Introduction

This is still a "work in progress". It was necessary for me to revise the notes before teaching them as I had to correct some errors and adjust some statements. There were a few obsolete passages as well. These last are being considered by the SARL. As are the questions in the exam being revised as well. This all takes time that nobody seems to have these days.

This should be 'readable' on almost any device - tablet or PC or phone. I think it will work on a phone but would not recommend it! The contents table has clickable links to the chapters and some chapters have WikiPedia links as well.

Anyway please email or WhatsApp me with any questions - whether as part of the notes or as aside to the subject. I am a retired Electronics Engineer who specialised in Radio and TV Communications. For example the notes don't mention that television transmissions are not AM nor SSB, but "Vestigial Sideband" transmissions.

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Chapter 11: Tuned Circuits

Inductors and capacitors can be combined in series and parallel to form circuits that have the ability to accept or reject signals of particular frequencies. These circuits, which are called tuned circuits, are of great importance in radio.

As you will see later, modern radios use digital techniques to achieve tuning, but until late in the twentieth century, tuned circuits were used universally for selecting frequencies to receive.

11.1 Reactances in Series

Both capacitors and inductors exhibit reactance in AC circuits. The reactance depends on frequency according to the formulae:

XC = 1 / (2 π f C)

and

XL = 2 π f L

When reactances are connected in series – for example, two capacitors or a capacitor and an inductor – then the reactances can be added to give the equivalent reactance of the two reactances in series. However, remember that the phase lag between voltage and current is opposite for capacitors and inductors. To make provision for this difference, we add inductive reactances and subtract capacitive reactances to determine the total reactance.

XL = XL1 + XL2 + …

and

XC = XC1 + XC2 + …

If capacitive and inductive reactance are both present:

XTotal = XL – XC

For example, suppose we connect two 100 pF (10^-10 F) capacitors in series. At a frequency of 10 MHz (10^7 Hz), the reactance of each of the capacitors individually is:

XC = 1 / (2 π f C)
= 1 / (2 x 3.14 x 10^7 x 10^-10) Ω
= 1 / 0.00628 Ω
= 159 Ω

So the equivalent reactance of the two reactances in series is:

XTotal = X1 + X2
= -XC1 – XC2
= -159 Ω -159 Ω
= -318 Ω

Remember that capacitive reactance must be subtracted, to compensate for its opposite **current** lag to that of an inductor.

Of course there is another way to find this result. Since we have two capacitors of the same value (100 pF) in series, the equivalent capacitance must be half the capacitance of the individual capacitors, or 50 pF (5 x 10^-11 F). We can calculate the reactance of this equivalent 50 pF capacitance at 10 MHz (10^7 Hz) as follows:

XC = 1 / (2 π f C)
= 1 / (2 x 3.14 x 10^7 x 5 x 10^-11) Ω
= 1 / 0.00314 Ω
= 318 Ω

or

XTotal = -318 Ω

As expected, we reach the same answer.

11.2 Reactances in Parallel

Similarly, the formula for the equivalent reactance of two reactances in parallel is:

1 / XTotal = 1/X1 + 1/X2 + …

For example, if we take our two 100 pF (10-10 F) capacitors, which each has a capacitive reactance of 159 Ω at 10 MHz, and connect them in parallel, the equivalent reactance is found as follows:

1/XTotal = 1/X1 + 1/X2 + …
= 1/-159 Ω + 1/-159 Ω
= -0.0126 /Ω

so

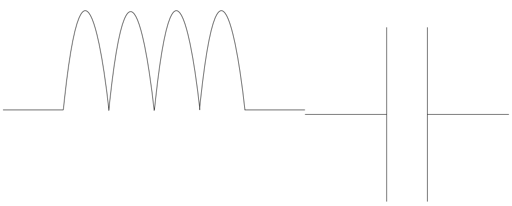
XTotal = 1/(-0.0126 /Ω)
= -79.5 Ω

Once again this makes sense since the two 100 pF capacitors connected in parallel are equivalent to a single 200 pF (or 2 x 10^-10 F) capacitor, with a reactance at 10 MHz of:

