structure of a klystron tube and explain how it operates as an oscillator.

The basic klystron consists of an electron gun, two cavities, a drift space between the cavities and a collector. The electron gun is similar in construction to the electron gun in a cathode ray tube inasmuch as it contains a cathode or an electron source, and also focus electrodes that focus a beam of electrons that travel down the length of the tube. A high negative potential is applied to the electron gun. This potential repels the electrons from the gun, down the length of the klystron tube. The collector is connected to the positive side of the power supply and re-



Basic Klystron Tube

turns electrons arriving at the collector to the positive terminal of the power supply.

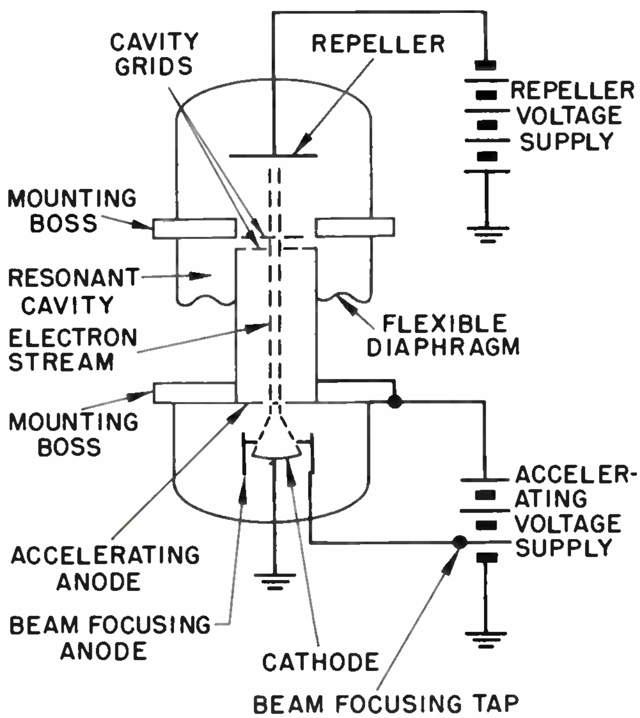
In operation, the input signal is applied to the input cavity. There is a small gap in the klystron called a window, and labelled the input gap. RF energy from the input cavity is coupled into the klystron tube through this gap. The rf signal, as it goes through its cycle, first slows down the electron beam travelling down the tube and then during the next half cycle speeds up the electron beam. Thus the electrons travel down the tube in bunches containing large amounts of energy.

The output cavity is similar to the input cavity except that it removes energy from the electron stream. A large amount of energy can be removed because of the high energy in the electron bunches and also because of the high Q of the output cavity. Electrons arriving in bunches more or less set up an oscillation in the output cavity, and large amounts of energy can be taken from the electrons because they are travelling at a high velocity and contain a great deal of energy. The modulated or bunched electron stream more or less acts as a very high current that induces a signal in the output cavity.

Klystrons make excellent vhf and uhf amplifiers. They can be made to handle large amounts of power. Peak cathode currents of several hundred amperes will be found in some large klystrons used in high power transmitters.

The Klystron tube can be used as an oscillator by feeding energy from the output cavity back into the input cavity. An attenuator is generally used between the output cavity and the input cavity to limit the amount of signal fed back from the output to the input.

Another type of klystron is the reflex klystron. The reflex klystron has a cathode and a beam focussing anode. The electrons are focussed into a narrow beam and then accelerated by an accelerator grid toward the cavity grids. The electron beam passes through the cavity grids which velocity-modulates the stream of electrons as in the case of the klystron described previously. The frequency at which the electron beam is modulated depends primarily upon the resonant frequency of the cavity. The cavity grids used in the reflex klystron provide better



Reflex Klystron Tube

coupling to the electron stream than the slots provided in the drift tube of the two-cavity klystron described previously. Reflex klystrons are usually low-power devices and therefore there is no danger of the electron streams burning up these grids.

The modulated electron stream travels down the tube, but instead of having a collector, at the far end of the tube with a positive voltage applied to it, we have a repeller with a negative voltage applied to it. The high negative voltage turns the stream around and the electrons pass back to the cavity grids. The trip towards the repeller and back has the same purpose as the drift space in the two-cavity klystron; it allows the velocity-modulated electron time to form bunches. The bunches arrive back at the cavity grids in such a way that they feed rf energy into the klystron cavity, just as the bunched electrons of the two-cavity klystron fed rf energy into the output cavity. In this manner a repeller electrode permits a single cavity to act both as the input and output cavity. For this reason, we can refer to the grids of the cavity as buncher grids, when they modulate the electrons moving toward the repeller, and as catcher grids when they receive energy from the electrons turning from the repeller.

Reflex klystrons are excellent uhf oscillators in applications where low power is required.

244. Draw a diagram showing the construction and explain the principle of operation of a travelling-wave tube.

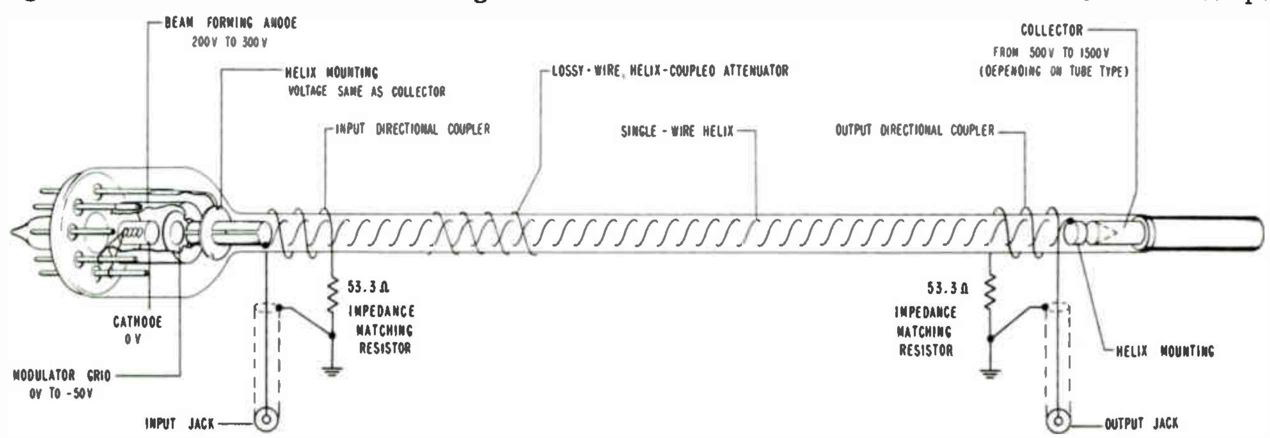
In the diagram shown you see that the travelling-wave tube has an electron gun at the left of the tube. The gun consists of a cathode made in the form of a cylinder closed at one end. The cathode, which has a heater inside of it, emits electrons when it is heated, from the oxide coating on the end of the cylinder. The electrons are accelerated down the tube, in a beam, by a beam-forming anode, which is shown in the diagram. Between the cathode and the beam-forming anode is a grid shaped like a washer, with a hole in the center. The grid can be used to control the intensity of the electron beam travelling down the tube.

The electrons are attracted down the tube to the extreme right of the tube by an anode called the collector. This anode has a high positive voltage applied to it.

In operation, the tube is placed inside a long, round magnet with a hole in the center. The magnetic field keeps the electron beam in the center of the tube so that it will travel down the center to the collector. This magnet is usually an electromagnet, and the current through it is carefully controlled to give exactly the correct field strength to keep the electron beam travelling exactly down the center of the tube.

In operation, the electrons travelling down the tube are bunched as they are in the two-cavity klystron. Notice that there is a coil called a helix wound inside the long neck of the tube. The helix is supported between two supports, one mounted nest to the electron gun and the other mounted next to the collector. Notice also that at the left end of the helix is a small coil marked "input directional coupler", and at the right end is another coil marked "output directional coupler".

The signal to be amplified is fed to the input directional coupler. This coupler is inductively coupled to the helix and couples the signal into the helix. The signal now travels down the neck of the tube along the helix as it would along the transmission line. Because the helix is wound like a spiral, the distance around the helix, in other words the distance the waves actually travel, is much greater than the distance between the turns of the helix. Therefore even though the signal is travelling along the wire in the helix at a very high speed, it is moving down the tube at a somewhat slower rate. The exact speed of



the signal down the tube depends upon the ratio of the distance between successive turns of the helix and the distance around the helix. Usually this is a ratio of about one to thirteen.

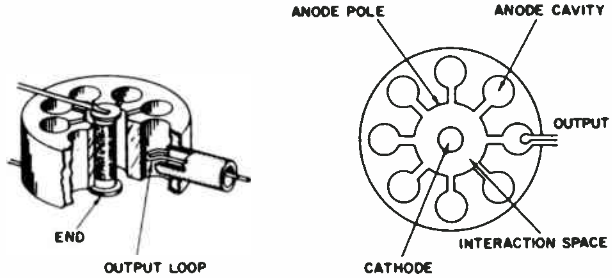
The signal moving down the tube on the helix produces an electric field which moves down the long neck of the tube. This electric field moves down the neck of the tube at a slightly slower rate than the electrons are travelling down the neck of the tube. When electrons travel through an electric field, they are slowed down as they approach the field, and then are accelerated after they pass through it. Thus the electrons travelling down the neck of the tube at a slightly higher rate than the electric field are continually approaching electric fields and hence are alternately accelerated and slowed sown by this field. Thus we have a bunching of the electrons as they travel down the neck of the tube. The bunching is increased as the electrons travel from the cathode to the anode so that at the anode end of the tube the electrons are bunched closely together.

The electrons that are slowed down as they travel down the tube, transfer energy into the electric field. When the electrons are accelerated by the field, they absorb energy from the field. Since the average speed of the electrons in the beams is greater than the speed of the electric field down the tube, there will be more electrons subjected to the retarding force of the field than there will be to the accelerating force of the field. Therefore the travelling wave will absorb more energy from the electron beam than it will give to the electron beam. Therefore the energy in the travelling wave grows. As the electrons are more and more tightly packed into bunches they deliver more and more energy to the wave. Therefore a radio wave of higher and higher amplitude results as the wave moves down the helix towards the anode.

The energy in the electric wave travelling down the helix is coupled to the output directional coupler and then fed through a transmission line to the load.

245. Describe the physical structure of a multi-anode magnetron and explain how it operates.

The construction of a magnetron is shown. As you can see the magnetron is basically a special type of diode. The magnetron is placed in a strong magnetic field which runs through the magnet parallel to the cathode and the anode cavities. The frequency at which the magnetron will oscillate will be determined primarily by the dimensions of the anode cavities and to some extent



upon the strength of the magnetic and electric fields within the tube. The magnetic field of course is controlled by the strength of the magnet and the electric field is controlled by the voltage applied between the cathode and anode of the magnetron.

In operation, the interacting electric and magnetic fields cause the electrons to trace circular paths just outside of the slots leading to each cavity. The cavities are shock excited into oscillation by the electrons travelling past the slots. Once the oscillation has started, and an electron beam moves past one of the cavities, the field across the cavity must have a polarity which slows down the electron stream. When this happens the stream gives up energy to the cavity to sustain oscillation. Energy is removed from one of the cavities and fed to an external load.

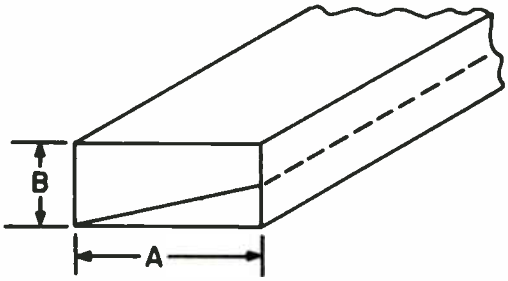
246. Discuss the following with respect to waveguides:

(a) Relationshp between frequency and size.
(b) Modes of operation.
(c) Coupling of energy into the waveguide.
(d) General principles of operation.

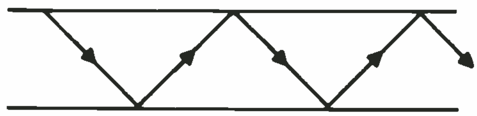
(a) The waveguide dimensions are determined by the frequency at which the guide is to be operated. In the example shown on the next page, A, which is the wider dimension, must be greater than one-half wavelength at the operating frequency, and B, which is the narrower dimension, must be less than one-half wavelength at the operating frequency. The length of the waveguide of course will depend on how far we want to transmit the rf signal. The length is not critical, but the longer the guide the greater the attenuation.

(b) Energy is propogated down a waveguide inside of the guide. The modes tell how the electric and magnetic fields arrange themselves inside of the waveguide. A TM mode (transverse magnetic mode) has the magnetic field in the direction transverse to the direction of propogation. A TE mode (transverse electric mode) has its electric field transverse to the direction of propogation.

(c) Coupling of energy into the waveguide is



WAVEGUIDE
DIMENSIONS



WAVE TRAVELLING
DOWN GUIDE

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usually done by means of a small probe or small loop inserted into the guide.

(d) The waveguide is able to completely contain the electromagnetic waves within its boundaries. You can more or less picture the waveguide as two parallel surfaces with the wave bouncing down the guide as shown. The energy is transmitted down the guide completely in the form of an electric wave rather than in the form of a current along the guide. No energy escapes from the guide through the waveguide walls although there is a certain amount of attenuation as the wave travels down the length of the guide.

247. Describe briefly the construction and purpose of a waveguide. What precautions should be taken in the installation and maintenance of a waveguide to insure proper operation?

A waveguide is a hollow rectangular or circular tube. The dimensions of the rectangular waveguide are determined by the frequency of the signal to be transmitted down the guide. The length of the guide is determined by the distance over which the rf wave must be transmitted. The purpose of the waveguide is to transmit rf energy in the form of a wave from one point to another.

There are a number of precautions that should be observed in the installation and maintenance of a waveguide to insure proper operation. First, in the installation, the inside of the guide must be kept clean, and any foreign matter that may fall into the guide should be removed, otherwise it will increase the attenuation of the guide. Long horizontal runs should be avoided since there is a tendency to collect moisture which would absorb rf energy. It is better to run the guide on some small angle and provide a means, if possible, for draining the guide so that any moisture can be removed. Joints in sections of the guide should be tight, both to prevent loss of rf energy and to avoid moisture seepage into the guide. At the bottom of a vertical run, some means should be provided for draining the guide in the event that moisture may get into the guide.

248. Explain the principles of operation of a cavity resonator.

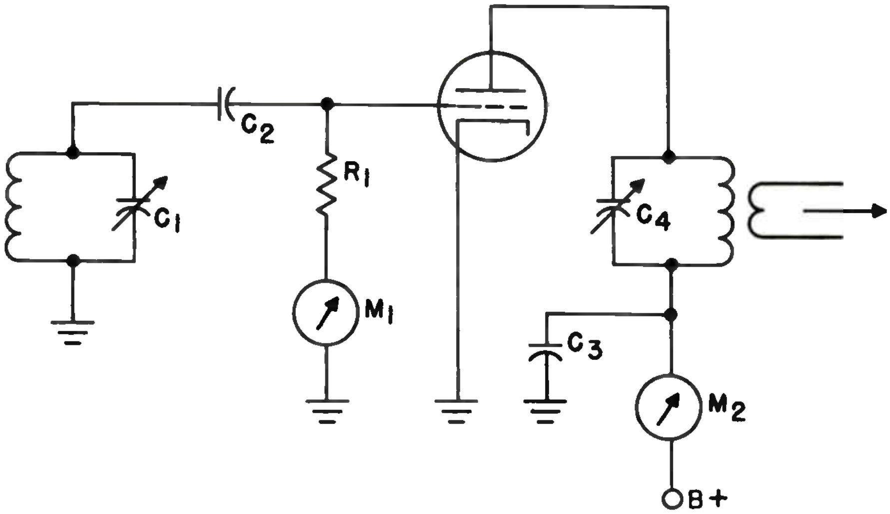
A resonant cavity may be rectangular, cylindrical or of a similar shape. For example the cavities in a magnetron are cylindrical-type cavities. A section of a waveguide may also be used as a resonant cavity. A resonant cavity is a small resonant circuit with a very high Q. The frequency at which the cavity is resonant will depend upon the size of the cavity. Oscillation in the resonant cavity is set up by a passing field, which may consist of a group of electrons passing by the entrance of the cavity. Oscillation in the cavity is maintained by successive groups of electrons passing by the entrance to the cavity at the correct time interval to reinforce the oscillation set up with the cavity.

249. How are cavities installed in vertical waveguides to prevent moisture from collecting? Why are long, horizontal waveguides not desired?

Cavities may be installed in vertical waveguides by means of sealed joints to prevent moisture from collecting. Also the waveguide can be pressurized by means of some dry inert gas, which will prevent moisture from collecting in the guide or the cavity. Long horizontal waveguides are not desired because any moisture that may be collected in the waveguide may settle at the bottom of the horizontal run. It may be difficult to get this moisture out of the guide and if it isn't removed it will cause excessive attenuation in the guide.

250. If in the circuit shown, which represents a tuned-plate, tuned-grid oscillator, C_1 should short, what would happen to the meter reading on M_1? What would happen to the meter reading on M_2? If R_1 should burn out, what would happen to the meter reading on M_1? What would happen to the meter reading on M_2?

If C_1 should short, then the resonant circuit in the grid circuit of the tube would be shorted out. This would mean that the oscillator would stop oscillating. When the oscillator stops oscillating



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there would be no bias developed across R_1 because there would be no grid current flow through the resistor. Therefore the meter M_1 which would normally show grid current would drop to zero. Since the bias on the tube would disappear, the plate current flowing through the tube would increase. Therefore the reading on meter M_2 would increase.

The resistor R_1 is the grid return resistor. If this resistor were to burn out, electrons attracted to the grid of the tube would charge the capacitor C_2 with a polarity such that the grid end of the capacitor would be negative. This charge would build up such a high value that eventually the flow of plate current through the tube would be cut off because there would be no way for C_2 to discharge. Therefore, with R_1 open, there is no current flow in the grid circuit, the reading on the meter M_1 would drop to zero. Similarly, since the high bias built up across C_2 would cut off the flow of plate current, the reading on the meter M_2 would also drop to zero.

National Radio Institute

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T-11: FCC

NRI Study Guide For FCC License Examination

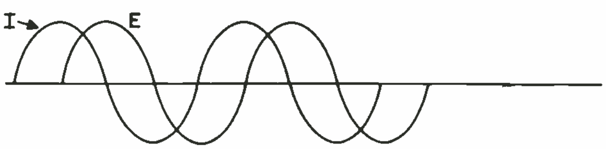
PART XII

The questions found in this section and in the following two sections of the study guide all deal with questions that may be asked in Element IV of your FCC License Examination. Element IV is the element that holders of the Second Class Radiotelephone Operator's License must pass in order to obtain a First Class Radiotelephone Operator's License.

Most of the questions covered in this and the following two sections of the study guide are not too difficult. However, be sure that you go over each question carefully and that you understand the material covered. If you find that there is something which you do not understand, be sure to spend whatever time is necessary to review the subject. Don't let up now; you are very close to reaching your goal of a First Class Radiotelephone Operator's License.

Save this section of the study guide along with those you have already received, so you can review them before taking your FCC License Examination.

251. Show by a simple graph what is meant when it is said that the current in a circuit leads the voltage. What would cause this?



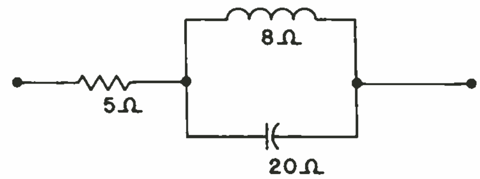
In the example shown, the current leads the voltage by 90°. This situation would be encountered in a purely capacitive circuit. In most circuits where the current leads the voltage, the current will lead the voltage by somewhat less than 90°. This, once again, is caused by a capacity in the circuit. The amount of capacitive reactance will affect the phase angle. If a circuit is a pure capacitive circuit, the current will lead the voltage by 90°. If a circuit is a pure resistive circuit, the current and voltage will be in phase.

In a circuit containing both capacity and resistance, the phase angle will be something between 0° and 90°, with the current leading the voltage.

252. List the fundamental frequency and the first ten harmonic frequencies of a broadcast station licensed to operate at 790 kHz.

Fundamental 790 kHz	Sixth harmonic 4740 kHz
Second harmonic 1580 kHz	Seventh harmonic 5530 kHz
Third harmonic 2370 kHz	Eighth harmonic 6320 kHz
Fourth harmonic 3160 kHz	Ninth harmonic 7110 kHz
Fifth harmonic 3950 kHz	Tenth harmonic 7900 kHz

253. A series-parallel circuit is composed of a 5-ohm resistor in series with the parallel combination of a capacitor having a pure reactance of 20 ohms and an inductance having a pure reactance of 8 ohms. What is the total impedance of the circuit? Is the total reactance capacitive or inductive?



The first step in solving this problem is to determine the impedance of the parallel circuit consisting of the inductance and the capacitance. We can do this by assuming a voltage across the two, and then by determining the current that will flow through each branch.

We then get the total current, and divide this into the assumed voltage to find the reactance of the branch. Since both 8 and 20 will divide into 40 evenly, let's assume a voltage of 40 volts across the parallel branch.

In solving this problem we can use the j operator if we wish to, but the problem is so simple it is entirely unnecessary to do this. We simply remember that the current through the inductance will be 180° out-of-phase with the current through the capacitance.

Assuming a voltage of 40 volts, the current through the inductance will be:

$$I_L = \frac{40}{8} = 5 \text{ amps}$$

The current through the capacitor will be:

$$I_C = \frac{40}{20} = 2 \text{ amps}$$

Since these two currents are 180° out-of-phase, the total current flowing in the circuit will be the difference between the two. Therefore:

$$I_T = 5 - 2 = 3 \text{ amps}$$

Since the inductive current is greater than the capacitive current, the total current flowing will be inductive.

Now we find the impedance of the parallel branch by dividing the total current flowing into the assumed voltage. Thus the impedance is:

$$Z = \frac{40}{3} = 13.3 \text{ ohms}$$

Now to find the total impedance in the circuit, we must get the impedance of the 5-ohm resistor in series with the 13.3-ohm inductive reactance. We can do this quite simply by using the formula:

$$\begin{aligned} Z &= \sqrt{R^2 + X^2} \\ &= \sqrt{5^2 + 13.3^2} \\ &= \sqrt{25 + 176.89} = \sqrt{25 + 177} \\ &= \sqrt{202} \\ &= 14.2 \text{ ohms inductive} \end{aligned}$$

In case you have forgotten how to get the square root of a number, you start at the decimal point and mark the number off in groups of two digits. Thus you mark 202 as 2'02. Setting up the problem, we find the square root of the number(s) in the first group on the left. In this case the square root of 2 is 1.

```
     2'02. |1
     1
 2 | 1 02  <---bring down the next two digits
```

Now we need to find a new divisor to divide into 2'02. We get the first digit by multiplying the numbers obtained in square root thus far by 2. Since 1 × 2 = 2, the first digit in our new divisor will be 2. To find the second digit, we have to experiment a little. Notice that 25 will go into 100 four times, but not five times. Twenty-four, however, will go into 100 four times, so we use four as the next figure and place 4 in the square root and four as the divisor. Now we multiply the divisor by 4 and subtract it from 1'02.

```
        2'02.  |14.2
        1
  24 |  1 02
          96
   282 |   600  <---bring down the next two digits
           564
            36
```

254. The 10-kilohm cathode resistor of a certain amplifier is bypassed to ground with a capacitor. If you want to operate this amplifier at a minimum frequency of 5 kHz, what size capacitor should be used?

(Assume $X_{Ck} \leq R_{k}/10$)

(Symbol $\leq$ means equal to or less than.)

