Project 741

High quality 10-channel

RADIO MICROPHONE

Part 2

This article goes into the alignment and suggests a method of encasing the unit to 'look like a bought one'. There's no reason you can't fashion a professional-looking unit for yourself with a little ingenuity and elbow grease.

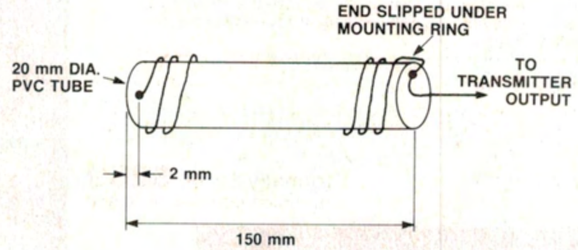
Ian Thomas

NOW WE GET DOWN to the nitty gritty of putting the unit on-air and making a suitable housing for it. What I describe here is simply how I housed the unit for myself. You can make a 'Chinese copy' or simply use what I've done as a guide. Then again, you might like to do something completely different! Enough. On with it.

Alignment

Tack a piece of wire on to the antenna lead point but at this stage it need only be 100 mm or so long. The next thing you have to decide is what part of the 88 to 108 MHz band you want to use. Tune your FM receiver across the band and find what seems to be a comparatively unoccupied region (they're getting a bit hard to find in Sydney) and note where you want to use the radio microphone. Then, using Table 1 cut the appropriate tracks to pins 9 to 13 on the board to select that band of frequencies in the synthesiser. Set the rotary switch to channel 4, connect a power supply to the appropriate terminals on the board and stoke the beast up. Monitor the node of R9, R11 and C6 with a 10M input impedance DVM or the like and slowly screw the ferrite adjusting slug into L1.

WINDING THE HELICAL ANTENNA



The appropriate number of turns (see Table 1) are wound evenly along the former using 1 mm wire. A length of heatshrink tubing will keep them in place and improve the appearance.

MC145112 DIN					CH0	CH9	ANTENNA TURNS
9	10	11	12	13			
	X			X	88.0	90.25	23.0
	X				92.0	94.25	22.0
		X	X	X	96.0	98.25	21.125
		X	X		100.0	102.25	20.25
		X		X	104.0	106.25	19.5
		X			108.0	110.25	18.75

Table 1: Cut the tracks to the PLL's pins as indicated (by the 'X') to get the desired synthesiser range. The antenna turns for each range are also given.

Before you start adjusting L1 the voltage on the DVM should be very close to ground but as the slug is part way in the voltage should start to rise. It may help if the cup core is loosely placed over the coil but it shouldn't be necessary. A word of warning about adjusting the ferrite slug: they're rather fragile and if the slug's right in the former and breaks it's nigh on impossible to get the pieces out. Before you start, get or make a brass or copper adjusting tool that fits nicely (it's kind of dumb to use iron!).

Adjust the slug until the DVM reads about two and a half volts give or take a volt. Tune your FM receiver across the band watching the signal strength meter. Around the frequency you expect to be radiating the receiver should go quiet. You should see a massive signal with no modulation. If there's a station on the frequency you're using, the radio microphone will probably swamp it out but if you want you can change channels to a vacant frequency (that's why you spent the money). When you do change channels the control voltage should go up or down by about 0.25 volts per channel.

Trouble shooting

If all this is happening OK then you're definitely cooking with gas and the whole

FREQUENCY SYNTHESISERS

With today's demands on the frequency spectrum it is essential that the frequency of all radio transmissions be precisely controlled to avoid interference between adjacent channels. In the past this was done by having each transmission frequency set by separate quartz crystals. If many frequencies were wanted in the one unit this meant that a considerable part of the unit cost was taken up by crystals and their associated circuitry. It also meant that an upper limit was placed on the number of available channels in a unit by space requirements even if cost alone was ignored. Clearly a better way of setting transmission frequency was needed.

It was inevitable that digital techniques would provide the answer in the form of the digital phase-locked loop (PLL) frequency synthesiser. This circuit enables an arbitrary number of frequencies to be generated which are all controlled by the one quartz crystal and which have the same stability (at least in the long term) as the crystal. Years ago the digital circuitry needed was far bulkier than multiple crystals and used *far* more power. Recent developments in high speed CMOS have reduced package count and power to the point that the frequency synthesiser is a good alternative to the multiple crystal transceiver.

The basic circuit for a PLL frequency synthesiser is shown in the diagram and may be divided up into seven separate blocks. The first is the **reference oscillator** which is usually controlled by a quartz crystal and gives out a frequency which we will call f_r. This frequency is divided down digitally by a **reference divider**, whose division ratio is normally fixed, to give out a much lower frequency which we will call f_c or the comparison frequency.

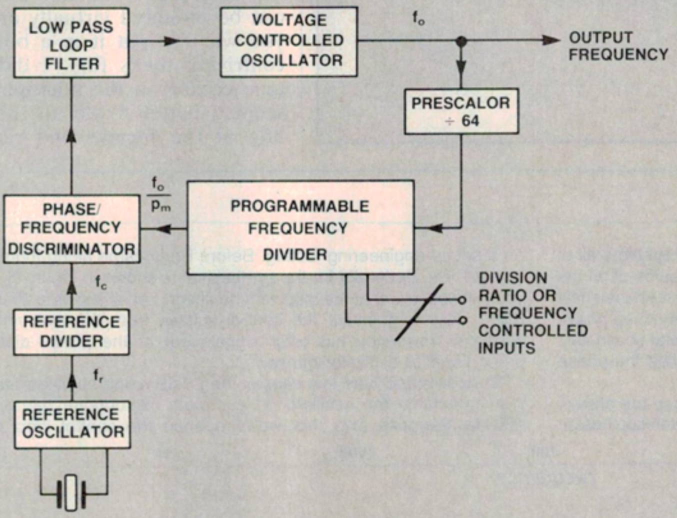


Figure A.

