Radio Electronics
A General Introduction

FOREWORD

The following is by no means an introduction to electronics, there are many such books that cover the subject, but intends to explore some of the ideas and concept involved in radio broadcasting that are relevant to the pirate radio operator on VHF FM. In particular we will go a step by step tour of a typical VHF FM transmitter system starting with the output from the tape recorder or mixer, and finishing with a brief discussion of aerials. At each stage we will discuss the pros and cons of various alternatives and additional background info, e.g. the use of equipment will be introduced.

Radio frequency signals have AMPLITUDE and FREQUENCY. The frequency is how fast the signal is oscillating from one extreme to the other and back again. Frequency is measured in cycles per second (cp/s), which these days are known as HERTZ (Hz), 1000 Hz = 1 kHz, 1000000 Hz = 1 MHz. The amplitude is to what extent the signal is oscillating. LEVEL or STRENGTH can be thought of as meaning the same as amplitude. Amplitude can be measured in Volts (V). There is more than one way of measuring amplitude.

INTRODUCTION

What we are trying to is get information from one place to lots of other. I'm using information here in a wider sense, meaning speech, music, etc., rather than phone numbers local hairdressers or whatever. Now I'm going to assume we're going to use radio broadcasting to achieve this, which immediately rules out things like standing on top of tall buildings and shouting out really loud. We'll also assume we've got this info in the form of an audio frequency signal, i.e. what comes out of a tape recorder or an audio mixer. You can't transmit audio frequency signals very easily so what we can do is import the info in the audio frequency signal onto a higher frequency carrier signal. Two ways of doing this are AMPLITUDE MODULATION and FREQUENCY MODULATION (AM and FM).

In AM the amplitude of the carrier is determined at every instant by the amplitude of the audio signal, the carrier frequency remains constant. In FM the frequency of the carrier is determined at every instant by the amplitude of the audio signal, and the carrier amplitude remains constant.

Frequencies between 30 MHz and 300 MHz are known as Very High Frequencies or VHF. This corresponds to wavelengths between 10 m and 1 m. To convert between wavelength and frequency use the formula: $wavelength(metres) = \frac{300}{frequency(MHz)}$.

FM

There are two sorts of FM, known as Narrow Band FM (NBFM) and Wideband FM. They differ by the maximum allowable frequency shift of the carrier when the transmitter is fully modulated. This frequency shift is known as the DEVIATION. Legal CB radios use NBFM with a maximum deviation of 3 kHz. Wideband FM is used by the national broadcasting companies for radio broadcasting and for studio to transmitter links. The standard maximum deviation for FM radio broadcasting in Europe is 75 kHz. There is no simple way to set the deviation of a transmitter without a deviation meter which is an expensive piece of test gear. Probably the best way to do this is to vary the level of the audio signal going into the transmitter (TX) and listen on a receiver, until your signal sounds about the same loudness as the other (legal?) broadcasting stations. If you use too high a deviation you'll use a bigger than necessary chunk of the radio spectrum and be more likely to cause interference with others, which will make you even more unpopular with the DTI.

The police use NBFM as well, which is why if you listen to them on an ordinary FM receiver, which is wideband, you can hear more than one channel at a time.

CHOOSING A FREQUENCY

If your first action could be to reach for your receiver and tune trough looking for a blank space, think again, for a kick-off the FM broadcast band is 88 to 108 MHz. What stations you can receive is determined by where you are, as well as by the nature of and positioning of your aerial. If you look our old friend the Maplin catalogue we find on P24 of the '88 issue a list of the frequencies and

locations of all FM broadcasting stations. What it doesn't say, of course, is the frequency of existing pirates. TX Magazine gives a good rundown of these. Armed with this info you should make a list of all frequencies in use in, say, a 50 km radius. If you write to the BBC or IBA's Engineering Info Offices they'll send you service maps of where their TX's are meant to be able to heard. Then its just a question of finding a big enough gap between stations, with the proviso that your station shouldn't be nearer than 200 kHz (0.2 MHz) to the frequency of any existing station. This is no problem as the band is half empty. Also don't choose a frequency which is 10.7 MHz away from any other station as for complex reasons (which involve the use of 10.7 MHz as intermediate frequency in FM receivers) reception will be hard for people listening to you and/or the other station.
Now let's take a little stroll through the whole system.