When we bypass a cathode resistor, we use a capacitor that will have a low enough reactance to effectively bypass the resistor, so far as the lowest frequency ac signal is concerned. You can't use too large a capacitor, because the larger the capacitor the better. However, there is a certain minimum size that should be used, and in most engineering applications it is assumed that a capacitor that has a reactance of 1/10th of the resistance of the cathode resistor, at the lowest operating frequency, will effectively bypass the cathode resistor. This is the information that the FCC applies to this question; it wants a capacitor that has a reactance of 1/10th or less than the resistance of the cathode resistor. Therefore, we know that the reactance of the capacitor should be 1000 ohms at 5 kHz, which is the lowest operating frequency. The reactance of a capacitor is given by the formula:

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

We can transpose this formula so that it becomes:

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

Since we know the value of f is 5000 Hz and the value of X_C is 1000 ohms, we can substitute these values in the formula and get the value of C. We also know that when the 6.28 is divided into 1 we will get .159. Now we can rearrange the formula substituting and get:

$$C = \frac{.159}{5000 \times 1000}$$

5000 × 1000 can be written as 5 × 1000 × 1 × 1000. This can be simplified into 5 × 100,000,000 or 5×10^6. Substituting this we have:

$$C = \frac{.159}{5 \times 10^6} = .0318 \times 10^{-6} \text{ farads}$$

Since 10^{-6} is one-millionth, we can drop the one-millionth, and we'll have .0318 mfd as our answer. In actual practice we couldn't find a capacitor exactly this value; the closest standard size that is larger than this, and thus one that will have a lower reactance, is a .033 mfd capacitor, and this would be the size we want to use in this application.

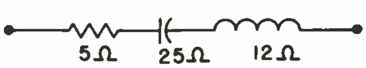
255. What effect does mutual inductance have on the total inductance of the two coils connected in series?

The total inductance of the two coils connected in series is equal to:

$$L_T = L_1 + L_2 \pm 2M$$

Thus if the two coils are connected in series-aiding, the total inductance will be equal to the inductance of the two coils plus twice the mutual inductance. If the two coils are connected in series-opposing, then the total inductance will be the inductance of the two coils minus twice the mutual inductance. Then the mutual inductance between the two coils either increases or decreases the inductance of the coils connected in series, and whether it increases or decreases the total inductance depends upon whether the fields of the two coils are aiding or opposing.

256. 10 amps ac is flowing in a series circuit composed of 5 ohms resistance, 25 ohms capacitive reactance, and 12 ohms inductive reactance. What is the voltage across each component? What is the total voltage? Why is the total voltage not simply the sum of the individual voltages? Explain.



The voltage across each of the components can be found using Ohm's Law in this form: the voltage is equal to the current × the resistance (or reactance), therefore the voltage across the components is:

$$E_R = I \times R = 10 \times 5 = 50 \text{ volts}$$
$$E_C = I \times X_C = 10 \times 25 = 250 \text{ volts}$$
$$E_L = I \times X_L = 10 \times 12 = 120 \text{ volts}$$

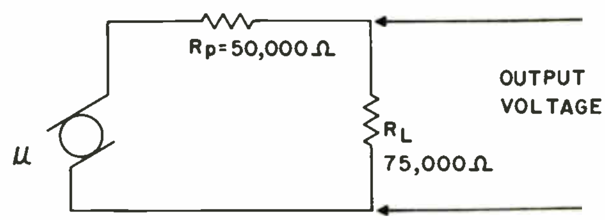
We cannot find the total voltage across these three factors simply by adding them together, because the voltages are not in phase. The voltage across the capacitor will be 180° out-of-phase with the voltage across the inductance. The voltages across the capacitor and inductance will be 90° out-of-phase with the voltage across the resistance. The voltage across the capacitor will lag the voltage across the resistor by 90°, and the voltage across the inductor will lead the voltage across the resistor by 90°. Therefore, we must perform a vector addition, and the simplest way to do this is by means of the formula:

$$E_T = \sqrt{E_R^2 + (E_C - E_L)^2}$$
$$E_T = \sqrt{50^2 + (250 - 120)^2}$$
$$E_T = \sqrt{50^2 + 130^2}$$
$$= \sqrt{2500 + 16{,}900}$$
$$= \sqrt{19{,}400}$$
$$= 139 \text{ volts}$$

257. What causes resistance noise in electrical conductors, and shot-effect noise in diodes?

Resistance noise in electrical conductors is caused by random motion of the molecules in the conductor. Shot-effect noise in diodes is caused by irregular current flow through the diodes.

258. Find the gain of a triode amplifier with a plate resistance of 50,000 ohms and a load resistance of 75,000 ohms. The amplification factor is 25.



The equivalent circuit of the tube to the plate resistance and its load resistance is shown. The tube acts as a generator with a gain of 25. The actual output voltage of the generator will be 25 times the grid voltage. However, this voltage is then fed to the series network consisting of the plate resistance of 50,000 ohms and the load resistance of 75,000 ohms. Only the voltage developed across the load resistance will be useful output voltage. Therefore we can get the voltage gain from the formula:

$$V_G = \frac{\mu \times R_L}{R_P + R_L}$$
$$= \frac{25 \times 75{,}000}{50{,}000 + 75{,}000}$$
$$= \frac{25 \times 75{,}000}{125{,}000}$$

We can divide the top and bottom of this expression by 1000, and simply cancel the three zeros off the 75,000 and the three zeros off the 125,-000. We can also divide 25 into 75 above the line and 25 into the 125 below the line, and get 5. The 5 then divides into 75 fifteen times, so that the gain of the amplifier will be 15.

259. List some of the precautions to be observed when soldering transistors and repairing printed circuits.

Transistors are easily damaged by excessive heat. Therefore, it is important when soldering a transistor into a circuit to make sure that the transistor leads are clean, and also to make sure that the component that the transistor lead is to be soldered to is clean. This will enable you to solder the transistor lead quickly, to avoid excessively heating the transistor. The transistor lead, since it is soldered, can be held between the jaws of a pair of longnosed pliers. The pliers will act as a heat sink and prevent an excessive amount of heat from traveling up the transistor lead into the transistor.

Printed circuits also are easily damaged by excessive heat. In a printed circuit board, the copper conductor is glued to a nonconducting base. If you apply too much heat to a printed circuit board when repairing it, you may cause the copper to break loose from the base. Remember that in removing parts you should avoid pushing from the base side to the conductor side, as this will have a tendency to push the copper off the insulated base. It is better to pull the lead from the insulated base side, so that the tendency is to pull the copper down against the insulated base.

260. What is the "gain factor" of a transistor?

The gain factor of a transistor used in the common-emitter circuit is usually referred to as the beta of the transistor. The beta is given by the formula:

$$\beta = \frac{\Delta I_C}{\Delta I_B}$$

This means that the beta of the transistor is equal to the change in collector current, divided by the change in base current that causes it.

In the common-base circuit, the gain of the transistor is referred to as alpha. Alpha is given by the formula:

$$\alpha = \frac{\Delta I_C}{\Delta I_E}$$

This means that the gain of the transistor in the common-base circuit is equal to the change in the collector current, divided by the change in emitter current that causes it. Since the collector current is always lower than the emitter current, the change in collector current will be less than the change in emitter current; therefore the alpha of the transistor is always less than one.

261. What are the main disadvantages of using transistors in circuits rather than vacuum tubes, assuming the cost is the same for both?

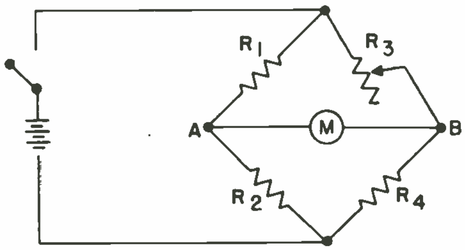
While transistors offer many advantages, such as small size, ruggedness, and the fact that they do not require any heater supply, they do have a number of disadvantages. Transistor parameters vary considerably from transistor to transistor --- you cannot expect nearly as constant a performance from one transistor to another as you can from one vacuum tube to another. In addition, transistor characteristics vary widely with temperature changes. Transistors are also susceptible to damage by transients. A sudden voltage surge, that might cause no trouble at all in a vacuum-tube circuit, may destroy a transistor. Often there will be a need for protective diodes in transistor circuits to protect the transistor.

In addition to the disadvantages inherent in the transistor itself, transistors are much smaller than vacuum tubes, and therefore more difficult to handle. It is comparatively easy to replace a vacuum tube in a piece of equipment; you simply unplug the old tube and plug a new one in. However, even where sockets are used, it is much more difficult to get the transistor plugged into the socket than it is in the case of a tube. In most cases, no sockets are used with transistors; to replace one you have to unsolder the old one in the circuit and then solder a new one into the circuit. Unless care is taken to prevent the transistor from overheating, you could accidentally destroy the new transistor while you are soldering it into the circuit.

262. If a standard broadcast station is licensed to operate at a frequency of 1260 kHz, what are the minimum and maximum frequencies at which it may operate and still be within the proper limits established by the Commission's rules?

The tolerance allowed a standard broadcast station is 20 Hertz. Therefore the carrier must be maintained within 20 Hertz of the assigned operating frequency. 1260 kHz is equal to 1,260-000 Hertz. Therefore, the station may operate at any frequency within 1,259,980 Hertz and 1,260,020 Hertz and be within the FCC rated frequency tolerance.

263. Explain the operation of a resistance bridge. If the known resistances in such an instrument are 5 and 10 ohms, and if adjusting the third resistance to 50 ohms produces a perfect balance, what is the unknown resistance?



In the drawing of the resistance and bridge, R_1 and R_2 are the 5-ohm and 10-ohm resistors respectively. R_3 is the adjustable resistance which is adjusted to 50 ohms in order to balance the bridge. R_4 is the unknown resistance which we are trying to measure.

In balancing the bridge, what we are actually doing is setting up conditions where the voltage between points A and B will be zero. This will result in no current flow through the meter; when this happens the bridge is balanced. In order to reach this condition we must have a ratio of R_1 to R_2 equal to the ratio of R_3 to R_4. We can set this up mathematically so that we have:

$$\frac{R_1}{R_2} = \frac{R_3}{R_4}$$

This equation can be turned around mathematically, so that we have:

$$R_4 = \frac{R_2}{R_1} \times R_3$$

Now substituting the values for R_1, R_2 and R_3 in the equation, we can get the value of R_4.

$$R_4 = \frac{10}{5} \times 50$$

$$= 100 \text{ ohms}$$

264. What is an audio frequency? What approximate band of frequencies is normally referred to as the audio frequency range?

An audio frequency is one that can be heard by the average human ear. We usually refer to this band of frequencies, from about 20 Hertz to 20,000 Hertz, as the audio frequency range.

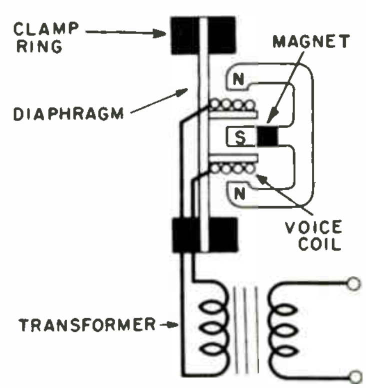
265. What causes sound, and how is it transmitted in the air?

Sound is caused by a disturbance in the air, which sets up sound waves in the air. For example, when we strike a note on a piano, a hammer strikes a string of a certain length, which is stretched on a frame. The string begins vibrating back and forth at a certain frequency, and it sets up a sound wave. As the string vibrates in one direction, it compresses the air molecules, and as it vibrates in the other direction it reduces the air pressure.

This wave travels out from the string in the form of sound waves, consisting of regions of compressed molecules followed by regions where the number of molecules is lower than normal. The actual change from the normal pressure to an increased air pressure, then reduced pressure is not abrupt, but rather follows a sine wave such as is in the case of a note produced by a musical instrument. When sound is produced by the human voice, we don't have such regular patterns, but we have areas of above normal pressure and below normal pressure, as in the case of a musical note. The changes, however, are more abrupt and do not follow a sinusoidal pattern. Sound is transmitted through the air by this sound wave, which travels from the object producing the sound, usually in all directions. The sound wave is gradually attenuated as it travels from the source, so that at some distance from the source it will disappear completely.

266. Sketch the physical construction of the following types of microphones and list their advantages and/or disadvantages. Which types are normally used in the broadcast studio? Why?

(a) Dynamic
(b) Ceramic
(c) Crystal
(d) Single-button
(e) Ribbon



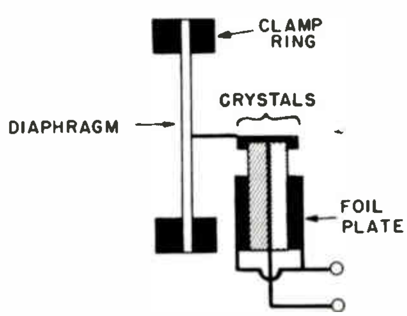
(a) A dynamic microphone is actually quite similar to a small, permanent-magnet dynamic speaker. As a matter of fact, you can use a permanent-magnet dynamic speaker as a dynamic microphone; however, the output will be lower than from a microphone and the frequency response will not be as good.

As you can see from the sketch shown, the dynamic microphone consists of a strong permanent magnet. A diaghragm, which is attached to a cylinder, is placed in the front of the microphone. The cylinder slides over one of the pole pieces of the magnet and a coil, consisting of a large number of turns of fine wire, is wound on the cylinder.

As the sound waves strike the diaphragm, the diaphragm vibrates in and out, causing the coil to move in the magnetic field. This induces a voltage in the coil. A reasonably high output voltage can be obtained from this type of microphone.

The advantages of the dynamic microphone are that it is comparatively lightweight, is reasonably rugged, and as we mentioned previously, has a reasonably high output. The frequency response of a dynamic microphone can be made quite wide by using a corrugated type of diaphragm. The corrugations are circular, and look something like the corrugations in the speaker coil of a dynamic speaker. Additional advantages of the dynamic microphone are that it does not require a power supply, and it is insensitive to changes in temperature and humidity. This type of microphone is widely used in broadcast studios.

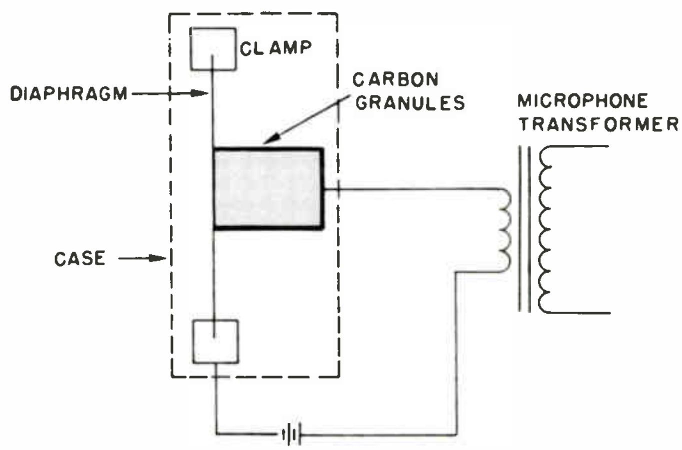
(b) The ceramic crystal operates on the piezoelectric effect of certain ceramic materials. A slab of ceramic is used, as shown in the diagram, and this in turn is attached to the metal diaphragm of the microphone. The vibrations of the diaphragm, produced by sound waves, stress the crystal material, and this in turn causes the latter to produce an electric output signal.



One advantage of a ceramic microphone is that it has a reasonably high output signal, usually about the same magnitude as that available from a dynamic microphone. Also this type of microphone does not require a power supply, and has a reasonably good frequency response (although usually not quite as good as that of a dynamic microphone). The ceramic microphone is considerably less expensive than the dynamic microphone.

The ceramic microphone is not used in broadcast work. Its frequency response is not as good as some other types, and in addition, because of its high output impedance it is quite susceptible to hum pickup. The ceramic microphone is usually made so that the ceramic element is in a moisture-proof container, because it is susceptible to changes in temperature and humidity. Ceramic microphones are not as rugged as dynamic microphones.

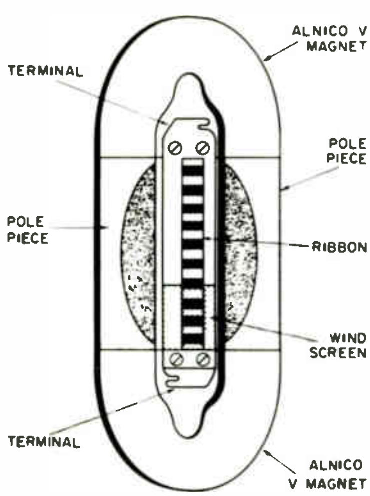
(c) The crystal microphone is essentially the same as the ceramic microphone, except that a Rochelle salt crystal is used to produce the output voltage instead of a ceramic material. The crystal microphone has all the advantages of the ceramic microphone: It usually has a higher output than a ceramic microphone. The crystal mike also has all of the disadvantages of the ceramic microphone, except that it is even more fragile than a ceramic microphone, and even more susceptible to damage by excessive humidity and temperature. Crystal microphones are not used in broadcast work.



(d) The single-button carbon microphone shown in the diagram works on this principle: The diaphragm is attached to the carbon button. The carbon button consists of a container packed with carbon granules. The sound waves striking the diaphragm cause the diaphragm to vibrate, and thus change the pressure on the carbon granules. When the diaphragm moves in and increases the pressure on the carbon granules, the resistance of the button decreases. When the diaphragm moves in the opposite direction and reduces the pressure on the granules, the resistance increases.

By using the single-button carbon microphone in a circuit, as shown along with the microphone transformer, the change in resistance produced in the button causes the current in the circuit consisting of the button, the primary of the microphone transformer, and the battery to change. This changing current, flowing through the primary winding of the transformer, induces a voltage in the secondary of the microphone transformer; this induced voltage is then fed to the first amplifier stage.

The advantages of the carbon microphone are its high output, low impedance and low cost. Among the disadvantages are the so-called carbon hiss which is present at all times. This hiss is due to small current variations caused by the current flowing through the carbon granules. The carbon microphone also is susceptible to moisture problems; it has a somewhat limited frequency response and requires some type of external dc power supply. This type of microphone is not used in broadcast work.



(e) A simplified sketch of a ribbon-type microphone is shown. In the ribbon microphone (also called a velocity microphone), a ribbon is stretched between the poles of a magnet. Usually the ribbon is a very light ribbon of corrugated aluminum, and is held at each end with two clamps. The sound waves striking the ribbon cause it to vibrate in the magnetic field, and a voltage is induced in the ribbon. This ribbon is connected to the primary winding of the microphone transformer, which is used to step-up the very low impedance of the microphone, and at the same time increases the voltage available.

The advantages of the ribbon-type microphone are that it has a good frequency response, a reasonably high output signal and a comparatively low output impedance. The ribbon-type microphone is also rugged, lightweight, and does not require an external power supply.

In addition, the microphone is not sensitive to changes in heat or humidity. One disadvantage of the ribbon microphone is that the ribbon must not be stretched too tightly or it will lose its elasticity. In addition, the ribbon is quite delicate and can be damaged by sudden blasts of wind or a loud nearby sound. The ribbon microphone is widely used in broadcast studios.

267. What is meant by the "phasing" of microphones? When is this necessary?

Phasing of microphones is connecting the microphones so that each microphone has the same output polarity for a given sound signal. For example, if we have two microphones that are connected to a mixer, the microphones should be connected, so that if a certain note is struck on a piano, the two audio signals produced by the microphones are positive during one-half cycle, and negative during the other half-cycle. If one microphone is connected with the opposite polarity, during the half-cycle when it is producing a positive output signal, the other microphone will be producing a negative output signal. This would result in reduction of output from the mixer, and at the same time a certain amount of distortion would be produced. Thus, phasing is necessary to get maximum output and to eliminate distortion.

268. What is the difference between unidirectional, bidirectional and omnidirectional microphones?

A unidirectional microphone is a microphone that has a major pickup lobe extending in only one direction. For example, a microphone may pick up sound coming directly at it from the front, but will not pick up sounds from the back, or two sides.

A bidirectional microphone is a microphone that has its major pickup lobes centered on two directions. A bidirectional microphone may pick up sound equally well from the front and back, but will not pick up sound from the two sides.

An omnidirectional microphone is a microphone that picks up sounds equally well in all directions.

269. What is a decibel?

A decibel is one tenth of a bel. The bel is a unit originally introduced by telephone engineers to measure the ratio between two sound-power levels. However, the bel was found to be too large a unit to use conveniently, so the decibel, which is a tenth of a bel, is more widely used. The decibel is used to express the ratio between two sound-power levels, and is also used to express the ratio between two electric power levels. The decibel is given by the formula:

$$db = 10 \log \frac{P_1}{P_2}$$

The decibel can also be used to compare voltage and current levels. Where the two voltages are across circuits having the same impedance, the number of decibels in terms of the voltage is given by:

$$db = 20 \log \frac{E_1}{E_2}$$

In circuits where the currents are being measured in circuits of equal impedances, the current ratio in decibels is expressed by the formula:

$$db = 20 \log \frac{I_1}{I_2}$$

270. VU meters are normally placed across transmission lines of what characteristic impedance?

600 ohms. A VU meter (volume unit) is a meter designed for operation across 600-ohm transmission lines. Zero VU is equal to 1 milliwatt across a 600-ohm line. 1 milliwatt across a 600-ohm line is equal to .775 volt. The VU meter is a logarithmic type meter that is highly damped, so that it gives a reading of the average volume level. It is widely used in broadcast studios.

271. Show by a circuit diagram a method of desensitizing a VU meter to cause it to read lower than normal.

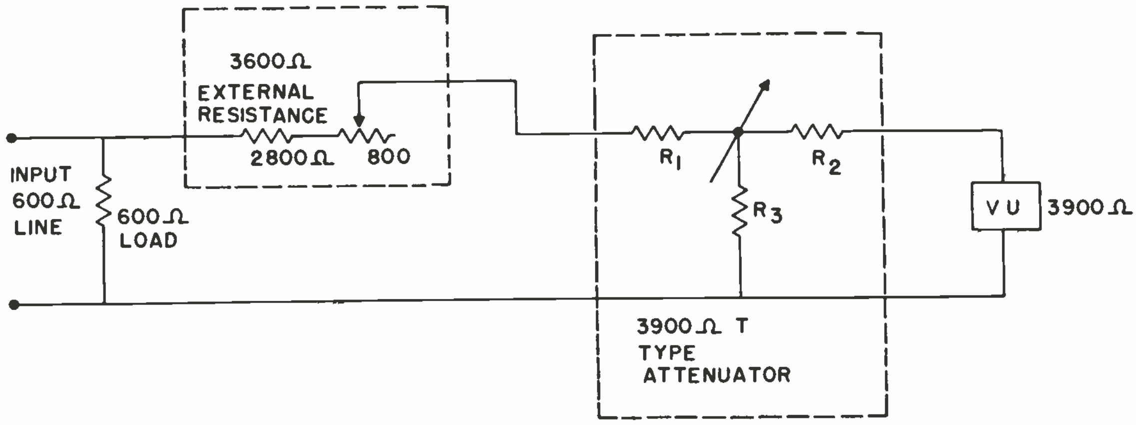
The total resistance of a VU meter is 7500 ohms, 3900 ohms of this resistance is in the meter itself, the balance of 3600 ohms is in an external resistance. To add an attenuator to the VU meter, the external resistance and the meter are separated, and a 3900 ohm T-type attenuator is inserted between the external resistance and the meter.

Although 0 VU is 1 milliwatt across a 600-ohm load, VU meters are made to read 0 with 1.225 volts applied. This is the standard signal used on 600-ohm telephone lines. It represents 4 VU above 0 VU. Thus the attenuator dial is made to read 4 VU when R_1 and R_2 are zero ohms and R_3 is open.

If the signal level exceeds zero reading on the VU meter, the attenuator is adjusted to give a zero reading on the meter. Then the signal strength is read on the attenuator dial rather than on the meter.

272. Why is it important to keep the contact points on attenuator pads, used in a broadcast studio console, clean? How are they cleaned?

It is important that the contact points on attenuator pads be kept clean in order to insure noise-free operation of the pads as well as reliable operation. The contacts may be cleaned by means of a suitable contact cleaner on a piece of clean cloth. Contact cleaners are available from most radio-TV wholesalers. Some technicians use carbon tetrachloride to clean contact points. This is an excellent cleaner and is non-flammable; but its fumes are very dangerous, therefore if you use carbon tetrachloride, avoid inhaling the fumes and be sure the room is well ventilated. There are also many other reliable, easily available cleaning fluids which remain harmless under ordinary safety precautions.