The next block, which is the heart of the synthesiser, is the **voltage controlled oscillator.** This is an oscillator whose frequency is set by an input control voltage and must be designed to cover the complete range of frequencies to be generated by the synthesiser. It must also have very good short term frequency stability as the PLL cannot correct short term frequency variations.

The voltage controlled oscillator output is fed to the next block of the synthesiser (as well as the rest of the transceiver) which is usually a high speed frequency divider called a **prescaler.** As the output frequency of the synthesiser may be many hundreds of megahertz (in our case about 100 MHz) it must be reduced to the order of kilohertz before it can be handled by the next block which is the **programmable divider** which actually determines the final output frequency.

The programmable divider takes the signal from the prescaler and further divides its frequency by a ratio which is digitally determined by digital control lines used to set the frequency. The output of the programmable divider output frequency is, say, too high then the discriminator output will go lower, and

This block, at least in the steady, locked state produces an output dc voltage which is proportional to the phase difference between its two input frequencies. The discriminator dc output is fed to a **loop filter** to remove any ac components then connected to the control input of the voltage controlled oscillator. If the programmable divider output frequency is, say, too high then the discriminator output will go lower and lower the voltage controlled oscillator frequency which, in turn, lowers the programmable divider output until the two frequencies are *exactly* the same.

To describe this in more detail we have already said that the reference oscillator frequency is f_r. If we say the reference divider divides by n then the comparison frequency is $f_c = f_r/n$. Similarly the voltage controlled oscillator is f_o(it's also the output frequency) and is divided first by the prescaler ratio p then by the programmable divider ratio m. Therefore the output frequency from the programmable divider is f_o/pm. The phase/frequency discriminator ensures that these two frequencies are exactly the same so we can write

$$\frac{f_r}{n} = \frac{f_o}{pm}$$

or rearranging things

$$f_o = \frac{pmf_r}{n}$$

or, putting things in words, the output frequency is eactly equal to the reference frequency multiplied by a fixed constant, p/n, and also multiplied by a digitally controlled variable number, m. As m is always an integral number (that is m never has fractions — that's the way digital dividers work) then the output frequency of the synthesiser can be stepped in frequency increments of pf_r/n and this is the channel spacing of the synthesiser.

To illustrate this more clearly take the example of the radio microphone. The reference frequency is 4.000 MHz and the fixed reference divider ratio used is 1024 so the comparison frequency is

$$\frac{4\,000\,000}{1024} \text{ or } 3906.25 \text{ Hz.}$$

The prescaler used has a division ratio of 64 (it's always easier i.e: cheaper to divide by 2^n where n is integral) so if we want an output frequency of 108.000 MHz then the programmable divider ratio must be set to 432. This gives a programmable divider output frequency of

$$\frac{108\,000\,000}{64 \times 432} = 3906.25 \text{ which is what is expected.}$$

If the division ratio of the programmable divider were to be changed to 431 then the phase discriminator would adjust the voltage controlled oscillator so the output frequency divided by 64 x 431 = 3906.25 Hz of f_o = 3906.25 x 64 x 431 = 107.750 MHz.

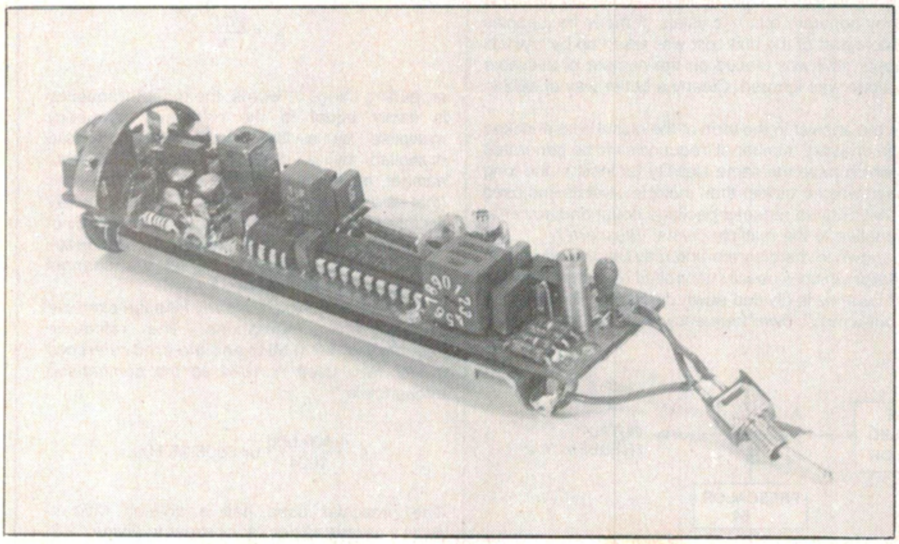
This shows that the channel spacing for the microphone synthesiser is 250 kHz which is the answer given from the formula.

system has come up fine. If, however, you can't get anything at all and the control voltage seems to be stuck to the positive rail then the oscillator probably isn't running. Check all the dc potentials around the circuit starting with the 1.2 volts at the top of the reference IC4. The emitters of Q2 and Q3 should be at 0.6 to 0.7 volts and the emitter of Q1 should be at about 2.5 volts.

Check that the crystal reference oscillator's running at 4 MHz by monitoring pin 4 of IC2 as pin 3 is a high impedance point and doesn't take kindly to load. If all this is correct then it's a pretty good bet the polarity of the inductor L1 is wrong and you'll have to reverse the secondary terminations. This should get things going but if you still have trouble check that the ends of the bypass capacitors C3, C17 and C23 are a good solid connection to the ground plane. The same applies to the tuning capacitor C13; if it's open circuit the oscillator will be a mile off frequency.