XC = 1 / (2 π f C)
= 1 / (2 x 3.14 x 10^7 x 2 x 10^-10) Ω
= 1 / 0.0126 Ω
= 79.5 Ω

11.3 The Series Tuned Circuit

Of course you might well ask, why bother to learn the formulas for reactances in series and parallel if we can calculate the same results using the formulas for capacitors and inductors in series and parallel that we already know? Good question; the answer can be found in the following circuit, which shows an inductor and a capacitor connected in series.



6.5 µH 39 pF
LC series tuned circuit

Suppose we want to calculate the equivalent total reactance of these two components at 10 MHz (10^7 Hz). We can't use the formula for inductors in series or the formula for capacitors in series, since the circuit contains one of each. So instead we must calculate the individual reactances of each component at a frequency of 10 MHz, and then use the formula for reactances in series.

The reactance of the inductor is found as follows:

XL = 2.π.f.L [2 * Pi * F * L]
= 2 x 3.14 x 10^7 x 6.5 x 10^-6 Ω
= 408 Ω

The reactance of the capacitor is given by:

XC = 1 / (2 π f C) [1/ 2 * Pi * F * C]
= 1 / (2 x 3.14 x 10^7 x 39 x 10^-12) Ω
= 1 / 0.006908 Ω
= 408 Ω

