

Introduction

Communications systems use transmitters to generate radio frequency carrier waves that carry intelligence through space using electromagnetic waves. These carrier waves are of little use without a means of intercepting them, and extracting the intelligence. The device used for this is a radio receiver. To better understand the operation of the receiver, an introduction to the transmitting devices and how they generate signals is important.

Basic block diagram of an AM (Amplitude Modulated) Transmitter

The block diagram of a transmitter used to produce amplitude modulated signals is shown in Figure 1.

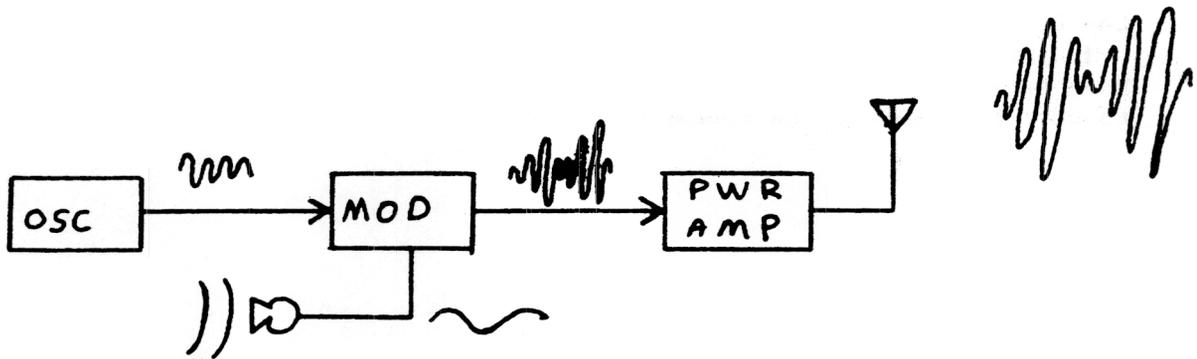


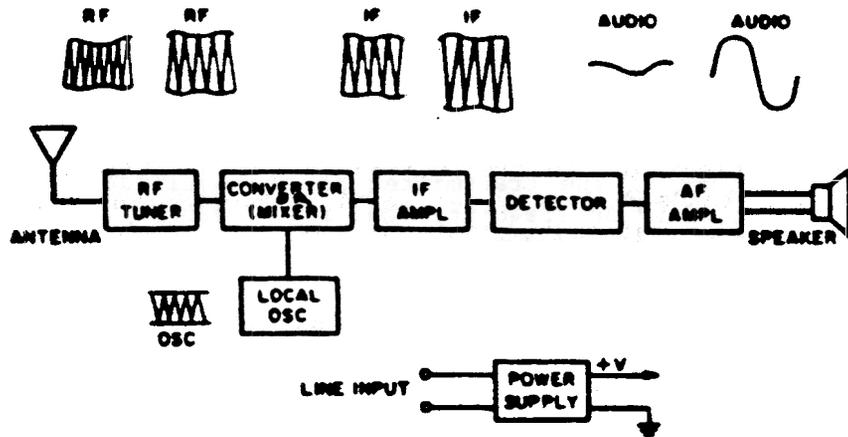
Fig 1.

The first block in the transmitter is the oscillator. The frequency range for the standard AM frequency band is 540 KHz to 1610 KHz. The frequency each radio station transmits is assigned by the Federal Communications Commission (FCC).

These frequencies are recognized by all the frequencies shown on the front of the radio dial. The oscillator is tuned to an assigned frequency and provides a continuous wave (CW) signal to the modulator. A second signal is applied to the modulator from the microphone. The microphone

can also be referred to as a transducer. A transducer is a device that changes one form of energy into another form. The microphone changes the audio, sound energy, into electrical energy. The modulator takes the two signals, CW from the oscillator and the audio from the microphone, and combines the two signals. This produces an amplitude modulated signal to be applied to the power amplifier. The power amplifier is required to amplify the signal to a level, that when applied to the antenna, will reach radio stations at considerable distances. The signal, amplitude modulated (AM) radio frequency (RF) is then transmitted into space by the antenna. This is the signal that will be intercepted by the receiver and allow the intelligence from the transmitter to be used.

Figure 2 is a block diagram of the circuits used in the radio receiver to be discussed. This type radio is called a superheterodyne receiver. The name superheterodyne is from the principle of operation of the receiver. This principle is the converting of a signal from one frequency to another frequency; it is done by heterodyning (mixing the incoming signal with a second signal) in a mixer or converter. The receiver selects the desired frequency coming in the antenna, converts it by mixing it with a higher frequency to produce a third frequency called the Intermediate Frequency (IF), amplifies this new frequency and extracts the intelligence to be reproduced at the speaker as it was sent from the transmitter.



Block diagram of superheterodyne receiver.

Fig. 2

The discussion of the Superheterodyne Receiver will start at the antenna. (Refer to figure 2)

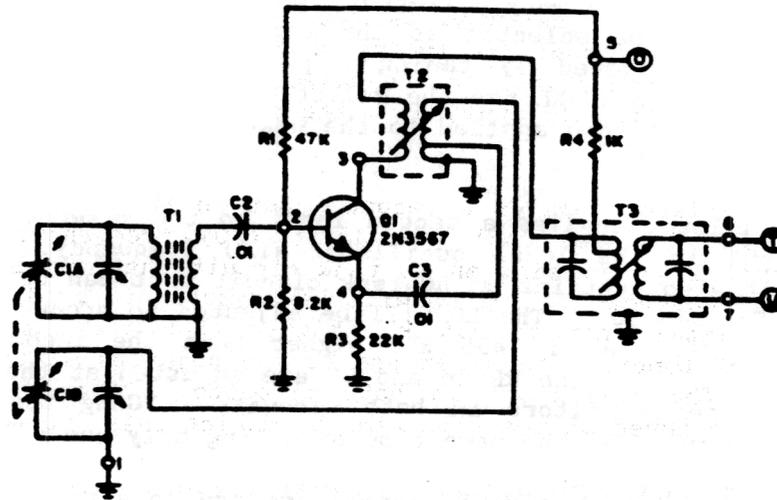
The electromagnetic waves that are transmitted by the radio station will cut through the antenna and induce small amounts of current in the antenna. Although there are many different signal currents flowing in the antenna, only one will be selected as the input to the mixer by the RF tuner. This is accomplished by tuning a parallel LC tank circuit to resonance. This is where $X_C = X_L$ for the selected frequency. The frequency selected by the tuner is then applied to the next block, the mixer or converter.

The local oscillator applies a second input to the mixer or converter block. This input is the local oscillator (LO) frequency. The local oscillator circuit also contains an LC tank circuit that can be tuned over a wide range of frequencies. The LO will be adjusted to produce a continuous wave (CW) signal that is 455 khz higher than the incoming signal selected at the RF tuner. The RF in and LO are adjusted at the same time by "gang tuning" the capacitors in both circuits. "Gang tuning" means both capacitors are tuned at the same time by making only one adjustment.