273. What is a pre-amplifier? Why are they normally used in the broadcast station?

A pre-amplifier is simply a high-gain audio amplifier that is designed to amplify a very low level signal such as the output from a magnetic type pickup or a microphone. They are used in broadcast stations to amplify these low level signals before the signal is fed to the mixer in the audio console. The pre-amplifier is usually placed as close as possible to the pickup device in order to build up the amplitude of the signal before it is transmitted from the pickup device to the audio console. The reason for putting the amplifier as close as possible to the pickup device is to prevent hum and other types of pickup in the transmission line. If the signal in the line is weak, and a small amount of hum or noise is picked up, the ratio of the strength of the signal to the hum or noise is small, and therefore hum or noise would become objectionable. However, if the signal is first amplified before it is fed to the transmission line and then a small amount of hum or noise is picked up in the transmission line, the signal will be so much stronger than the hum or noise that it will simply override it and not cause any objectionable interference.

274. Given the gain of an amplifier, amplifying feedback and the overall voltage gain of the circuit, how is it possible to determine the amount of feedback used? State the formula used and solve a sample problem.

The gain of an amplifier with feedback can be determined from the formula:

$$a = \frac{A}{1 + BA}$$

where:

a = gain with feedback
A = gain without feedback
B = fraction of output voltage feedback

Let us assume that the gain of the amplifier without feedback is 25 and the gain of the amplifier with feedback is 15. What we want to do is find the percentage of feedback used. To do this we must rearrange the gain formula so we can solve for B. Therefore starting with the gain formula:

$$a = \frac{A}{1 + BA}$$

cross multiplying:

$$A = a + BAa$$

rearranging we have:

$$BAa = A - a$$

dividing through by Aa we get:

$$B = \frac{A}{Aa} - \frac{a}{Aa}$$

$$B = \frac{1}{a} - \frac{1}{A}$$

substituting the gain with feedback, 15, for "a" and and the gain without feedback, 25, for "A" we get:

$$B = \frac{1}{15} - \frac{1}{25}$$

and bringing these to a common denominator which is 75 we have:

$$B = \frac{5 - 3}{75} = \frac{2}{75}$$

now to express this as a percentage we multiply by 100 and we have:

$$B = \frac{2}{75} \times 100 = \frac{8}{3} = 2.67\%$$

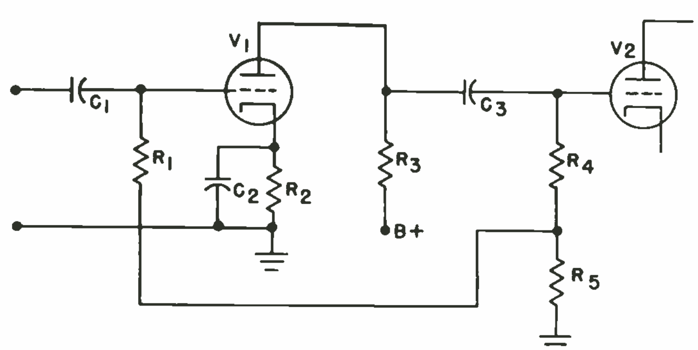
Thus the amount of signal fed from the output back to the input is 2.67%.

275. What is the technical requirement for negative feedback? Show by a circuit diagram how this is achieved for:

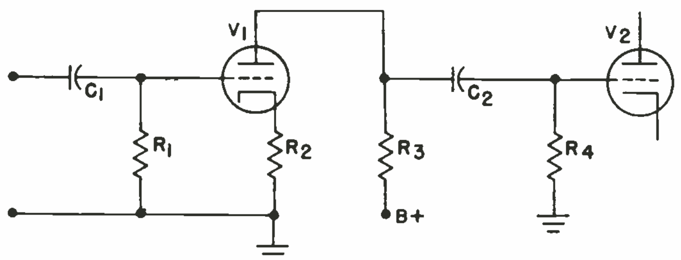
(1) Negative voltage feedback.
(2) Negative current feedback.

Negative feedback is used to improve amplifier stability and to reduce distortion. In order to provide negative feedback, the voltage used as the feedback voltage must oppose the input voltage.

There are two types of feedback, voltage feedback and current feedback.



In the circuit shown, the voltage developed across R_5 is the feedback voltage. This type of circuit is a voltage feedback circuit. The input signal is amplified by V_1 and the amplified voltage in the output will be 180° out-of-phase with the input voltage. The coupling capacitor C_3 couples the signal to the network consisting of R_4 and R_5. A portion of the total voltage (that voltage appearing across R_5) is fed back into the input circuit through R_1 and opposes the input signal voltage.



In the circuit shown, we have current feedback. The cathode resistor R_2 is unbypassed. As the input signal swings in a positive direction and causes the current through the tube to increase, the current through R_2 will increase. This will cause a positive voltage to appear at the cathode of the tube. Thus, the net grid-to-cathode voltage applied to the tube will be reduced. This, therefore, is negative feedback. This type of feedback is called current feedback because the strength of the feedback signal will depend upon the current through the tube and R_2.

276. What is meant by the "fidelity" of an audio amplifier? Why is good fidelity an important consideration when replacing amplifiers in the broadcast station?

When we speak of the fidelity of an audio amplifier, we are referring to its ability to reproduce in the output an amplified version of the input signal which is a true reproduction of the input signal without any distortion added. This refers to amplitude distortion, frequency distortion, phase and intermodulation distortion.

Good fidelity is important in an amplifier in order for the quality of the signal broadcasted to be good. When replacing an amplifier, if you do not use an amplifier with good fidelity, the quality of reproduction will suffer and hence the quality of the signal transmitted by the station. With amplifiers being continually improved, there is no reason why a replacement amplifier should not have as good fidelity as the original one, in most cases it is possible to obtain an amplifier with improved fidelity.

277. What type of playback stylus is generally used in broadcast station turntables? Why?

A diamond-tipped stylus is generally used in broadcast station turntables. The diamond-tipped stylus will last a long time and provide good reproduction with a minimum of distortion. Therefore the maintenance requirements are quite nominal. At the same time, the diamond stylus does not cause excessive wear on the records so that maximum life can be expected from them.

278. How does dirt on the playback head of a tape recorder affect the audio output? How are such heads cleaned?

Tape recorders are widely used in broadcast stations because of their flexibility. An entire program or a single musical selection can be recorded on the tape and played later. The tape recorder is capable of better fidelity than a conventional record. In a tape recorder, the information is recorded on a tape in the form of magnetic variations. When the tape is played, it is moved past a head which picks up the signal recorded on the tape. Dirt on the playback head may cause distortion, it may cause a reduction in the amplitude of the output and it may also cause the tape to move across the head at an uneven speed. Changes in the speed of the tape will produce a low frequency type of modulation referred to as wow or a high frequency modulation on the tape referred to as flutter.

Dirt can be removed from the head by means of a special tape that is used in place of the conventional tape. The tape is put on the tape recorder and allowed to pass over the head. The tape has been treated with a cleaning solvent which will clean the head. The head may also be cleaned with a commercial cleaner or with carbon tetrachloride. If you use carbon tetrachloride, avoid inhaling the fumes, and use it in a well ventilated space.

279. What are "wow" and "rumble" when referred to turntables? How can they be prevented?

Wow is a low-frequency modulation. It is caused by uneven rotation of the record. This may be caused by improper lubrication, improper tension on any drive belts that may be used, or by warped or damaged parts in the drive mechanism. In some cases, wow may be due to a manufacturing defect where the speed of the record changes slightly as it rotates. In this case the only thing you can do is to find the defective part and replace it. If a turntable that has operated without any sign of this type of distortion should suddenly develop wow, it is an indication of improper maintenance. Lubricating the drive mechanism and properly adjusting tension on drive belts, etc. should clear up the problem.

Rumble is somewhat similar to wow inasmuch as it is also a low-frequency modulation. It is caused by turntable vibration. This is frequently due to vibrations from the motor or drive mechanism causing the turntable to vibrate. In the design of the equipment, rumble is kept at a minimum by using a heavy turntable and by shock mounting the motor and parts of the drive mechanism so that any vibration produced by them is not transferred to the turntable. Worn or dried out rubber shock mounts may be a cause of rumble.

280. What factors can cause a serious loss of high frequencies in tape recordings?

Normally tape recordings should have excellent frequency response. Both the low-frequency and the high-frequency response should be good. However, poor high-frequency response can be caused by either electrical or mechanical problems.

In the case of electrical problems, incorrect bias current in the tape head may cause a loss of high frequencies. Of course, any defect in the amplifying equipment used with the tape recorder could cause poor high-frequency response. If the high-frequency response from the microphone or record is normal and it is poor from tape recordings, then the chances are that the pre-amplifier used with the tape recorder has a defect that is causing this problem, if the problem is electrical.

Mechanical problems may also cause a serious loss of high frequencies. Dirt on the head will affect the high-frequency response.

The tape-head gap must be exactly perpendicular to the direction of travel of the tape. Therefore any misalignment of the tape head can upset the high-frequency response. Also failure of the tape to maintain a contact with the tape head might affect the high-frequency response. If the tape moves across the head, there is some wear on the head; uneven wear on the tape head will affect the high-frequency response, and if it becomes excessive, can cause a serious loss of high-frequencies.

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T-12: FCC

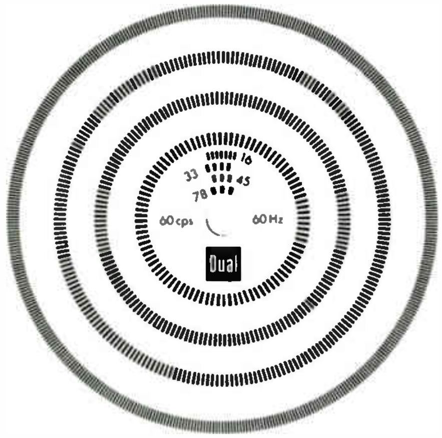
NRI Study Guide for FCC License Examination

PART XIII

The subjects covered in this section of the study guide are topics which cover the questions you are likely to be asked in the examination for Element IV of your FCC License Examination. The topics cover a wide range of subjects dealing with broadcasting. They are covered in sufficient detail in the study guide so that you shouldn't have any difficulty in answering the questions on these subjects in your examination.

Be sure to save this part of the study guide and those you received previously so that you can review them before taking the examination for Element IV of the FCC License Examination.

281. Explain the use of a stroboscope disc in checking turntable speed.



A stroboscope disc is a disc having a circle of bars on the disc. It is used for checking the speed of a turntable. A stroboscope disc designed to check a 33-1/3 rpm turntable should have a total of 216 bars around the circle. If the turntable is turning at exactly 33-1/3 rpm, there will be 216 X 33-1/3 = 7200 bars passing a given point every minute. On a 60-cycle power line there will be 60 X 60 = 3600 cycles per second. A light that is operated from this power line will have 7200 peaks of brightness per minute, one peak on each half cycle. Thus if a light is used to illuminate the disc and the disc is traveling at exactly 33-1/3 rpm, the bars will appear to stand still. If the turntable is turning too fast, the bars will move in the direction of rotation, but if the turntable is turning too slowly the bars will appear to move in a direction opposite to that of rotation.

In using the stroboscope disc, a record is placed on the turntable and then the disc is placed on top of the record. The tone arm is then placed on the record so the turntable speed will be checked under normal operating conditions. A fluorescent or neon lamp (neon is better) is used to light the disc to see whether the bars are standing still or moving in one direction or the other. An incandescent light is not suitable because the filament in the incandescent light stays at almost a constant brightness and hence it is difficult to tell whether the bars are stationary or moving.

Turntables such as those found in broadcast studios have speed controls. These controls provide a means of adjusting the speed of the turntable to the exact speed required. It is usually impossible to get the turntable at exactly the correct speed, but the slower the bars move in either direction, the closer you are to the correct speed. Obviously, in adjusting the turntable speed, you should try to get it as close to the correct speed as possible.

Stroboscope discs with different numbers of bars are available for checking 78 rpm, 45 rpm and 16 rpm turntables. Some discs have these different numbers of circular bars on the same disc so that a single stroboscope disc can be used for checking any turntable speed.

282. Show how frequency response of a pickup unit of either a tape recorder or turntable is tested.

The pickup unit of a tape recorder is usually used with its associated amplifier; in most cases we are interested in evaluating the frequency response of the entire system. In this case you use an audio oscillator that has a constant amplitude output. If the oscillator does not have a good quality amplitude output, you will need some type of amplitude indicating device across the output of the oscillator so you can adjust the amplitude as you change frequency.

In either case the procedure is to make a tape of numerous frequencies between approximately 20 Hertz and 20,000 Hertz. As we mentioned, the input signal to the tape recorder must be held constant at all times.

After the tape is made, then the tape is played back and the amplitude of the output from the amplifier used with the tape recorder is measured and the curve can be plotted showing the response of the amplifier at the various different frequencies. If you want to determine the frequency characteristics of the pickup head alone, you follow the same procedure and develop a curve for the head plus the amplifier. Then you check the amplifier itself by feeding a signal from the audio oscillator directly into the amplifier and develop a curve for the frequency response of the amplifier. For the curve of the amplifier and head and for the curve of the amplifier alone you can develop a curve of the pickup head.

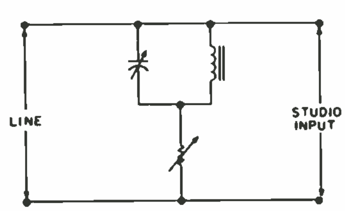
Essentially the same procedure is used in checking the frequency response of a turntable pickup. Standard frequency-test records are available. These records contain constant amplitude recordings of various test frequencies within the audio frequency spectrum. The output from the pickup and amplifier is measured using a test record and then a curve is plotted showing the frequency response of the pickup and the amplifier. After you have this curve you use an audio oscillator to feed signals of the same frequency into the amplifier and measure the frequency response of the amplifier alone. From the two curves of the audio amplifier and the amplifier pickup device you can develop a third curve showing the frequency response of the pickup device alone.

283. What are line equalizers? Why are they used? Where, in the transmission line, are they normally placed?

When an audio signal is transmitted over a long line, there is considerable attenuation of the high frequency part of the audio signal due to line capacitance. The longer the line, the greater the attenuation will be. The result is that frequency distortion occurs along the transmission line.

This frequency distortion can be overcome by means of a line equalizer. A line equalizer is a form of a high-pass filter. The high-pass filter is designed to pass the higher frequency signals without attenuation and to attenuate the low frequency signals. Thus it tends to compensate for the frequency distortion in the line and hence eliminates frequency distortion.

Telephone lines are frequently compensated by the telephone company up to a certain frequency. Usually in standard broadcast work the line is equalized up to 5000 Hertz. In FM broadcasting, where higher quality reproduction is desired, the line is usually equalized up to 15,000 Hertz. However, if the line is not equalized and equalization is required, it is usually performed where the line enters the studio input equipment. An example of a line equalizing circuit is shown in the schematic.



284. What are limiting amplifiers? Why are they used in broadcast stations? Where are they normally placed in the program circuit?

In a live pickup for broadcasting, the amplitude of the signal to be picked up usually varies over quite a wide range. An example of this would be the pickup of a program by a symphony orchestra; some of the passages would be quite weak, others very strong. A limiting amplifier is one which automatically reduces its gain when the amplitude exceeds a certain level. The limiting amplifier is used to prevent overmodulation on the passages of very high amplitude. Overmodulation, of course, would cause the production of spurious signals and distortion.

When a limiting amplifier is used it is usually part of the audio system of the transmitter. The limiting amplifier is connected between the input and the audio driver which drives the modulator.

285. Explain the operation of limiting amplifiers.

A limiting amplifier is an amplifier in which the gain is controlled. The circuit usually consists of some type of peak limiting device, for example, a biased diode. If the peak amplitude of the signal exceeds a predetermined level, the limiter conducts and develops a dc voltage. This voltage is used more or less as an automatic control voltage. It is applied to a variable μ pentode tube and is used to reduce the gain of the stage. The rectified peaks of course are filtered by a suitable quick-acting filter network so that the gain of the stage is rapidly reduced to prevent overmodulation and then smoothly returned to normal as the peak amplitude of the signal goes down.

286. What are the uses of peak limiting amplifiers?

Peak limiting amplifiers are used to prevent overmodulation of the transmitter on instantaneous loud or strong peaks. As pointed out previously, in a pickup of a live broadcast, particularly musical programs, there may be sudden bursts which reach a high amplitude. The peak limiting amplifier clips these peaks off to prevent overmodulation. This permits a high level of modulation on the average program amplitude, without the inherent danger of overmodulation on the peaks.

287. What are agc amplifiers and why are they used?

AGC amplifiers used in broadcast studios are amplifiers in which the gain is controlled automatically. These amplifiers are particularly useful in helping to maintain a constant level of modulation. For example, in a broadcast pickup where an announcer is interviewing another person, one of the two may speak with a soft voice and the other with a comparatively loud or strong voice. The only way to modulate the transmitter equally when the two are speaking is to provide some means of controlling the gain of the audio amplifier in the transmitter. This can be done by riding the volume control. In other words, turning the gain control up when the person with the softer voice is speaking and turning it down when the person with the stronger voice is speaking. This works out satisfactorily providing the person on the gain control is alert and providing the control is not turned up when the person with the softer voice is speaking and the one with the stronger voice suddenly interrupts to insert a few words - this could result in overmodulation.

In an amplifier where the gain is controlled automatically, the gain will be turned up automatically when the person with the soft voice is speaking and reduced automatically when the one with the stronger voice is speaking. The gain should be turned up while the one with the softer voice is speaking and when the one with the stronger voice suddenly interrupts, the amplifier gain will immediately and automatically be reduced so that the level of modulation will be maintained.

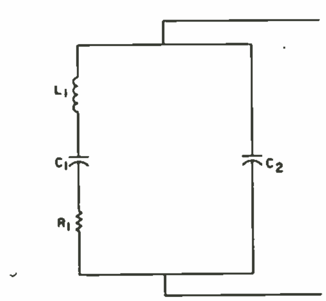
288. Explain the operation and use of compression amplifiers.

Compression amplifiers are amplifiers that are used to compress the dynamic range of a broadcast. By this we mean that the amplitude of the loud passages is reduced so that there is less of a variation between the amplitude of the softer passages and the amplitude of the louder passages in the program.

In operation a rectifier is used to rectify the signal envelope and to develop a negative voltage. The amplitude of this voltage depends upon the amplitude of the modulating signal. The stronger the modulating signal the greater the negative voltage and the weaker the modulating signal the smaller the negative voltage. This negative voltage is then fed to the grid of a variable μ tube and is used to control the gain of the tube.

Compression is different from limiting inasmuch as compression operates at all times. All signals are compressed so that the entire dynamic range is reduced. In peak limiting, there is no change in the signal amplitude until a certain level is reached. When this level is reached the signal is clipped off to avoid overmodulation of the transmitter. Dynamic compression permits a more constant modulation level. As we mentioned previously, the sound level in a symphony program varies over wide ranges. Therefore it is desirable to give more amplification to the very weak passages so they can be heard while at the same time giving less amplification to the very strong signals so that very high or excessively high modulation is avoided.

289. Draw the approximate equivalent circuit of a quartz crystal.



The quartz crystal itself is roughly equivalent to a series-resonant circuit consisting of L_1, C_1 and R_1. The reactance of L_1 is large compared to that of C_1 and the resistance of R_1 and hence the crystal has a very high Q. C_2 represents the capacitance of the plates in the holder between which the quartz crystal is sandwiched. The value of C_2 is large compared to C_1 and hence has very little effect in determining the resonant frequency of the crystal.

Crystals are widely used in transmitters to maintain the frequency stability of the transmitter. When they are used in an oven where the temperature of the crystal is constant, the frequency of the transmitter can be held to within a few cycles of its assigned frequency. Where crystals are used to control the operating frequency of a transmitter they usually are used in circuits similar to the tuned grid-tuned plate circuit, where the crystal has been used to replace the L-C circuit in the grid circuit of the tube.

290. What factors affect the resonant frequency of a crystal? Why are crystal heaters often left on all night even though the broadcast station is not on the air?

The basic frequency at which a crystal is resonant depends on the type of cut of the crystal and on the dimensions of the crystal. Thus a manufacturer may be able to grind a crystal to operate on a certain frequency, and the operating frequency of the crystal will be close to that value.

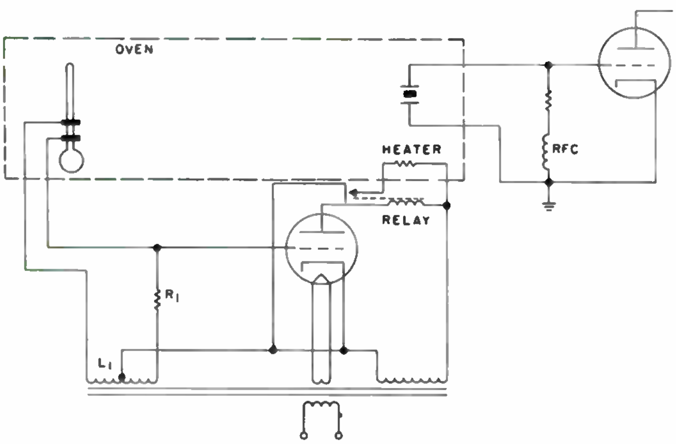
Although the primary factors controlling the resonant frequency of the crystal are the dimensions of the crystal and the type of cut, there are other factors that have some affect on the crystal frequency. Different cuts have different temperature coefficients. Therefore if a very close frequency tolerance must be maintained, the crystal must be operated at the temperature which the manufacturer specifies. To keep the crystal and the components associated with it at a constant temperature, the crystal heaters may be left on all night even though the transmitter is not on the air. Leaving the crystal heaters on has another advantage in that it prevents moisture condensation within the crystal oven. Moisture absorption in the crystal and in other components associated with the crystal may cause some frequency instability, particularly when the transmitter is first turned on. The same is true of the temperature. If the temperature is not up to the rated value when the transmitter is first turned on, the transmitter frequency may drift until the crystal temperature comes up to its normal operating temperature.

The crystal frequency may also be affected by the holder. Both the pressure on the crystal and the capacitance of the holder affect the operating frequency of the crystal. For that matter, any shunt reactance across the crystal, whether it be capacitive or inductive, will have some effect on the crystal frequency.

291. Explain by the use of simple drawings the physical construction and the operation of a mercury thermometer and thermocoupled types of crystal heater controls.

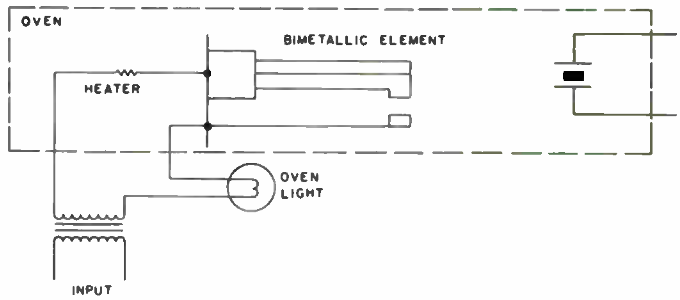
The mercury thermometer is actually a very sensitive form of thermostat. In the mercury thermometer type of thermostat the mercury is placed in a hollow glass tube as in a conventional thermometer, but a hole in the glass tube is stretched out along the operating range at which the thermostat is to operate. This results in a very sensitive device since a small change in temperature will cause an appreciable change in the height of the mercury column.

In the mercury thermometer one wire is placed in the bottom of the thermometer which makes contact with the mercury column. The other contact is placed in the column at the desired operating temperature. If the mercury expands and reaches the upper contacts, the circuit closes.



The mercury thermometer is used in a circuit similar to that shown. In this circuit, the winding L_1 on the transformer is arranged so that when the grid is connected to the one end through the grid resistor R_1, the grid will be driven in a positive direction at the same time the plate of the tube is driven in a positive direction. This will cause current to flow through the tube and when the current is sufficient to close the relay the heater circuit will be energized. The heater heats up the oven causing the temperature to rise and the mercury to expand. When the mercury expands sufficiently to close the contacts, which it will do when the oven reaches the correct temperature, the grid of the tube is connected through the mercury column to the other side of the winding L_1 on the transformer. When it is connected to this side of the transformer the voltage applied to the grid of the tube will be 180° out-of-phase with the plate voltage. This will cause the plate current to drop to zero or to a low value that is insufficient to keep the relay closed. When the relay in the plate circuit of the tube opens, the current through the heater is cut off and there is no further heating of the crystal oven.