So the combined reactance of the inductor and capacitor in series at 10 MHz is

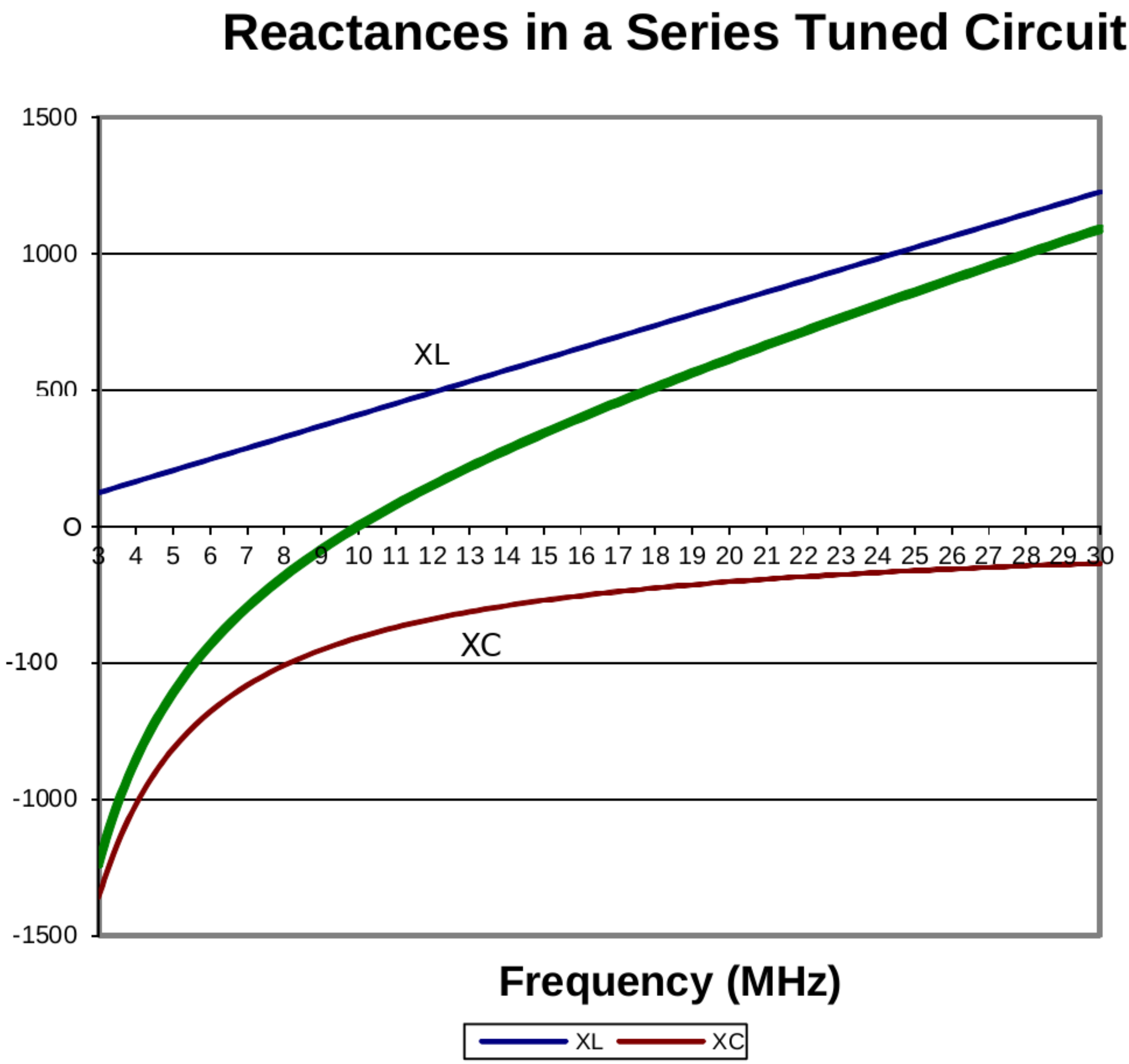
XTotal = XL - XC
= 408 Ω – 408 Ω
= 0Ω

That's right—zero! The capacitor has reactance, and the inductor has reactance, but at this frequency (10 MHz) the positive reactance of the inductor exactly cancels out the negative reactance of the capacitor, leaving no reactance at all! The frequency at which the positive and negative reactances cancel out is known as the **resonant frequency** of the circuit. The circuit itself is called a series resonant circuit or a **series tuned circuit**.

Since the reactance of the inductor increases with frequency, while the reactance of the capacitor decreases with frequency, this canceling out will only happen at one specific frequency. At any other frequency, the circuit will exhibit either inductive (positive) or capacitive (negative) reactance.

Real tuned circuits also contain a little resistance, due to the wire from which the inductor is wound and the leakage current of the capacitor. However, good quality components will result in very low resistance.

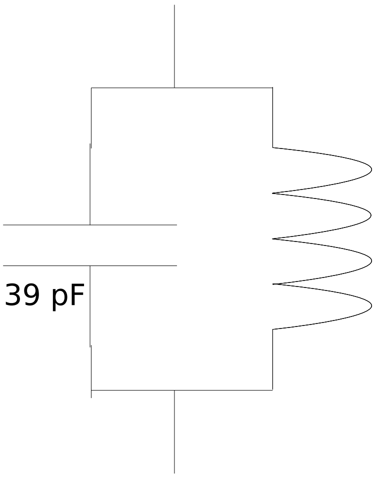
The graph below shows the inductive reactance XL (which is always positive), capacitive reactance XC (always negative) and the combined reactance of the series circuit XS. As you can see, the combined reactance is negative (capacitive) below the resonant frequency of 10 MHz, and positive (inductive) above the resonant frequency.



The series tuned circuit is very useful in radio electronics as the low reactance near the resonant frequency means that current can easily flow in the circuit near this frequency; while the high reactance at other frequencies will oppose the flow of current at frequencies other than the resonant frequency. In this way, a series tuned circuit can be used to accept signals with frequencies near the resonant frequency, while rejecting other signals.

11.4 The Parallel Tuned Circuit

Having seen the strange and interesting behaviour we get when we connect an inductor and capacitor in series naturally raises the question of what would happen if we were to connect them in parallel. To save us unnecessary calculations, we choose the same values of L = 6.5 µH and C = 39 pF.



