# 18 The Colt 80 m receiver – Part 1

## Introduction

This is the first of four parts detailing the design and construction of a simple radio capable of good reception of amateur radio signals on the 80 metre band. If you can solder and have some basic hand tools, at the end of Part 4 you will have a working receiver, in which you can take pride, and start some serious listening on 80 m! For testing, all you will need is another receiver and a multimeter.

## Description

The radio will be built in three modules, or stages, as illustrated in **Figure 1**. Each of these is built on a printed-circuit board (PCB) or matrix board and can be tested in its own right. The case will need some holes drilled, and care will be needed to mount the PCBs in the case.

## Get to it!

Surprisingly enough, when you are building a radio section by section (which is *always* the best way), it is easiest to start at the output and work

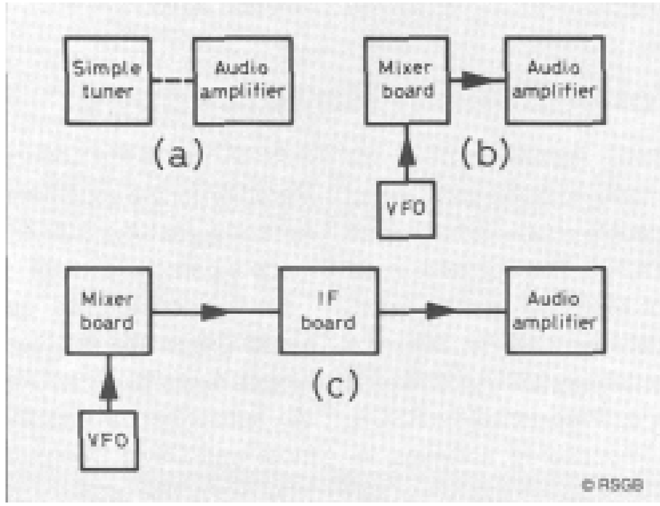



**Figure 1** Build your Colt and watch it grow. A simple crystal set (a) becomes a direct conversion receiver (b) and finally an 80 metre amateur band superhet (c)

towards the input, and you can test what you have done stage by stage. You will see what this means as you progress with your construction.

Every receiver needs some audio frequency (AF) amplification to make the sound signals big enough for you to hear. The circuit for the AF amplifier is shown in **Figure 2**. It uses the TDA7052 integrated circuit (IC) plus a handful of extra components. R1 in conjunction with C2 and C3 *decouple* the battery supply, preventing any audio signals getting through to it and affecting other parts of the radio, when they are connected. C1 acts to prevent high frequencies (above the 3 kHz bandwidth) going into the amplifier input. A volume control, VR1, is connected across the amplifier input, so that the amplifier can accept signal inputs over a wide range. **Figure 3c** shows the connections, which are made with screened cable. The

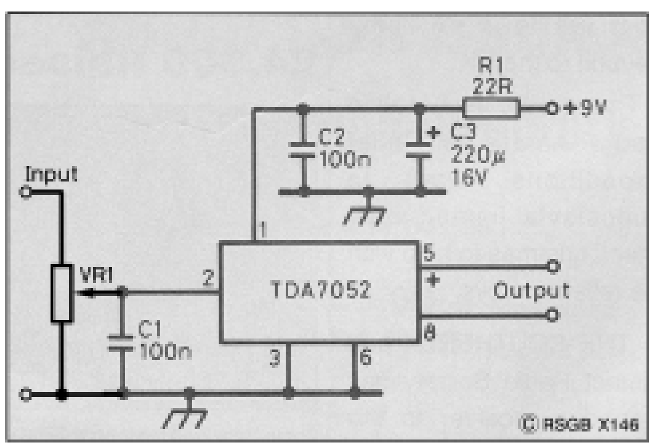



**Figure 2** The Philips TDA7052 integrated circuit (IC) used in the audio amplifier needs very few extra components. It has a signal voltage gain of 100 times and the output is suitable for a loudspeaker or headphones

centre conductor of the left lead goes to the point marked 'input' in Figure 2, the braid being connected to the amplifier earth (0 V) tag. The right lead to VR1 goes to whatever signal source you have for testing – see later. VR1 is *not* mounted on the PCB.

The circuit is constructed on a small PCB or matrix board. Make sure that the electrolytic capacitor, C3, is soldered into the board the correct way – its positive and negative connections are shown in **Figure 3b** for reference. If you are **at all** concerned about soldering the IC into the board, enlist some help, *or*, obtain an 8-pin DIL socket, which you can solder in and then carefully insert the chip into the socket, making sure that is the correct way round. The markings on the chip are shown in **Figure 3a**.

The output leads from the amplifier go to a plastic 6.3 mm (¼ inch) mono jack socket, so that *neither* output lead is connected to the metal case. The amplifier will drive a pair of headphones or a small 8 Ω loudspeaker.

The battery leads must be the right way round also; the battery itself can be a PP3 or PP9, or you can use a small DC power supply.

## Testing

First, check that all the components are in the right places, that your soldering is good, and that you have headphones or a loudspeaker connected. Set VR1 about halfway along its travel. Connect the battery.