TAPE OR LIVE
What we are going to feed to our TX? The obvious possibilities are:
A) A tape or cassette player.
B) Live, either directly from the mixer or via some kind of link from studio to TX site (highly recommended).

TAPE. This is the safest approach in that you can put a tape on and then retire to a safe distance. Links are now being traced and studios busted, and some of the biggest pirates (e.g. the LWR) are going back to taped broadcasts. If the DTI trace your transmission and turn up all they can do to confiscate your tape player, TX and aerial, i.e. no arrests (unless they catch you changing the tape). Its also the most inflexible alternative as tapes will have to be prepared in advance. Time checks, if you're into that, will be difficult and live phone ins are right out.
Give a little thought to your choice of tape recorder, as it will probably be the weakest link in terms of sound quality. In an old clapped out one the heads will be worn flat. Maybe you can use a 'Walkman' type of player, which are small, can be battery powered and have a OK sound quality and are cheap. An amateur radio rally I was at recently were selling off very slightly damaged ones for £2 each. To reduce 'noise' or 'tape hiss' on such recorders, if you're doing programmes with quiet passages, you can use a circuit known as a Dynamic Noise Limiter (DNL), which is placed on the output and cuts off the 'noise' just in quiet pauses. DNLs are sometimes used in the soundtracks of old films. You can find a DNL circuit in part of the 'Audio Embellisher' project in the Jan. 84 issue of 'Elektor' magazine.
If you want to go upmarket you could use a proper 1/4" reel to tape recorder, though few pirates do. The latest and greatest is to use 'Stack machines' which will change the tapes for you. Whatever you use get one that can be battery powered as you may not always have access to mains power.

MONO OR STEREO
The advantages of mono are that the TX is kept as simple and cheap as possible, and you don't need as much power as on stereo to get same result. The disadvantages are you don't sound as professional, quite small pirates are now using Stereo Encoders, and maybe people might dial past when the red stereo light on their receivers doesn't flash. With stereo the listener can get quality the same of legal stations. Weigh against this is the extra cost, extra circuitry and more output power needed for the same signal.
What you need is a STEREO ENCODER, which combines the left and right stereo signals into a single composite stereo signal which is then fed into your TX.
For those interested a brief description follows. The left (L) and right (R) signals are fed into a summing and differential amp to get a L+R and L-R signal respectively. The L-R signal is mixed in a balanced modulator with a 38 kHz sub carrier to produce an amplitude modulated double sideband suppressed carrier signal. The 38 kHz signal is derived from the same source as the 19 kHz pilot tone. The composite output is formed by mixing the L+R signal, the sidebands containing the info of the L-R signal, and a bit of 19 kHz pilot tone. The pilot tone switches on the STEREO DECODER in peoples' receivers.
Back in the receiver, once the stereo decoder has extracted the L+R and L-R signal the original left and right signals are easily got by

$$(L+R)+(L-R)=2L$$
$$(L+R)-(L-R)=2R.$$

The reason L+R and L-R signals are encoded rather than L and R is so that a mono receiver can just demodulate the L+R bit and ignore the rest of the signal. If L and R signals were encoded a mono receiver would only be able to hear the left channel. The 19 kHz pilot tone is usually got from a crystal oscillator, to be quite accurate and stable. A crystal resonating on 4.8640 MHz is convenient as 4864 divided by 2 eight times is 19. This can easily be done by digital logic chips, but its highly unlikely that you'll be able to buy a 4.8640 crystal off the shelf, so you'll have to have one made for order.

It doesn't matter if you didn't understand all of the above but one thing is important. The standard FM broadcast audio bandwidth extends only to 15 kHz and stereo encoders are designed to assume this figure. If you put signals into them with frequencies above that the L+R signal and the lower side band of the L-R signal could spread into each other and you will get a right bloody mess. With a tape recorder you can't really get over 15 kHz, but if you're live its quite possible. In that case you need a LOW PASS FILTER on each input to a stereo encoder. Maplin have a high quality design on page 243 in summer 86 issue. The pot could be replaced with a 500k resistor to wire the circuit permanently for max. roll off. If you're using a link between studio and TX and you want stereo you'll have to know the bandwidth of the link. If its 53 kHz (=38+15) or more you can use it after the encoder. Otherwise you'll need two links and have to encode at the TX end.