Figure 3 shows that both circuit capacitors are located on one shaft. As the shaft is turned both capacitors are changed at the same time. This will maintain the same frequency difference of 455 khz, as the LO and RF tuner are tuned across the entire frequency band of the radio. (540KHz to 1610 KHz)

You can identify the set of plates for the local oscillator and the set for the RF section by comparing the size of each side. The larger size or greater number of plates is the RF input side, and the smaller size or lesser number of plates is the local oscillator side.

The RF input cap, C1a and the LO cap, C1b work together. The dotted lines represent the mechanical connection for gang tuning.



LOCAL OSC. AND CONVERTER PC 38

Fig 3

The mixer or converter will use the two input signals, one from the RF tuner and the other from the LO (refer to figure 2), mix these signals and produce four frequencies at its output. In figure 4 the four frequencies are shown when the RF tuner has selected a radio station at 550 KHz. The LO is tuned 455 KHz above this frequency, therefore its frequency is 1005 KHz (550 KHz + 455 KHz). The output of the mixer is these two original frequencies and two additional frequencies. They are the sum and difference of the two original frequencies (F1&F2).

$$\text{Sum} = 1005\text{KHz} + 550\text{KHz} = 1555\text{KHz}$$

$$\text{Difference} = 1005\text{KHz} - 550\text{KHz} = 455\text{KHz}$$

The difference frequency is the output desired from the mixer. It will be referred to as the Intermediate Frequency (IF). The IF is a frequency between the incoming RF and the audio output. Using this IF will allow for uniform selectivity across the entire receiver band. Selectivity is the ability of the amplifier to select only one frequency and amplify it. This is accomplished by tuning the circuits to the 455KHz difference frequency.

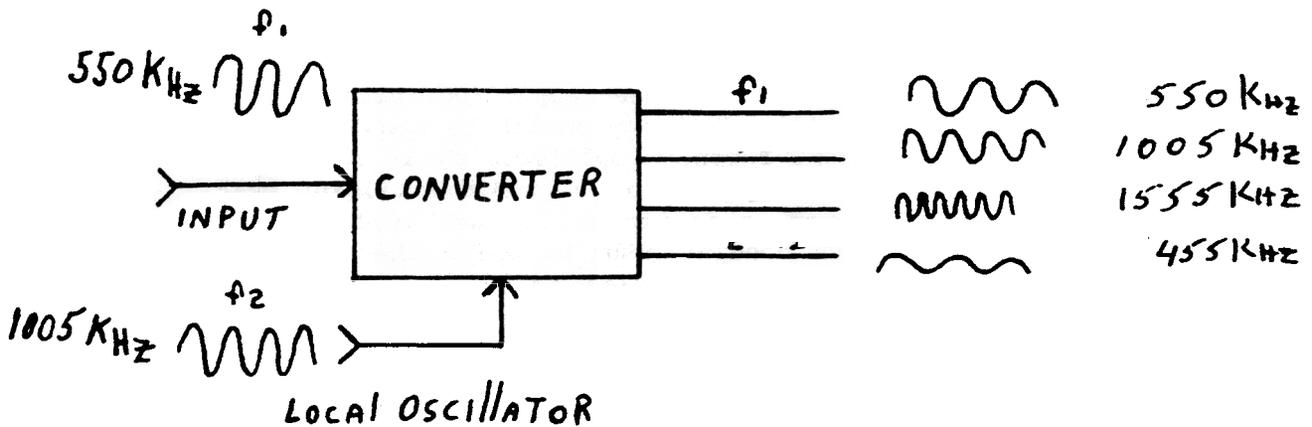


Fig 4

The IF amplifier block in figure 2 consists of one to three stages of cascaded high gain amplifiers. Each stage has a tuned circuit in its output. These will be tuned to the IF frequency (455KHz). The receiver's sensitivity and bandwidth is determined by these stages. Sensitivity and bandwidth will be covered later in the lesson. After the amplitude modulated signal has been amplified, it is then applied to the detector. This circuit first rectifies the modulated signal then filters the unwanted signal which is the 455KHz signal frequency. This will leave only the audio signal. The detector is also called the demodulator because it prepares the signal for the audio amplifier. Figure 5 displays the sequence that leads up to demodulation.

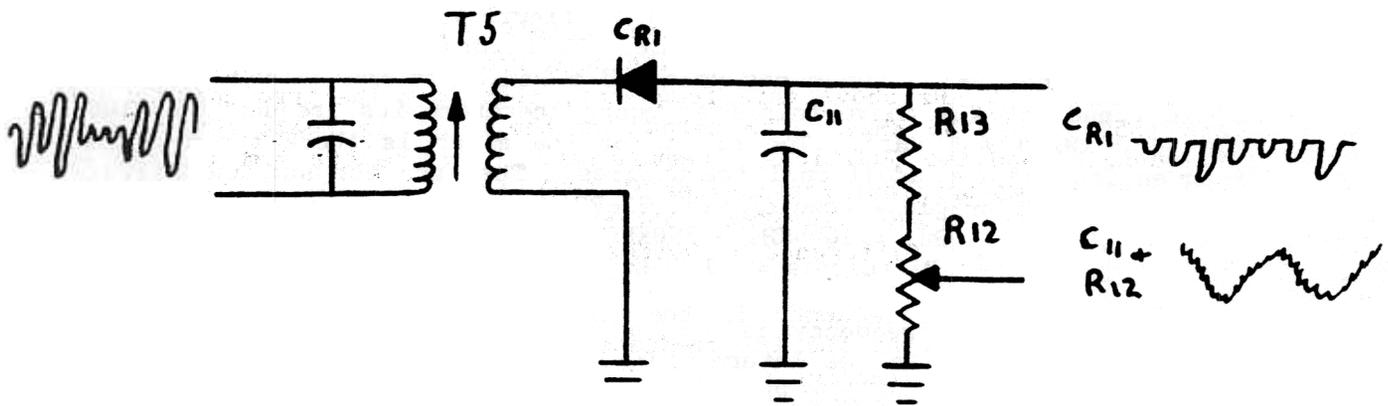


Fig 5

The filtered signal is the audio that is going to be sent to the audio amplifier.

Even though the modulated RF has been amplified, rectified and filtered, the audio signal is still not large enough to produce an output from the speaker, therefore, some further amplification is needed. The audio signal from the detector is applied to the audio amplifier section which consists of an audio pre-amp, a driver and a push pull power amp. This will produce a signal with enough power to drive the speaker cone that produces the sound. The speaker is also called a transducer because it changes electrical energy to sound energy.

From this point on, we will follow a signal through a complete super-heterodyne receiver schematic to see how the signal is accepted and changed to meet the needs of each individual circuit. The first circuit that will be discussed is the converter. The converter is one of two circuits that can be used to accomplish the heterodyning action. The other is a mixer and L.O. A brief introduction to a mixer will be given first. Refer to fig 6.