An LC parallel tuned circuit

Once again, we will calculate the combined reactance at 10 MHz – since this was the resonant frequency for the series tuned circuit, perhaps it will also show some interesting behaviour in this parallel tuned circuit.

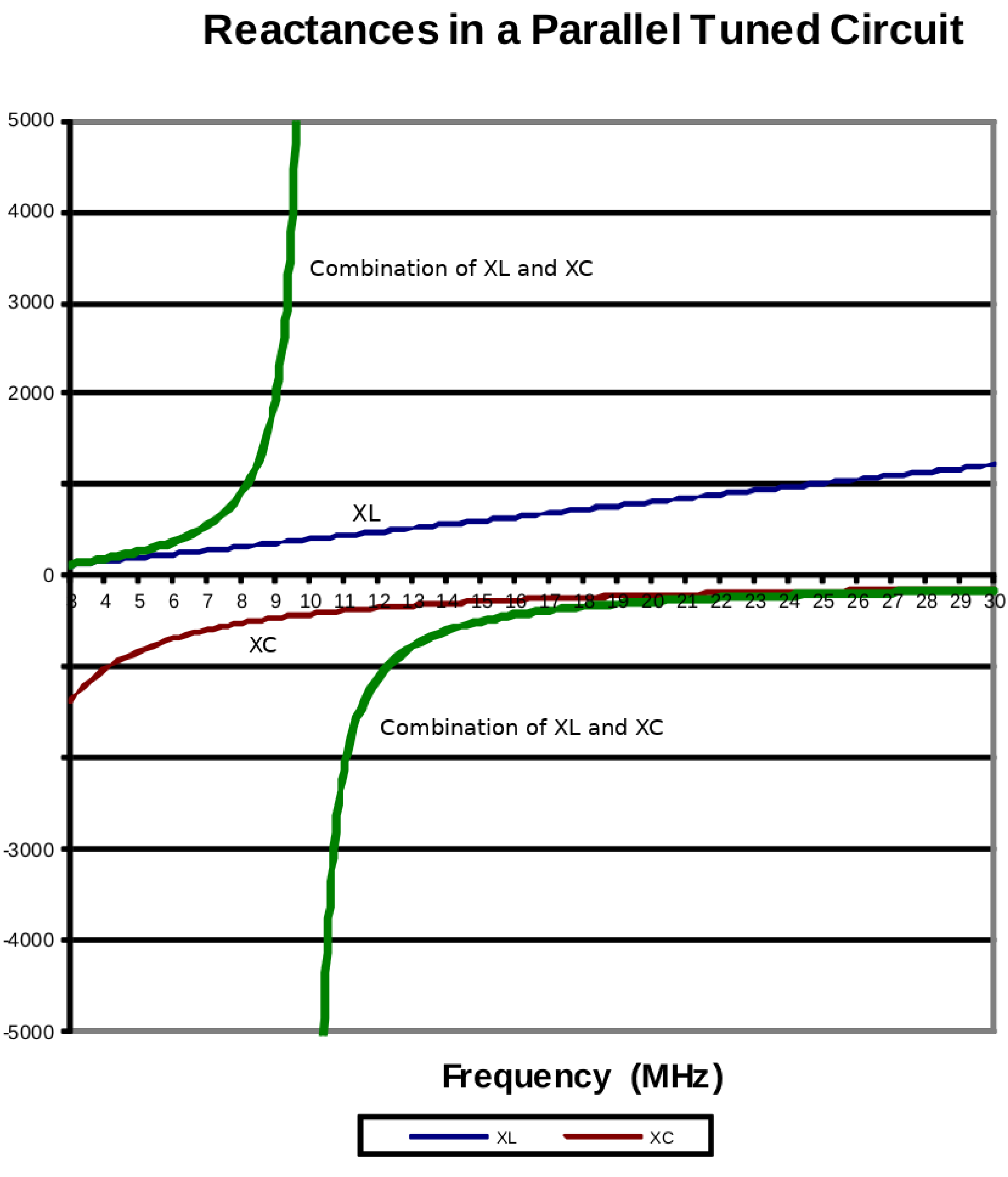
From the formula for reactances in parallel, we know that