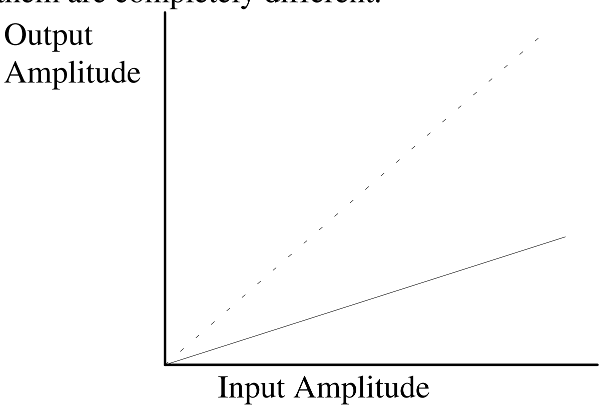
PRE-EMPHASIS

In a typical audio signal the high frequency sounds have less energy than the low ones and so produce less deviation of the carrier. This in turn makes them susceptible to noise when received. To avoid this high frequencies are boosted before being transmitted by PRE-EMPHASIS. In the receiver the frequencies are cut by the same amount by DE-EMPHASIS. So the overall frequency response of TX to receiver stays flat, but the level of background noise is reduced a lot.

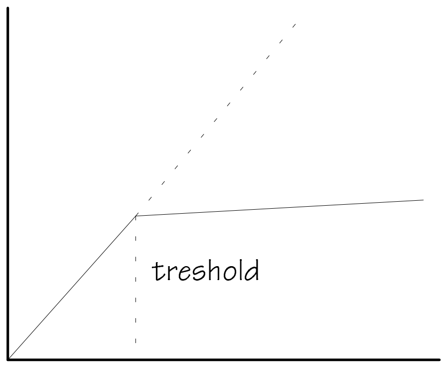
Pre- and de-emphasis networks are characterised by their TIME CONSTANT. In the USA the standard is 75 us, but in UK its 50 us so anything designed or bought from there needs slight modification. In a mono TX the pre-emphasis network can be built into the front end of the exciter. For a stereo TX such a network must not be in the exciter or it'll play hell with the composite stereo signal from the encoder. Instead you need 2 networks, one for each channel, on the inputs of the stereo encoder. They're actually often built into the studio encoder.

COMPRESSORS AND LIMITERS

Compressors and limiters operate on the same principles, but their effects and the reasons for using them are completely different.



COMPRESSOR



LIMITER

A compressor compresses, it reduces the DYNAMIC RANGE of its input signal. This means as the input amplitude varies over a certain range, the output amplitude varies only a fraction of that range. The graph shows a 2:1 compression characteristic. In this case with every change in the input amplitude the output changes only half as much. The dotted line shows a 1:1 non compressed characteristic.

A limiter passes its signal unaffected till the input amplitude reaches its THRESHOLD. At this point the limiter prevents the output increasing much by compressing its input much more strongly than in compressors e.g. 10:1.

Some American music stations and some pirates compress their programmes to make it seem louder and more upfront than other stations. This occurs cos the compressor keeps the average level of the signal high, even in quiet parts of the prog. The flip side of this is listeners can soon get 'listener fatigue' as constant compression can become boring and irritating to the ear, as if the music were rammed into it!
Compression has other uses, you might compress your programme as you transfer it to tape to stop quieter bits fading into background tape hiss when played. The process of recording and playing does this to some extent anyway. Don't compress the output of a tape recorder as it'll make tape noise worse. Guitar effect units, labelled compressors, are unlikely to be much use. Compressors intended for use in home studio recording are worth experimenting with. A stereo compressor with a 2:1 characteristic can be simply constructed around a NE571 IC.
Limiters are used to stop a signal's amplitude going over a certain level. E.g. when cutting a master disc in record manufacture, large PA systems at gigs to stop loudspeakers blowing every time someone burps in a mike and, surprise surprise, in broadcasting. In FM particularly, as the signal level increases so also does the bandwidth of the transmitted signal, risking interfering with other stations. With tape input to the TX its different the output is inherently limited by the recording process, no limiter needed. With live input to the TX its different. Though you might set the levels right to start, along comes a loud record or voice and you could be interfering with the next station. Use a limiter.
Any limiters based on 2 back to back diodes is a little more than a guitar fuzz box and will sound like one. A suitable high quality limiter was described in the May 83 issue of 'Electronics Today' International Magazine.