Final set-up

Once things are running OK you can paint the oscillator coil windings with lacquer. This should be done reasonably generously; if the coil windings can move around, the radio microphone will be microphonic in ways you don't want. If it's knocked the coil geometry will change slightly and change oscillator frequency. Use superglue ▶



to stick down the cup core over the lacquered coil and solder that can cover the lot. Realign L1 so the control voltage is 2.5 volts for channel 4 and try putting some tone on the audio input. It should take about -46 dBm or 4 mV to give full modulation which will be clearly audible on your receiver. On the prototype I found that the total harmonic distortion going through the microphone and a good (well — sort of) receiver was -47 dB or 0.45%. It was interesting to note that the THD got worse rapidly as I wound the deviation over the ±75 kHz, presumably as the receiver started cutting off sidebands. This type of modulator will quite happily give MHz of deviation so I assumed the bad things were happening in the receiver.

Mechanical

The board as assembled and tested so far can be mounted virtually anywhere and if you want to put it in a box of your own contriving that's fine. I didn't put a volume control in the microphone itself as it seemed better to do all the gain controlling at the receive end rather than risk

PHASE-LOCKED LOOPS

The phase-locked loop is only one of a whole class of control systems all of which have an essential requirement. That is that the sum of gains of all the blocks in the loop *must* be less than 0 dB when the sum of all the phase shifts reaches 360°. How much the loop gain is less than 0 dB when the loop phase shift reaches 360° is called the gain margin and how much the total phase shift is less than 360° when the total loop gain reaches 0 dB is called the phase margin.

When a phase-locked loop is designed the loop gain and phase are always plotted on a Bode plot, named after the bloke who really got control loop design on a sound engineering footing. Before Bode came along fudge factors were high art. The Bode plot for the synthesiser is shown in Figure B. The first thing you will notice is that the loop with no filtering at all shows a 20 dB per decade rolloff. This is because the control voltage from the discriminator controls oscillator *frequency* but error signals out of the phase discriminator are proportional to oscillator *phase*.

To understand how this causes the 20 dB rolloff suppose the audio signal that modulated the oscillator in our radio microphone had a frequency of 100 Hz. Suppose also that we've opened the control loop and magically

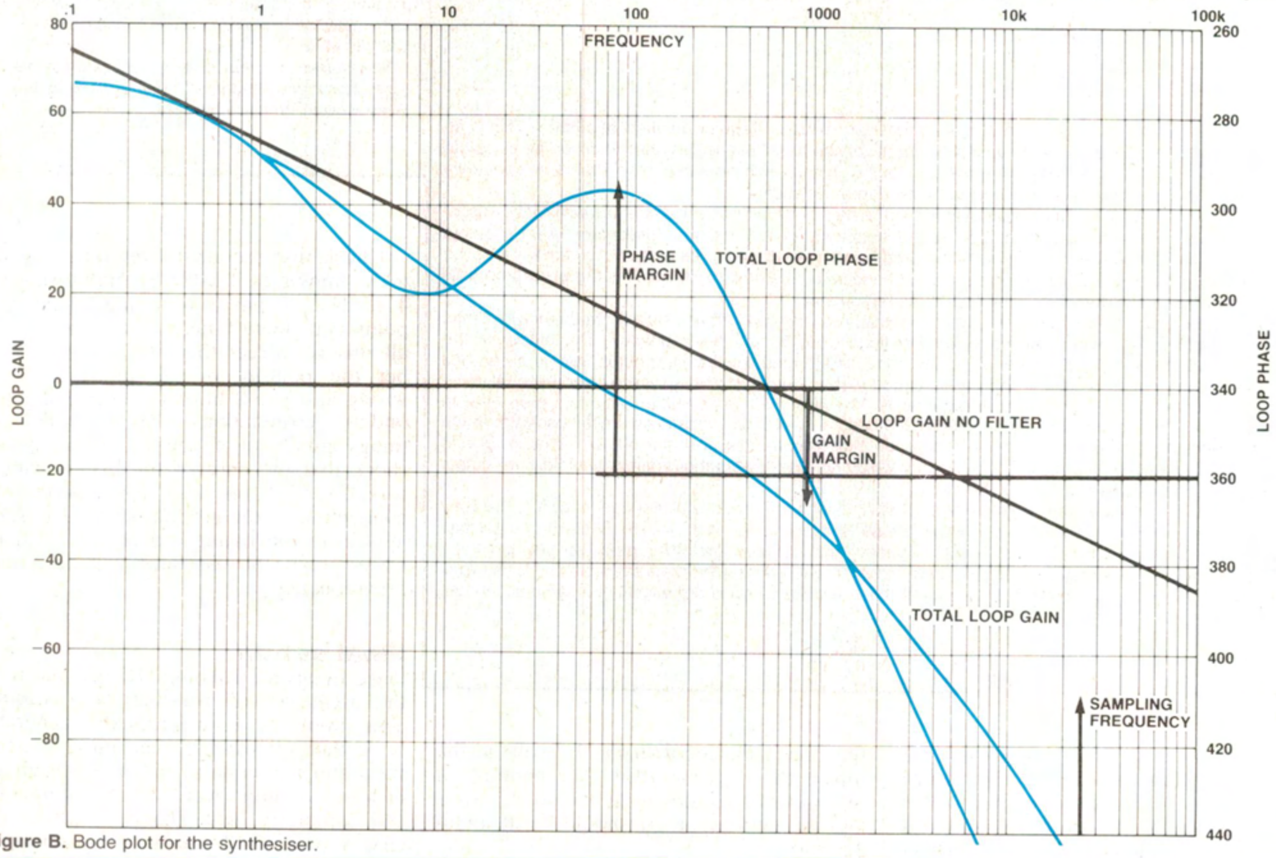


Figure B. Bode plot for the synthesiser.

having someone wind the modulation up to ±500 kHz and wonder why it sounds poorly. This makes it necessary to find out what sort of level comes out of the microphone you intend to use and adjust R1 and R2 to suit. If you do mount the transmitter in a box and want a gain control then R8 could be replaced with a pot to give some variation but it should be 1k or less to avoid causing the audio response to roll off too soon.