Because the mercury in the column is very thin at the operating range, a very small change in temperature will cause sufficient contraction or expansion of the mercury column to open or close the contacts, thus either turning the heater on or off. Therefore this type of control is able to maintain the crystal oven temperature within a very close tolerance.



The thermocoupled type of heater control is simpler and less expensive than the mercury thermometer type of control. However, it is not as sensitive a control.

In the thermocoupled type of control we make use of a bimetallic element. This is made up of two thin strips of different types of metal. In the example shown, the bottom strip of the bimetallic element has a greater coefficient of expansion than the upper strip. Thus as the bimetallic element heats, the lower strip will expand more than the upper strip forming an arc that is bent upward so that the contacts will open. When the bimetallic element cools, the lower strip will contract more than the upper one. Thus an arc that is bent downward is formed, closing the contacts on the thermostat.

When the thermostat is closed, current supplied by the transformer flows through the heating element and through the thermostat contacts. When the oven reaches its rated temperature, the bimetallic element is supposed to have expanded sufficiently to open the contacts, thus opening the heater circuit and preventing further heating of the crystal oven.

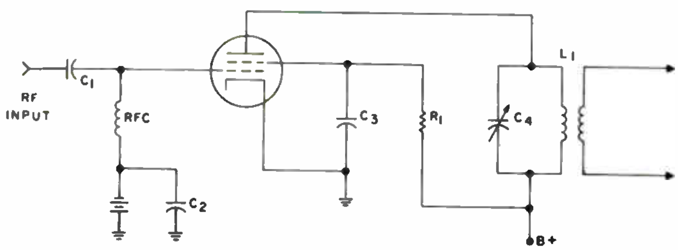
292. Why are tubes used in linear rf amplifiers normally not biased Class A?

A Class A power amplifier has a relatively low efficiency. In a linear rf amplifier we are usually interested in obtaining the best possible efficiency and therefore the tube is usually biased as a Class AB amplifier or as a Class B amplifier.

The maximum theoretical efficiency of a Class A power amplifier is 50%. However, in actual practice, it is seldom possible to get an efficiency much better than 10% or 12% from a Class A rf power amplifier. However, in the case of a Class AB or Class B power amplifier, efficiencies in the order of 30% to 35% are easily realized. You can see from this that there is quite a savings in power input and tube cost by operating the tube as a Class AB or Class B amplifier rather than as a Class A amplifier. This is particularly true of high power stations where the cost of the electrical power consumed by the station is substantial.

293. Draw circuit diagrams of the following rf amplifiers and explain their operation.

(a) Class C rf power amplifier with battery bias.
(b) Tetrode rf power amplifier with grid-leak bias.
(c) RF power amplifier with two tetrode tubes in parallel.
(d) RF power amplifier with two tetrode tubes in push-pull.
(e) Plate-neutralized triode rf amplifier.
(f) Grid-neutralized triode rf amplifier.
(g) Triode-frequency doubler stage.
(h) Push-push frequency doubler stage.
(i) Grounded-grid rf amplifier.



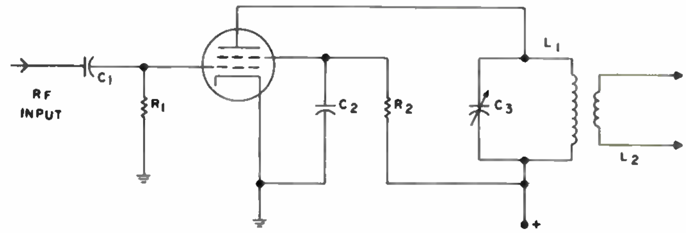
(a)

(a) In the circuit shown, the bias voltage applied between the grid of the tube and the cathode is sufficient to cut off the flow of plate current. In a Class C amplifier, the grid bias is usually from two to four times the cutoff voltage. Thus with no input signal applied, there will be no plate or screen current flowing in the circuit.

The input signal applied to the grid of a Class C power amplifier is a high amplitude signal which is sufficient to drive the grid positive. During the positive peak of the rf input signal, a high plate current flows through the tube. The voltage between the plate and the cathode of the tube drops to a low value. The current flows through the tube in the form of a series of high amplitude pulses.

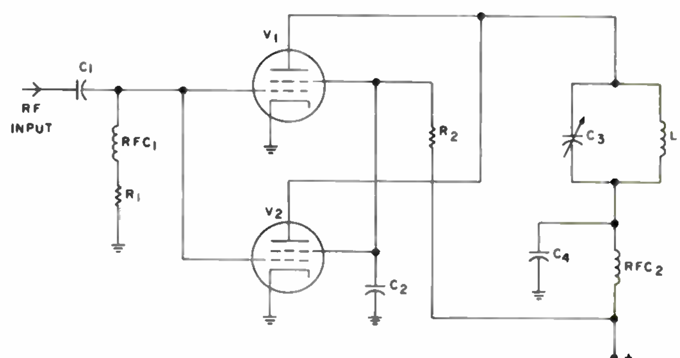
The plate current pulses are fed to the tank circuit consisting of L_1 and C_4. These pulses shock excite the tank circuit into oscillation. The tank circuit in an amplifier is tuned to the frequency of the incoming signal. The bursts of plate current sustain the oscillation in the tank circuit so that the output from the Class C amplifier is a sine wave.

The high efficiency of a Class C amplifier is due to the high plate current flowing through the tube at a time when the plate-to-cathode voltage of the tube is low. Thus the actual dissipation in the tube is equal to a relatively small amount of the total power input. One of the characteristics of a Class C power amplifier is its high efficiency. However, it can be used only with a cw input signal.



(b)

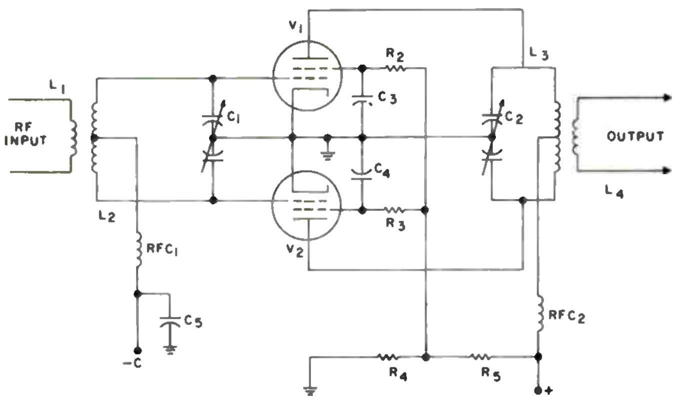
(b) The circuit shown in (b) is essentially the same as the circuit shown at (a) except that grid-leak bias is used in the circuit in (b). In this circuit, the grid of the tube is driven positive and grid current flows through the tube and through the grid resistor R_1. The current flowing through the grid resistor R_1 develops a voltage across it which biases the tube. The size of the grid resistor is selected on a basis of the average or dc grid current of the tube and the dc grid bias required on the tube. The disadvantage of this type of bias is that if the excitation to the stage fails, the bias will disappear and very high plate and screen currents will flow. Unless some protective devices are used, the high plate and screen currents could destroy the tube.



(c)

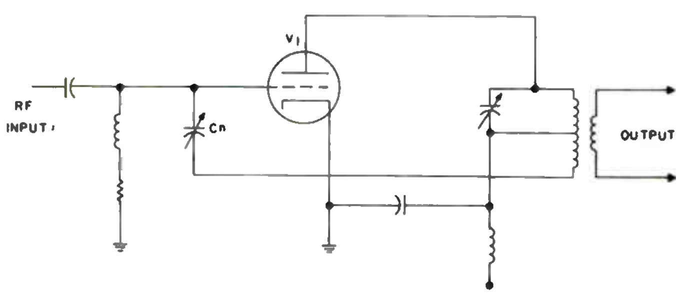
(c) The circuit shown here is essentially the same as the one shown at (b) except that two tubes are used in parallel in place of the one. In an arrangement of this type, the grid current will be twice the grid current of the circuit shown at (b). Thus twice the driving power is required to drive the two tubes. Since the tubes are connected in parallel, the output from an arrangement of this type is double that which can be obtained from a single tube. The disadvantages of parallel-operated tubes are increased input and output capacitances. Also, because of slight variations in the characteristics of the tubes, one tube may handle more current than the other. If the tubes are operated close to or at their rated value, this could mean that one tube would be overloaded which might result in shortened tube life.

In the circuit shown, bias for the two tubes V_1 and V_2 is developed across the resistor R_1. Since we have the grid current from both tubes flowing through this resistor, the resistor would be half the value used to bias a single tube.



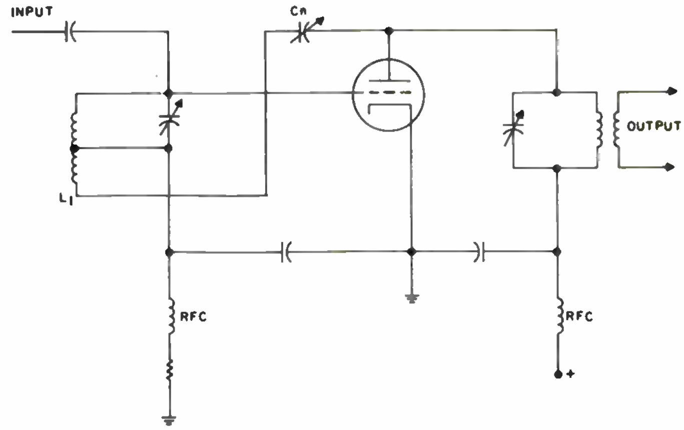
(d)

(d) In a push-pull circuit when the grid of V_1 is driven in a positive direction by the input signal, the grid of V_2 will be driven in a negative direction. Thus V_1 will conduct heavily while V_2 is driven still further negative so that no current flows through it. The current pulse from V_1 is supplied to the output tank circuit consisting of L_3 and C_2. During the next half cycle when the polarity input signal reverses, the grid of V_2 will be driven positive and the grid of V_1 negative. During this half cycle V_2 will conduct and feed a strong current pulse to the tank circuit. The output of the push-pull power amplifier is double that which can be obtained from a single stage. The push-pull power amplifier also has the characteristic of cancelling even-order harmonics generated in the stage.



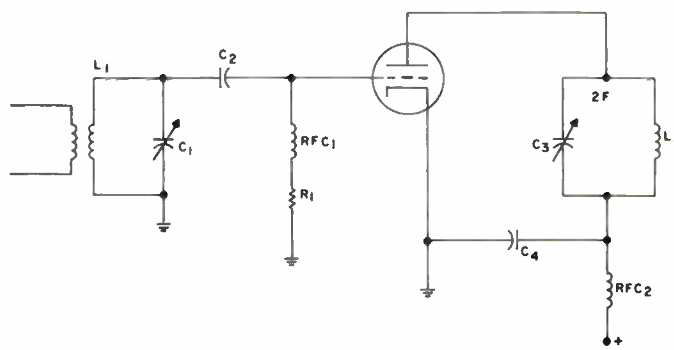
(e)

(e) In the circuit shown, a certain amount of energy will be fed from the plate of the tube back to the grid through the grid-to-plate capacity of the tube. If the resonant circuit in the plate circuit is inductive, this will cause the stage to go into self-oscillation. To overcome this, we feed a signal from the plate circuit through C_n back into the grid of V_1. This signal is 180° out-of-phase with the signal fed by the grid-to-plate capacity of the tube to the grid and hence cancels it out. C_n is made adjustable so that the amount of signal needed to cancel the signal fed back through the tube can be fed back through C_n. Notice that the neutralizing signal is derived from a tapped tank circuit in the plate circuit of the tube.



(f)

(f) In the diagram shown, the capacitor C_n is the neutralizing capacitor. Energy is fed through this capacitor into the lower end of the grid tank circuit L_1. The voltage fed through C_n into the lower half of L_1 induces a voltage in the upper half which is 180° out-of-phase. Therefore the signal at the grid end of the coil is 180° out-of-phase with the signal fed into the lower end of the grid coil through C_n. This signal is fed to the grid of the tube to neutralize or cancel the signal fed back to the grid through the grid-to-plate capacitance of the tube.

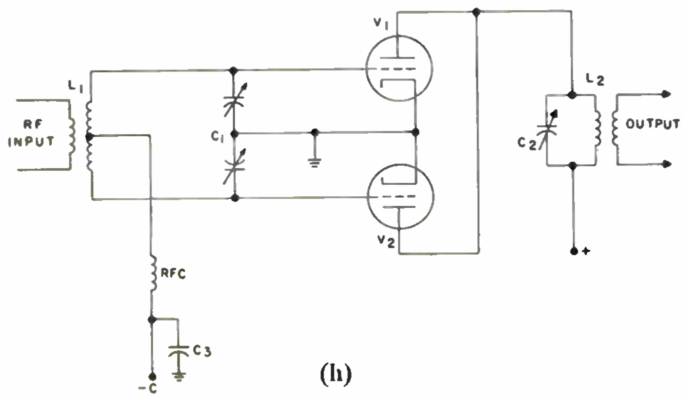


(g)

(g) The schematic diagram of the triode rf doubler stage may resemble a power amplifier simply tuned to amplify the signal. However, there are a number of differences. First of all, a much higher bias is used on a doubler. Usually the bias is in the order of ten times the cutoff voltage in order to get very sharp current pulses through the tube. The input tank circuit consisting of L_1 and C_1 is tuned to the frequency of the incoming signal. The output tank circuit consisting of L_2 and C_3 is tuned to twice the input signal frequency if the circuit is to be used as a doubler. The sharp current pulses shock the tank circuit into oscillation at twice the input frequency.

The circuit goes through two cycles before another pulse arrives to make up for energy lost in the tank circuit.

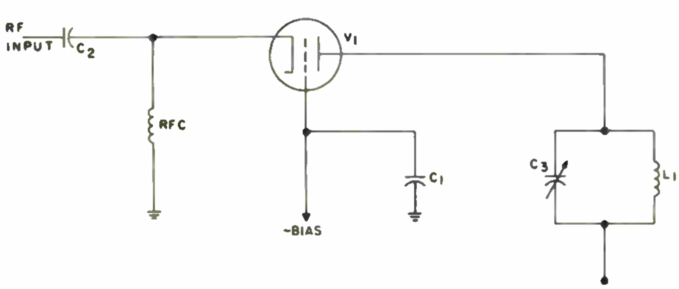
One big difference between a frequency doubler and a straight through amplifier is that a frequency doubler usually does not require neutralization. This is due to the fact that the power developed in the output circuit is twice the frequency of the input signal and hence any energy fed back into the input circuit is not likely to cause oscillation.



(h)

(h) In the circuit shown, notice that the grid circuit is arranged in push-pull, but the plate circuit of the two tubes is arranged in parallel.

In the operation of a push-push frequency doubler, when the input signal causes the grid of V_1 to swing in a positive direction, the grid of V_2 will swing in a negative direction. There will be a current pulse through V_1 which is fed to the tank circuit consisting of L_2 and C_2, shock exciting the circuit into oscillation. During the next half cycle, when the grid of V_2 is driven in a positive direction, V_1 is driven in a negative direction, and a second current pulse is fed to the tank circuit consisting of L_2 and C_2. The tank circuit consisting of L_2 and C_2 is tuned to twice the input frequency. It receives a pulse during one half of the input cycle from V_1 and a pulse during the other half of the input cycle from V_2. These pulses arrive at the correct time to keep the circuit consisting of L_2 and C_2 oscillating at twice the input frequency.



(i)

(i) In the grounded-grid amplifier, the grid of the tube is operated at signal ground potential. The rf input signal is fed to the cathode of the tube and the output signal is taken from the plate circuit as shown in the diagram. In some grounded-grid amplifiers, the grid bias may be developed in the cathode circuit of the tube, but in the circuit shown the dc grid bias is fed into the grid of the tube. The grid is held at signal ground by capacitor C_1.

The advantage of the grounded-grid amplifier is that it requires no neutralization. The grid of the tube acts as an effective shield between the input and output circuits so that energy cannot be fed from the plate circuit back to the cathode circuit to produce oscillation. One of the disadvantages of the grounded-grid amplifier is that a somewhat higher driving power is required. However, in spite of this, grounded-grid amplifiers are quite widely used, particularly at the higher frequencies where effective shielding between the output and input circuits is very important.

294. Explain in a general way how radio signals are transmitted and received through the use of amplitude modulation.

In amplitude modulation a carrier wave is transmitted. This carrier wave is varied in strength or amplitude by means of a modulator. The amplitude of the carrier varies at a rate determined by the frequency of the signal to be transmitted. The amount of variation in amplitude is determined by the amplitude of the modulating signal. Thus we have a signal that is varying in amplitude at different frequencies depending upon the amplitude and the frequencies of the signal to be transmitted.

In the receiver, the signal is intercepted and picked up by the antenna. It is amplified and then fed to a detector which is actually a form of a rectifier. The output from this rectifier varies in amplitude with the amplitude of the incoming signal. The frequency at which the amplitude varies is determined by the frequency of the incoming signal. Thus we develop a signal voltage in the output of the detector which is the equivalent of the original signal used to modulate the transmitter. This signal is amplified by audio amplifiers in the receiver and then fed to a speaker where it is converted back into sound.

295. In amplitude modulation, what is the relationship between sideband power, output carrier power, and percent modulation? Give an example of a problem to determine sideband power if other necessary information is given.

If an AM transmitter is to be modulated 100%, the audio modulator must supply a power equal to 50% of the carrier power. Thus the power in the sidebands is 50% or one half of the carrier power. The power for any given percentage of modulation can be determined from the formula:

$$P = \frac{m^2}{2} \times P_C$$

where m is the modulation index and P_C is the power in the carrier. The modulation index can be determined by dividing the percentage of modulation by 100. For example, when a transmitter is 100% modulated, the value of the modulation index will be:

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

and therefore the value of m equals 1. If the transmitter is 50% modulated, the value of m will be:

$$\frac{50}{100} = .5$$

Thus if a transmitter has a carrier power of 1000 watts, and the modulation is 50%, we can get the power of the sidebands from the formula by substituting the known values. Thus we have:

$$P = \frac{.5^2}{2} \times 1000$$

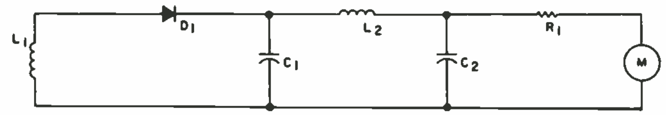
$$= \frac{.25}{2} \times 1000$$

$$= 125 \text{ watts}$$

296. What is a carrier shift? How is it measured? Show by a circuit diagram one method of measuring carrier shift.

Normally when a carrier is amplitude modulated, the two halves of the modulating signal are symmetrical. This means that during one half of the modulating cycle the amplitude of the carrier increases and during the next half cycle the amplitude decreases. If the change is symmetrical, there will be no change in the current reading of the milliameter in the plate circuit of the modulated stage. However, if the modulation is nonsymmetrical so that the positive half of the modulation exceeds the negative half, then the plate current meter will kick up indicating a higher current value. If during the modulation process the meter kicks down, then we have a negative carrier shift. This indicates that the negative half cycle of modulation is greater than the positive half cycle and the dc plate current of the modulated stage is decreasing. Carrier shift does not refer to any shift in frequency of the carrier, but refers only to a change in the plate current reading due to a nonsymmetrical modulation of the two halves of the carrier.

Carrier shift can be measured or indicated by watching the plate current meter in the plate circuit of the modulated stage. It can be also measured by a circuit similar to the one shown.



In the circuit shown, a small amount of power is taken from the final amplifier stage and fed to the diode rectifier. The filter network smooths the rectified current to a pure dc. The coupling between the indicating device of the final amplifier stage is adjusted by moving L_1 closer to or further from the transmitter to give some reading less than full scale on the meter. Usually it is convenient to adjust the coupling to give you a half-scale reading. Then, as modulation is applied, if the meter kicks upscale, we have positive carrier shift, and if the meter kicks downscale, we have negative carrier shift.

297. Explain the direct and the indirect methods of calculating operating power of broadcast stations. Give an example of each method.

In the direct method of measuring the operating power of a broadcast station we must know the transmission line impedance. If we know this, we can measure the line current and use the formula:

$$P = I^2 \times R$$

Thus if we have a transmitter where the antenna line current is 10 amps and the line impedance is 52 ohms then we can determine the output power using the formula and we will have:

$$P = 10^2 \times 52$$
$$= 100 \times 52$$
$$= 5200 \text{ watts}$$

In the indirect system we use the plate voltage and plate current of the tube times the efficiency of the final stage. As an example, suppose that the plate voltage applied to the final amplifier stage is 5000 volts and the current is 2 amps. If the efficiency is 36%, then the power output can be determined from the formula:

$$P = E \times I \times \text{Eff}$$
$$= 5000 \times 2 \times .36$$
$$= 3600 \text{ watts}$$

298. In relation to the safety of the radio operator, explain the function of:

(a) Interlocks
(b) Circuit breakers
(c) Bleeder resistors

All of the devices referred to in this question are subject to failure. Therefore none of them should be solely relied on to protect the operator. The operator should see to it that the power is removed from all dangerous circuits, that all filter capacitors are discharged, and that all other safety precautions have been observed before working on a transmitter.

The operator who observes all of the safety precautions can count on being around to collect his social security benefits. The radio operator who ignores these safety precautions is simply ignoring his responsibility to protect his own life.

(a) Interlocks are designed to remove the operating voltages from the transmitter when the screens or the doors of the transmitter are removed without turning off the power. The various interlocks in the transmitter are usually connected into a series circuit so that if any one of them is opened, all dangerous voltages are removed from the transmitter.

(b) Circuit breakers are used to prevent transmitter overloads or excessively high temperatures. They are primarily designed to protect the equipment but will also protect the operator by preventing a dangerous fire.

(c) Bleeder resistors are connected across filter capacitors. Their purpose is to discharge the filter capacitors when the power has been removed from the transmitter. Thus the danger of the technician being shocked by a charged capacitor is avoided.

299. Explain the method of cleaning relay contacts. Why is it necessary that the original contact shape be maintained?

Relay contacts, if they are only slightly dirty, can be cleaned with a burnishing tool. A burnishing tool is simply a very hard piece of metal that has been roughed so that it will remove a very small amount of material from the contact. Thus if the contact is simply dirty or slightly oxidized, the burnishing tool will clean the contacts satisfactorily.

If a contact is pitted or charred then it must be filed with a very fine file. The file must be very clean; for that matter it is a good idea to get a fine file and save it for the purpose of cleaning contacts only.

The contact shape must be maintained. The relay manufacturer has designed the contacts for a specific operation. For example, if a relay must open and close a circuit where there is a fairly large current, there will be a large, flat area on the relay giving the contact surface sufficient cross section to handle the current. If you should file the contacts so that only a small area is making contact, then the area will be insufficient to handle the current. This will cause overheating and in a short time the contacts will be pitted and burned again. Before filing any relay contact, you should carefully note the shape and make sure that you remove equal amounts of metal over the entire surface of the contact in order to maintain the original shape of the relay contact.