THE OSCILLATOR

At the heart of everything is the OSCILLATOR that generates the VHF signal. The frequency of this is modulated by applying an audio signal to it. The most common way of doing this is using one or two VARICAP diodes. When a varicap diode is operated with a reverse bias the capacitance of the diode varies with that bias. The diode(s) is/are connected to a frequency determining part of the oscillator. The audio signal is connected across the diode to achieve frequency modulation. Also by varying the DC reverse bias the oscillator can be fine tuned. The higher the voltage, the lower the capacitance, the higher the frequency.
The VHF signal can be generated directly, or the oscillator can oscillate on a lower frequency e.g. a third or half that desired and then followed by a TRIPLER or DOUBLER stage. There are three main types of oscillator:

a) Variable Frequency Oscillator (VFO)
b) Crystal Oscillator
c) Phase Locked Loop oscillator (PLL)

VFO's

These are simple oscillators which can be built round a single transistor. This can be a Bipolar Junction Transistor (BJT) or a Field Effect Transistor (FET).
The problem with oscillators based on BJT's is that the frequency is too dependent on the temperature of the transistor. i.e. a few degrees temperature change will result a significant change in transmitting frequency. For this reason oscillators based on BJT's are UNSUITABLE for serious use as a TX. FET's don't suffer from this problem so badly, so they can be used, but you should still bear it in mind.
The FET's will heat itself up slightly, and other bits of the TX, like the power amps, will be fair old chucking heat out, and are usually built into the same case as the oscillator. The frequency will drift most when the TX is first switched on as all the components will be at the same temperature as the air outside the TX's case, this is known as the AMBIENT TEMPERATURE. After the TX is turned on the heat from the amps will warm the air in the case directly or indirectly. As the FET warms the frequency will drift a bit. When heat loss equals heat gain you get THERMAL EQUILIBRIUM and it won't drift more. Keep your TX out of drafts to avoid messing this up. If you have a frequency counter plug it in to a dummy load and see how long it takes for the frequency displayed to settle down, maybe about 15 minutes. If you have time you can arrive at the TX site early and run your

TX for the warm up time with no input to a dummy load. This avoids listeners who tune in immediately having to retune as your frequency drifts.

CRYSTAL OSCILLATORS

This is also simple oscillator but incorporates a crystal into the frequency determining network. There are various types of crystal (fundamental, 3rd overtone, 5th overtone etc.) and various ways of using them (series mode, parallel mode) but their basic properties are the same. They're resonant on one frequency which is determined by the crystal's characteristics when made. This is their problem, whereas a VFO's are not very stable crystal oscillators are too bloody stable and it's a job to get enough deviation. You'll probably lose the higher frequencies of your programme and stereo is right out. Also chances are you'll have to get a crystal made order for your desired frequency so if you want to change it you'll need a new one.