Since I'm a sucker for trouble I decided to mount the board in a tubular case so it could be used as the handle of a microphone. This meant that everything finished up absolutely jammed in but made for a nice neat unit. I'll describe how I did it but *be warned:* mounting the transmitter this way takes a fair bit of sheet metal work and you'll need some basic tools to do the job right. The very basic essentials are a good bench vice, drill press and selection of drills to go with it, assorted files, a 2.5 mm x 0.45 mm thread cutting tap, a multiple size hole saw and lots of patience.

The first thing to do is make the mounting bracket to mount the board in the tube. Cut and clean a strip of the sheet brass 5 mm wide and 50 mm long then bend it around a mandrel (see Figure 3) so it will fit exactly inside the copper tube. Next bend in the two mounting tabs so the distance from the outside of the mounting tabs to the bottom of the curve is exactly 14 mm (see Figure 4). Cut the two mounting tabs off and file them down so there is enough material to hold the mounting screws but not so much that the bracket fouls any components. Hold the bracket where it's finally to go on the board and mark where the mounting holes on the board go. Drill and tap the two mounting tabs with the 2.5 mm tap. The spacer that goes on the audio end of the board is also about 5 mm diameter and exactly 14 mm long drilled and tapped 2.5 mm.

Temporarily attach the bracket and spacer and try the assembly for fit in the copper tube. The first thing you'll see is that the channel select switch fouls the tube wall but this is correct — don't worry. To mount the board start by trimming one end of the tube off clean and square to become the antenna end of the final unit. Next mark a point on the tube

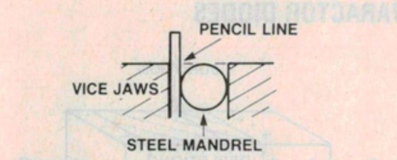


Figure 3. Bending the sheet brass around a mandrel.

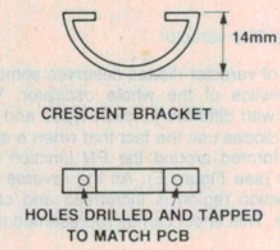


Figure 4. The bracket as it should end up.

arranged that the correct dc voltage is applied to the dc control input of the voltage controlled oscillator so it doesn't drift. The resultant output from the oscillator has 100 Hz frequency modulation with deviation of, say, ±1 kHz. The frequency deviation peaks will correspond exactly to the peaks in the modulating signal. The phase error as seen by the phase discriminator is not so straightforward however.

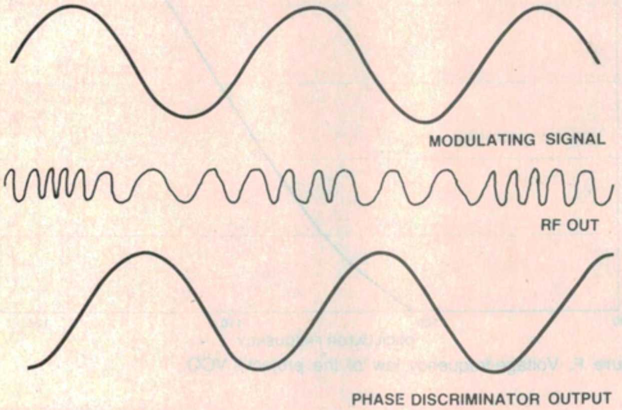


Figure C. The frequency deviation peaks correspond with the peaks in the modulating signal.

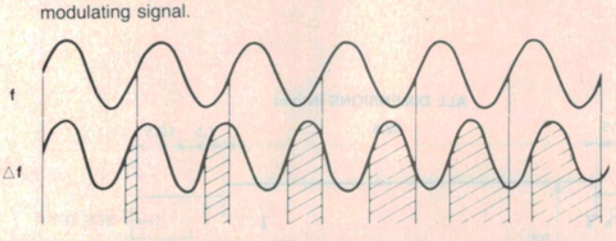


Figure D. Showing how two slightly different frequencies have a phase difference that increases linearly with time.

Herein lies the nub of the peculiar behaviour. If two signals have frequencies that differ slightly by an exact fixed amount the phase difference between them increases linearly with time. This effect is shown in Figure D where two sine waves with fractionally different frequencies are drawn. The *phase error* is given by:

$$\phi = \int f \, dt$$

in our example, with a signal modulated by a 100 Hz tone, the phase error is zero when the modulating signal passes through zero but for all the time that the modulating signal is positive the phase error is adding up and causes the output of the phase discriminator to continue going positive. This means that when the modulating tone passes through zero going negative the phase discriminator output is at its peak positive point. So the value of the discriminator swing is proportional to *the area under the positive swing of the modulating signal.* This causes two effects. The first is that the peaks of the phase discriminator output correspond to the zero crossings. This corresponds to an inherent 90° phase shift between the input phase error signal and the phase discriminator output.

The second effect is shown by considering what happens if the modulating signal frequency is increased to 200 Hz without altering its amplitude. As the discriminator output is proportional to the *area* under the excursions of the input error signal, the peak output will be only half the peak for 100 Hz.