$$\begin{aligned} 1/\,X_{Total} &= 1/X_L + 1/X_C \\ &= 1/408\ \Omega + 1/{-408}\ \Omega \\ &= 0.00245\ /\Omega - 0.00245\ /\Omega \\ &= 0\ /\Omega \end{aligned}$$

so $$\begin{aligned} X_{EQUIV} &= 1\ /\ 0/\Omega \\ &= ???? \end{aligned}$$

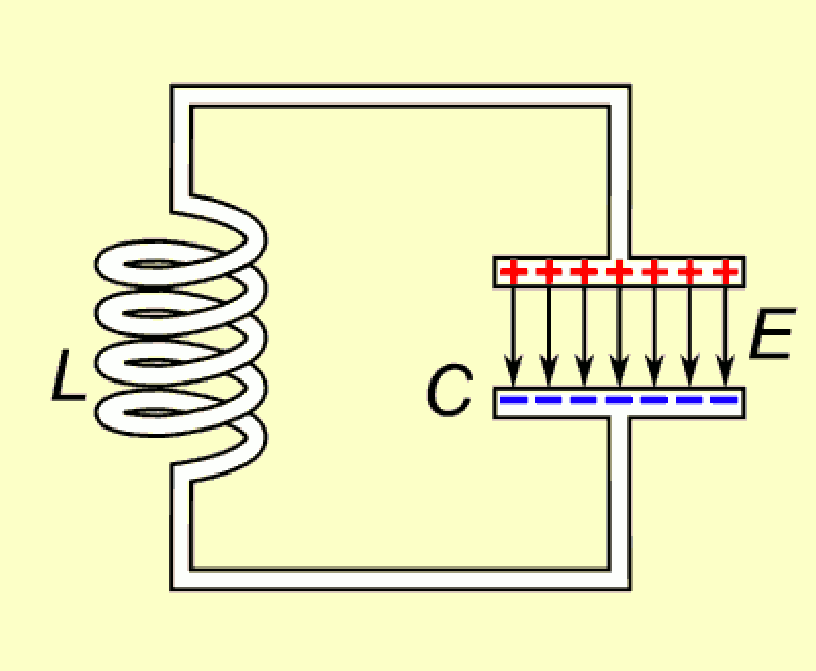
What has happened here? Once again the positive inductive reactance has cancelled out the negative capacitive inductance, but this time it has left the zero in the denominator (bottom) of a fraction, which means that the result is undefined.

However, if we plot a graph showing the reactances for a range of frequencies, we will understand what is happening better.



Once again the inductive reactance is always positive, while the capacitive reactance is always negative. This time, however, the combined reactance of the tuned circuit starts slightly positive (inductive) and rapidly gets more and more positive as the resonant frequency is approached. However, at the resonant frequency it instantaneously transitions from being a very high positive (inductive) reactance to being very high negative (capacitive) reactance. No wonder the exact value at resonance is undefined.

As a result, a parallel tuned circuit has a high reactance near resonance while its reactance is small away from the resonant frequency. This means that a parallel tuned circuit can be used to block signals near its resonant frequency, while allowing signals of other frequencies to pass relatively easily.



11.5 Circulating Current in a Parallel Tuned Circuit

A parallel tuned circuit has two components that are capable of storing energy. The inductor stores energy in its **magnetic field**; and the capacitor stores energy in the **electric field** between its plates. At resonance, energy is constantly being transferred from the capacitor to the inductor and back again. (see above animation)

As the capacitor charges up, a voltage develops between its plates. This voltage causes a current to flow through the inductor, which generates a magnetic field. As the capacitor discharges the voltage across its plates drops, which tends to reduce the current flowing through the inductor. However, an inductor will resist any attempt to change the current flowing through it. The magnetic field of the inductor collapses, inducing a potential difference into the inductor that acts to keep the current flowing in the same direction as it was before. This current flow now charges the capacitor up again, but with the opposite polarity to before. As the capacitor charges a voltage develops across its plates. This voltage causes current to flow through the inductor in the reverse direction, which generates a magnetic field, and so on.