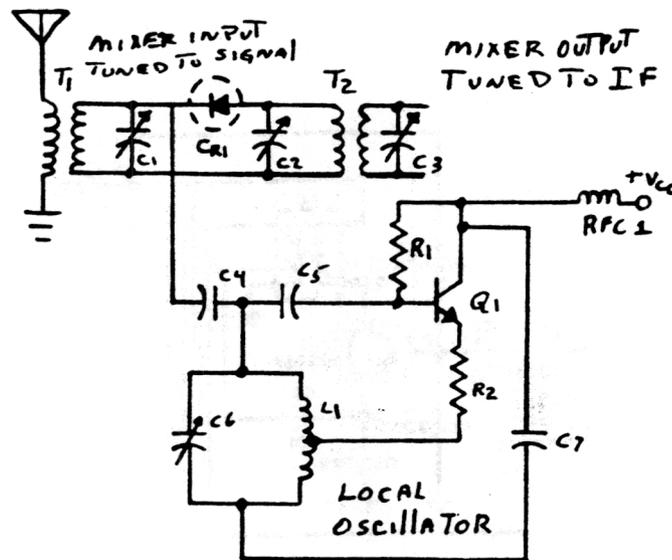


Fig 6

The mixer in this circuit is a semi-conductor diode (CR1). Both the incoming modulated RF coupled by T_1 and the local oscillator frequency coupled by C_4 to the cathode of CR_1 beat together in the non-linear rectifier producing a difference frequency which is the intermediate frequency (IF). This type of circuit is not used much because a diode mixing circuit lacks the ability to amplify. Therefore we will be using a single stage circuit called a converter in order to have the amplification needed for future circuits. As we discuss the converter circuits refer to figure 7.

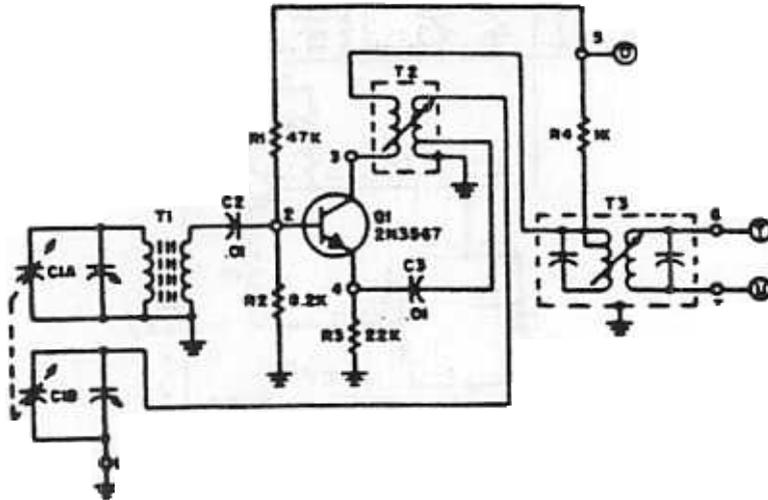


Fig 7

The first component the RF sees is the antenna. T1 in this circuit is the antenna. But, it also serves as a transformer coupling the input signal to Q1. The common name for this type of component is Loop stick antenna. An extra piece of wire can be connected to this device to have better reception.

To start this discussion, the most logical place to begin is the input circuit being a combination of the primary winding of T1 and C1a. These two components make up a tank circuit or frequency determining device. Adjusting C1a tunes this tank circuit to the desired input frequency or Radio Station. Then, T1 inductively couples this signal to the base of Q1 via RC coupling consisting of C2 and R2.

C1B and T2 secondary winding make up the second tuned circuit, the local oscillator. The oscillator is tuned to 455KHz above the incoming frequency or received signal. This is accomplished by gang tuning C1A and C1B. When C1A is adjusted C1B will follow, maintaining the same frequency difference across the AM Receiver Band. This is called receiver selectivity.

Radio Station is 1000KHz, L.O. is 1455KHz, the difference is 455KHz

Radio Station is 1230KHz, L.O. 1685KHz, the difference is 455KHz

Radio Station is 1600KHz, L.O. 2055 KHz, the difference is 455KHz

In each case the L.O. changed the same amount in respect to the radio station you tuned in to hear.

The input frequency F_1 that is coupled to the base of Q1 controls the bias from one direction. F_2 the L.O. frequency, is sent to the emitter of Q1 by C3. This frequency controls the bias at a different rate. Figure 8 shows this change.

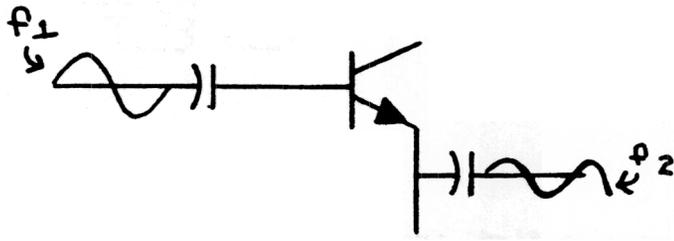


Fig 8

F_1 is lower than F_2 , thus giving us conduction changes at two different rates.

Heterodyning occurs because collector current is controlled by the combination the input bias changes, not just one. The resulting output will be four frequencies. The two originals, (received signal and the local oscillator frequency), the sum of the two and the difference of the two. This complex signal is then sent to the primary of T3 which is our first IF Transformer. These different currents that are at the collector of Q1, (T1) can now be sent to a circuit component that will select, by adjustment, the desired frequency. The frequency that will be selected is the intermediate frequency, (IF), of 455 KHz.

The IF amplifier circuit is the next step of our signal flow. Figure 9 will allow us to follow the signal through it.

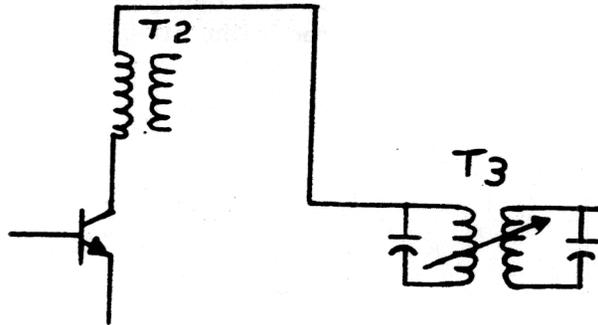
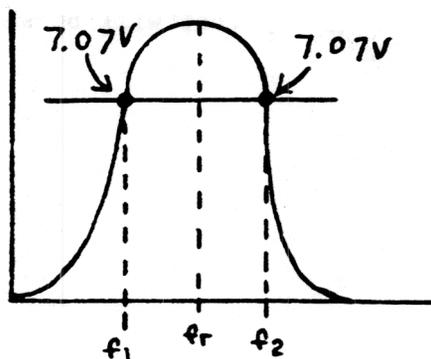


Fig 9

The IF amplifiers are fixed frequency amplifiers that accept the output of this converter from the secondary winding of T2 located at the collector of Q1. It is then transformer coupled to Q2 base by T3. This transformer T3, is tuned to the modulated 455KHz signal.