All this rather complicated description says is that the phase discriminator output is 90° phase shifted and shows a 20 dB per decade rolloff with frequency — just what is shown on the Bode plot! It may seem rather baffling but if you don't allow for it in the design your circuit won't work!

The final upshot of this discussion is that at zero frequency a phase-locked loop has infinite gain (really!). But at high frequencies the loop gain naturally rolls off and in many cases no loop filter is necessary for stability. This is such a case. I used a loop filter, but only to remove the crud from the discriminator output.

We must now look at the problem of phase shifts around the loop. The first phase shift must be 180° because we want any disturbance in the loop to be corrected by a signal in the opposite direction. The next inherent phase shift in the loop is the frequency-to-phase 90° shift I just described.

One of the basic rules of this game is any filter whose frequency response falls with rising frequency shows a phase shift the wrong way for our case (Murphy again) so if we want to include a loop filter it must roll off very slowly with frequency so as not to give too much phase shift (the faster the rolloff the larger the phase shift during the rolloff).

Hence the loop filter has only one simple RC stage at low frequencies with a second resistor (R10) in series with the capacitor to ensure that the phase shift in the loop never gets too near 90°. As the loop gain nears the critical 0 dB point most of the roll-off is removed to give plenty of phase margin. As soon as the loop gain is comfortably below 0 dB we can do anything we want to the phase so C8 is placed across R10 and a second stage, R11 and C5, is added to attenuate the discriminator nasties. Both the total loop gain and loop phase shift are shown on the Bode plot to show how they are optimised.

The crystal reference is a normal parallel mode 4.000 MHz device. The crystal is packaged in a small can to keep things compact. As it has to resonate with a parallel 30 pF capacitance both sides really need 60 pF to ground but you can vary this a bit as it isn't absolutely necessary to have the reference right on 4.000 000 MHz. The main thing the crystal provides in this case is a dead stable reference frequency with temperature and voltage changes having no effect (almost). The trimmer is provided as a nicety as I like to see the output frequency come up on the counter as say 105.750 000 MHz.

VARACTOR DIODES

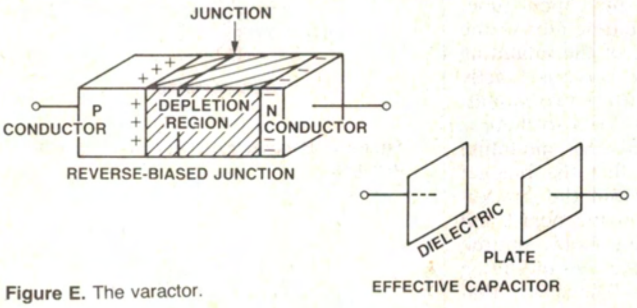


Figure E. The varactor.

The use of varactor diodes deserves some discussion as this device sets the characteristics of the whole oscillator. Varactor diodes come in several varieties with different junction types and hence different characteristics. All varactor diodes use the fact that when a diode is reverse biassed a depletion layer is formed around the PN junction which acts as the dielectric for a capacitor (see Figure E). As the reverse bias is increased the thickness of this depletion region is increased and causes the junction capacitance to decrease. This effect can be represented mathematically in a rather simplified form by

$$C_d = \frac{C_O}{(V+\phi)^{\gamma}}$$

where ϕ is normally about 0.7 V at room temperature, C_O is a function of the diode geometry and assorted other parameters. It's constant for any diode. γ is dependent on the dopant profile in the diode. γ is really the most important parameter in designing voltage controlled oscillators as it determines the way the diode capacitance changes with voltage and hence the way the oscillator frequency changes with voltage.

By far the easiest type of diode to make is the abrupt junction diode where γ is equal to about 0.5. Most of the varactors you can buy are of this sort. However for some applications a diode with a tuning range greater than that given by a γ of 0.5 is needed and for this special diodes are used called hyperabrupt junction diodes (now there's a really impressive term — try laying that one on your non-technical friends!). The diodes are fabricated using special epitaxial growth or ion implantation techniques and show the peculiar property that the dopant concentration increases the nearer you are to the actual junction. In this way diodes with γ of 1 or 2 can be made.

"So what!" did I hear you say? Well all this has a point. The frequency of oscillation of a tuned circuit is given by the recipe:

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

Now if we could find a varactor whose capacitance is proportional to $1/V^2$ then the resonant frequency of a tuned circuit using such a diode would be given by

$$C \propto \frac{1}{V^2}$$

and therefore

$$f_O \propto \frac{V}{\sqrt{L}}$$

or the frequency is directly proportional to the varactor bias voltage — a very nice thing to have. Usually the γ of a diode will not be exactly 2 but in general the higher the γ of the varactor diode the more linear the voltage-frequency characteristics of a tuned circuit using it. This means that oscillators using such a varactor as the tuning control element will have a gain that's constant (more or less) with bias voltage (the gain of a voltage controlled oscillator is the change in frequency divided by the change in voltage required to produce the frequency change or

$$A_{vco} = \frac{\Delta f}{\Delta V}$$

and for our case is given in MHz/V). If an abrupt juntion varactor was used in a voltage controlled oscillator then the frequency would change very fast with voltage for low biases. But as the bias increased the rate of change would slow. We could probably live with this and design the phase-locked loop to allow for the gain variations, but the major stumbling block would be that the same modulating voltage is being applied to the varactor to give frequency modulation no matter what dc bias is being applied to it. *This* means the radio microphone would have different modulation deviation as the channel was changed. Using a hyperabrupt junction diode avoids both problems and gives nearer constant gain and modulation. You do lose a little in circuit Q with hyperabrupt junction diodes but the advantages far outweigh the problems.