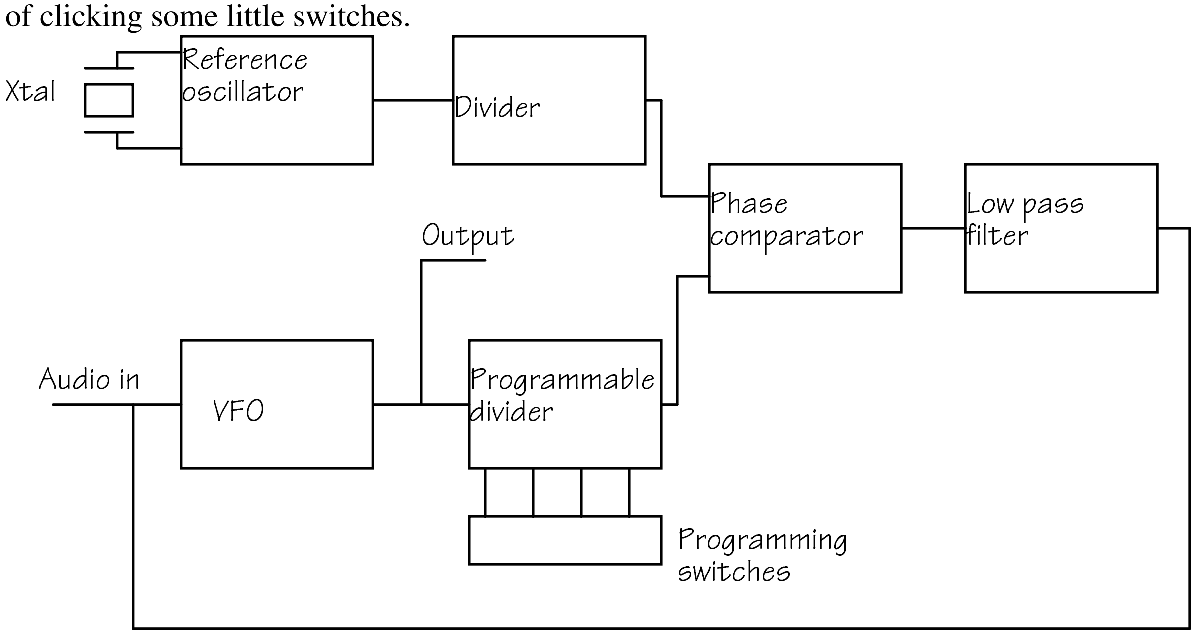
PHASE LOCKED LOOP (PLL) OSCILLATORS

The way its done properly is with the phase locked loop oscillator. This combines the ease of tuning and wide deviation of a VFO with the frequency stability of a crystal oscillator. It works thus: A crystal oscillator is used to provide a reference frequency. This is digitally divided by logic chips to a relatively low frequency, say 25 kHz. A VFO provides the output, which is also digitally divided to give another relatively low frequency. These two low frequencies are presented to a PHASE COMPARATOR which basically decides which frequency is higher by comparing the phases of the two signals. The phase comparator generates an ERROR VOLTAGE which is connected back to the input of the VFO through a low pass filter. This is the loop bit.

If the VFO is running too fast the phase comparator decreases the error voltage so as to slow it down till the phases at its input are the same. If its running too slow the error voltage is increased to speed it till the phases are the same. All this happens instantaneously of course so the output frequency remains constant.

In this way the temperature stability of the VFO isn't important and it can be built round a BJT, as its output frequency is phase locked to the crystal oscillator, and the frequency is very good.

Two more things to explain. How do you change the output frequency? By making the VFO's divider programmable. Say its set to divide by the number N. The phase comparator is a simple minded sort of soul, concerned only with equalising the phases at its inputs, it doesn't know what's really coming out of the VFO, which is N times the divided reference signal. Because this signal is so low compared to the VFO frequency N can be made to have hundreds of different values, giving hundreds of different output frequencies from the VFO. So changing the frequencies is just a matter of clicking some little switches.



Hang on a sec, the VFO is being frequency modulated by the audio input, so its frequency at any given instant depends on the voltage of the audio output. We don't want this variation of the VFO's frequency to be ironed out by the PLL system, so we 'iron out' the error voltage from the phase

comparator, so it just contains the underlying trend rather than what's happening any split second. This is purpose of the low pass filter.
The system can be simplified by leaving out the dividers. If this is done you end up with an output frequency determined solely by the crystal. You've still got the wide deviation capability of course, which distinguishes this system from one based on a simple crystal oscillator. This sort of fixed frequency oscillator is used for things like wireless mikes and could be used for studio to TX links. Programmable PLL oscillators are used in all manner of professional communication equipment, including broadcast TX's.

BUFFERS
Any oscillator, regardless of its type, is followed by a buffer. This is usually one or two transistors operating in what is known as class A mode. Its function is to protect the oscillator from what is going on further along the circuit, especially from changes in its 'load' as the following stage is tuned. The combination of oscillator and buffer together is called the EXCITER and is a small but fully fledged TX. Small in respect to its output power. Typical values are in the region of 100 - 500 mW.

AMPLIFIERS
To increase the power output of our fledging TX we need to add an amplifier. Obviously we are talking about radio frequency (RF), not audio amps. RF amps have certain important characteristics: a) Bandwidth, b) Gain and maximum power output c) Input and output impedance
BANDWIDTH. This is the range frequencies the amp will amplify properly. The bandwidth is ultimately limited by the characteristics of the active devices in the amp (i.e. transistors or valves), but more specifically by its type, LINEAR or a TUNED amplifier.
A linear amp will amplify quite a large range of frequencies and they have a good bandwidth, commonly 1.8 - 30 MHz which covers all of the amateur shortwave broadcast bands... no good for a VHF pirate, but could be useful for a MW pirate. They operate in class A or B mode and have the advantage that they don't need adjusting when the frequency is changed. Their disadvantage are they're more complex and dearer than tuned amps and are much harder to design, requiring extensive knowledge of the transistors round which the amp is constructed. Linear amps for VHF are uncommon.
Tuned amps only amplify a narrow band of frequencies, they have a small bandwidth, centred on one frequency which is determined by the TUNED CIRCUITS in the input and output networks of the amp. Tuned circuit have a RESONANT frequency. This can be adjusted by variable capacitors known as trimmers, to the desired frequency. The amp will produce max. output when the tuned circuit resonant frequency is the same as the input frequency from the exciter. Tuned amps often operate in the class C mode, which is more efficient than A or B. This means more of the power being drawn from the battery or whatever turns into watts up the aerial rather than heat the amp. They are relatively simple circuits, and are easier to design. The bandwidth is a trade-off with gain, the wider the bandwidth, the less the gain. The disadvantages of a tuned amp is of course you have to tune it to the frequency you're using and if you change the frequency you'll have to retune to maintain the gain of the amp.