300. Explain the operation of the following relays:

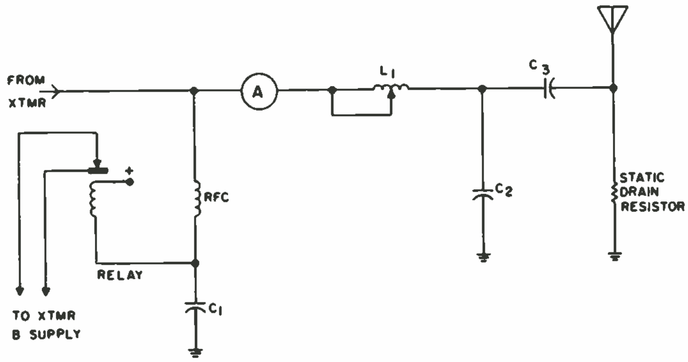
(a) Overload
(b) Time-delay
(c) Recycling

(a) An overload relay is a relay which is designed to operate if a current through it exceeds a certain value. For example, an overload relay might be used in the plate circuit of a vacuum tube. Under normal operating conditions, the current through the relay is insufficient to cause the relay to operate and the operating voltages are applied to the plate of the tube. However, if a defect should develop in the stage, the relay will operate. For example, in the case of an rf power amplifier using grid-leak bias, if the excitation should fail, the plate current will go up to a high value. If there is no overload relay in the plate circuit of the tube, the current rating of the tube will be exceeded. If there is an overload relay, the high current will cause it to operate and remove the plate voltage from the tube.

(b) A time-delay relay is a relay that does not close until a certain time has elapsed. Time-delay relays are frequently used in power supplies using mercury-vapor rectifier tubes. In a power supply that uses mercury-vapor rectifier tubes, it is desirable to allow the filament of the rectifier tubes to come up to operating temperature to evaporate any mercury that may have condensed in the tube. The mercury will then turn to vapor and the tube will be ready for operation. The time-delay relay starts operating when the power is first applied. The rectifier tubes have time to heat up and evaporate any condensed mercury and then the time-delay relay closes applying the power to the rectifier tube.

(c) A recycling relay is a relay that recloses after it has been opened by an overload. For example, an overload relay might open due to some temporary transient. If the relay simply remains open, the transmitter will be off the air. In a recycling relay, after a few seconds the relay closes applying power to the vacuum tube. If the overload has disappeared, the relay remains closed and power will still be applied to the tube. However, if the overload is still present, the relay will open again. If the overload persists, the relay will open and close several times. Most recycling relays are arranged to open and close two or three times and then if the overload is still present the relay will remain open so that the operator will have to look into the cause of the overload.

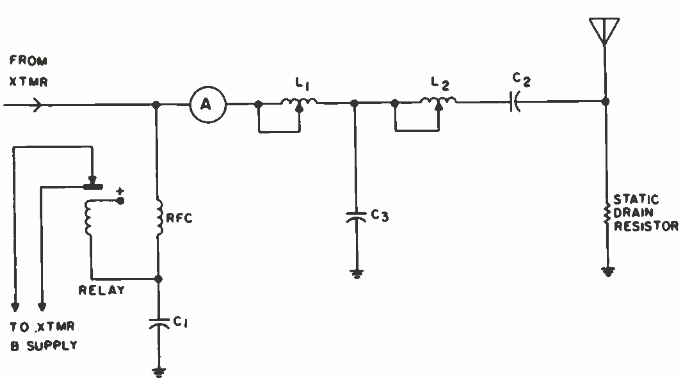
301. Show by a circuit diagram two methods of coupling a standard broadcast transmitter output to an antenna. Include a provision for matching impedance, attenuating harmonics, and guarding against lightning damage.



In the first circuit we have used an L-type impedance matching network to couple the output of the transmission line to the antenna. This type of coupling network is also a low-pass filter and hence will discriminate against frequencies above the operating frequency. Therefore it will be effective in reducing harmonics.

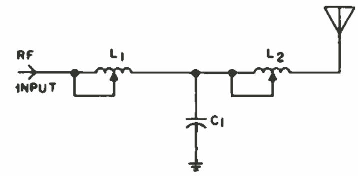
The arc suppression circuit consists of a relay which is used to remove the power from the transmitter in the event that the antenna is struck by lightning. If lightning strikes the antenna or strikes near the antenna there will be a flashover in the coaxial line or in the coupling network. Once this flashover occurs the power output stage will continue supplying power and sustain the flashover. This could cause the final stage to draw excessive power and damage the equipment.

Notice that in the arc suppression circuit the relay coil is connected to the positive side of a power supply. There is no dc return to the negative side of the power supply which is ground. However, when a flashover occurs the return path is completed through the arc. When this happens the relay operates, opening the power to the final power amplifier stage. This happens so quickly that there is very little interruption to the broadcast; as a matter of fact, it is almost impossible to detect the interruption.



In the second circuit the same type of flashover protection is used. However, in this circuit we have used a T impedance matching network to couple the transmission line to the antenna. The T network is also a low-pass filter and hence it attenuates signals above the operating frequency and therefore is effective in the suppression of harmonics. The static drain resistor shown in both circuits is used to prevent a voltage buildup on the tower due to static pickup from nearby lightning flashes. The buildup could be sufficient to cause a flashover and operate the arc suppression relay if we did not provide some means for it to drain off the antenna.

302. Explain the method of adjusting a T network of two tunable coils and a fixed capacitor in order that a standard broadcast station operating at 1340 kHz will be properly coupled to the antenna.



In the diagram shown we have included only the T match and the antenna to simplify the circuit. The two coils used in the T match are labelled L_1 and L_2 and the capacitor C_1. The capacitor C_1 is a fixed capacitor and therefore all of the adjustments must be performed with the two coils L_1 and L_2.

The first step in matching the antenna to the transmission line is to measure the impedance of the antenna using an rf bridge. You should know the impedance of the transmission line since it is readily available from the manufacturer of the line. Once you have the impedance of the antenna and the impedance of the transmission line, you can set coils L_1 and L_2 using the rf bridge.

The procedure is as follows. Let's assume that the antenna measures a resistance of 35 ohms and a capacitive reactance of −20 ohms. Let's assume that the cable is a 50 ohm transmission line. The reactance of the coils L_1 and L_2 can be found from the formula:

$$X_L = \sqrt{R_1 \times R_2}$$

where R_1 is the resistance of the transmission line and R_2 is the antenna resistance. Substituting these values in the formula we get:

$$\begin{aligned} X_L &= \sqrt{50 \times 35} \\ &= \sqrt{1750} \\ &= 41.8 \text{ ohms} \end{aligned}$$

Thus to cancel out the capacitive reactance of −20 ohms of the antenna we need to add an equal amount of inductive reactance to L_2. Therefore L_2 should be set using the bridge to 41.8 + 20 = 61.8 ohms. L_1 is set to 41.8 ohms. After you have the coils adjusted you can use the rf bridge to check the impedance match. With the transmission line disconnected and with the antenna connected at the output of the T network, you should read 50 ohms at the input to L_1. If you get this value then you know that the transmission line will be correctly terminated in its characteristic impedance. If the input to L_1 reads slightly off from 50 ohms you can make whatever minor adjustments are necessary to correct the discrepancy.

303. Define field intensity. Explain how it is measured.

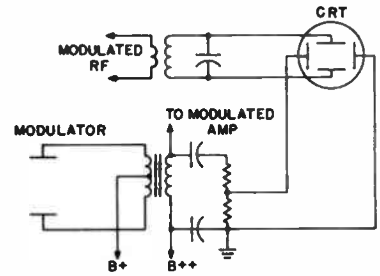
Field intensity is a measure of the strength of a radio wave at a given location. It is usually expressed in terms of microvolts per meter. The field strength tells you what voltage will be induced per meter of antenna length. For example, if we state that the field strength of a signal is 10 microvolts per meter this means that an antenna that is 1 meter long would pick up a signal of 10 microvolts.

Field intensity is measured by means of a field strength meter. A field strength meter is simply a receiver with a loop antenna and a microammeter in the second detector circuit. The receiver is calibrated so that the meter indicates the field strength in microvolts per meter. In using the field strength meter, rotate the antenna for maximum signal pickup. At this point the signal strength can be read off the microammeter.

304. Cathode-ray oscilloscopes are frequently used to register percentage modulation. Sketch the visual displays of:

(a) 0% modulation
(b) 50% modulation
(c) 100% modulation
(d) 120% modulation

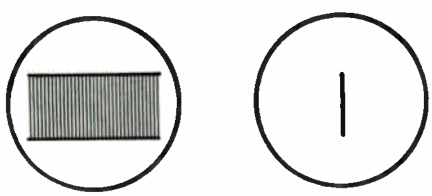
Two sets of waveforms are shown for each percentage of modulation. The first waveform is a so-called envelope waveform. This type of waveform is obtained by using some type of pickup coil and feeding the signal directly to the vertical deflection plates of the oscilloscope. The horizontal sweep on the oscilloscope is adjusted to some convenient frequency in order to obtain a stationary pattern on the screen. Usually an audio signal generator is used to modulate the transmitter to get a constant audio frequency so the modulation percentage can be examined. Speech displays produce an irregular waveform and very little can be determined about the modulation percentage from this type of waveform.



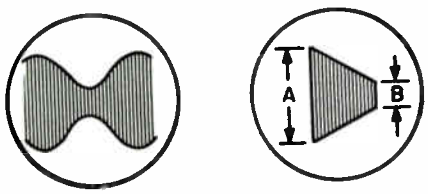
The second waveform in each case is the so-called trapezoidal pattern. This type of pattern can be obtained using the circuit shown. The modulation percentage can be quite readily determined from this type of waveform by the amplitude of (a) and (b). The percentage of modulation is equal to

$$\frac{a-b}{a+b} \times 100$$

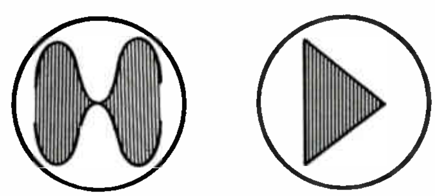
The trapezoidal pattern may be somewhat easier to evaluate, but since it requires additional connections to the transmitter it is easier to use the waveform pattern.



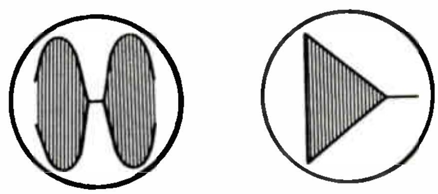
(a)



(b)



(c)



(d)

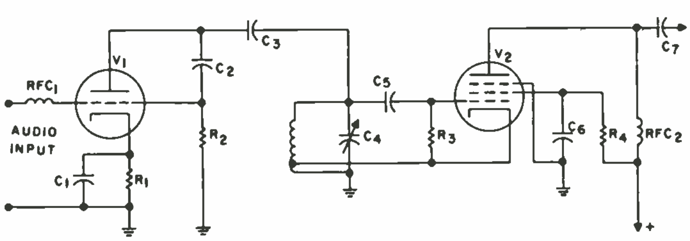
305. Explain in a general way how radio signals are transmitted and received through the use of frequency modulation.

In a frequency modulating system, the power output of the transmitter does not change. The frequency of the carrier is varied in proportion to the strength of the modulating signal and at a rate determined by the frequency of the modulating signal. Thus if we are transmitting a 400 Hertz audio signal, this will cause the output frequency to swing above and below the carrier frequency at a rate of 400 Hertz. The amount that the carrier frequency will swing depends upon the strength of the signal. In standard FM broadcasting the maximum deviation or swing, as it is sometimes called, is limited to 75 kHz above and below the carrier frequency. In the FM sound used in television, the deviation is limited to 25 kHz above and below the carrier frequency. In the narrow-band systems used for FM mobile communications, the deviation is limited to 5 kHz above and below the resting frequency.

The maximum rate at which the carrier is varied depends upon the fidelity required of the system. In FM broadcasting, the modulating frequency is limited to 15,000 Hertz. This means that the carrier frequency may be changed at a rate up to 15 kHz. Of course, in most parts of an FM broadcast, the frequency of the modulating signal will be considerably less than 15,000 Hertz and therefore the frequency will be deviating at a lower rate.

In the receiver, the signal is picked up and amplified in essentially the same way as an AM broadcast is handled. It is fed to a mixer where it is converted to a lower i-f frequency and then amplified by the i-f amplifier. However, the last stage of an i-f amplifier is often a limiter. This stage is designed to clip the signal so that the signal at the output has a constant amplitude. Thus any atmospheric noise that could override the signal is flattened out and will be no stronger than the signal. The signal is then fed to a detector which responds to the frequency variations. Some FM detectors are also sensitive to amplitude variations. Thus the limiter used as the last i-f stage prevents any amplitude variations on the signal from reaching the detector.

306. Draw a circuit diagram of a reactance-tube modulator and explain its operation.



The oscillator, V_2, is a conventional electron-coupled Hartley oscillator. Part of the signal from the oscillator tank circuit is fed through C_3 to a phase-shifting network consisting of C_2 and R_2. In this circuit the values are selected so that the reactance of C_2 is much larger than the resistance of R_2. As a result, the circuit will act like a capacitive reactance. Thus the signal voltage fed from the oscillator tank circuit to this network will cause a current to flow through the network that will lead the oscillator signal voltage by 90°. Therefore a voltage will be developed across the resistor R_2 which leads the oscillator tank voltage by 90°. This voltage is applied to the grid of the reactance tube V_1 and produces an rf current that leads the oscillator voltage by 90°.

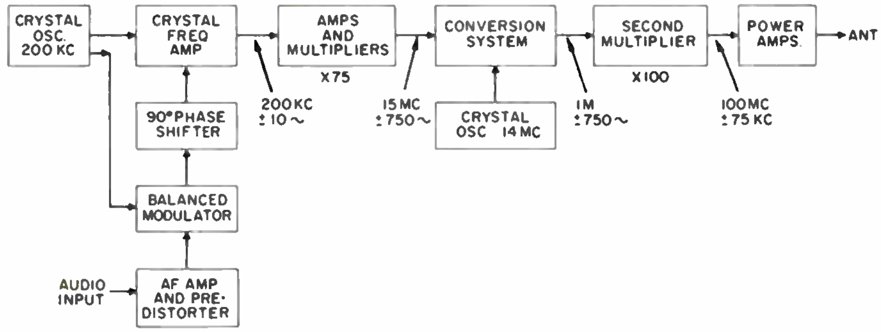
The exact amount of current that will flow through the reactance tube will be determined by the bias on the reactance tube. This bias is developed across R_1. When an audio signal is applied to the input of the reactance tube it will cause the current flowing through the reactance tube to vary. Thus the varying leading current that flows through C_3 is added to the oscillator tank circuit current. This will have the effect of changing the current flowing through the capacitor, since this leading current will be in-phase with the capacitor current. If the current increases, it will act like the value of the capacitor has increased because the capacitive reactance must have decreased. Conversely, if the current decreases then it will act like the value of capacitance has decreased because the capacitive reactance has increased. This effect of changing the capacitance in the tank circuit by means of the reactance tube current will change the resonant frequency of the oscillator.

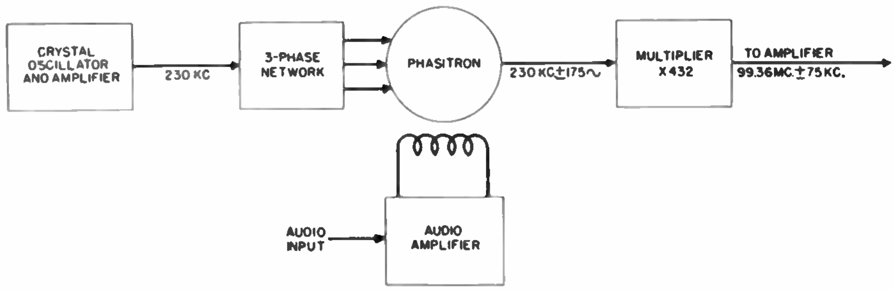
Reactance modulators may introduce either leading or lagging currents into the oscillator tank circuit depending upon the combinations of resistance and capacitance or resistance and inductance used in the phase shifting network.

307. What is the difference between frequency and phase modulation?

There actually is very little difference between frequency and phase modulation. For example, when the frequency of a carrier is increased or decreased the phase is changed at the same time. Similarly, as the phase relationship between two cycles of a carrier is changed there is a corresponding change in frequency.

In the so-called direct FM system, the frequency of the oscillator is changed with modulation so that the oscillator frequency actually increases above and below the carrier resting frequency. In the indirect system, the phase of the signal is changed rather than the frequency. Thus, there is actually very little difference between the two types of modulation. An FM receiver detects frequency modulation and phase modulation equally well.





308. Describe briefly the operation of the Armstrong and phasitron methods of obtaining phase modulation.

In the Armstrong system shown in the block diagram, the output of the crystal oscillator supplies two channels with the signal. One signal goes directly to the crystal frequency amplifier and serves as the center frequency in the formation of an FM signal. A second signal from the crystal oscillator is supplied to a balanced modulator. This forms amplitude-modulated sidebands when the audio signal is simultaneously applied to its input. The balanced modulator removes the carrier components supplied to it originally, and only the sidebands are fed to the phase-shifter at its output. The sideband output of the phase shifter is then recombined with the center frequency in the crystal frequency amplifier to form a phase-modulated signal.

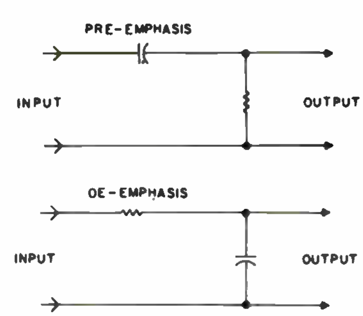
The phase-modulated signal with its limited amount of frequency deviation is now applied to a group of frequency multipliers and a converter to develop a final frequency-modulated signal with adequate deviation. The function of the conversion system is to reduce the frequency of the carrier and its sidebands so the signal can be fed to additional frequency multipliers. In this manner, it is possible to get full deviation of the carrier, although the original deviation at the output of the phase modulator is small.

The phasitron is a special tube that was designed specifically to obtain wide-angle phase modulation. A simplified block diagram of a transmitter using a phasitron is shown. A crystal oscillator is used to generate a stable center-frequency signal. In the example shown, this 230 kHz crystal-controlled component is applied through a 3-phase network to the phasitron tube. The audio signal also is coupled to the phasitron tube. The output signal will have a center frequency of 230 kHz ± 175 Hz. By feeding this signal to a number of multipliers to multiply the frequency 432 times, we obtain a center operating frequency of 99.36 MHz and a maximum frequency deviation of ± 75 kHz.

As mentioned previously the phasitron is a special tube. The tube has three beam-deflecting anodes in addition to the usual cathode, control grid, screen grid, suppressor grid, and anode. The output from the 3-phase network is fed to these deflecting anodes. The tube is completely surrounded by a coil which is energized by the modulating signal. The field formed within the tube by the three deflecting anodes and the coil produces a frequency modulated output.

309. What is the purpose of pre-emphasis in an FM transmitter? Of de-emphasis in an FM receiver? Draw a circuit diagram of a method of obtaining pre-emphasis.

The basic purpose of pre-emphasis is to provide a better signal-to-noise ratio in the upper audio range. There is very little energy in the high frequency audio signals and these signals could be lost or masked by noise. To overcome this problem, the higher audio signals are given extra amplification. This extra amplification is provided by means of a pre-emphasis circuit such as shown.



In both circuits a small value of C is used. In the pre-emphasis circuit this passes the high audio frequencies readily but attenuates the low frequencies. In the de-emphasis circuit the capacitor reduces the strength of the high frequency signals without affecting the low frequencies.

The pre-emphasis circuit makes it possible for us to have a good signal-to-noise ratio at the higher audio frequencies and thus transmit a more natural sounding signal.

De-emphasis is used in the receiver to restore the original ratio between the signals. The high frequency signals which are emphasized by the pre-emphasis circuit are de-emphasized by the de-emphasis circuit so that the amplitude of the higher frequency signals is correct in relationship to the amplitude of the middle and lower frequency signals.

310. What is effective radiated power? Given transmitter power output, antenna resistance, antenna transmission line loss, transmitter efficiency and power gain, show how ERP is calculated.

The effective radiated power is the power that is actually delivered to the antenna multiplied by the power gain of the antenna. Thus if the power actually delivered to a nondirectional antenna is 1000 watts and the power gain of the antenna is 5, the ERP is 5000 watts. In the case of a directional antenna, the effective radiated power is the actual power delivered to the antenna multiplied by the gain of the antenna in the direction in which the antenna radiates.

In most cases, the loss in the transmission line and the gain of the antenna are given in db. Therefore in order to determine the effective radiated power, we have to convert these to a ratio. For example, suppose that the power gain of the antenna is 18 db and the loss in the transmission line is 5 db. Subtracting the transmission line from the antenna gain we have an effective gain of 18 - 5 = 13 db. To convert this to a power ratio we use the formula:

$$\text{db} = 10 \log \frac{P_1}{P_2}$$

Now substituting 13 in the formula, we have:

$$13 = 10 \log \frac{P_1}{P_2} \quad \text{or}$$

$$1.3 = \log \frac{P_1}{P_2}$$

Using a logarithm table we find that the log of 2 is .3, therefore 1.3 must be the log of 20. This means that we have a power gain of 20. If the actual power output of the transmitter is 1000 watts then the ERP would be:

$$\text{ERP} = 1000 \times 20 = 20{,}000 \text{ watts}$$

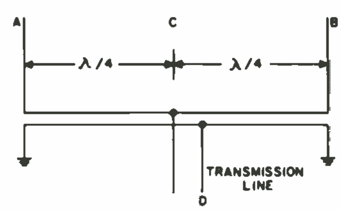
311. What type of antenna site is technically best for an AM broadcast station? For an FM broadcast station? For a vhf television station? For a uhf television station?

The most widely used antenna for AM broadcast service is the Marconi-type antenna. This type of antenna gives best operation in areas where there is good ground conductivity. Therefore the best location for this type of antenna would be where the ground is damp.

Since vhf and uhf television signals as well as FM signals travel only in straight lines because of the high frequencies used, the best thing you can do with this type of antenna is put it as high as possible to get maximum coverage. Therefore the antenna should be placed on the top of a hill, if there is one nearby. If there isn't, it should be put on the top of the tallest building or on the top of a tall tower.

312. How does a directional antenna array at an AM broadcast station reduce radiation in some directions and increase it in other directions.

In a directional array system the antenna elements are arranged so that the signals from the elements reinforce each other in the direction in which we wish to transmit and oppose each other in the direction in which we wish to prevent radiation. In other words, in one direction the signals are in-phase so they add. In other directions they will be out-of-phase so they cancel.



In the example shown, the two antennas, A and B, are placed one-half wavelength apart. Since they are fed from a common transmission line which is equidistant from the two antennas, the signal will arrive at the two antennas at the same time. As a result, the signal at A and the signal at B will be exactly in-phase. If they travel in the direction of C-D, the two signals will add together and reinforce each other. However, let's consider what happens when the signal travels from A to B. During the time it would take to get from A to B, the signal at B will be 180° out-of-phase with the signal at A. Since the two signals are of equal amplitude they should cancel, since there is no radiation in the direction from A to B. Similarly, a signal leaving B and traveling towards A will arrive at A after the signal at A has undergone a 180°

phase shift so they will cancel in this direction. Thus we have no radiation in the direction A-B, but a maximum radiation in the direction C-D.

Variations in this complete cancellation can be made by varying the amount of power fed to the two antennas. In other words, if the power fed to element B is only half that fed to A there will not be complete cancellation in the direction A-B, but the radiation will be reduced. At the same time, in the direction of C-D, the two signals will still add to increase the radiation in this direction. By using several elements and controlling the phase of the signal fed to each element and the amount of power fed to these elements, almost any desired radiation pattern can be produced.

313. What factors can cause the directional antenna pattern of an AM station to change?

Any factor that changes the phase or the amplitude of the signal fed to any one of the elements of an antenna system will cause the pattern to change. Thus, changes in tuning which may be caused by a change in the temperature or humidity can produce changes in the antenna pattern.

Large changes in the radiation pattern can also be caused by the construction of buildings containing large amounts of metal close to the antenna site. This is particularly true if the length of any of the metal structures in the building approach a quarter wavelength. Thus the construction of a large building near the antenna site may necessitate a change in phasing and amplitude of the signal fed to some of the antenna elements in order to maintain the desired radiation pattern.

314. What adjustable controls are normally provided at an AM broadcast station to maintain the directional pattern?

Since the radiation pattern is affected by the power fed to the various elements and by the phasing of the signal fed to these elements, controls which permit variation of the power and the phasing to each element are effective in controlling the directional pattern.

The control of the power to the individual elements is usually managed by adjustable tapping points on the tank inductance in the transmitter.