The input to the first IF amplifier is extremely weak, therefore, more than one stage of amplification is required. Input and output coupling for each IF stage is by resonant transformer action. The transformers are tuned to 455KHz, with bandwidth of 10KHz. Bandwidth can be explained as the range of frequencies that fall between a minimum and maximum point that a receiver will let pass through its' tuned circuits. An example is given in figure 10. The two points are called half power points. (.707 of Peak Voltage) the peak is obtained when the tuned circuit is resonating at its designed frequency.



Peak is 10V
 Half Power Points = $.707 \times 10 = 7.07V$
 $f_R = 1000 \text{ KHz}$
 $f_1 = 995 \text{ KHz}$
 $f_2 = 1005 \text{ KHz}$

Fig 10

F1 will be the lowest frequency that will be usable. F2 is the highest frequency that will be usable. FR is the frequency you selected to listen to. Bandwidth is then calculated as.

$$\begin{aligned}
 BW &= f_2 - f_1 \\
 &= 1005 \text{ KHz} - 995 \text{ KHz} \\
 &= 10 \text{ KHz} = BW
 \end{aligned}$$

By putting the frequency values in place of F2 and F1 we can then subtract. Bandwidth in this example is 10KHz. All frequencies between these points will be passed.

These circuits, (IF amp) provide the majority of receiver gain. Thus, it has the major influence on its sensitivity. This tells us how a small signal will be accepted and result in a good audio signal out.

Q2 and Q3 are two common emitter amps and will be treated as such. The major difference is the type signal we will send through them. It is not the audio tones you are used to using. It is R.F. (Radio Frequency) at 455KHz. See fig. 12

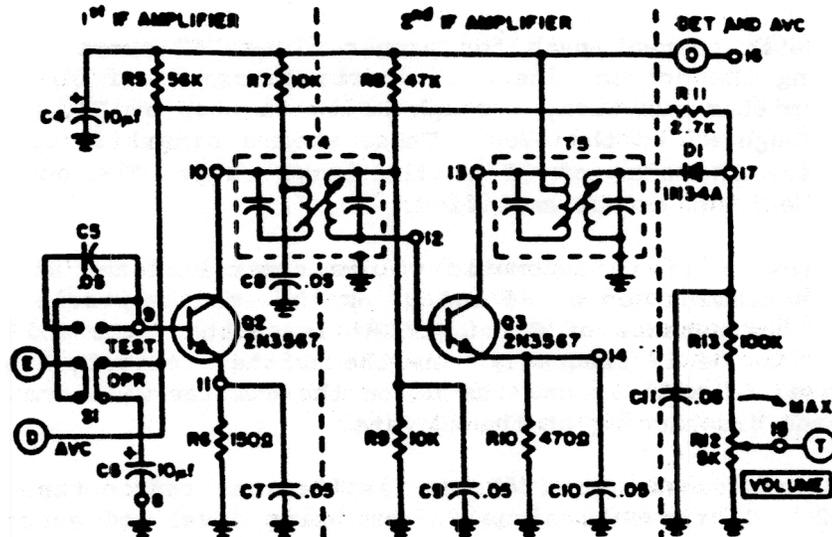


Fig 12

In figure 9, with switch S1 in the OPR, position, the IF signal from T3 is sent to the base of Q1.

Note: Keep in mind Pins E and D are connected to the secondary winding of T3. See fig 13.

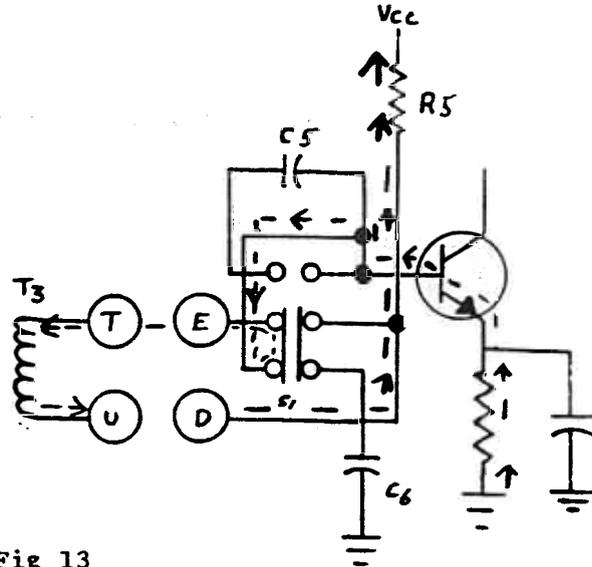


Fig 13

The static current path for proper class "A" operation can be traced by following the dotted line. It starts at ground of Q2 emitter, through R6, Base emitter junction, through S1 to the top of T3, out terminal "U" and up through R5 to the +Vcc. Thus, when a signal is coupled across T3 it will vary the bias of Q2 at the input rate. The output will be an image of the input except amplified.

The gain of Q2 is Automatic volume controlled and will be described later. The combination of R6 and C7 on the emitter provides bias stabilization. The reactance of C7 at 455KHz is about 7 ohms and provides a low reactance for the IF frequency. As the emitter tries to change in respect to the base, C7 will insure the DC on the emitter will remain constant by charging and discharging at the AC rate.

R7 in the transformer (T4) top, is there to reduce the collector voltage of Q2. C8 prevents signals from being developed across R7. The IF signal is then inductively coupled to the base of Q3. It is very important to restate that the Transformer, T4, is tuned to the IR frequency and it is a high "Q" circuit. The "Q" of a circuit can be considered to mean "quality". This term is used where inductance and capacitance are involved. When A-C circuits are involved, inductors and capacitors have resistance. Thus, "Q" is a ratio of its' reactance to it resistances.

$$Q = \frac{XL}{R} = \frac{2 FL}{R}$$

$$Q = \frac{XC}{R} = \frac{1}{2 FCR}$$

The Q3 circuit is similar in operation to Q2 except that it is not AVC controlled. Bias is provided by R8 and R9 and their function is decoupled by C9. R10 and C10 make up the emitter circuit. The output signal is developed across T5, again tuned to the IF frequency and tapped for impedance matching. The output is inductively coupled to the secondary and applied to the detector.

In most cases, we use step - down turns - ratios to reduce the voltage and increase the current. This is common in transistor circuits to provide the driving currents for the next step. Once the modulated IF is amplified to a sufficient level, the signal is sent to a detector circuit. This circuit will pass the audio frequencies and filter out the RF frequency, (455KHz). We no longer need this RF. The audio frequency range for receiver circuits is from 50Hz to 20 KHz. Figure 14 can be followed while the detector section is discussed.