Diodes are made just to suit operation in the 88-108 MHz band by Motorola and I chose one of these to use in the oscillator. It has a γ of about 1 so the voltage/frequency law of the final oscillator isn't perfectly linear but is quite good enough (see Figure F). As the oscillator doesn't have to span the full 88 to 108 MHz but only about 4 MHz the actual varactor diode is padded out with a series capacitor C12 and a parallel capacitor C13. This reduces the total frequency range but increases the tuned circuit Q.

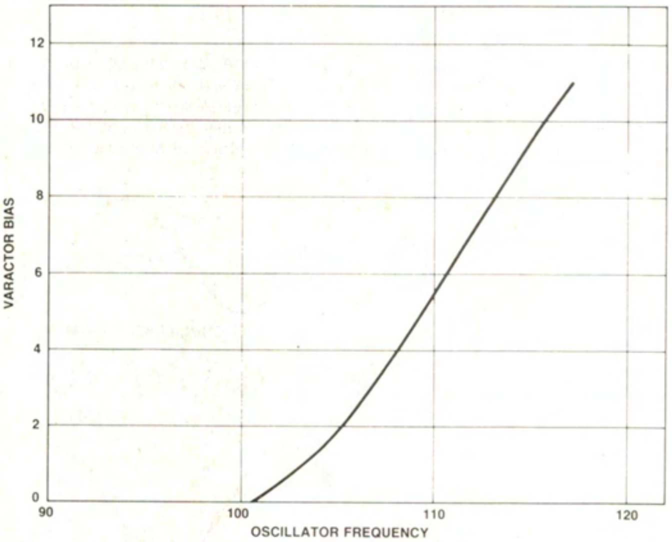


Figure F. Voltage/frequency law of the project's VCO.

where the spacer should go so that if the board is mounted from this screw the transmitter end of the board will be 7 mm from the end of the tube. Drill a hole to clear a 2.5 mm screw and countersink it as deep as possible. Don't cut right through the tube though. Next hold the board against the outside of the tube and mark off where the channel select switch will go.

When you're satisfied with the switch hole, screw a 2.5 mm diameter x 5 mm long countersink head screw into the spacer to hold the board in place in the tube. With a thin piece of rod of some sort measure exactly the distance from the antenna end of the tube to the crescent bracket with the board mounted in the tube. With the board still mounted in the tube drill two holes through both the tube and the mounting bracket. When drilling

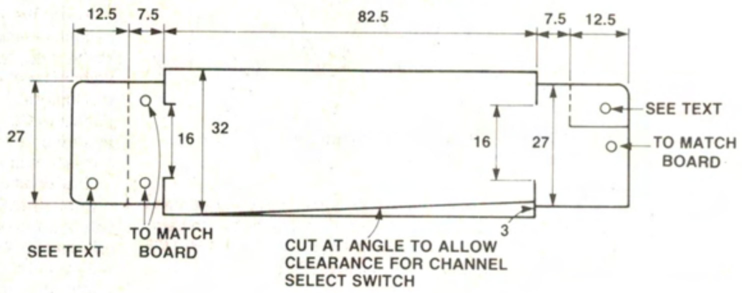


Figure 5. Battery holder.

through the bracket it's necessary to hold it hard up against the wall of the tube. Remove the board from the tube, tap the two holes in the bracket, open out the holes in the tube to 2.5 mm and countersink them as before and the board mounting in the tube is complete.

Battery holder

The battery holder is the next part to be made and is kind of fiddly so I hope you've got lots of patience. The batteries used are AAA cells that fit (just) in the free space beside the board in the tube. The terminals and contact springs can be salvaged from the cheap plastic AA cell holders you can buy from most electronics suppliers. Using the same sheet brass as before or perhaps a little thinner cut out the shape shown in Figure 5. Using the board as a template mark and drill the three mounting holes to clear 2.5 mm screws and make sure it will fit OK. Bend up the end with the two holes where the sketch shows a dotted line *sharply at right angle.* To get a nice sharp bend hold the sheet in a vice so the point to be bent lies exactly on the edge of the vice jaw and clamp it *hard*. Hammer the bit that protrudes down against the vice jaw and you get a good clean bend! Next bend up half of the other end the same way and the two battery terminal mountings are done.

To stop the batteries rattling around in the final unit I bent up a piece of mild steel around the mandrel in a nice even curve. Try placing four AAA batteries in the channel you've made and if it's right they should fit snugly with almost no free movement at all. If they won't lie down flat against the bottom of the channel the sides have been bent up too sharply and you can fix it by opening the sides out a bit. If they are very loose the battery holder may not fit in the tube and it's necessary to repeat the bending process with the mandrel moved down a bit.

The next thing to be done is to attach spacers to the bottom of the battery holder so it clears the cut off leads on the board. First work all over the board cutting *all* the leads off as short as possible. Next find three small brass nuts no thicker than 2 mm and no wider than 5 mm in any dimension. Align these nuts over the three holes in the battery holder on the back of the holder and solder them in place. The nuts must be exactly in line with the holes already drilled in the sheet brass or when everything's assembled the nuts may short out tracks. Finally dress down the surfaces of the nuts that touch the board with a file so the thickness of the nuts *plus* the sheet brass is 2.5 mm. Any more and there's a risk the batteries won't fit. When all this is done try mounting the completed metalwork on the board with 2.5 mm diameter x 8 mm long countersink head screws and try the assembly for size in the tube.