The phasing is usually controlled by the variable inductors in the phasing network. These are usually located at the base of the antenna. Sometimes these phasing devices must be adjusted manually, and other times they can be controlled remotely by motors that can be operated from the transmitter.

315. Define polarization as it refers to broadcast antennas.

Polarization refers to the direction of the electric field radiated by the antenna. The electric field is always parallel to the physical direction of the antenna. Since Marconi antennas (which are vertical antennas used in the broadcast band) radiate a vertically-polarized electric field, a broadcast antenna is a vertically-polarized device.

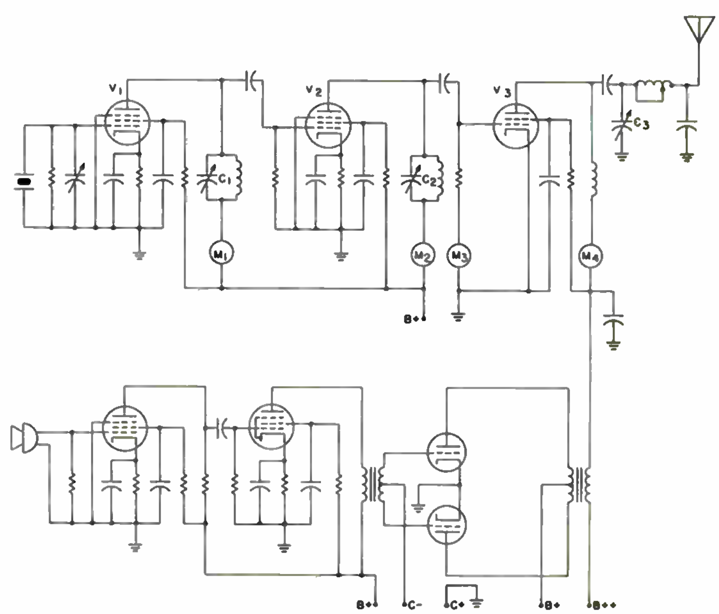
In other services where the radiating elements may be horizontal to the earth's surface, the electric field radiated by the antenna will be horizontal to the surface of the earth and hence horizontally polarized.

316. What is the importance of the ground radials associated with standard broadcast antennas? What is likely to be the result of a large number of such radials becoming broken or seriously corroded?

Ground radials are a series of wires running from the base of the antenna out in the form of spokes in a wheel. These radials, which are buried in the ground, are usually made as long as possible because the longer they are the better the ground and the better the radiation from the antenna will be for a given amount of power. Ground radials are particularly important in locations where the conductivity of the soil is poor. Broken radials will introduce losses and reduce the efficiency of the antenna system, thus degrading the transmission from the station.

317. Draw a circuit diagram of a complete radiotelephone transmitter composed of the following stages:

(a) Microphone input connection.
(b) Pre-amplifier.
(c) Speech amplifier.
(d) Class B modulator.
(e) Crystal oscillator.
(f) Buffer amplifier.
(g) Class C modulated amplifier.
(h) Antenna output connection.
(i) Insert meters in the circuit where necessary and explain, step-by-step, how the transmitter is tuned.



In the schematic diagram shown an electron-coupled crystal oscillator is used. Heater voltages should be applied to the transmitter and to the crystal oscillator and the oscillator allowed to come up to operating temperature. Next, apply B+ voltage to the oscillator, V_1, and adjust the oscillator tuning capacitor C_1 until the reading on M_1 is a minimum. Next, apply operating voltages to the buffer, V_2, and adjust the tuning capacitor in the buffer tank circuit, C_2, for a minimum current indication on the meter M_2. At the same time, observe the meter M_3 in the grid circuit of the Class C power amplifier, V_3. It should reach a maximum at the same time that M_2 is at a minimum.

Finally, to tune the Class C power amplifier apply plate and screen voltages to the tube and then adjust C_3 for a minimum current reading on M_4. Now adjust L_1 and the current reading on M_4 should begin to increase. When this happens, readjust C_3 for a minimum reading. Continue adjusting C_3 for a minimum and L_1 for a maximum, until the meter M_4 indicates that V_3 is drawing its rated plate current.

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NRI Study Guide for FCC License Examination

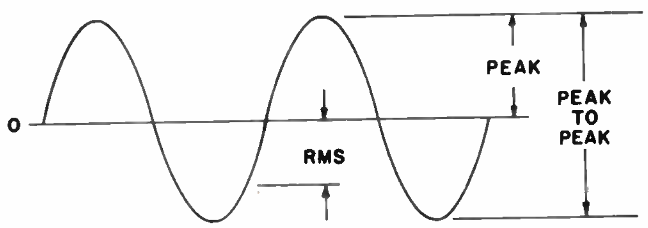
PART XIV

The subjects covered in this final section of the study guide are more of the topics which cover the questions you are likely to be asked in the examination for Element IV of your FCC License Examination. The topics cover a wide range of subjects dealing with broadcasting. They are covered in sufficient detail in the study guide so that you shouldn't have any difficulty in answering the questions on these subjects in your examination.

Be sure to save this final part of the study guide along with those you received previously so that you can review them before taking the examination for Element IV of the FCC License Examination.

318. In relation to ac circuits, what is the relationship between (1) rms values (2) maximum and minimum values (3) peak values and (4) peak-to-peak values?

In the illustration shown, the ac value swings above and below zero so that the polarity of the voltage or current is continually reversing. The rms (root mean square) value is the effective value. In other words, it has the same effect as the equivalent dc. It is equal to .707 times the peak value; the peak value is 1.414 times the rms value. When we say that the ac current is 1 amp we mean that it has the same heating effect as 1 amp of dc. The peak value of the current flowing would actually be 1.414 amps but the effective or rms value is 1 amp.

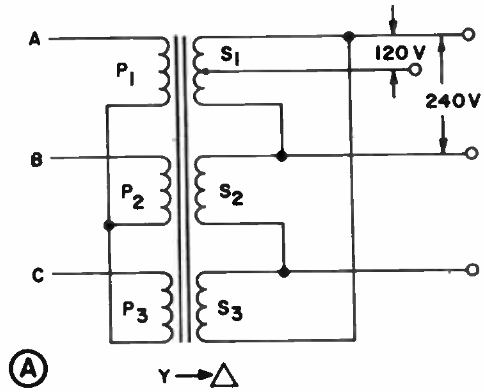


The maximum value is shown. It is the greatest or peak value of the current. This is labeled on the diagram. The peak value is the same as the maximum value. The minimum value in the case of the sine wave shown is zero. In some nonsinusoidal ac signals the minumum value will not necessarily go to zero.

The peak-to-peak value is labeled. In the case of the sine wave ac signal it will be 2.828 times the rms value. It's the amplitude between the positive peak and the negative peak.

319. Show by diagrams the delta method and the wye method of connecting transformer secondaries in a power distribution system. Show also how various output voltages might be obtained from each.

(a) The wye-delta transformer connection is quite frequently used in stepping down the higher line voltages used to transmit electric power to the value used in residences and businesses. An example of the wye-delta three-phase connection is shown.

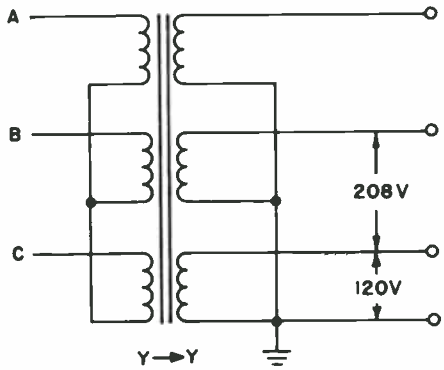


In this example the primary is connected in a wye configuration. Notice that the voltage between lines A and B is applied to both P_1 and P_2 in series. The actual voltage appearing across either P_1 or P_2 will be equal to the line voltage between A and B divided by the square root of 3. This is an advantage in stepping down the voltage because with the secondary connected in delta, even with a 1:1 turns-ratio, we would get approximately a 1:.58 step-down between the primary and the secondary. Thus the difference between the number of turns on the primary and the number of turns on the secondary is kept at a minimum for a given step-down in voltage.

In the example shown, the delta-connected secondary has sufficient turns to develop 240 volts across each winding. Each winding is center-tapped so that 120 volts can be obtained between the center tap and the ends of each winding. This is an arrangement similar to that used in the average residence where a three-wire system is used to obtain a total of 240 volts to operate such appliances as electric stoves and dryers and 120 volts for lighting purposes and miscellaneous small appliances.

(b) In the wye-wye connected transformer we do not get the advantage of the step-down due to the different primary and secondary configurations that we do in the wye-delta arrangement. Here the step-down is obtained entirely by having the required turns-ratio on the primary and secondary.

In the example shown, where the common connection of the three secondaries is grounded, we can design the transformer to give us 120 volts between the ground and any one of the outside windings. With this arrangement the voltage across two windings will be $\sqrt{3} \times 120$ volts which is equal to approximately 208 volts.



In radio transmitters, three-phase power is often used for the power supply which produces the plate voltage used to power the final amplifier stages. In this case the problem is generally of taking the available line voltage, which may be 220 or 440 volts, and stepping it up to a much higher voltage. A plate transformer designed for this purpose often uses a delta-wye connection. In other words, the primary is connected in delta and the secondary is connected in a wye.

With this arrangement even with a 1:1 turns-ratio we will obtain a step-up of $\sqrt{3}$ times the primary line voltage across any two of the wye-connected secondaries. This arrangement is often used in the full-wave bridge-type rectifier to reduce the number of turns required on the secondary of the transformer to get the required voltage.

320. Why is plate modulation more desirable than grid modulation for use in standard broadcast transmitters? Why is grid modulation more desirable in television video transmitters?

Plate modulation provides much greater efficiency than grid modulation. When you plate-modulate a Class C amplifier efficiencies as high as 75% may be obtained. Thus for a given power output smaller tubes can be used and the overall efficiency of the transmitter is improved. Plate modulation does have the disadvantage that it requires a relatively large amount of audio power. To fully modulate a Class C amplifier the audio power must be 50% of the input power to the Class C amplifier. However, in transmitters up to about 5 kW it is not too difficult to develop this much audio power. For higher power transmitters, an intermediate power amplifier stage is usually plate-modulated and then the modulated signal is fed to a linear amplifier.

Grid modulation does have the advantage that very little audio power is required. However, the efficiency of the Class C stage is low, usually in the vicinity of. 35%. Thus, since better than double the efficiency can be obtained from a plate-modulated stage, it is more desirable in broadcast transmitters.

In television transmitters the output impedance of the final amplifier stage must be very low. This is due to the wide range of frequencies which must be accommodated by the final amplifier stage. You will remember that the video carrier and sidebands have a bandwidth of over 5 MHz.

When a stage is plate-modulated, the modulator impedance must be matched to the impedance of the final amplifier. With such a very low impedance, it would be difficult to match the modulator to the final stage in the television transmitter, and even then extremely high modulation currents would be flowing. This would introduce large losses and generally provide unsatisfactory performance.

By using grid modulation in a television transmitter, we don't run into this problem of high modulating currents and much less video driving power is required to modulate the Class C stage. Therefore grid modulation is more practical.

321. Explain why dry air or inert gases are often used in rf transmission lines which link broadcast transmitters and antennas.

Dry air or inert gases are often used in rf transmission lines to prevent moisture absorption in the lines. Moisture absorption will increase the losses in the line and at the same time lower the flashover voltage. In other words, the moisture may cause the line to flash over. With dry air or inert gases pumped into the line, moisture cannot collect in the line and hence these problems are avoided.

322. Describe the procedure for installing transmission lines between broadcast transmitters and antennas. Include information as to the characteristic impedance, bends, kinks, cutting, and connections. Discuss both the solid dielectric and the gas-filled lines.

Both solid dielectric and gas-filled transmission lines may be installed either above or below ground. When installed above ground they are usually supported on small frames two or three feet above ground. The cables are usually hung on messenger wires rather than secured directly to the frame.

When cables are installed underground, it is good practice to install them in concrete trenches covered by some means to provide suitable protection. They can be buried directly in the ground, but this is not as good as putting them in trenches because if any defect develops in the line, it is very difficult to find the defect if the line is buried.

The outside conductor of both the solid and gas-filled line should be grounded every thirty to forty feet to prevent undesirable radiation from the line and also to provide maximum protection in the event that the line is struck by lightning.

When installing a transmission line you should run it as straight as possible from the transmitter to the antenna. If you have to bend the line, you should use as large a radius as possible to prevent distorting the shape of the line. Distortions in the line shape will upset the impedance of the line. This will cause increased line losses due to standing waves that may be introduced.

A kink in a line is formed by an extremely sharp bend. This will greatly distort the shape of the line, resulting in an appreciable impedance change at the point which will introduce losses and also reduce the breakdown voltage of the line.

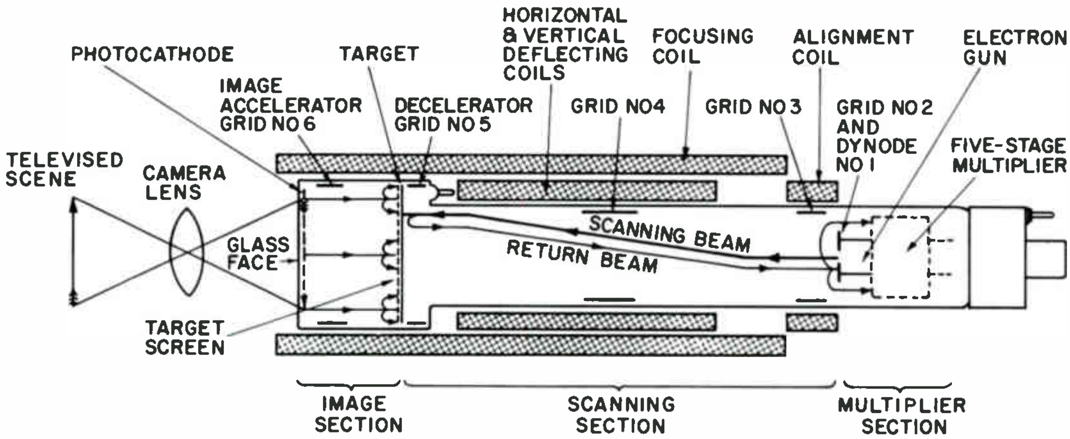
When you are cutting a solid dielectric line, you must be very careful to avoid nicking the center conductor. This could introduce some standing waves, and if the nick is deep enough it could physically weaken the line. When cutting a gas-filled line it is important that you make a square cut. If the cut leaves any burrs, the burrs should be removed and all filings carefully wiped from the inside of the line. If any filings get left inside of the line they will distort the line impedance and also reduce the voltage breakdown of the line.

There are special fittings available for use on solid dielectric lines. The correct type of fittings should always be used. Connections to gas-filled lines are made with special connectors with gaskets or else by connectors that are used to solder the pieces of line together. In the case where you solder the connector, you use as little solder as possible and essentially sweat the connections together. If you use too much solder or flux, it may work inside of the line and result in a change in characteristic impedance and reduce the voltage breakdown.

After you have completed a connection in a gas-filled line, the connection should be tested by filling the line with nitrogen or dry gas and then using a soap-and-water mixture around the connection. If there are any leaks, bubbles will appear at the leak, and the connection should be repaired to make it air-tight.

323. Explain the operation of the image orthicon camera tube. Include in your explanation a schematic diagram of the tube which shows the focusing and scanning details.

The image orthicon consists of three main sections, which are the image section, the scanning section and the multiplier section. A camera lens focuses the televised scene on a photo cathode just inside the glass face of the tube. The photo cathode is a thin plate of glass with a continuous electrically conducting photosensitive surface on the side opposite the source of illumination.



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Illumination on the photo cathode creates an electron image on the photosensitive surface, which immediately releases electrons in proportion to the varying light intensity of the optical image. Under the influence of the external focusing coil and the internal image accelerator, these electrons move perpendicularly from the photosensitive surface toward the target electrode.

The target is a very thin piece of low resistivity glass, about 0.0001 inch thick. When the accelerated electrons from the photosensitive surface strike the glass, secondary electron emission occurs. The electrons emitted are gathered on the target screen. The surface of the target then contains a pattern of positive charges, which correspond to the electron image. Because of the low resistivity of the target glass, the image charge on the target is available for scanning on the reverse side.

The scanning beam from the electron gun scans the rear of the target. The velocity of the beam is decreased in the immediate vicinity of the target by the decelerator grid No. 5, so that it impinges on the target at low velocity and in proper focus at all points of the scanning motion.

Electrons from the beam are deposited on the target in proportion to the positive charge at each point. This is, of course, a spot-by-spot action. Full scanning of the electron image on the target is accomplished by the deflection circuits. If there is no charge on a particular element in front of the beam, all electrons in the beam reverse their direction and proceed back toward the electron gun. This occurs when the corresponding element on the photo cathode is not illuminated (conveying black). If the element has a positive charge, it attracts some electrons from the beam, and fewer electrons return toward the electron gun. For a very brightly illuminated spot on the photo cathode (intense white illumination), the corresponding element on the target will have a high positive charge. As a result, this element withdraws the maximum number of electrons from the scanning beam, and relatively few electrons return toward the gun.

When the scanning beam first leaves the electron gun, it contains a nearly constant number of electrons. The returning beam, however, is minus those electrons deposited on the target. The varying density of the return beam therefore represents modulation by the electron image.

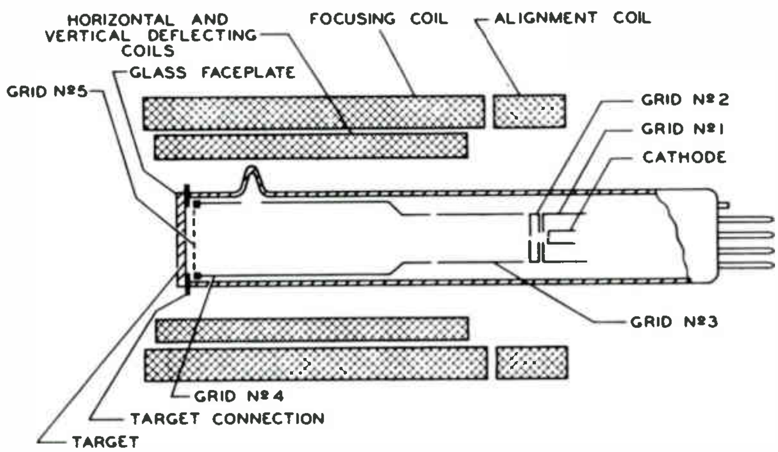
When the modulated return beam reaches the area in front of the electron-gun assembly, it strikes the first dynode of an electron multiplier. This impact causes the emission of secondary electrons. The combined electron stream, which now consists of primary and secondary electrons, moves to the second dynode. This dynode, in being bombarded by the intensified electron stream, produces an even greater number of secondary electrons. This process of electron multiplication occurs at each succeeding dynode in the multiplier. The output, which may represent an overall signal gain of as much as 2000, is finally removed from a signal screen (anode) placed in front of the fifth dynode. This signal current, in passing through a load-resistance, develops the video signal voltage that is applied to the input of the first stage of a video amplifier.

It is essential that the electron gun be aligned correctly with respect to the tube axis during manufacture. Any tilting would cause a spiral electron beam, which would introduce geometric distortion in the viewed picture. Therefore, an alignment coil is mounted around the tube in front of the electron gun to compensate for such tilting.

To complete the discussion of the operation of the image orthicon, grid No. 2 and dynode No. 1 accelerate the scanning beam and form the first section of the multiplier assembly. Grid No. 4 is a focusing electrode which acts in conjunction with the external magnetic focusing coil. Grid No. 3 focuses the electron multiplier.

324. What are the advantages and disadvantages of the vidicon TV camera tube in comparison to the image orthicon tube?

The vidicon camera tube is smaller and less complex than the image orthicon. It differs from the image orthicon in that it uses a photoconductive, rather than a photoemissive, layer as its light-sensitive electrode. The target consists of a transparent conducting film serving as the signal electrode. This film is deposited on the inner surface of the face plate. A thin photoconductive layer is then deposited on this film. A fine-mesh screen, grid 5, is mounted parallel



324.

to and behind the photoconductive layer. Grid 4, the focusing and accelerating grid, is connected to grid 5 so that both operate at the same potential. The voltage applied to grid 3 is made adjustable to act as a vernier focusing adjustment.

Operation is similar to that of the image orthicon. The photosensitive element takes on a pattern of positive charges as the result of the electron emission caused by the optical image. A low-velocity beam from the electron gun strikes this element and discharges it back to zero potential. Unlike the image orthicon, however, the video information in the vidicon is taken directly at the photosensitive element. Excess electrons, instead of returning to the rear of the tube, are collected on the screen, grid No. 5, and returned to the cathode indirectly.

The advantages of the vidicon over the image orthicon tube are several. It is smaller, weighs less and is simpler in construction. The associated circuitry used with the vidicon is simpler and these tubes are comparatively rugged when compared to the fragile image orthicon tube. The power requirements of the vidicon are quite modest.

The disadvantages of the vidicon are that the output signal is lower than that obtainable from the image orthicon. The vidicon requires more illumination and does not have as good a signal-to-noise ratio as the image orthicon. Another disadvantage is that the picture obtained from the vidicon usually has less detail than that obtained with an image orthicon. However, this is partially due to the fact that most vidicons are smaller in size than the image orthicon tube.

325. Describe the scanning technique used in United States television transmissions. Why is interlacing used?

In transmitting a TV picture the picture is broken down into horizontal lines. There are a total of 525 horizontal lines in one complete picture, which is called a frame.

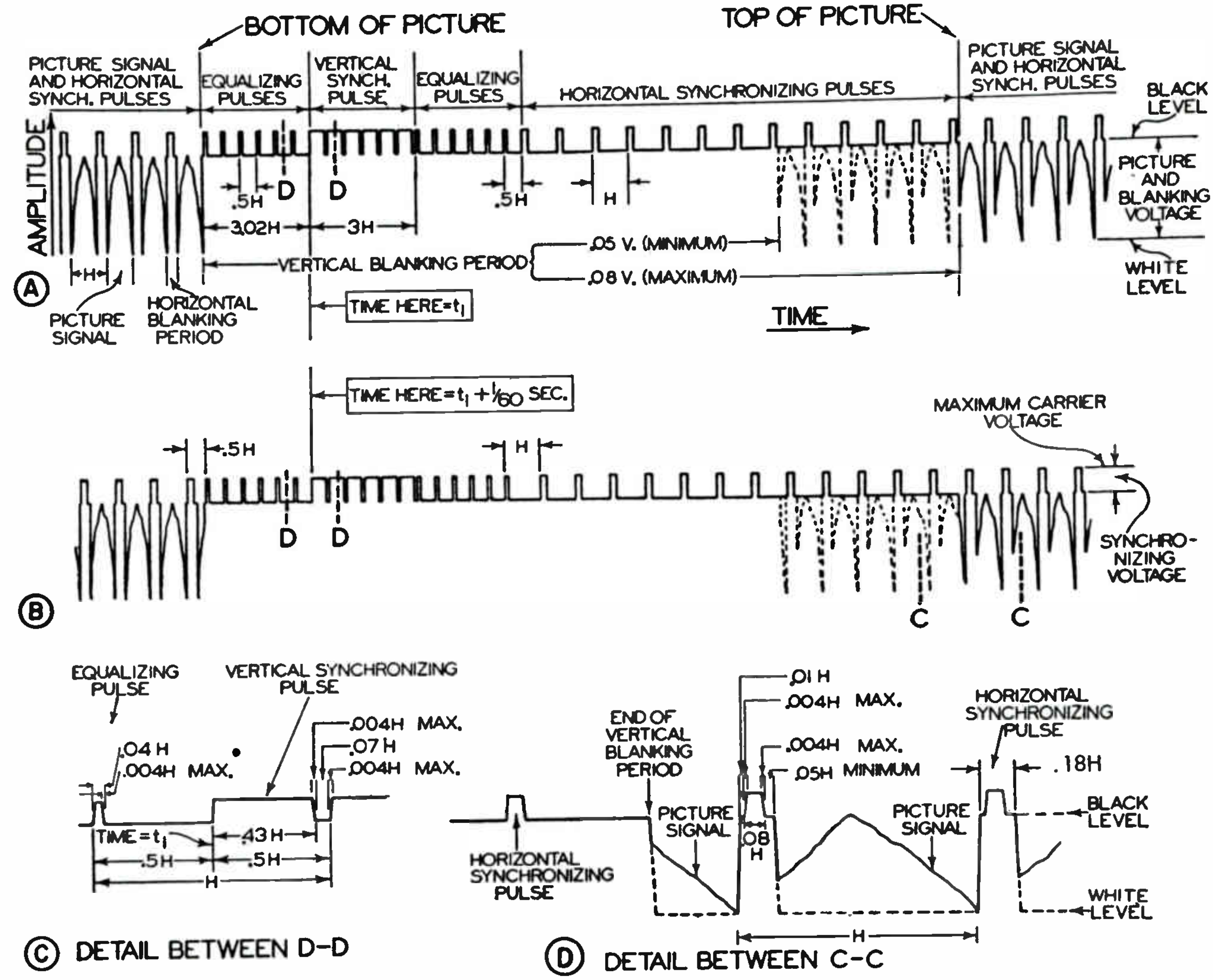
Instead of scanning the picture horizontally so that we scan first line 1 and then line 2, we scan line 1, then line 3 and then line 5, etc. until we have scanned a total of 262-1/2 lines. This is half of 525. Then the electron beam is moved back to the top of the screen where we scan a half a line and then go and pick up lines 2, 4, 6, etc., until we have completed a second field of 262-1/2 lines. These two fields form a single frame or complete picture where we scanned 525 lines. Interlaced scanning is used to get the effect of producing 60 frames per second while actually only producing 30. The higher frame rate reduces flicker in the TV picture.