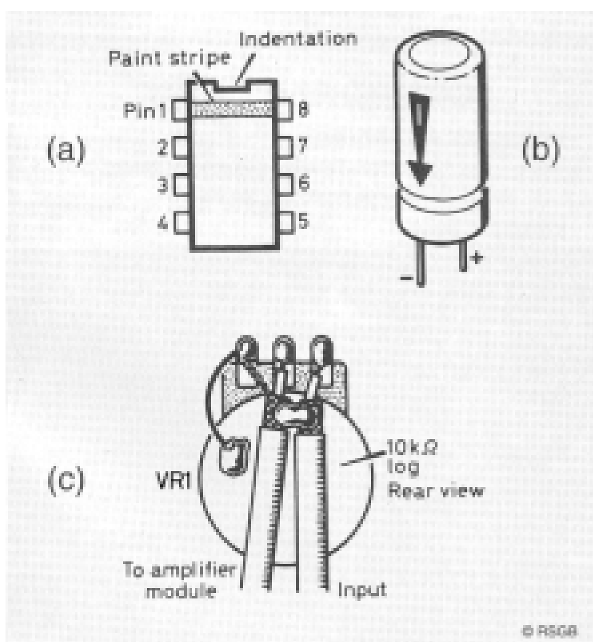



**Figure 3** It is important to check the component connections carefully. The diagram shows (a) top view of the IC, (b) electrolytic capacitor and (c) volume control (VR1) connected across the input of the amplifier

A slight hissing noise should be heard; touching the input lead to the amplifier (the centre of the three connections on VR1) should produce a loud buzz. Touching the shaft at the same time will make the buzz quieter. This is the quickest way of confirming that your amplifier *seems* to be working. The only real test is to give it something meaningful to amplify! See the design of our *Crystal Radio Receiver* for full details.

## Parts list

Resistors: all 0.25 watt, 5% tolerance

| | |
|---|---|
| R1 | 22 ohms (Ω) |
| VR1 | 10 kilohms (kΩ) log |

Capacitors

| | |
|---|---|
| C1, C2 | 0.1 microfarad (μF) |
| C3 | 220 microfarads (μF) electrolytic 16 V |

Integrated circuit

| | |
|---|---|
| IC1 | TDA7052 audio amplifier |

Additional item

| | |
|---|---|
| PCB | (see below) |

Component suppliers:
Maplin

## The next part . . .

The metal case will be marked out ready to receive the completed modules.

# 22 The Colt 80 m receiver – Part 2

## Introduction

In Part 1 we constructed the audio amplifier module for the system and tested it in a very simple way. If you did what was suggested and built the simple Crystal Set to use as a signal source, you will know just how well the amplifier works.

## The case

Metal cases for the project are available from Maplin, telephone 01702 554 161 (code XB67). From the photograph on p. 72 you can see the way the components are mounted. The audio amplifier is seen at the top right of the base, to the right of the tuning capacitor VC1. The next in this series will deal with the variable-frequency oscillator (VFO) and VC1. The current part deals with preparing the case to receive the components.