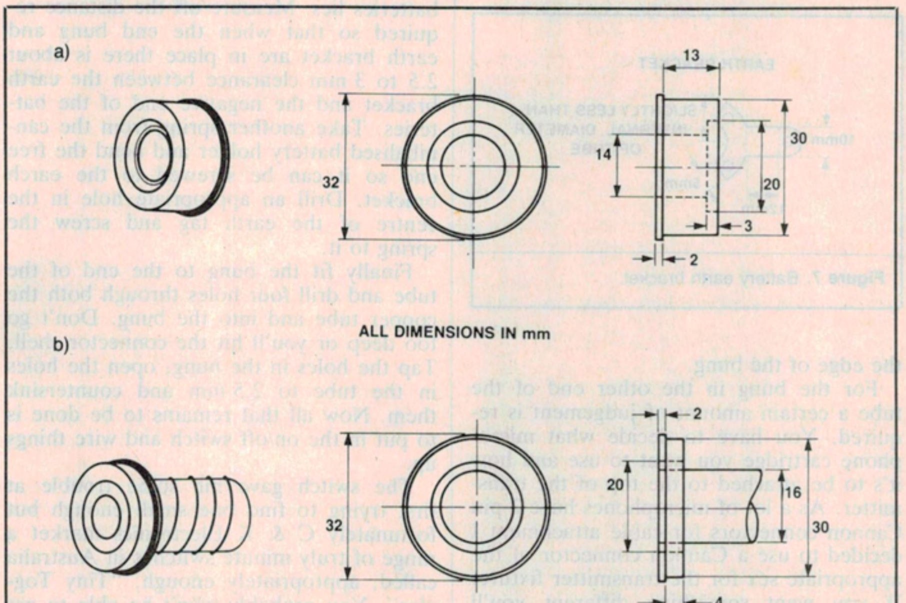


Figure 6. Making the end plugs. I cut these to suit the tube used to house the project. The top one shows the plug at the microphone end. I fitted an XLR socket into this. The lower one takes the plastic tube that carries the helical antenna.

The channel select switch needs to be eased into place and to give room to do this the side of the battery holder is deliberately slightly cut away. If the terminal ends of the battery holder foul the tube they should be trimmed away until they fit but don't get carried away — take off small bits and try again until it's right.

Battery terminals

The battery terminals are made up from 3 mm pan head screws for the positive terminals and springs salvaged from the commercial holder for the negative terminals. Mount the 3 mm screws in one end of the battery holder using a plastic insulating washer from a T0220 transistor mounting kit. A solder tag should be placed under the nut to allow connection.

The other end of the battery holder needs a piece of printed circuit board to act as a spacer. Cut out a piece so it fits exactly over the end of the whole bent up end and just clears the two mounting screws. The board should be singled sided copper laminate with the copper side away from the brass. Make a positive terminal diagonally opposite the positive terminal you've just made. Then take one of the springs that will act as negative terminal and bend the bottom end until it forms a neat loop to go under the screw. The spring should line up with the end of the battery.

Antennas

The two ends of the case are made from sheets of 6.4 mm thick plastic. Make up the antenna end of the microphone first by cutting out a disc of sheet plastic with a one and one half inch hole saw. Don't cut too fast or you'll melt the plastic and mess it up. Cut and file until you have a disc the same diameter as the outside of the tube. Then, leaving a 1 mm wide rim, cut away the rest of the edge of the disc until it just fits into the tube. This gives you a neat bung that closes off the end of the tube (see Figure 6).

Next cut a 22 mm diameter hole in the centre of the disc. This hole should be an interference fit with a 22 mm conduit (that is it shouldn't be a sloppy fit but rather you should have to push *hard* to get it in). Using Araldite, glue in a piece of conduit exactly 150 mm long so when the bung is in the end of the tube the conduit sticks out 144 mm to form the antenna.

To make the actual antenna you'll need about 1.5 m of 1 mm diameter copper wire. Drill holes at either end of the conduit. These two holes are to terminate the anntenna wire. The antenna needs 19.5 turns to operate correctly at 106 MHz. If you've decided to use a different part of the 88 to 108 MHz band refer to Table 1 to see the correct number of turns.

Once the wire helix is tight and even with 19½ turns (count them again!) slip a piece of heat shrink tubing over it and gently heat it over a gas torch or stove. Be patient here as if you try to rush it you'll burn the tubing and have to try again. Finally trim off the wire at the outer end of the helix and the antenna's finished. To fit it to the body of the microphone fit it to the end of the copper tube and drill three holes through both the copper tube and

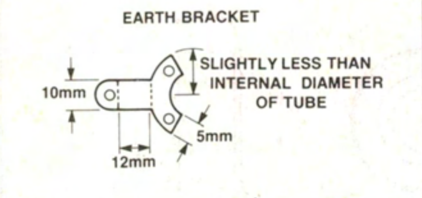


Figure 7. Battery earth bracket.

the edge of the bung.

For the bung in the other end of the tube a certain amount of judgement is required. You have to decide what microphone cartridge you want to use and how it's to be attached to the top of the transmitter. As a lot of microphones have 3 pin Cannon connectors for cable attachment I decided to use a Cannon connector of the appropriate sex for the transmitter fixture. If you want something different you'll have to ad lib a bit from here on.

I started by gluing two bits of the plastic sheet I used for the bottom bung together to make a 12 mm thick sheet. Then I made another bung exactly the same as before. I cut a hole in it to accept the outer shell of the connector. The shell was cut off so that when the cut end was flush with the inner end of the bung the screw that holds the connector in the shell was just free of the bung. The cut off shell was then Araldited in place. The much thicker bung ensured that the connector would hold the heaviest microphone. Testing was done with an AKG D330BT microphone which is a great lump of a thing and plenty of strength seemed a good idea.