Not all of the 525 lines are usable when reproducing part of the picture. Part of each line is lost in transmitting a horizontal synchronizing pulse. This pulse is used to synchronize the electron beam in the TV receiver with the electron beam in the television camera. Thus the two beams will be moving across the face of the tube in exact unison, so as the TV camera beam picks up the variations in brightness the TV tube in the receiver is reproducing these variations in brightness in the correct position along each horizontal line.

In addition to the synchronizing pulses at the end of each horizontal line, a vertical blanking and synchronizing pulse is transmitted at the end of each frame. The purpose of the blanking pulse is to blank the TV screen so that the beam in the TV receiver picture tube is cut off while the electron beam is moved from the bottom of the screen up to the top. The time it takes for the beam to move from the bottom to the top is much longer than it takes for the beam to retrace from the right side of the picture back to the left side in the horizontal retrace interval. Thus several horizontal lines are lost while the electron beam is moved from the bottom to the top of the screen. As a result of the lines lost during the two blanking and synchronizing intervals at the end of each field, a complete picture actually consists of about 485 lines. The remaining lines are lost during the blanking interval.

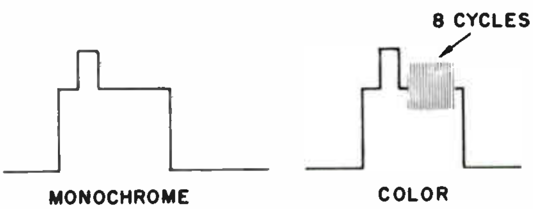
To summarize the scanning technique, the beam in the camera tube and the picture tube are synchronized so that they start at the left side of the screen and move at a constant speed across the face of both tubes until they reach the end of the line. Then a horizontal blanking pulse is transmitted that cuts off the beam and then a horizontal synchronizing pulse, which triggers the sweep circuits and rapidly moves the beam from the right side of the screen back to the left so it is ready to start the next scan. The scanning beam scans first line 1 then line 3 and then 5 until all the odd lines are scanned as the beam moves down from the top to the bottom of the screen. Then the vertical synchronizing pulse is transmitted which cuts off the beam and moves the beam from the bottom back to the top of the tube where the even-numbered lines are then filled in on the next field. Each vertical scan is called a field; two fields make up a complete frame and there are 30 complete frames, or pictures transmitted per second. Once again, the purpose of using interlaced scanning is to in effect increase the number of fields transmitted so that, as far as flicker is concerned, the reproduced picture acts like there are 60 complete pictures per second instead of only 30.

326. Make a sketch showing equalizing, blanking, and synchronizing pulses of a standard U.S. television transmission.



The illustrations shown are for standard 525-line pictures transmitted at a rate of 30 frames per second with standard interlaced scanning giving 60 fields per second. These are the standards used for black-and-white TV transmission in the United States. Notice in the first field, labeled A, the timing of the first equalizing pulse coincides exactly with one of the horizontal sync pulses. Thus the vertical retrace starts at the end of the horizontal line. In the illustration shown at B, we have shown the first equalizing pulse appearing a half line away from the horizontal synchronizing pulse. Thus this vertical retrace starts at the center of the horizontal lines.

327. Make a sketch which shows the difference between blanking and synchronizing pulses used for color and those used for monochrome.

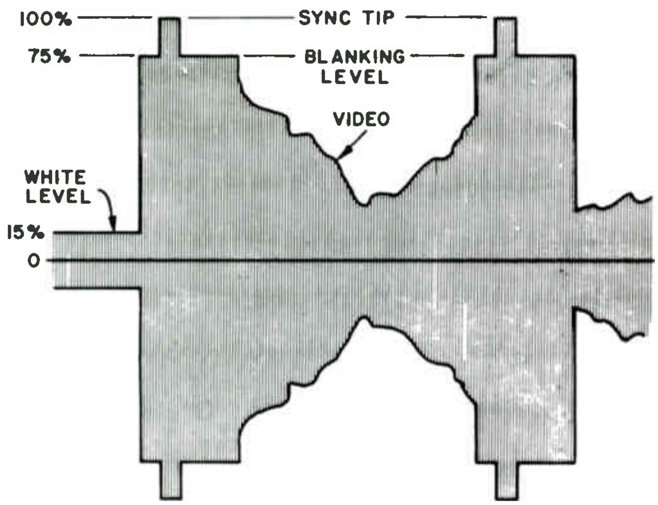


The color TV signal is essentially the same as the black-and-white signal except for the addition of the color subcarrier and the color burst. The color burst consists of an 8-cycle burst transmitted on the rear porch of the horizontal synchronizing pulses. The purpose of this color burst is to synchronize the color oscillator used in the TV receiver. The color oscillator must generate the carrier signal, which is mixed with the transmitted color sidebands to detect the original color information.

The number of horizontal lines transmitted in monochrome television is 15,750 lines per second. This comes about as the result of there being 525 lines in each picture and 30 complete pictures consisting of two fields per picture are transmitted each second. In color television a slightly different line rate of 15,734.264 lines per second is used, and there are 59.94 fields per second, instead of 60. The color subcarrier is usually referred to as a 3.58 MHz color subcarrier, but the exact frequency is 3.579545 MHz.

The slight differences between the line and field rates in monochrome and color television transmissions are so small that both signals are compatible with either monochrome or color television receivers.

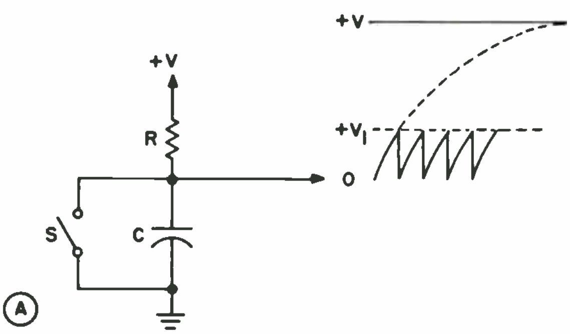
328. Sketch the amplitude characteristics of an idealized picture transmission of a television station in the United States.



Notice that in the modulation envelope the sync tips represent the highest amplitude. This is 100% modulation. The blanking level and the black level have an amplitude of 75% and the white level, which represents the brightest part of the picture, has an amplitude of 15% of the maximum level. The various shades of the video signal range in amplitude from the 15% white level to the 75% black level.

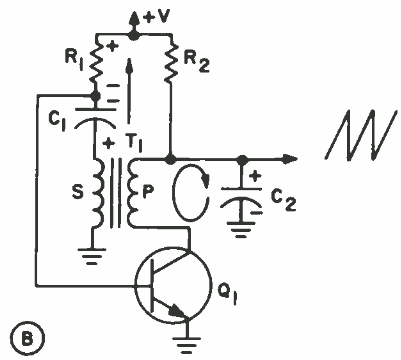
329. Show by simple circuit diagrams at least two ways of obtaining a sawtooth wave. Explain how the wave shape is formed. Where, in television transmitters, are sawtooth waves employed? Why?

The circuit at A shows basic sawtooth generation. Capacitor C charges exponentionally towards +V through R. During the first part of the capacitor charge time, the voltage rises almost linearly as shown. If we close switch S when the voltage reaches $+V_1$, C will discharge to zero volts. Opening S will allow C to begin charging again. If S is operated rapidly enough, a sawtooth wave with an amplitude of $+V_1$ volts will be generated.



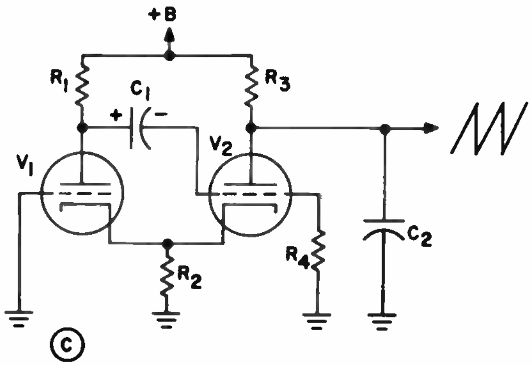
In practical sawtooth generating circuits, switch S is replaced by a vacuum tube or a transistor.

The circuit at B is an example of a transistor blocking oscillator used as a switch to discharge C_2. When the circuit is first turned on, Q_1 base current flows through R_1 to +V and causes Q_1 collector current to increase. The collector current of Q_1 flows through the primary of T_1 and induces a positive voltage in the secondary (base) winding of T_1 which drives the base more positive, increasing the collector current of Q_1 until it becomes saturated. When this happens, the collector current is no longer changing, so no voltage is induced in the primary of T_1. Capacitor C_1, however, has been charged up with the polarity shown during the regenerative turn-on time of Q_1, and now begins to discharge through R_1 toward +V as shown. This discharge current of C_1 makes the base end of R_1 negative, which reverse-biases the base-emitter junction of Q_1, turning Q_1 off.



With Q_1 off, C_2 begins to charge toward +V through R_2. C_2 is allowed to charge for a time determined by the R_1C_1 time-constant in the base circuit. When C_1 has discharged sufficiently, the reverse bias is removed from Q_1 and it once again conducts and discharges C_2 rapidly as shown. Q_1 will quickly saturate as first discussed. This cycle will repeat as long at the +V is applied to the circuit, generating the sawtooth wave shown across C_2.

The circuit at C shows how a cathode-coupled multivibrator can be used to generate a sawtooth voltage wave. When the equipment is first turned on, C_1 will charge to the value of +B through R_1 and R_4. When the tubes warm up, V_1 will start to conduct and its plate voltage will decrease. This will cause C_1 to discharge through R_4, making the grid of V_2 negative. At the same time, cathode current of both tubes through R_2 makes the cathodes of both tubes positive. The combination of a positive voltage on the cathode and a negative voltage on the grid of V_2 will cut off V_2 and allow C_2 to charge toward +B through R_3. The sizes of R_3 and C_2 are chosen so that C_2 charges only a small amount, like C in circuit A, so the voltage rise is quite linear.



When C_1 has discharged, the negative voltage across R_4 will disappear and V_2 will begin to conduct and discharge C_2. This will cause the voltage drop across R_2 to increase, which will in turn decrease the conduction of V_1. When this happens, the voltage drop across R_1 will decrease so C_1 will begin to charge through R_4, making the grid of V_2 positive, which further increases its conduction, cutting off V_1.

When C_1 is fully charged, we are at the same state as at the beginning of the cycle. C_1 will begin to discharge, and the cycle repeats.

In television transmitters, sawtooth generators are used to generate horizontal and vertical sweep signals. They will be found in cameras. They are also used in monitors used to check the video signal.

Sawtooth waves are employed because they provide a linear sweep. In other words, when the picture is scanned from left to right, the slow constant buildup of the sawtooth will move the electron beam from the left side of the camera tube across the face of the tube to the right side at a constant rate. Then it will rapidly move back from the right side to the left side to start scanning the next line.

330. What is an STL system?

An STL system is a studio-to-transmitter-link by means of radio transmission. It is used for remote pickups. For example, an STL system might be used by a television station to pick up a football game. By means of a highly directional antenna, the pickup is fed from the game to the station by means of microwaves. At the station it is picked up, detected and then used to modulate the vhf or uhf TV transmitter.

331. For what purpose are reflectometers or directional couplers used in TV transmission systems?

Reflectometers are usually used in both the aural and visual transmitter output circuits. They are generally installed at the input to the transmission line.

The reflectometer consists of two pickup coils that are used to feed energy to a meter. One coil is adjusted to intercept the incident wave; that is, the wave going from the transmitter out the line to the antenna. The other coil is oriented to intercept the reflected wave. The two waves are used to give an indication of the standing-wave ratio.

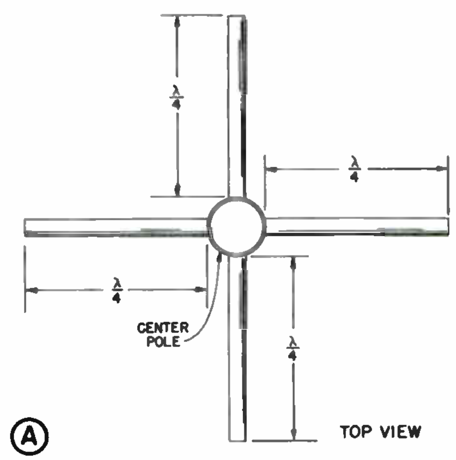
The reflectometer is also used to measure the power going to the antenna. To do this only the incident coil is used.

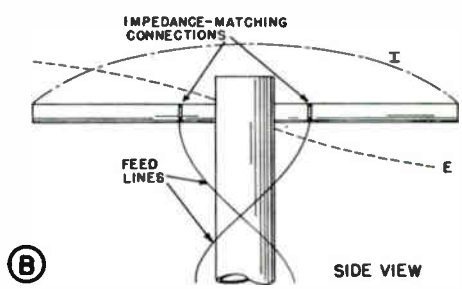
Frequently the reflected wave coil is connected into a protective circuit. This circuit is designed to protect against high surges of current.

Thus the reflectometer serves three purposes. It is used to measure the standing wave ratio, it is used to measure the relative power output of the transmitter and it is used to protect the transmitter from high current surges coming down the transmission line.

332. Explain the operation of a turnstile TV antenna.

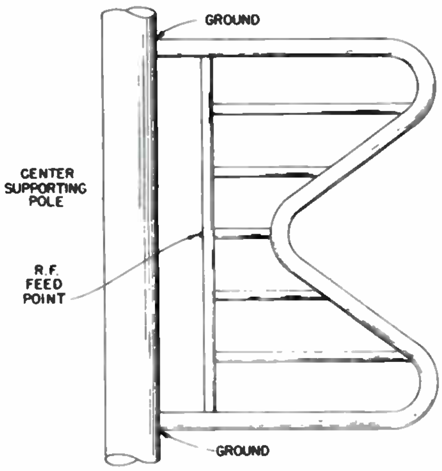
The basic turnstile antenna consists of four horizontal quarter-wave elements arranged to form two half-wave dipole antennas at right angles to each other, as shown at A. The current and voltage waves for each dipole are shown at B. The center of the dipole is at zero voltage with respect to ground, so the radiating rods are fastened directly to the grounding (center) pole at this point.





The rf power is shunt-fed to each dipole by open-wire or coaxial lines, connected as shown at B. The feed lines are connected to the elements at the proper distance from the center pole to provide an impedance match from the line to the antenna.

Most TV antenna installations use an antenna called the super turnstile rather than the turnstile. The super turnstile has a broader frequency response. In this antenna, each of the four radiating elements in A is replaced by a single current sheet radiator as shown.



Although the radiator is of tubular construction to reduce wind resistance, it acts electrically as a solid sheet of metal, thereby providing a broad-frequency response characteristic.

Four of these elements, arranged and operated as a turnstile, form one bay. The rf feed point is located at the middle of each element, and each end of the element is grounded to the center pole as shown. Higher gain is achieved by stacking, with the number of bays separated vertically one-half wavelength.

Both the turnstile and the super turnstile antenna radiate a horizontally polarized signal for both aural and visual signals.

333. What type of polarization is generally used in the transmission of the aural portion of television signals?

Horizontal polarization is generally used in the transmission of both the aural and visual portion of television signals. Horizontal polarization means that the electric field is horizontally polarized.

334. Why is a diplexer a necessary stage of most TV transmitters?

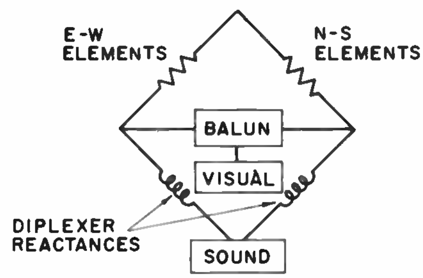
A diplexer combines the output of both the picture and sound transmitters to excite a single antenna system. The diplexer unit reflects the proper load to both transmitter outputs. At the same time, it is arranged so that a minimum of crosstalk occurs between the two transmitters. In other words, the sound signal will have no adverse influence on the operation of the picture transmitter and vice versa.

335. Draw a circuit diagram of a typical bridge-type diplexer used in a television transmitter for the purpose of transmitting both video and audio from a turnstile antenna.

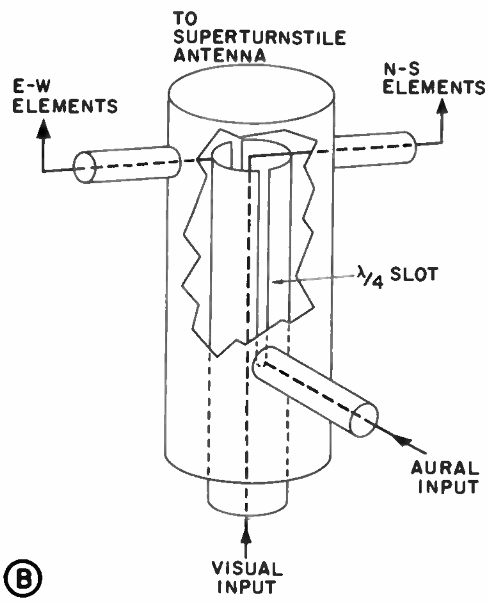
The diplexer is made up of transmission line sections and a balun that permits proper feeding of the two pairs of radiating elements that form the usual turnstile type of transmitting antenna.

The electrical equivalent of a diplexer unit is shown at A, and B shows a common physical arrangement of the components that make up a diplexer unit. Electrically, the diplexer antenna system breaks down into a bridge circuit such as shown at A. The resistance presented by the two antenna sections, referred to as north and south elements and east and west elements, form two equal legs of the bridge. The diplexer itself displays two equal reactive legs to complete the bridge.

The picture transmitter, through its balun, feeds the two antenna sections in push-pull. Since the bridge is



(A)



(B)

balanced, the picture energy does not feed into the sound portion. The sound transmitter supplies energy to the two antenna sections in a push-push arrangement. The balanced bridge action again prevents the sound transmitter from reflecting any disturbance of the picture transmitter.

A typical balun arrangement for supplying signal from the picture transmitter is shown at B. The visual transmitter supplies energy to the inner conductor of the coaxial line. The quarter wavelength top section of the outer conductor is slotted.

Since the open top of this quarter-wave section is separated a quarter wavelength from the shorted bottom section, where the slot begins, it displays a high impedance. In effect, the outer conductor is no longer grounded and will feed signals to the inner conductor of one of the coaxial elements, the one supplying energy to the east-west radiators. The inner coax conductor supplies signals to the north-south elements. The coaxial shield surrounding the entire diplexer assembly serves as the ground for the single-ended to push-pull balun.

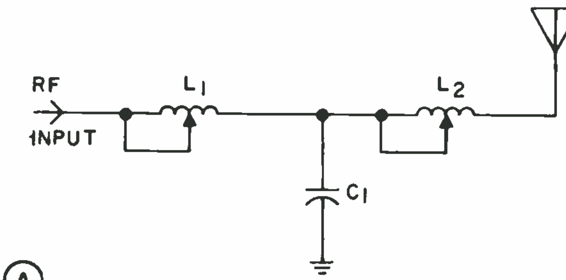
The visual signal does not interfere with the sound signal, because looking back into the sound input it sees a high impedance, and looking toward the antenna element, it sees its characteristic impedance. Consequently, all of the energy is fed to the antenna elements and not into the sound input line.

The inner conductor of the sound input is connected to the bottom of the slot. In effect, it now supplies components of equal energy and of the same polarity along the two sides of the slot to both antenna elements.

The outer shield again acts as the ground, the outer conductor of the sound input line being attached to this shield. The sound energy does not feed into the picture line because it sees a high reactance looking in that direction.

336. Describe how to tune a broadcast antenna by (1) the rf bridge method and (2) the substitution method.

(a) The diagram shows a typical T-match network such as might be used to match an rf transmission line to the antenna. To match the transmission line to the antenna using the bridge, the first step is to use the bridge to measure the antenna impedance. Let's assume that you do this and that the antenna measures a resistance of 35 ohms and a capacitive reactance of 20 ohms. You should know the impedance of the transmission line, since the manufacturer's specifications will give this information. Let's assume that the transmission line is 50-ohm coaxial cable.



Ⓐ

The reactance to which the coils L_1 and L_2 must be set can be found from the formula:

$$X_L = \sqrt{R_1 \times R_2}$$

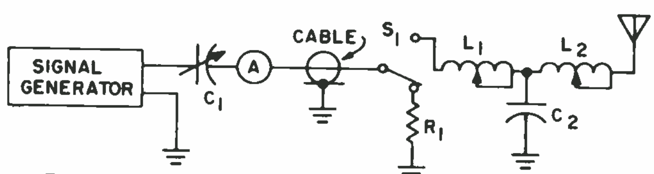
where R_1 is the resistance of the transmission line and R_2 is the antenna resistance. Substituting these values in the formula, we get:

$$\begin{aligned} X_L &= \sqrt{50 \times 35} \\ &= \sqrt{1750} \\ &= 41.8 \text{ ohms.} \end{aligned}$$

To cancel out the capacitive reactance of 20 ohms of the antenna, we need to add an equal amount of inductive reactance to L_2. Therefore L_2 should be set using the bridge to 41.8 + 20 = 61.8 ohms. L_1 is set to 41.8 ohms.

After you have the coils adjusted, you can use the rf bridge to check the impedance match with the transmission line disconnected but with the antenna connected at the output of the T-network. You should read 50 ohms at the input to L_1. If you get this value, then you know that the transmission line will be correctly terminated in its characteristic impedance. If the input to L_1 reads slightly off from 50 ohms, you can make whatever minor adjustments are necessary at this time to correct the discrepancy.

(b) In the substitution method, a switching arrangement is used so that a noninductive resistance equal to the resistance of the transmission line can be connected to terminate the line. In the preceding example, with a 50-ohm line, R_1 should have a resistance of 50 ohms. The T-matching network is connected to the antenna as shown.



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The first step in the adjustment procedure is to carefully set the signal generator to the exact frequency of the transmitter. Next, the noninductive resistor is connected to terminate the transmission line, the signal generator is turned on and C_1 is adjusted for a maximum reading on the rf ammeter. The output from the signal generator is adjusted to give a reasonably high indication on the meter.

Next, switch to the antenna tuner and readjust C_1 for a maximum reading on the rf ammeter. If you find it necessary to reduce the capacity to get a maximum reading, the load is inductive. If you have to increase the capacity, the load is capacitive. If no change is required for a maximum reading, the load is resistive.

Even with a pure resistive load, the reading may not be the same as with the resistor connected into the circuit. This is unimportant; the important thing is to adjust for a maximum reading to determine if the load is resistive, inductive or capacitive.