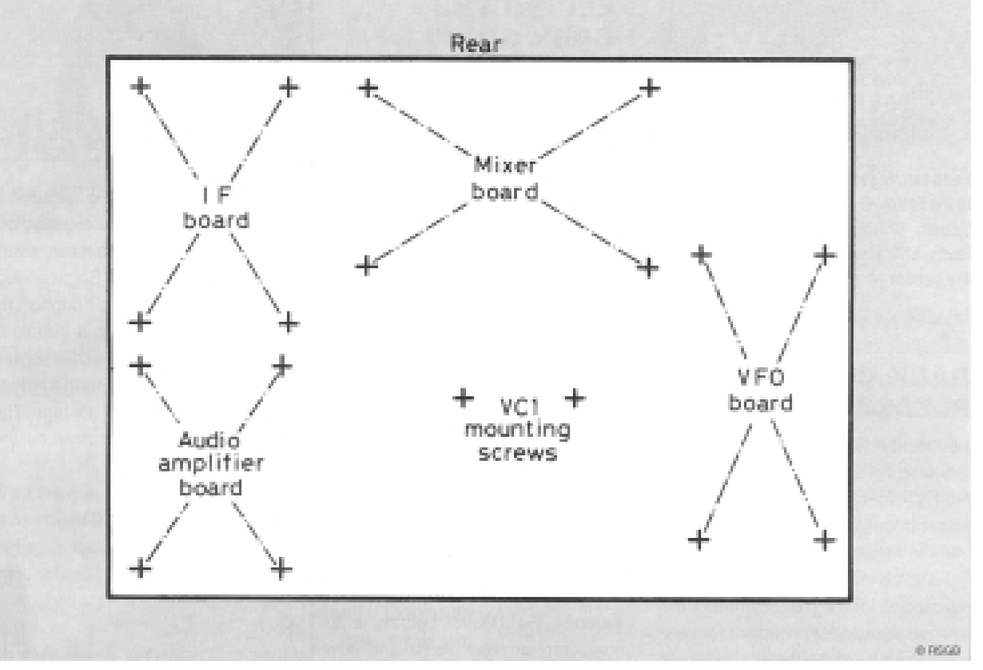



**Figure 1** Fixing holes for each module are best measured from each printed circuit board or matrix board

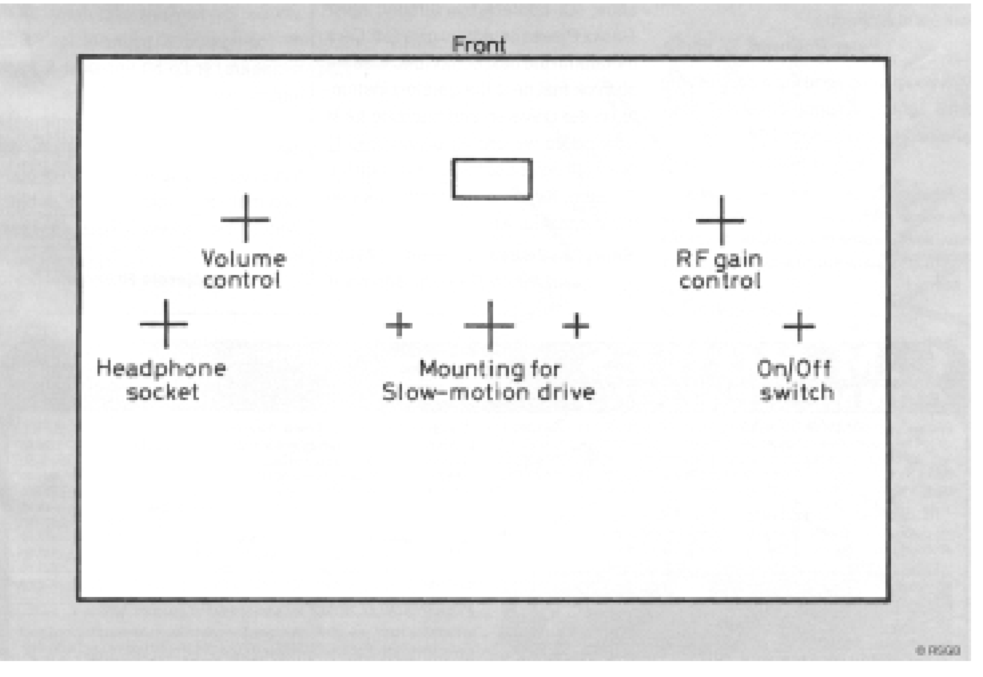



**Figure 2** Position the slow motion drive to allow viewing of the tuning dial

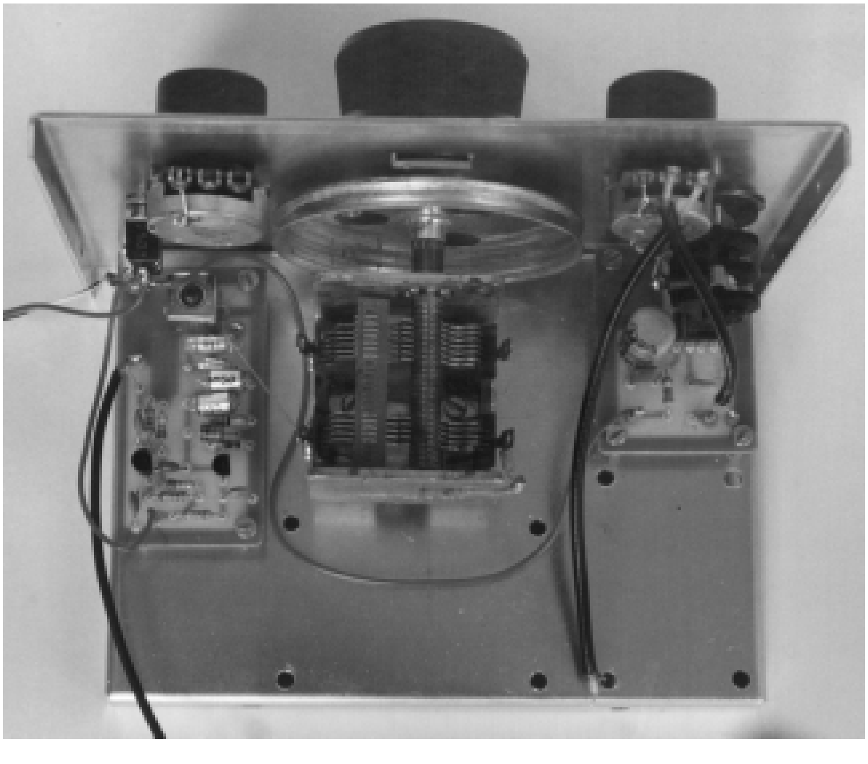
**Figures 1** and **2** show the markings for preparing the front panel and base. All the circuit boards and the tuning capacitor are mounted on the base using 10 mm stand-off pillars with 6BA bolts. The board locations are shown in Figure 1. The front panel control positions are shown in Figure 2, together with the small rectangular hole for viewing the tuning dial.

The best way to mark out the holes for the boards is to lie the boards flat on the base (before you've started soldering the components in) and marking the base through the holes in the boards. This minimises the scope for errors!

A *reduction drive* is used between the tuning knob and the capacitor shaft. This is simply a gear mechanism that slows down the capacitor shaft by a factor of six compared with the tuning knob, and makes tuning very much easier. The recommended variable capacitor also has a pulley wheel mounted on the shaft. Glued to this wheel will be a scale marked with frequency and is visible through the rectangular hole in the front panel.

## The next part . . .

The variable-frequency oscillator and mixer will be added to the project.



# 25 The Colt 80 m receiver – Part 3

## Introduction

In Part 2 we marked out the case ready for installation of the modules when they are completed. In this part, the building of the variable-frequency oscillator (VFO) and mixer will be described. This will produce a type of receiver known as *direct-conversion*, because it converts the radio-frequency (RF) signal directly into an audio-frequency (AF) signal which we can hear in a loudspeaker after amplification. A block diagram of the system is shown in **Figure 1**. The modifications needed to make a full superheterodyne receiver will be left until later.