Earth clamp

The final part to be made is the earth clamp and connection for the battery holder. Cut out a piece of sheet brass according to the drawing given (Figure 7) and bend it sharply at right angles where the dotted lines show. Drill and tap 2.5 mm threads in the inner end of the top end bung so that when the bung is inserted the bent over tag is next to the earth end of the batteries. Before the bung is fitted to the tube the tube must be trimmed off to the correct length. With all the works in the tube and the four AAA cells in place carefully measure where the end of the batteries lies. Measure off the distance required so that when the end bung and earth bracket are in place there is about 2.5 to 3 mm clearance between the earth bracket and the negative end of the batteries. Take another spring from the cannibalised battery holder and bend the free end so it can be screwed to the earch bracket. Drill an appropriate hole in the centre of the earth tag and screw the spring to it.

Finally fit the bung to the end of the tube and drill four holes through both the copper tube and into the bung. Don't go too deep or you'll hit the connector shell. Tap the holes in the bung, open the holes in the tube to 2.5 mm and countersink them. Now all that remains to be done is to put in the on/off switch and wire things up.

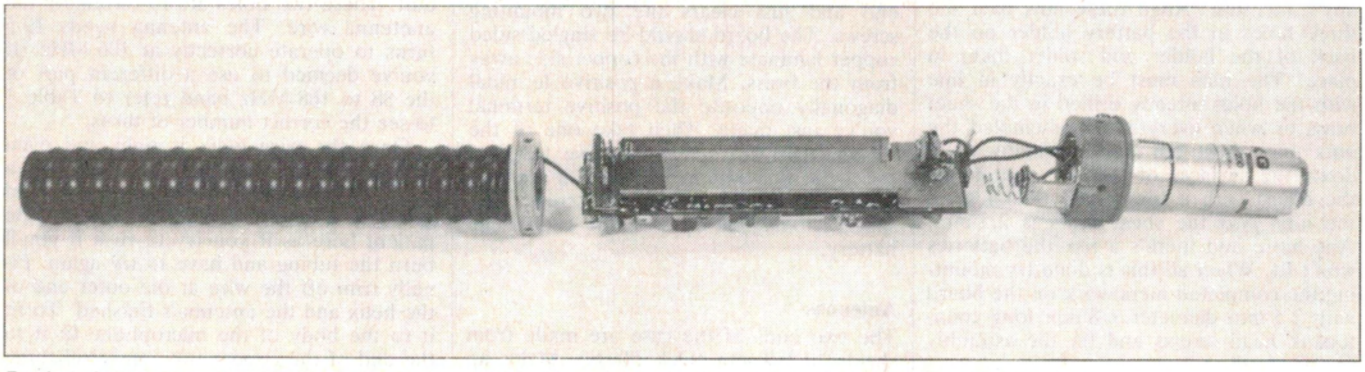
The switch gave me some trouble at first trying to find one small enough but fortunately C & K Electronics market a range of truly minute switches in Australia called, appropriately enough, "Tiny Toggles". You probably won't be able to get them from most of the distributors but C & K themselves at Harris Park in Sydney or any of the C & K distributors will be able to help you. I was able to get just a few and their policy is to help anyone at all with no minimum order size (nice one C & K!). Even though the switch is small I still had to trim off the terminals before it would fit. The switch mounts in a 4.8 mm (3/16″) hole so to mount it all that's necessary is to drill a hole through the copper tube next to the battery positive terminal and screw in the switch. After the switch is in, attach insulated wires to the mic input pads, the positive and negative pads and the antenna pad on the board and also a lead to the positive terminal on the battery holder. Screw it in place in the tube and trim and connect the positive lead from the battery holder to the centre terminal of the switch. Do the same for the positive lead from the board to the upper terminal of the switch. The antenna lead can be cut short and soldered to the tinned end of the antenna and the whole antenna screwed into position. Connect the earth pin of the microphone socket to the earth bracket and the earth bracket to the earth lead from the board using as short a wire as you can. Trim off the two mic leads and solder them to the connector pins. Finally insert the four batteries and screw on the top bung with four screws and the radio microphone's ready to test.

Testing

I found that there's a free space in Sydney that corresponds to channel 8 or 9 on the microphone at about 106 to 106.5 MHz. Tune your FM receiver to this space or, if it isn't free where you live, find a free space and tune to it and adjust the transmitter to that frequency. Turn on the project with a microphone plugged in and you should see the signal strength meter or tuning indicator slam over against the stops. You're only transmitting 25 to 30 mW but you're a hell of a lot closer than the FM stations. In fact, if you inadvertently tune to the same frequency as an FM station you'll completely swamp it out but DON'T DO IT DELIBERATELY; you'll annoy the hell out of the neighbours. Just talk into the mic and listen to see if it sounds OK.

To test for range, arrange for someone to listen to the radio and walk down the street turning the transmitter on and trying it occasionally. The person listening can wave to you if they can hear you. I found that I could get about 400 metres range before the signal got too weak. If you want you can drill a hole in the transmitter over L2 and adjust the inductor to optimise transmitter power but I found it hard to do as the transmitter always gave out heaps of power and swamped the receiver.

Once everything seems to be going as desired you may like to embellish the transmitter by having the copper body plated either with nickel or chrome (or if you want a *really* class job, gold!) and it's all finished. Just a final warning though: don't ever use the transmitter on the same frequency as a commercial or community station. You've got a choice of 10 frequencies while they have but one. Do the right thing and find a free channel — that way you don't bother them and they probably won't bother you. If you do interfere with boadcasting the Department of Communications MUST act to find you and at the very least confiscate the results of all your labours! If you use it wisely nobody will be bothered. ●



Ready to house. The project showing battery holder and mounting bracket and pillar, ready to slip in the tube used to house it.