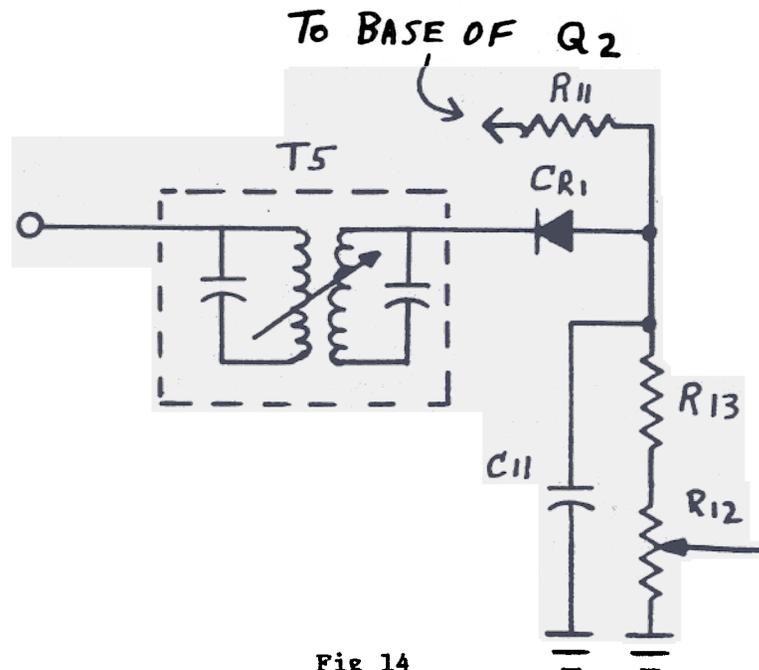


Fig 14

The output of the IF amplifier is coupled to CR1, diode detector by way of Transformer T5. CR1 rectifies the positions alternations of the secondary signal. C11 and R12 makeup the RF filter. The time constant of C11 and R12 will be such that they will filter out the carrier frequency, but will follow the audio frequency variations. R13 is an isolation resistor, and has little effect on the detector operation.

R12, R13, R11 and R5 form a voltage divider between Vcc and ground. A small positive voltage will be present at the detector anode. This will hold CR1 at the conduction level and will increase its sensitivity to small input signals. The secondary of T5 couples the modulated 455KHz to the cathode of CR1. This input signal will cause CR1 to conduct.

The direction in which the detector diode is connected would normally make the signal ride on a negative DC voltage level. But because of the voltage divider action, the DC level at the output of the diode will be positive. Diode conduction will change the DC level all along the voltage divider. Small signals will make little change but large signals will make larger voltage changes. Even larger signals can cause the voltage to become slightly negative. The voltage level at the bottom end of R5 (fig 10) is the bias voltage for Q2. This voltage is changed or controlled in accordance with the amplitude of the signal applied to the detector diode CR1. This controlled positive voltage will control the conduction of Q2.

When S1 is in its normal position (operate), C6, which is connected between R5 and R11 will charge up to the average DC level to prevent a change of Q2 bias due to short duration signal changes. For low amplitude signals, Q2 prevents a high gain. As signal amplitudes increase, the bias on Q2 is reduced proportionately and its gain decreases.

The automatic volume control circuit is an extension of the detector circuit, so is usually not shown on the block diagram. However, all receivers include an automatic volume control. You will also hear it called (AGC) or automatic gain control. These terms are used interchangeably. AVC is placed in a receiver because there are many factors that affect the strength of the radio waves that arrive at the receiver antenna. Two of the more pronounced factors are atmospheric conditions and the terrain between the transmitter and receiver. Both of these factors will cause the signal strength to get stronger or fade. When the signal strength changes, the volume level from the speaker changes also. By incorporating AVC circuits the signal strength will be kept relatively constant. Thus, what the AVC is doing is controlling the bias of Q2.

The larger the signal sent through the IF Amplifiers the larger the negative voltage that is coupled back to Q2 base. When this happens, the amount of forward bias on Q2 will decrease resulting in a smaller signal sent through the receiver. The opposite will occur when the input signal is small.

The final circuits are the audio frequency amplifiers. They are called the audio pre-amp, audio drivers, and audio-push-pull amps.

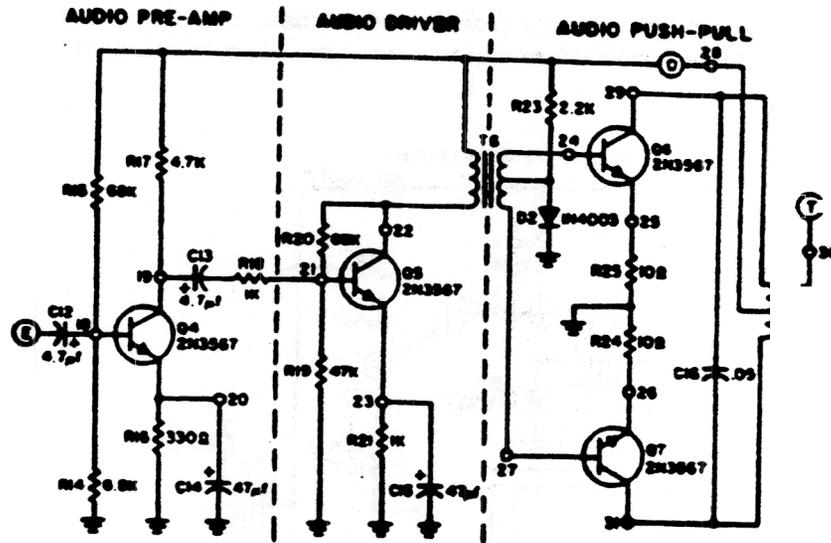


Fig 15

Fig 15 shows a typical audio amplifier common to many receivers. The small audio signal from the detector is amplified in voltage level and power content to drive a speaker at the rated power level. The first stage this small signal encounters is the audio pre-amp Q4. It is a common emitter amplifier biased class A. So that there will be no distorted signal at the collector output TP19. The combination of R14 and R15 set the base operating bias. Along with R14 and R15, emitter resistor R16 provides bias stabilization. C14 by-passes R16 to prevent signal degeneration. The input signal is coupled to the base by C12, R14 RC coupler. The output is coupled to the next stage by C1B, R18. R18 is placed in series to the bias to reduce the signal amplitude to prevent over-driving Q5. If the signal input is too large to Q5 the output will be distorted.

The driver amp Q5 is the next step the signal encounters. This step is necessary to develop the driving current for power amp. stages as output power increases, input current increases proportionately. Q5 is a common emitter, class A amplifier, but uses the primary winding of T6 as its load. The voltage drop across a transformer is quite low, so the Q5 collector voltage will be near Vcc. Its output is the varying current through the primary windings of T6. R19 and R20 form the base voltage divider to provide forward bias. A small amount of signal degeneration may occur through R20, but will help to stabilize the circuit. R21 and C15 are the same as for Q4. The varying current through the T6 primary induce the signal voltage in the secondary. Transformer coupling provides impedance matching and driving current for Q6 and Q7.