The direct-conversion receiver covers the 80 metre amateur band and will receive both Morse (CW) and speech (SSB) signals. The audio amplifier was covered in Part 1, so your *Colt* is rapidly taking shape! By the time your construction has reached the end of this part, you will have a receiver ready to use, even if the project is not yet complete!

## The direct conversion process

Like most things in radio, the principles of direct conversion are not difficult. From the aerial, the signal we want to hear is selected by the *tuned filter*, which rejects the signals we don't want. The signal then enters the

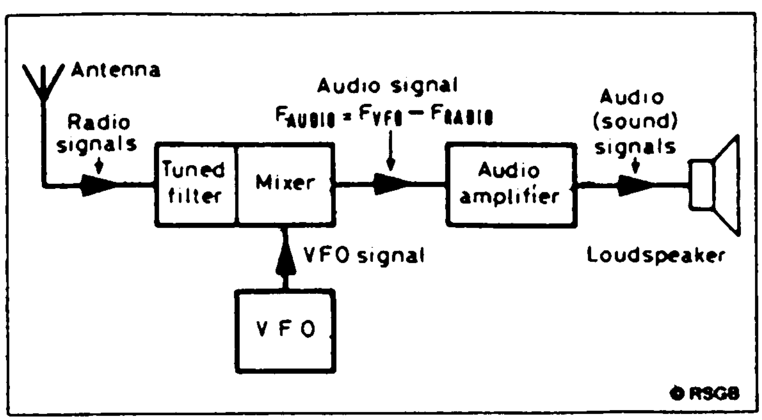



**Figure 1** Stages of direct-conversion receiver

*mixer*, along with the signal from the *VFO*. The VFO produces a sine wave whose frequency can be varied across the whole of the 80 metre amateur band (3.5–3.8 MHz), by turning the knob on the tuning capacitor, VC1. The mixer produces, at its output, *two* signals; one signal is at the frequency of the **sum** of the signal and VFO frequencies, the other is at the **difference** of the two frequencies. It is the latter that we want. Let's look at the numbers involved. If the signal is at 3650 kHz and the VFO is at 3651 kHz, then the sum frequency is 7301 kHz, and the difference frequency is 1 kHz. If we feed the output of the mixer into our audio amplifier, the 7301 kHz signal is automatically removed (it is far too high to be considered an audio signal!) and the resulting 1 kHz signal is amplified and fed to the speaker, producing a note which we can hear!

## Building the VFO

**Figure 2** shows the circuit of the VFO. It is a tried and tested circuit, and should work first time. It uses a *field-effect transistor* (FET) for TR1, the oscillator itself. RFC is a *radio-frequency choke*, a coil of wire which will pass a direct current (DC) but which will prevent radio-frequency (RF) signals getting through.

All VFOs have a tuned circuit which, in this case, is formed by the coil L1 and the capacitors C1 and VC1. The frequency will also be affected to some extent by C2, C3, C4 and C5. Transistor TR2 is an *emitter follower*, a stage which gives no voltage gain but provides a good *buffer* stage, isolating the VFO from the effects of the stages that follow it. When building a VFO, the

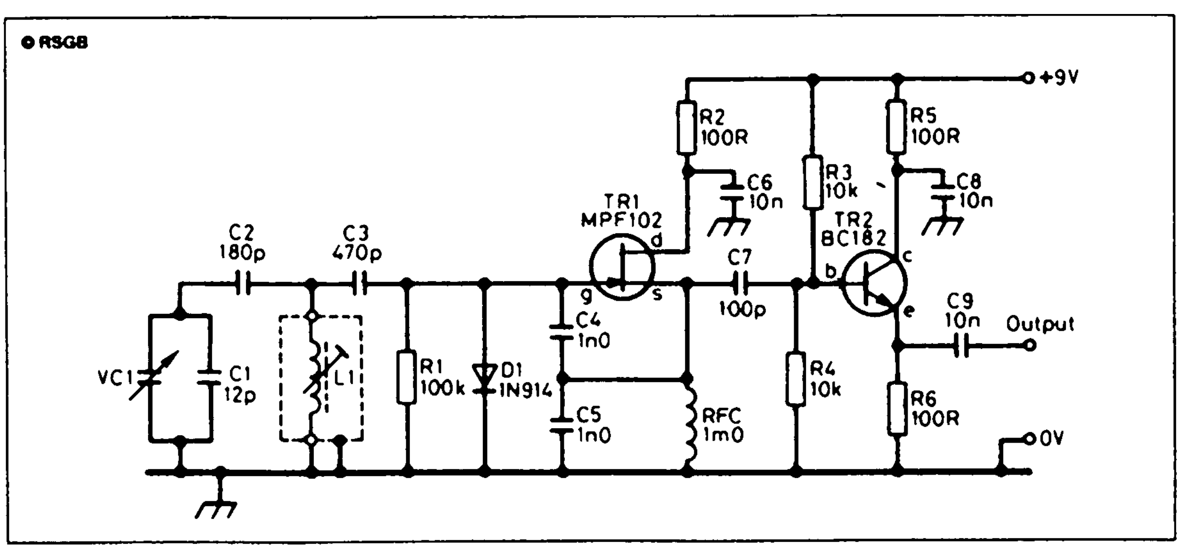



**Figure 2** The variable-frequency oscillator uses a field-effect transistor (FET)

parts must be securely mounted. If components move, so does the frequency! At worst, the oscillation will become unstable and the VFO will be useless. Keep the component leads as short as possible – this improves their mechanical stability as well as their electrical stability!