GAIN AND MAXIMUM OUTPUT POWER
The POWER GAIN (as opposed to a voltage or current gain which is different) of an amp is defined as a ratio: $Powergain = \frac{inputpower}{outputpower}$ and is a measure of the amps ability to make its input bigger.
Power gains are often expressed in DECIBELS (dB) which are defined: $powergain(dB) = 10\log\frac{outputpower}{inputpower}$.
Amps also have a max. output power. When this is reached increasing the input power won't result in more output power and may damage the amp.
In the case of single stage (i.e. one transistor) class C tuned amps the gain and max. output power of the amp is basically the gain and max. output power of the transistor. Knowing these we can calculate the power necessary to produce the max. output power. e.g. lets consider the popular

MRF237 transistor. According to the makers data sheet this has a max. output power of 4 watt and a gain of 12 dB. First we've to convert the gain in dB to ordinary gain: $gain = 10\wedge\frac{gain(dB)}{10}$

for example: $gain = 10\wedge(\frac{12}{10}) = 10\wedge 1.2 = 15.85$

$$inputpower = \frac{outputpower}{gain} = \frac{4}{15.85} = 0.25W = 250mW$$

So for 4 watt output power we need 250 mW input power. Most exciters can manage this, hence the popularity of the MRF237 in the first amp after the exciter. The joker in the pack is that all these figures are for a frequency of 175 MHz, that on which the transistor was designed. You can't predict what happens at 100 MHz and have to experiment.
The MRF238 has 30 watt output power and a gain of 9 dB, so it needs 3.8 watt input power. This can be had from the MRF237. That's how the makers (Motorola Corpse.) planned it.

INPUT AND OUTPUT IMPEDANCE
Impedance is the alternating current (AC) version of resistance. The standard impedance of exciters and inputs and outputs of amps is 50 Ω. The impedance of the input and the output networks of an amp is altered by the tuned circuits which you recall also tune the circuit in a tuned amp. The INPUT IMPEDANCE is important as it effects the LOAD the amp has on the stage before it. Max. power is transferred between stages when the impedance of the output and input are equal. If the impedances aren't equal a MISMATCH is said to occur and in this case some energy is reflected back from the input of a stage into the output of the preceding one, where its wasted as heat.

THE VSWR METER
Some of you may know that we can use a VSWR meter (also known as Voltage Standing Wave Ratio meter, SWR meter or a Reflectometer) to detect mismatch between TX and the aerial, but a VSWR meter is just as much at home doing this between amp stages. VSWR is the ratio of the forward (or incident) and reflected power. Except for dear ones they work the same. The switch is set to forward or the SET button is pressed. The knob is then adjusted to make the meter read full scale. The switch is then set to reverse or the button is pre-released. It now indicates the VSWR. A VSWR of 1:1 is perfect (no reflected power) and so unlikely. One of 00:1 shows all the power is reflected back into the amp, you'll get this with a VSWR connected to the amp output with nothing on the VSWR output (unless its got a built in dummy load). You'll also get it if there's a short circuit in the VSWR meter. In either case switch off **IMMEDIATELY** or you'll blow your power transistor.
The point of all this is to get the max. power output from the amp into the aerial, instead of a hot TX and a bad signal.
To tune such an amp you need a load connected to the output (or it'll blow up). We could use an aerial but this introduces an extra unknown quantity... the characteristics of the aerial. As well as the fact that we'd be broadcasting. What we need is a DUMMY LOAD.