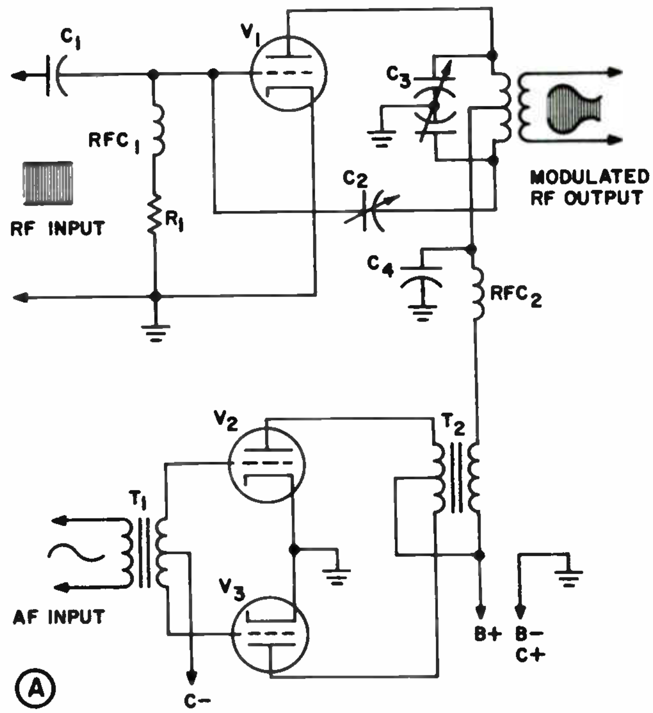
The next procedure is a try-and-guess method of adjusting the tuner. If the antenna is capacitive, adding additional inductance in L_2 should correct this situation, whereas if it is inductive, reducing the inductance of L_2 should help. By manipulating L_1 and L_2 and switching back and forth between the antenna and the resistor, you should be able to eventually reach a setting where you get the same reading in either position of the switch without having to readjust C_1 for a maximum reading when switching back and forth. When you have reached this setting, the input impedance to the T-network is 50 ohms and the transmission line will be correctly matched to the antenna by the matching network.

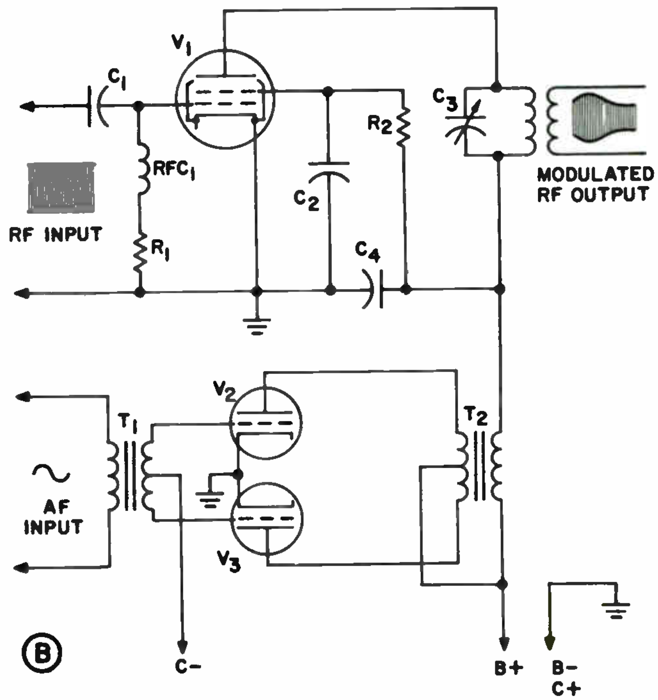
337. Draw circuit diagrams of:

(a) A triode Class C amplifier properly coupled to a push-pull power amplifier (modulator).
(b) A beam Class C amplifier coupled to a push-pull Class B power amplifier.

For both cases, show the modulating signal input, the rf exciting voltage input and the modulated output. Include neutralization for the triode case. Explain the operation of both the above types of Class C plate-modulated amplifiers.

(a) In the Class C triode amplifier, a constant amplitude rf signal is fed to the input of the stage. At the same time, the B+ to the plate of the tube is supplied through the secondary of the modulation transformer, T_2. The audio input, which is fed through T_1 to the Class B modulators, produces an amplified audio signal in the output which adds to and subtracts from the B supply voltage fed to the Class C amplifier. When the signal across the secondary of T_2 has a polarity such that it adds to the B supply





voltage, the voltage fed to the Class C amplifier is doubled. When it opposes the B supply voltage, it will be equal and opposite to it so that the voltage applied to the Class C amplifier will drop to zero. Thus we have a modulated signal appearing at the output as shown, and the amplitude will vary in accordance with the audio modulating signal.

(b) In the beam tetrode rf amplifier, we have essentially the same situation except that in this case the modulated signal is applied to the screen as well as to the plate. The modulating signal is applied to the screen through R_2. This will cause the screen voltage to vary in amplitude as well as the plate voltage. This is necessary in order to obtain 100% modulation.

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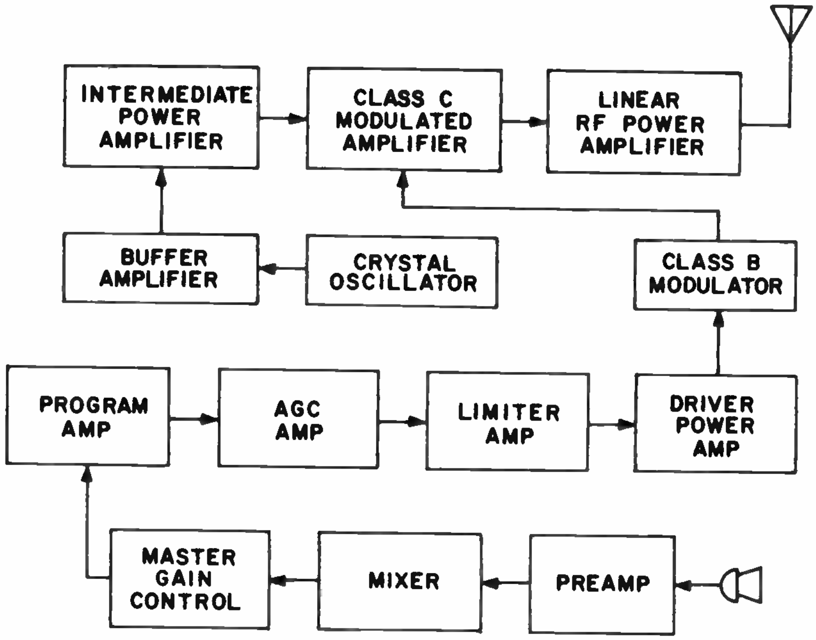

338. Draw block diagrams of the following transmitters complete from the microphone (and/or camera) inputs to the antenna outputs. State the purpose of each stage and explain briefly the overall operation of the transmitters.

(a) Standard (AM) broadcast.
(b) FM broadcast.
(c) Multiplex FM broadcast.
(d) TV broadcast.
(e) Color TV broadcast.

(a) The crystal oscillator generates a stable rf signal which must be within 20 Hz of the transmitter's assigned frequency.

The signal from the crystal oscillator is fed to a buffer amplifier. The primary purpose of the buffer amplifier is to isolate the crystal oscillator from the power amplifiers to prevent any loading of the crystal oscillator that might affect the oscillator's frequency.

The intermediate power amplifier amplifies the signal from the buffer amplifier and builds it up to sufficient amplitude to drive the Class C modulated rf amplifier.

The Class C rf amplifier builds up the amplitude of the signal still further. This stage is modulated by the audio signal from the Class B modulator.

The linear rf power amplifier takes the modulated rf signal and amplifies it still further to provide the rf power that is fed to the antenna. In lower power transmitters, the signal may be fed directly from the Class C modulated rf amplifier to the antenna, but in the higher power transmitters such as the 50 kW clear-channel stations, linear amplifiers are used after the Class C modulated rf amplifiers to avoid the need for large amounts of audio power.

The microphone picks up the audio signal and from there it is fed to a preamplifier. The preamplifier builds up the weak audio signal before it is fed to the mixer. From the mixer the signal is fed to the master gain control and then to the program amplifier. The program amplifier builds up the signal before it is fed to the agc amplifier. The agc amplifier is used to provide essentially constant amplitude signals. If the signal is weak, the agc amplifier will give it additional gain, but if it is a strong signal, the gain will be reduced automatically.

The signal from the agc amplifier is fed to a limiter amplifier. The limiter amplifier is used to prevent overmodulation.

From the limiter amplifier the signal is fed to a driver. The driver is a power amplifier which provides the power required to drive the Class B modulator.

The Class B modulator develops the audio power required to modulate the Class C rf amplifier. The power output from the Class B modulator must be equal to 50% of the power input to the Class C modulated amplifier stage in order to obtain 100% modulation.

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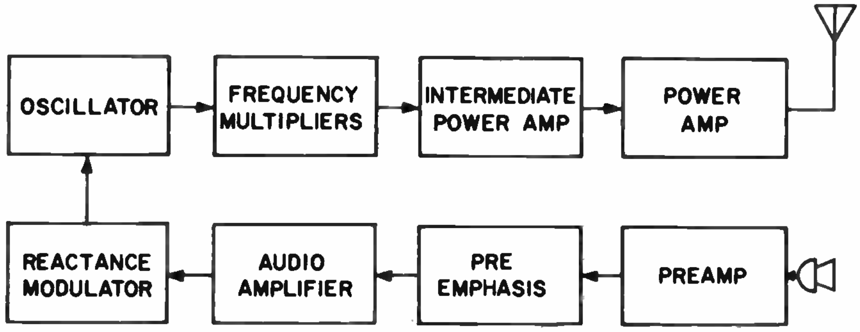

(b) The audio signal picked up by the microphone is fed to a preamplifier where it is amplified to provide a good signal-to-noise ratio.

From the preamplifier, the audio signal is fed to a pre-emphasis network which gives increased amplification to the higher audio signal frequencies. This is necessary because the high audio frequencies contain very little power, and by giving them additional amplification we will get a better signal-to-noise ratio at the higher audio signal frequencies.

The audio from the pre-emphasis network is fed to audio amplifiers and from there to a reactance modulator.

The main FM oscillator is designed to have very good frequency stability. The output resting frequency of the FM transmitter must be held within 2000 Hz of the transmitter's assigned frequency. Since the output signal from the oscillator is going to be multiplied many times by frequency multipliers, the actual frequency of the FM oscillator must be very stable. The reactance modulator is used to vary the frequency of the FM oscillator in accordance with the modulating signal.

The modulated output from the FM oscillator is fed to a series of frequency multipliers that are used to increase the frequency of the oscillator up to the operating frequency of the transmitter.

The output from the last frequency multiplier is fed to an intermediate power amplifier. The intermediate power amplifier provides the power required to drive the final amplifier. The final power amplifier builds up the power to the rated power output of the transmitter, and the signal from it is fed to the FM antenna.

(c) In the multiplex FM transmitter, the rf section in the main audio program channel is the same as in the FM broadcast transmitter. However, we have the multiplex channel in addition to the main audio channel.

In the multiplex channel, the signal is fed through a preamplifier, pre-emphasis network and audio amplifiers as in the main audio channel. The signal is then fed to a reactance modulator, which modulates a 67 kHz oscillator which produces the subcarrier. The 67 kHz oscillator is FM-modulated, and this in turn is fed to a subcarrier reactance modulator which modulates the main FM oscillator.

In the output from the FM transmitter, in addition to the modulation produced by the main audio channel, you also have a 67 kHz subcarrier which in turn is FM-modulated by the subcarrier modulation.

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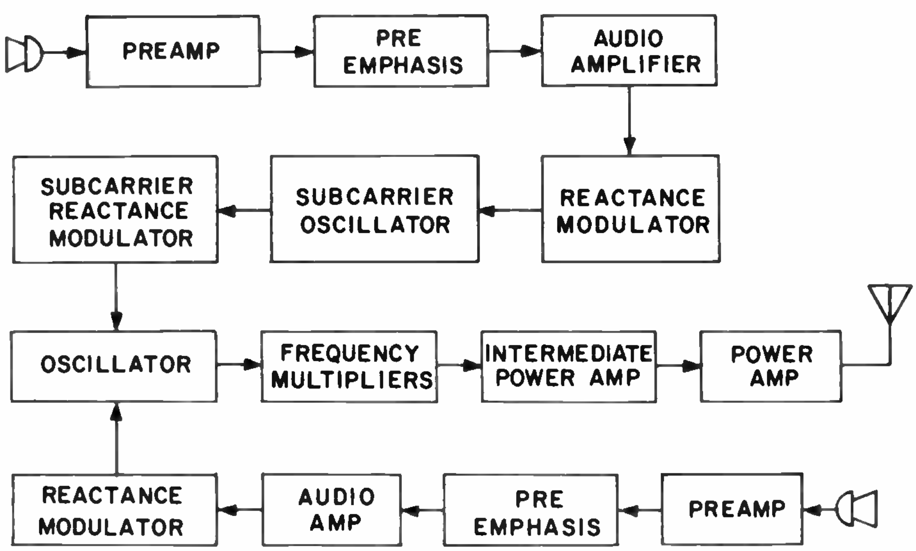
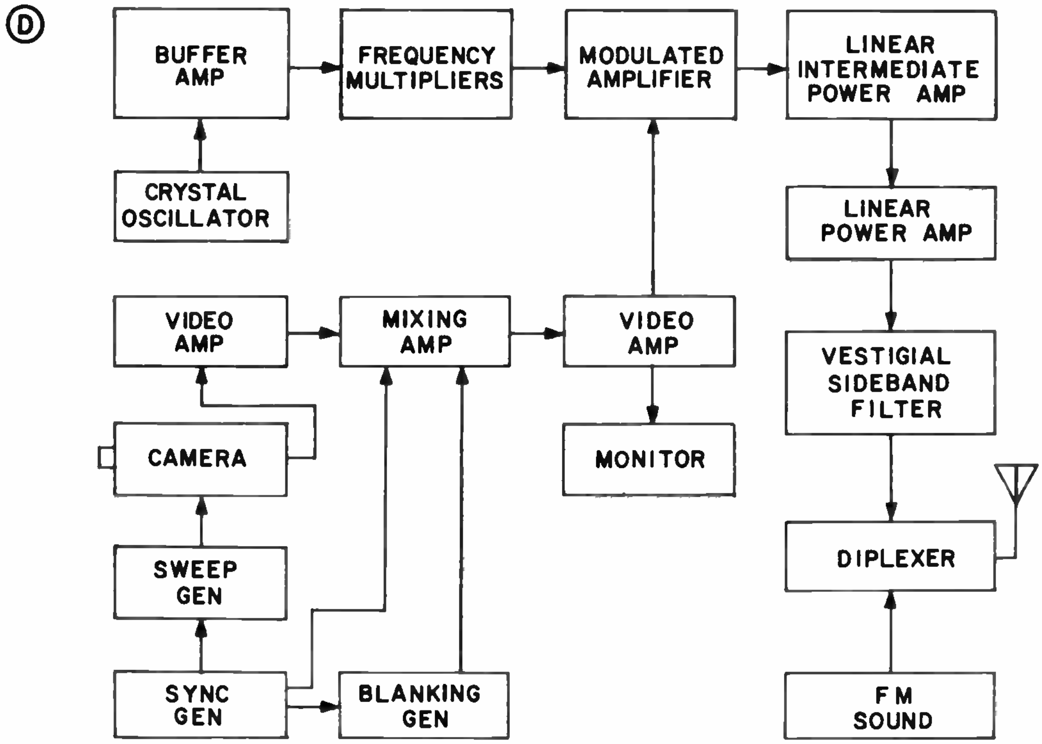




(d) In the rf section of the transmitter, a crystal oscillator is used to generate a stable crystal frequency. The oscillator operates at a frequency substantially below the output of the transmitter and must have sufficient stability so that the final output frequency will be held within 1000 Hz of the assigned transmitter frequency.

The output from the crystal oscillator is fed through a buffer amplifier which isolates the crystal from the following stages.

The output from the buffer amplifier is fed to a number of frequency multiplier stages that are used to increase the frequency to the required output frequency of the transmitter. Frequency multipliers may be doublers or triplers.

The output from the last frequency multiplier is fed to the modulated amplifier which combines the composite video signal with the rf signal.

The modulated stage of the transmitter is usually grid-modulated because of the special requirements of a TV transmitter and the wide range of video signals that must be handled.

The output from the modulated stage is fed to linear intermediate power amplifiers to build up the amplitude of the rf signal. The signal is fed from the intermediate power amplifiers to a final linear rf power amplifier to provide the required power output from the transmitter.

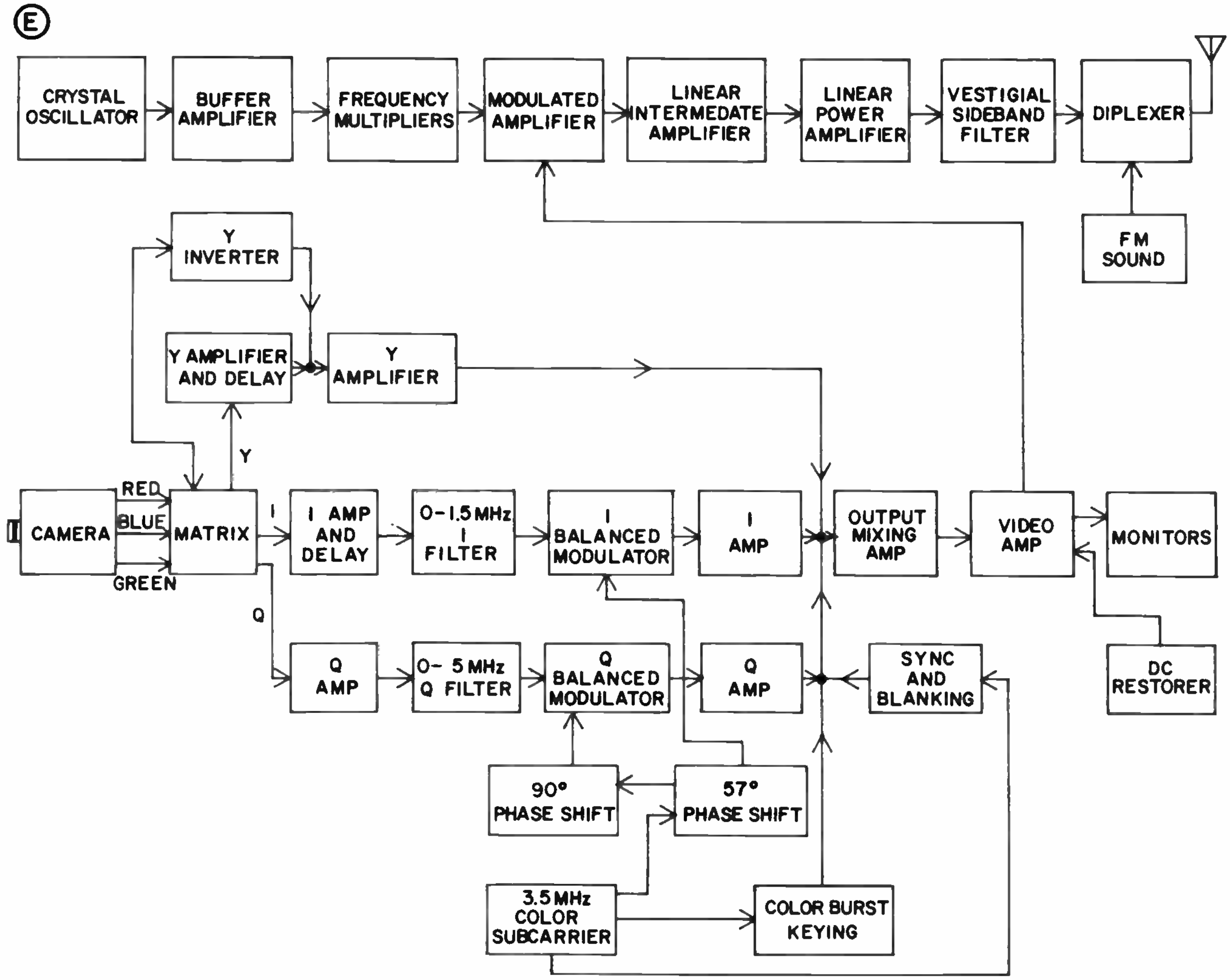
The signal from the power amplifier is fed to a vestigial sideband filter that is designed to shape the output signal as specified by FCC regulations. Modulating signals below 1.25 MHz from the carrier frequency are highly attenuated.

The signal from the filter is fed to a diplexer where the signal from the FM sound transmitter is combined from the diplexer to the antenna. The FM sound transmitter is essentially the same as the sound transmitter shown in (b) of this section.

In the camera, sweep signals are fed to the deflection yoke around the camera tube to sweep the face of the tube. The video signal from the camera is amplified and fed to a mixing amplifier. Sync and blanking pulses are added and the signal used to modulate the modulated Class C amplifier.

A signal from the modulator is also fed to a video monitor so that the quality of the signal picked up by the camera and fed to the modulated stage may be continually monitored.

(e) In the color TV broadcast transmitter, the rf section is the same as the black-and-white transmitter. We have a crystal oscillator which must maintain the



stability so that the output frequency is within 1 kHz of the assigned transmitter frequency. The signal from the crystal oscillator is fed to a buffer and amplifier and from there to frequency multipliers.

The signal from the last frequency multiplier is fed to the Class C modulated amplifier which is grid-modulated. The modulated TV signal is then fed to the intermediate power amplifiers, from there to the vestigial sideband filter. From the sideband filter, the signal is fed to the diplexer circuit and from there to the antenna.

The primary difference in the two TV transmitters is in the video chain. In the color transmitter the signal is picked up by a color camera which generates red, blue and green signals. These signals are fed to the matrix circuits where .59 of the green signal, .30 of the red signal and .11 of the blue signal are matrixed to produce a Y signal. The Y signal is fed to a Y amplifier and delay circuit. The delay circuit is needed because the Y amplifier has a wider bandpass than the color circuits. The Y signal is then fed to a Y amplifier and from there to an output mixing amplifier. You will remember that the Y signal represents the black-and-white signal and it is used to modulate the modulated amplifier in order to produce a black-and-white signal so that a color transmission can be picked up as black-and-white on a monochrome television receiver.

An output signal from the Y amplifier and delay circuit is inverted through a Y inverter and fed as a –Y signal back into the matrix circuits. The –Y signal is used with the signal from the color cameras to produce the I and the Q signals.

In the I channel, the signal is first fed to an amplifier and delay circuit. Again a delay circuit is required in the I amplifier because this amplifier has a wider bandpass than the Q amplifier. The output from the first I amplifier and delay circuit is fed to the filter which cuts off the signals above 1.5 MHz. The signal from the I amplifier is then fed to an I balanced modulator. The output from the I balanced modulator is then fed to an I amplifier.

In the Q channel, the signal is fed from the matrix circuits to a Q amplifier. Since this circuit has the narrowest bandpass, no delay is required. The output from the Q amplifier is fed to a filter which attenuates the signals above .5 MHz. The output from the Q filter is fed to a Q balanced modulator.

The 3.58 MHz color subcarrier oscillator signal is fed to the I and Q balanced modulators. The signal fed to the I modulator is displaced 57° from the subcarrier frequency and the signal fed to the Q balanced modulator is displaced 147° from the subcarrier oscillator frequency. Thus the signals fed to the I and Q balanced modulators are 90° out-of-phase.

The output from the Q balanced modulator is fed to the Q amplifier and from there to the output mixing amplifier.

In addition to the I and Q signals, a color burst signal is fed to the output mixing amplifier. This signal is keyed so that eight cycles of it will appear on the back porch of each horizontal sync pulse. The conventional sync and blanking pulse generators are fed also to the output mixing amplifier.

The output from the mixing amplifier is fed to video amplifiers to build up the amplitude of the signal to a high enough level to modulate the Class C modulated amplifier. A dc restorer is used to restore the dc component of the signal which may be lost.

A signal is also taken from the video amplifier to feed to the monitors. Both color and black-and-white monitors are usually employed in order that the station operator may continually monitor both the color and the monochrome signals transmitted by the transmitter.

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