Mount the parts on the printed-circuit board (PCB) or matrix board and, when completed, the VFO should look like the one shown on the left in the photograph on p. 72. Make sure that TR1, TR2 and D1 are the right way round.

On completion, check the component positions then mount it in the case as shown in the photograph, to the left of the tuning capacitor when viewed from the rear. The VFO coil, L1, will need some adjustment, but that will have to wait until the mixer is built. Connections to the other boards are made with screened cable.

## The mixer board

So far, we have an audio-frequency (AF) amplifier and a VFO; the addition of a mixer board gives us a complete direct-conversion receiver for 80 m. The mixer circuit diagram is shown in **Figure 3**. Let's follow the signal path.

- From the aerial, the RF signal goes to the gain control potentiometer, RV1. This reduces very strong signals, to prevent them overloading the mixer.
- To select the required band of signals, a *bandpass* filter is made up of RF transformers T1 and T2; these are tuned by C1 and C3, and are coupled together by C2.

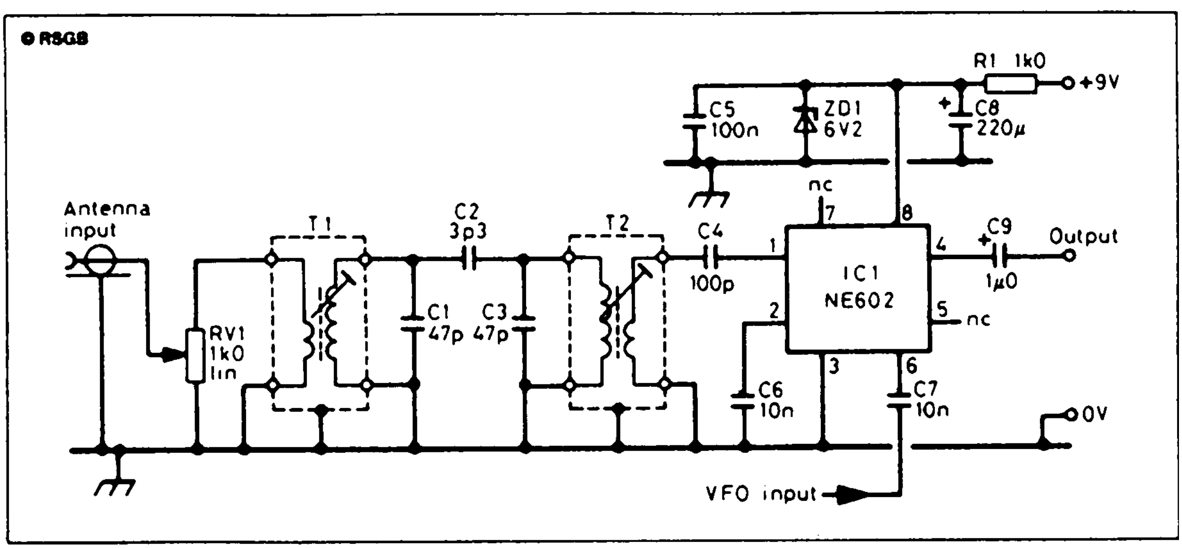



**Figure 3** The mixer board has a bandpass filter and stabilised supply

- After the filter, the signal is coupled into the integrated circuit (IC) mixer type NE602, by the capacitor C4. Capacitors C5 and C8 *decouple* the supply line, to prevent unwanted signals on the supply from disturbing the VFO operation. The use of the term *decouple* is exactly the opposite of *couple*; when two circuits are *coupled* together, the signal passes from one to the other; when two circuits are *decoupled*, signals **cannot** pass from one to the other.
- The NE602 works with a 6 V supply; it is produced here from the 9 V supply by the *Zener diode* ZD1 and the resistor R1. ZD1 operates at 6.2 V, and gives a steady output for the mixer. The audio output from the mixer appears at pin 4 or IC1. This is taken to the audio amplifier board via C9 and the volume control (see Part 1).

Care must still be taken to insert some components the right way round. These are the electrolytic capacitors, C8 and C9, the Zener diode, ZD1, and the integrated circuit, IC1. Check all component positions and make sure all your soldered joints are bright and shiny.

## Putting it together

The interboard wiring, shown in **Figure 4**, uses screened cable; ideally, this should be thin coaxial cable, but screened microphone cable is suitable. The diagram shows how the two controls, the RF Gain and Volume, are connected to the boards. The leads marked '+9 V' are all connected to the battery supply via a miniature on/off toggle switch. Double check all connections before connecting the battery.

## Setting the VFO

Very little adjustment is needed to get the receiver going. Firstly, the VFO must be adjusted to cover the required band, in this case 3.5–3.8 MHz.

If you have a frequency counter, connect it to the output of the VFO. If you haven't, read this part anyway so you understand the process, then another means of setting the VFO will be given especially for you! Rotate VC1 until the vanes are fully meshed. Very carefully, adjust the core of the VFO coil (L1) with a plastic trimming tool so that the frequency approaches and settles at 3.500 MHz. When you rotate VC1, the frequency should increase to **at least** 3.800 MHz at the far end of its travel.