THE DUMMY LOAD
This is basically a resistor, made so it presents a load to the amp's output independent of frequency (unlike the aerial). The 3 things about a dummy load we're interested are:
a) It should be suitable for the frequency we're interested in, about 100 MHz.
b) it should be rated to take the power we're trying to make.
c) It should have a resistance of 50 Ω to match the output network of the amp.
When buying ask for one for the 2 meter band, amateur radio, centred on 145 MHz. Most test gear for this band will work on frequencies we're interested in.
The amp should first be tuned with reduced input power and supply voltage. Adjust the network for the best input match (lowest reading on a VSWR meter connected to the input side) and adjust the output trimmers for max. output power. Be sure the extra power is in the frequency you want and not in the HARMONICS. Check with a wave meter (more of this coming up). Another VSWR meter can be used for a relative indication of the output power, or the RF PROBE will give an

absolute indication. The pairs of trimmers are very interdependent, adjust one and you'll have to adjust the other, and so on.
This done, if all OK, increase the input power by increasing the voltage supply to the previous stage, and the voltage supply and repeat the tuning. Do all this a few times till you reach the required levels. Listen on a nearby (but not too near) receiver. The signal should be in just one place on the dial with no funny noises or modulations going on. Check with a wavemeter. Altering the trimmers and varying the input power and supply voltage should result in smooth variations of the supply current and output power with no steps or jumps. The exception is, as the input power is reduced at some point the amp will switch off, a characteristics of class C amps.
To vary the supply voltage you need a Variable Stabilised Power Supply Unit. If you can't get hold of one you could build one. They're not expensive and are well handy, and give you some experience, if needed, of electronic construction.

HARMONICS
Harmonics are multiples of the transmitting frequency. For a frequency of 100 MHz, the first harmonic, known as the FUNDAMENTAL, is 100 MHz, the second is 200 MHz, the third is 300 MHz etc. They're produced as side effects in various parts of the circuit and will interfere with other users of these frequencies if let escape from the TX. Known as RADIO FREQUENCY INTERFERENCE (RFI). Tuned class C amps don't amplify harmonics, as they're out of the range of the amps abilities. But the use of class C means that harmonics are generated by the amp along with the desired frequency. The strongest ones (apart from the fundamental) from such amps are usually the third, then the fifth etc. The amplitude of harmonics is minimised if the output networks are tuned properly, but they're still there. Oscillators and buffers can also make harmonics if not set up right.

WAVEMETERS
To detect harmonics we need an ABSORPTION WAVEMETER, usually called just a wavemeter. Or we can use a GRID DIP OSCILLATOR (GDO) or a gate dip oscillator, both of which are known as DIP METERS. Most dip meters have a switch which turns them into wavemeters. A wavemeter has a tuning knob, calibrated in frequency, a meter showing signal strength, and some kind of aerial. You hold the aerial near a coil in the bit of the circuit you're interested in, and tune the wavemeter. It shows how much signal is present on the frequencies shown in the scale. So you can see what frequencies are being generated in that part of the circuit. Ideally you'll just find the fundamental, unless the circuit is a frequency tripler or something.
If you buy a wavemeter be sure it covers the right range, from below 100 MHz to get the fundamental to above 300 MHz to get the third harmonic.
Even with all tuned right you're still going to have some harmonics generated by the last stage. A sensible pirate won't let these reach the aerial, e.g. if you're using a frequency of 100.35 MHz the third harmonic us 307.05 MHz which happens to be that used by USAF Upper Heyford's Control Tower. You might think this is funny but you won't stay on the air for long. To stop harmonics reaching the aerial we need a BANDPASS FILTER.
Each amp bumps up the power some more, cos the transistor in each one can only supply so much gain. So if you're the proud owner of a 5 watter and you're offered a 1000 watt amp its useless as you'd need probably 100 watt input to drive it so you'd need amps in between.
To tune a series of amps on your TX you must break in, physically if needed, to tune each one at time. Do this by unsoldering components and soldering in short bits of co-ax with plugs to connect to dummy load and VSWR meter.

BANDPASS FILTER
This filter only allows through a narrow band of frequencies, i.e. it has a narrow bandwidth, a good one would be less than 1 MHz. It needs standard 50 Ω input and output impedance and be able to take power you're using and be tuned to the frequency you want to let through. Other frequencies are reduced drastically, by an amount known as INSERTION LOSS. It reduces also the desired frequency slightly. To keep this loss low bandpass filters for high output powers are usually pretty chunky numbers.