The last amps that the signal will see are the Push-Pull amps. Q6 and Q7. Fig 16 Push-Pull amplifiers require input signals that are equal in amplitude, but opposite in polarity, therefore the name is given push-pull. Depending on how they are biased, when one transistor increases conduction, the other transistor decreases or stops conduction.

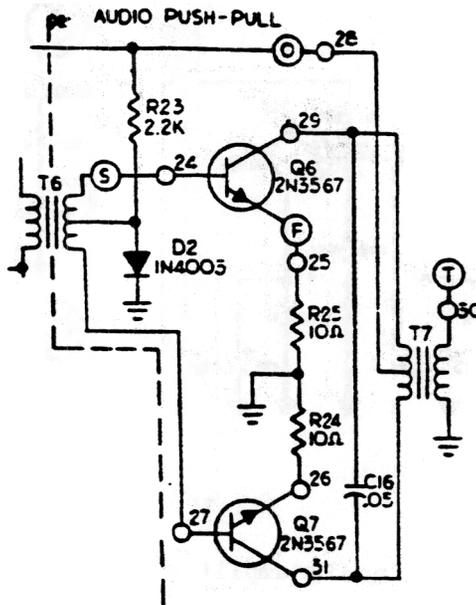


Fig 16

On the next input half cycle, their condition reverses. Push-pull amplifiers can be operated class A or class B. For ideal operation, no output distortion, class AB should be used. The distortion that is observed is called cross-over distortion. Figure 17 shows this.

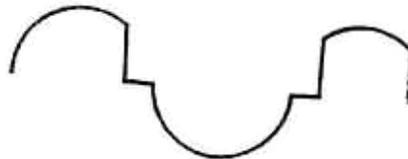


Fig 17

This cross-over distortion is caused by a small portion of the signal used to forward bias the opposite transistor. The output current flowing in opposite directions through the output transformer, T6 primary, combine to develop the complete sinewave signal in the secondary and distortion does not occur. T6 is center-tapped there fore two signals, equal in amplitude and opposite in polarity, are applied directly to the bases of Q6 and Q7. The emitters are connected to ground through 10 ohm swamping resistors. You will also note that there are no by-pass caps on the emitters, therefore, small degenerative signals are developed at the emitters. This aids in stabilizing the circuit and helps to equalize their gain across the audio frequency range. The DC voltage that will be measured on the emitter will be almost 0 volts.

The base bias network for Q6 and Q7 is made up of R23 and diode CR2. CR2 is a silicon diode that drops about 0.6 volts when conducting. The 0.6 volts is applied to the bases of Q6 and Q7 through the T6 secondary. When power is applied to the circuit both transistors will be conducting in the absence of an input signal. When the input signal is applied, one transistor increases conduction and the other turns off on each alternate half-cycle. Fig 18 shows path for each half-cycle. The output load for each transistor is one-half the T7 primary. The Q6 and Q7 currents flow in opposite directions to the center top. These currents provide inductive coupling to the secondary and the complete, undistorted signal, appears at the speaker. T7 is a step-down, impedance matching transformer. Secondary current is increased to drive the speaker voice coil. Output volume is directly proportional to the secondary driving current.

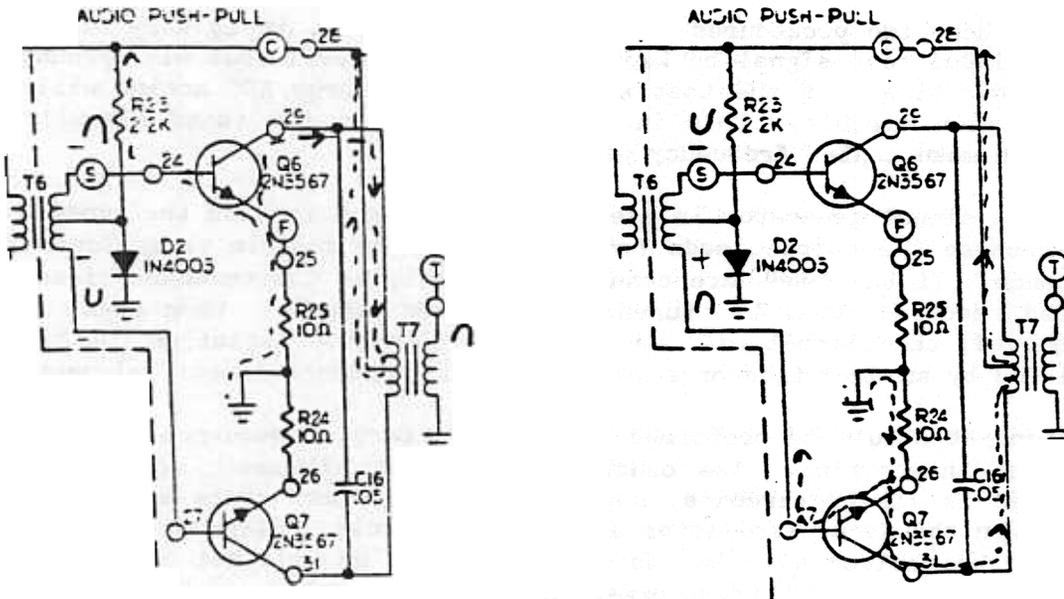


Fig 18

The capacitor C16, is connected in parallel with the primary winding of T7. It has a high reactance for audio frequencies, so audio current flows through the transformer. However, high frequency noise impulses see a low reactance and are shunted around the transformer. C16 functions as a noise filter to keep static and other noise from being reproduced by the speaker.

When all of these circuits are connected together, there is one final step to do before this receiver will be effective. This step is receiver alignment. The purpose of receiver alignment is to adjust all resonant circuits to correct frequency. Receiver alignment should never be attempted unless an accurate signal generator is available and it is known that alignment is necessary due to parts replacement or excessive loss in sensitivity. Receivers do not misalign themselves. If alignment is to be performed, it should only be performed when the receiver is fully operational. Alignment should never be used as a part of the troubleshooting procedure.

When alignment of a receiver is needed, a sensitive indicator must be used. The human ear is not a satisfactory alignment indicator. The type of indicator that will be used will depend on the type of test signal used. If a CW signal is used, a low range voltmeter is connected to measure the AVC voltage. Alignment is indicated when the voltage peaks at maximum value.

If a modulated test signal is used, an output meter or an oscilloscope can be connected to any audio stage or the speaker terminals. Alignment is indicated when the signal voltage peaks at maximum value.

When these two procedures are used for alignment, it is very important that the input test signal be kept at the lowest level that will produce a usable indication. If the test signal is too large, LO large AGC action will reduce the gain circuit. This in turn will flatten the tuned circuit response and make center frequency peaking impossible.