In the absence of a frequency counter, borrow a communications receiver, set it for SSB reception (USB or LSB) on exactly 3.500 MHz. Set VC1 with the vanes fully meshed and turn the core of L1 in both directions until you hear a whistle in the communications receiver. Rotate the core so that the whistle reduces in frequency. It will eventually fade out at around 200 Hz;

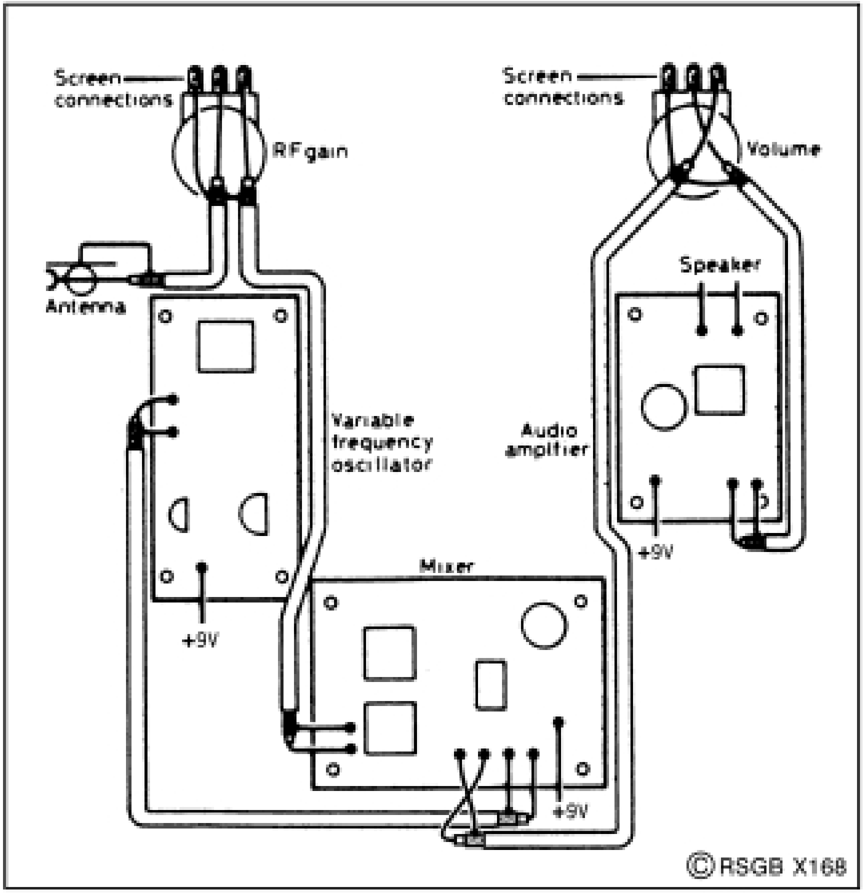



**Figure 4** Make sure that interconnections between the boards are correct, including cable screens

turn the core a little further and then leave it at that position. Rotate VC1 right to the other end of its travel, and search for the whistle with the tuning knob of the communications receiver. Check that the VFO frequency is at least 3.800 MHz.

Then, whichever method you are using for frequency measurement, mark the dial with frequency steps of 50 kHz. Setting and calibration are finished!

## Setting the mixer

Again, there are various ways of doing this. If you have a signal generator, inject a signal at a frequency within the 80 m band, and adjust the cores of T1 and T2 sequentially for maximum output.

If you haven't a signal generator, connect an aerial to the mixer input, set the RF Gain to maximum (fully clockwise), and find a consistent signal. Adjust the volume control to a comfortable level. Rotating the core of T2 with your trimming tool, maximise the output. Then do the same with T1, although this will have much less effect. Find another station, and check that the positions of the cores aren't too different for a maximum signal.

You may find that your receiver benefits from the insertion of an aerial tuning unit (ATU) between the aerial and the input, to compensate for the impedance of your aerial not being 50 Ω. A design for such an ATU is presented in another part of this series. If the signals are still weak, connect the ATU to the junction of C1 and C2 via a 100 pF capacitor.

## Try listening!

Remember that 80 m is a variable band. During daylight hours, your will hear Morse signals at the lower end of the band, and some British and closer continental stations between 3.7 and 3.8 MHz. In the evenings, stations up to 1000 miles away should be heard. Look for Novices around 3.7 MHz!

## Parts list – VFO board

| | | |
|---|---|---|
| Resistors: all 0.25 watt, 5% tolerance | | |
| R1 | 100 kilohms (kΩ) | |
| R2, R5, R6 | 100 ohms (Ω) | |
| R3, R4 | 10 kilohms (kΩ) | |
| RV1 | 1 kilohm (kΩ) linear | |
| Capacitors | | |
| C1 | 12 picofarads (pF) polystyrene | |
| C2 | 100 picofarads (pF) polystyrene | |
| C3 | 470 picofarads (pF) polystyrene | |
| C4, C5 | 1 nanofarad (nF) polystyrene | |
| C6, C8, C9 | 10 nanofarads (nF) polystyrene | |
| C7 | 100 picofarads (pF) min. ceramic | |
| VC1 | 140 + 140 picofarads (pF) variable | |
| Semiconductors | | |
| TR1 | MPF102 | FET |
| TR2 | BC182 | npn |
| D1 | 1N914 | silicon |
| Inductors | | |
| L1 | Toko KANK3334 | |
| RFC | 1 mH | RF choke |

## Parts list – mixer board

Resistors: 0.25 watt, 5% tolerance

| | |
|---|---|
| R1 | 1 kilohm (kΩ) |
| VR1 | 1 kilohm (kΩ) potentiometer (linear) |