Pirate gear doesn't have this filter built into the final stages so if you need one you have to add it on. It needs a well screened case to stop harmonics leaking out. In fact your whole TX should be well screened for the same reason. Say e.g. you used a shoebox and had your oscillator on a third of a frequency of 92.25 MHz you could be interfering with pagers of a local hospital as they use 31.75 MHz. Proper screening and a bandpass filter will eliminate such possibilities.

CONNECTORS

As you may have guessed you can't use any connectors on VHF as they have to match the amp and feeder. Use BNC or the UHF series. UHF is best for higher powers as you can get a wider cable into the plug. N type is also good but dearer.

FEEDERS

So you've got your nice clean harmonic free signal coming out of your bandpass filter... we're on the home run. All that's left is to get the signal up the aerial feeder to the aerial and we're away. BUT the aerial cable needs to MATCH the TX's output stage at one end and the aerial at the other. The cable like the TX's output, the connectors and the aerial has an impedance and to match this should be 50 Ω. It also needs a LOW LOSS or your watts will escape as heat. Not the same as a bad VSWR where you lose energy in the TX, a good VSWR does not mean the cable's okay. Decent cables for short runs are UR76 and RG56U. For longer runs or higher powers use UR67.

AERIALS

At last, the aerial! You can run a pirate knowing a little of TX's, but if you know nothing of aerials you'll have a few listeners. So you must read a book on it. I recommend 'The Two Metre Antenna Handbook' by FC Judd G2BCX. Lot's of it isn't useful but he goes into things like propagation, matching, VSWR in better detail. All the dimensions he gives are for the two meter amateur band, centred in 145 MHz. To convert to other frequencies all dimensions (including diameter of aerial element etc.) should be divided by your frequency in MHz and then multiplied by 145.

POLARISATION

One thing to decide is what polarisation to use. The main ones are HORIZONTAL and VERTICAL. To simplify you can say a horizontally placed aerial produces horizontally polarised radio waves and a vertically placed one vertically polarised ones. To receive a horizontally polarised signal you need a horizontally polarised aerial, and for vertical one you need a vertically polarised aerial. Most receivers on FM have horizontally polarised aerials, but all car aerials are vertically polarised. So what polarisation you go for depends on the audience you expect. E.g. on Sunday afternoon you'd expect people at home so use horizontal, while in rush hour you might favour vertical. You can build an aerial which splits the power between both, as used in legal stations, known as MIXED polarisation. But the effect of radio waves bouncing off buildings etc. tends to twist the polarisation of your signal from horizontal to vertical and vice versa, so your signal could still be picked up by the wrong aerial.
Your transmitting site will affect you choice of aerial. In the middle of the area you want to cover you'll need an OMNIDIRECTIONAL aerial which transmits equally each ways, while outside your coverage area you can beam the signal in with a DIRECTIONAL aerial.
The simplest possible aerial for VHF is known as the HALF WAVE DIPOLE. The elements can be bits of thin aluminium or copper tube. The lengths of each dipole you get from your frequency by: $L = \frac{71}{f(inMHz)} metres$. The impedance is about 75 Ω which is close enough to 50 to be fed from 50 Ω cable without too much power loss. A half wave dipole used vertically is omnidirectional, but when used horizontally it has a fig of eight coverage which isn't very useful. Also a dipole needs a balanced feed. You need a BALUN (BALance to UNbalance) transformer. These can be easily made out of bits of co-ax cable. If you don't do this power will be radiated from the feeder. An aerial with an impedance greatly different from 50 Ω needs an IMPEDANCE TRANSFORMER also made out of bits of co-ax cable.
Before going on air get a low VSWR by adjusting the position of the aerial and any adjustable pieces. Aim for 2:1 or less. Use low power into the aerial when tuning it up and adjusting, if using a

100's of watts and a bit came off in your hand the VSWR could be so bad as to blow the final transistor. For the same reason check the continuity of the aerial with an ohmmeter before plugging in, to be sure its what its meant to be, either a short circuit or an open one, depending on the type. A dipole should be an open circuit.

SITING

Siting is very important. Height is the main factor, even more than watts! Since VHF radio waves go almost in straight lines, 100 watt in your front room will only reach your neighbours, while 5 watt up high and unblocked will go 10 km's or more. The waves do bend a bit so you'll cover more than you can see but its hard to say how much.

GO FOR IT!!!!!