When a signal generator is used we must isolate it from the tuned circuits because the output leads from the generator contain capacitance and inductance. If the leads are connected directly to the resonant circuits, the lead reactance will be "tuned in" with the circuit. When these leads are removed, the circuit will no longer be aligned. Isolation can be accomplished by an amplifier or some other high impedance circuit element.

Alignment should be performed in a set pattern or sequence to minimize adjustment interacting. The order in which the alignment should be accomplished is all IF circuits, starting with the last stage working backwards, then the local oscillator and the RF circuit. Alignment procedures for the lab receiver will be described although we will not be performing this task. The small transformer cores are fragile, and if cracked,

chipped, or broken will render the receiver inoperable. We'll set the RF Generator for a modulated output and we'll monitor the output across the speaker terminals. Adjustments will be made for maximum signal voltages display. As we go through this procedure use the schematic diagram to follow. Fig 19 (schematic of receiver.)

The step by step procedure will be outlined as follows: The first step is to set the volume control at maximum. Set the RF signal generator for a modulated output at precisely 455KHz. Connect the scope across the speaker. Second connect the signal generator to the base of Q3. Reduce the output level until the scope display is 0.5V P/P or less. Then adjust the core of T5 for a voltage peak. After peaking T5, remove the generator leads from the base of Q3 and place the lead to the base of Q2. Reduce the output as necessary and adjust T4 for a voltage peak. This was step 3. When this step is complete, you are ready to remove the generator leads and connect them to the base of Q1. Most alignment procedures require disabling the oscillator. This is easily done by connecting a grounded jumper lead to C1B. Reduce this generator output and adjust T3 for an output voltage peak. Then remove the oscillator disabling jumper. Adjust the tuning capacitor fully closed. Set the signal generator frequency at 540 KHz and connect it to Q1 base. Reduce the signal level as necessary, adjust T2 for a voltage peak. Next turn the tuning capacitor fully open. Set the signal generator at 1610 KHz. Adjust the trimmer capacitor on the side of C1B for a voltage peak.

A note must be taken here before we progress any further. There are some receivers that will not tune the complete AM band. This may require using 550 or 600 KHz at the low end and 1600 or 1550 KHz at the high end. This condition can be recognized by inability to peak both end frequencies.

The adjustments that were just discussed should be repeated until no further improvement can be made. The oscillator should be able to "track" across the broadcast band. Next form the generator hot lead into a single turn loop and place it over the loopstick. Connect the ground lead to a circuit ground. Turn the tuning capacitor fully open and set the signal generator at 1610 KHz (or the highest frequency obtainable in the above example. Set the generator level as required. Adjust the trimmer capacitor on the side of C1A for maximum output voltage. This adjustment will not provide a narrow peak.

When all of this procedure has been accomplished, alignment has been completed. The receiver should now have maximum sensitivity and selectivity. It can be seen that if generator frequencies are not accurate, the alignment will not be accurate. This results in an operational receiver, but at a reduced sensitivity and selectivity level.

Now that we have our receiver working to its maximum ability, there are times when it will break down. Transistors are very durable and we know that the current levels are very low. But, this does not mean the component will last forever. There will be times when a transistor will open or short or some other component break down. The result is we have to fault isolate and repair it. Therefore there is a troubleshooting procedure we must follow in order to isolate and repair the problem. The procedure must follow a systematic and logical sequence to be effective. The procedure we will use is accomplished in six steps. The order is first, operational test, then sectionalize, localize, isolate, repair and finally another operational test.

The initial operational test involves symptom collection. Some of the questions that you will ask yourself are: is power on, does the power lamp light, are switches and controls set properly, are all connections made, and what is the nature of the trouble?

To sectionalize a trouble, to its location as to a compartment, drawer, module, or circuit card is determined. This test is accomplished by signal tracing or signal substitution either or both.

Signal tracing is accomplished by using an oscilloscope. The progress of the signal is followed to note its presence or absence, shape, amplitude, frequency, etc. Signal tracing begins at the input circuit and continues to the chassis, compartment, drawer, module, or circuit card, where the signal is lost or is incorrect.

Signal substitution uses a test signal from a signal generator of the correct type and frequency and applies it to a circuit under test. The results are indicated on a scope or self contained indicators, such as a meter or speaker. Signal substitution begins with the output circuit and proceeds toward the input. As the appropriate signal is substituted, the output is noted. This signal substituting continues until the chassis, compartment, drawer, module or card that blocks or distorts the signal is located.

When signal tracing and or signal substitution narrow the trouble to a circuit card, or stage localization has been accomplished.

Now that the trouble has been narrowed down to a card or stage you are ready to find the bad component. This step is isolation. The trouble will be found by using voltage and resistance checks. The actual defect should be determined, if possible, before unsoldering and resoldering is attempted during isolation testing.

After determining the defective component we are ready to repair it. Before this step can be accomplished the correct tools, equipment, and parts must be available. Only exact or authorized replacement parts should be used.

The final step is another operational test. Its purpose is to insure the system operates properly and no other troubles are present. This test should always be performed before equipment is returned to normal use.

To be of any value, all tests made must provide usable circuit information as to good or bad conditions. This is commonly called circuit analysis. The schematic diagram must be observed to select appropriate test points and provide the basis for evaluating test indications. Before reliable diagnosis can be made, normal circuit operations must be known. Reliable trouble analysis depends on the ability to read and interpret schematic diagrams, correctly use test instruments, and evaluate the test indications.

The step by step procedure has been outlined for you. Now we will look at the most practical method of testing and troubleshooting a radio receiver. The method used is signal substitution. The speaker provides the indicator and the test signal is supplied by the signal generator. A modulated signal is used and the indication is an audible tone from the speaker.

In testing the receiver, the number of tests required can be reduced by a procedure called Bracketing on half splitting. The receiver is divided into half, then into half again, and so on until the bad stage is located. This procedure will be demonstrated using the schematic diagram of the lab receiver. Fig 20

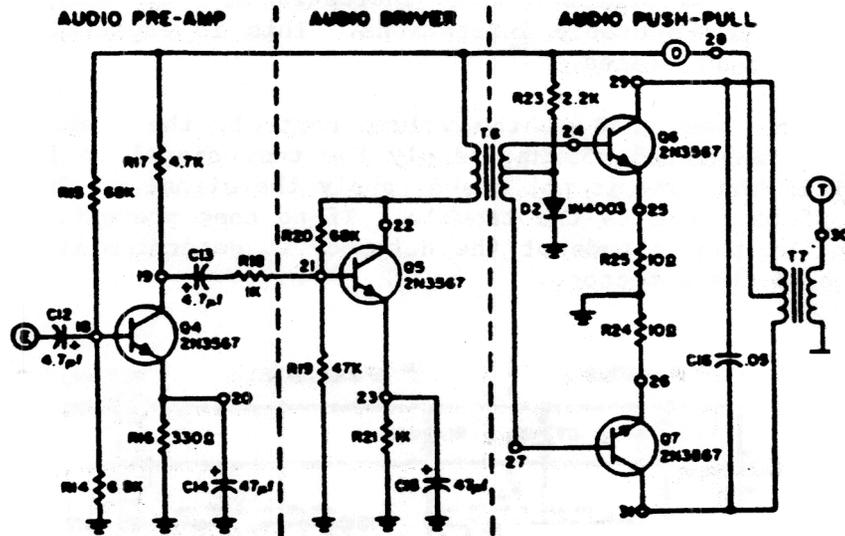


Fig 20

Before testing begins be sure power is applied and all switches and controls are correctly adjusted.