Capacitors

| | |
|---|---|
| C1, C3 | 47 picofarads (pF) min. ceramic |
| C2 | 3.3 picofarads (pF) min. ceramic |
| C4 | 100 picofarads (pF) min. ceramic |
| C5 | 100 nanofarads (nF) min. ceramic |
| C6, C7 | 10 nanofarads (nF) min. ceramic |
| C8 | 220 microfarads (µF) electrolytic 16 V |
| C9 | 1 microfarad (µF) electrolytic 16 V |

Integrated circuit

| | |
|---|---|
| IC1 | Philips NE602 or NE602A |

Additional items

| | |
|---|---|
| T1/T2 | Toko KANK3333 |
| | On/off switch |
| | Miniature toggle switch |

## The next part . . .

The IF amplifier and the Beat-Frequency Oscillator will be added to convert the *Colt* into a superheterodyne receiver.

# 27 The Colt 80 m receiver – Part 4

## Introduction

In Parts 1 to 3 of this series, the design of an 80 metre direct-conversion receiver has been described. In this final part, we are going to change the circuit to operate as a superheterodyne receiver, or *superhet*. Most radio receivers are superhets, and they change the incoming signal to another frequency, known as the *intermediate frequency*, or IF, before producing an audio signal. The use of a superhet in a good receiver is mandated by the requirements for good *sensitivity* and *selectivity*.

## Sensitivity and selectivity

**Figure 1** shows the block diagram of the receiver. If you look closely, you will see that it is two similar circuits, one after the other. The incoming

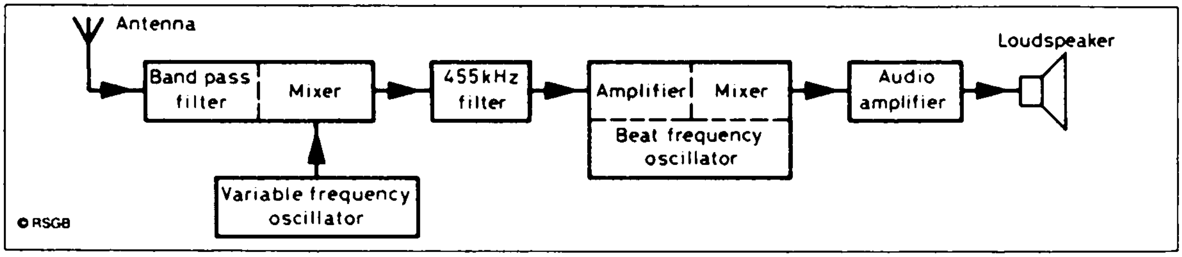



**Figure 1** This is how the different stages of the Colt go together to make a superhet receiver

signal is filtered and fed to a mixer where it is combined (mixed) with the signal from the variable-frequency oscillator (VFO). This oscillator operates at a frequency which is 455 kHz higher than the incoming signal from the aerial, and the mixer output is therefore at a frequency of 455 kHz. If you are not sure about this, please refer to the section 'The direct conversion process' in Part 3 of this project. This new frequency is called the *intermediate frequency*, or IF. This frequency doesn't change; the tuning is accomplished by the VFO, and the mixer output is *always* at 455 kHz. The extraction of the audio signal from the IF signal is identical with the direct conversion process which is used in your existing receiver.

This may seem a long-winded way of doing things, but it has its advantages. A receiver must have a good *sensitivity*, or gain, so that it can receive very weak signals. In very general terms, it is easier to handle low-frequency signals than it is to handle high-frequency signals. We are changing our signal frequency from around 3.6 MHz down to 0.455 MHz (455 kHz), which is much lower and can be filtered and amplified relatively easily. A second advantage is that it is easier to provide gain at a fixed frequency than at a variable frequency. Remember that the IF is fixed, and providing gain is, again, relatively simple.

Our receiver also needs good *selectivity*, the ability to separate (or select) one station from another very close to it in frequency. This requirement is significantly simplified by the fact that the IF is fixed, and a good filter in the IF circuits can do wonders for the rejection of adjacent-frequency stations! Several stages of IF amplification and filtering are possible in more adventurous designs.

The filtered signal at 455 kHz passes to another mixer which has an associated oscillator, usually called a *beat frequency oscillator* (BFO). When receiving CW (Morse) signals, the BFO is usually tuned about 1 kHz above or below the IF (i.e. at 454 or 456 kHz) to produce a 1 kHz beat note as the audio signal. This signal is then amplified and fed to a loudspeaker or headphones.

# The circuit

**Figure 2** shows the circuit of the IF section shown in the photograph on p. 94; this section is added to the existing circuit to make it a superhet. The existing mixer board (described in Part 3) is used as the first mixer. Between it and the audio amplifier is connected the new IF board.

The signal output from the first mixer (at 455 kHz IF) is fed into a *crystal filter*, the most expensive part in the whole receiver. It provides the selectivity which makes the Colt such a good receiver. Another NE602 mixer/oscillator chip follows the filter. The oscillator section is controlled by the tuned circuit in T1, the frequency of which can be altered by rotating the core inside the coil. Once set, it remains fixed.

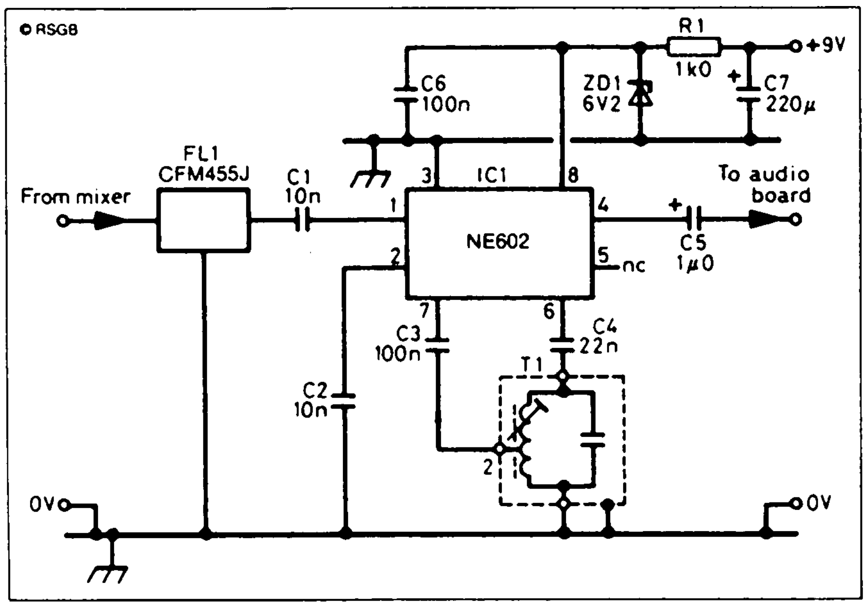



**Figure 2** The IF board is connected between the mixer and audio amplifier

At this stage you should have built the first mixer and the audio amplifier, and proved that they *both* work by using the circuit as a direct-conversion receiver. When you have finished constructing and checking the IF board, you will need to add it to your existing circuits.

# Adjusting and testing

There are three pairs of connections to the IF board – the 9 V supply leads, the input leads and the output leads. The IF board should be mounted on the metal baseplate, along with the other boards. **In making the following connections, make absolutely sure that the braid of each piece of coaxial**

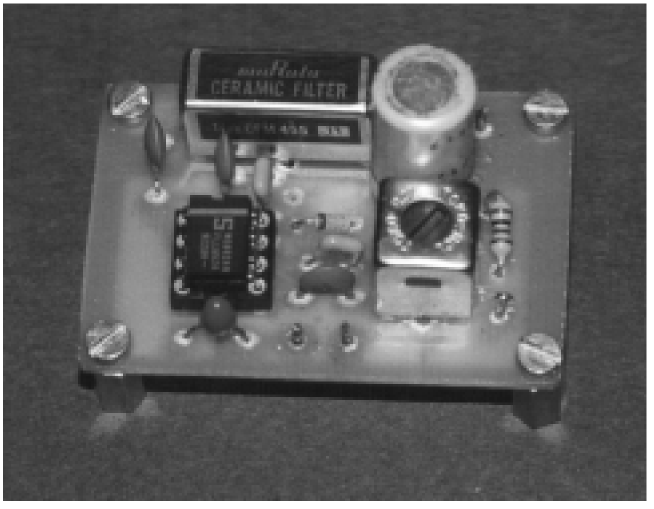
**cable is soldered to the correct connection on each board. The same applies to the polarity of the 9 V leads.** Disconnect the screened lead which presently goes from the first mixer board to the AF amplifier at the amplifier end. Connect it instead to the *input* of the IF board. Connect a new piece of screened cable from the IF output to the vacated AF input. Now connect the 9 V supply leads to the same points as the supply leads from the other boards. Check these newly made connections.

You should have confidence at this point that things should be right. After all, you *have* tested the direct-conversion process and you *know* it works. All that you are now testing is the IF board. This is the attraction of building a receiver in modules, working from the speaker backwards, and testing as you go!

The VFO and BFO need to be correctly adjusted. Mesh the vanes of the tuning capacitor and, using the same frequency measurement method as you did originally, set the VFO frequency to 3.955 MHz (which is 3.500 MHz + 0.455 MHz, if you hadn't guessed!). The BFO can be set using a frequency counter, but it is just as good to set it by listening to SSB or CW signals. A high-pitched hissing sound should be heard in the speaker. As you rotate the core, the pitch should reduce, go through a minimum, then increase again. Set the core at the minimum pitch position. You may want to readjust the two cores a little as your listening skills improve but, once you are happy, they will never need to be altered again!

## In conclusion . . .

You should now have built a superhet receiver capable of excellent results. It uses the same type of circuit as that found in far more advanced receivers.



The superhet is far more sensitive than the direct-conversion type, and can weed out those elusive DX stations. You may have found that a station will appear at two places on the dial of the direct-conversion receiver; you will have no such problem with the superhet.

## Parts list

| | |
|---|---|
| **Resistor** | |
| R1 | 1 kilohm (kΩ), 0.25 watt, 5% tolerance |
| **Capacitors** | |
| C1, C2 | 10 nanofarads (nF) min. ceramic |
| C3, C6 | 100 nanofarads (nF) min. ceramic |
| C4 | 22 nanofarads (nF) min. ceramic |
| C5 | 1 microfarad (μF) 16 V electrolytic |
| C7 | 220 microfarad (μF) 16 V electrolytic |
| **Integrated circuit** | |
| IC1 | NE602 |
| **Additional items** | |
| ZD1 | Zener, 6.2 V 0.5 watt |
| FL1 | Crystal filter, Murata CFM455J |
| T1 | Tuned inductor, Toko YHCS11100AC |