The first test is made at the input of the audio section which is Pin E of PC3 (volume control wiper arm TP15). If the audio section is defective, there is no need to change test frequencies. If the audio section is operational, there is no further need of the audio signal. The audio test signal is usually 600Hz or 1000 Hz and can be furnished from the function generator or from the audio frequency output terminal of the RF signal generator.

If the tone is not heard, the trouble is between the volume control and the speaker. The next logical test is at the base of Q5. If the tone is heard, the trouble must be in Q4 or its circuits. If the tone is not heard, the trouble is in Q5 or the power amplifiers. Inject the test signal at the base of Q6 and Q7. If a weak tone is heard, the trouble is in the Q5 stage. If a tone is not heard, the trouble is in the Q6-Q7 stage, but must be common to both stages. If Q6 side of the push-pull amp is bad and Q7 side good, there would be an output that is distorted on the positive half cycle output and the amplitude would be reduced. If Q7 were good and Q6 bad, the negative half cycle would be effected.

We must also remember that when the injection point moves toward the speaker, there is less amplification and the level of sound decreases.

It may also be necessary to increase signal strength from the generator. However, if too strong a signal is used, it can be passed (coupled) by open stages and provide false indications. Use only enough signal amplitude to provide usable indications. This is especially true in the high gain IF and RF stages.

If the tone was good at the volume control, the test signal must be changed to a modulated 455KHz. Apply the test signal to the base of Q2. See fig 21 if the tone is not heard, apply the signal to the base of Q3 to determine if Q2 stage is the trouble. If no tone present, then apply the test signal at the cathode of the detector to determine if the trouble is in Q3 stage of the detector.

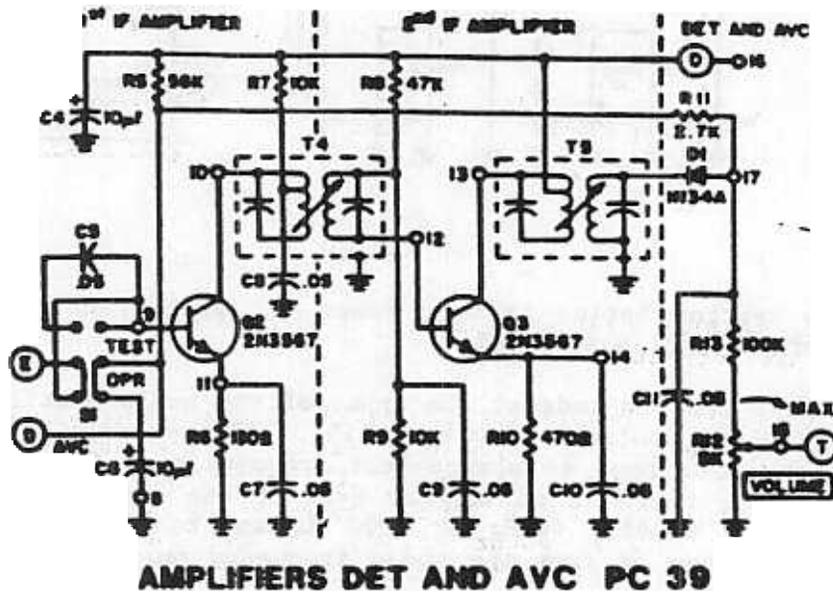


Fig 21

The IF signal can be used to test the circuits up to the Base of Q2. If the tone is heard from the base of Q1, the trouble is in the RF oscillator stage.

To test the RF section the tuning capacitor position and the RF frequency must be in agreement. Fully closed is about 540 KHz and full open is about 1610 KHz. Use Fig 22 to follow along.

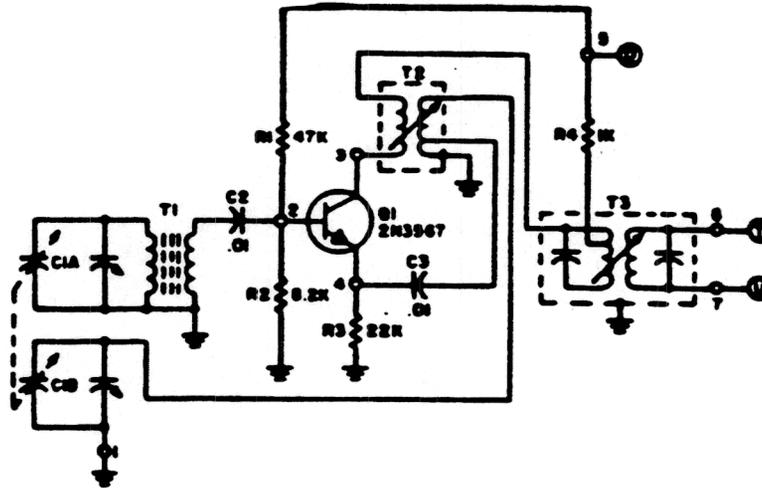


Fig 22

Testing the oscillator by signal substitution is a bit more involved. Set the tuning capacitor to a known station frequency.

Set the signal generator 455KHz higher than the received signal frequency and connect it to the emitter of Q1. If the station can be heard, the oscillator is defective. If a scope is available the oscillator signal can be observed at the emitter of Q1.

If a good oscillator signal is present on the emitter of Q1, but there is no station heard - the RF amp is bad. To make the DC voltage checks around the RF amp, you must first ground out the local oscillator.

Testing the RF section requires close attention to signal generator and the tuning frequencies. Do not use too strong a signal or coupling may occur and give false indications.

Effective troubleshooting of a superheterodyne receiver is based upon the proper use of the schematic diagram, test equipment and testing by signal substitution. When the defective stage is located, testing within the stage continues with voltage checks to find the defective component. The trouble is isolated to a specific component with the ohmmeter. All trouble can be designated as open, short, or changed value.

The schematic must be carefully analyzed to prevent wrong conclusions due to parallel paths and transistor junctions. The Bracketing (half-split) technique is the preferred system of troubleshooting because it saves time and reduces the number of unnecessary tests.

SUMMARY: The basic purpose of the superheterodyne receiver is to enable us to tune to a radio station, accept the frequency, amplify the signal, detect and amplify the audio intelligence carried by the RF carrier.

(The material covered in this presentation is UNCLASSIFIED)