So the parallel tuned circuit acts somewhat like a pendulum, continually transferring energy between two different forms. In the pendulum, these forms are the potential energy when the pendulum is stationary at the top of its arc, and the kinetic energy when the pendulum is moving at maximum speed at the bottom of its arc.

Remember that the parallel tuned circuit has lots of reactance at the resonant frequency. It therefore does not allow a lot of current to flow from the surrounding circuit. However, the circulating current that flows in a parallel tuned circuit – that is, the current flowing around the circuit containing the capacitor and the inductor – can be much larger. In practical circuits, it is not uncommon to have a circulating current that is 100 times the input current.

11.6 Calculating the Resonant Frequency

We have seen that in both a series tuned circuit and a parallel tuned circuit, something interesting happens at the resonant frequency which is where the reactance of the capacitor and inductor have the same magnitude (value) but one is positive and the other is negative.

The reactances counteract one another. We can derive a formula for the resonant frequency as follows:

At resonance, the magnitude of the capacitive and inductive reactances are equal, so

$$XL = XC$$

$$2 \pi f L = 1 / (2 \pi f C)$$

so

$$f^2 = 1 / (4 \pi^2 L C)$$

and

$$f = 1 / (2 \pi \sqrt{L C})$$

You do not need to know the derivation, but you should be able to apply the result. For example, let us calculate the resonant frequency of a series or parallel circuit consisting of a 6.5 μH inductor and a 39 pF capacitor:

$$f = 1 / (2 \pi \sqrt{LC})$$

$$= 1 / (2 \times 3.14 \times \sqrt{6.5 \times 10^{-6} \times 39} \times 10^{-12}) \text{ Hz}$$

$$= 1 / (6.28 \times \sqrt{253.5 \times 10}^{-18}) \text{ Hz}$$

$$= 1 / (6.28 \times 1.59 \times 10^{-8}) \text{ Hz}$$

$$= 1 / 10^{-7} \text{ Hz}$$

$$= 10^{7} \text{ Hz}$$

$$= 10 \text{ MHz}$$

This answer agrees with the resonant frequency in the series and parallel resonant circuits above.

11.7 Circuit Losses and the Quality Factor

The discussion so far has ignored circuit losses. For example, all practical inductors have some resistance as well as their inductance, and capacitors also have some losses although these are typically negligible compared to the losses caused by the resistance of the inductor.

The effect of these losses is that in a practical series tuned circuit, although at resonance the reactance would be zero, there would still be some small resistance. In a parallel tuned circuit, the effect of circuit losses is to limit the reactance at resonance to a high but finite value, rather than being completely undefined (or “infinite”) as predicted by the maths.

The extent of circuit losses is expressed by a number called the “Quality Factor”, or “Q Factor” or just the “Q” of the tuned circuit. A high Q means low circuit losses, while a low Q means high circuit losses. The Q is defined as the reactance of either the inductor or the capacitor at resonance (remember they are equal?) divided by the circuit resistance. So

$$\mathbf{Q} = X_L / R$$
$$= X_C / R$$

The Q of practical tuned circuits is typically between 50 and 200. **[BUT! can be effectively higher...]**

The Q is related to two other properties of the tuned circuit:

1. The ratio of circulating current in a parallel tuned circuit to the current drawn by the tuned circuit is the same as the Q. So in a parallel tuned circuit with a Q of 100, the **circulating current** will be 100 times greater than the current drawn from the rest of the circuit.

2. The **selectivity** of the circuit – that is, its ability to allow desired signals through while blocking undesired signals. The greater the Q of the tuned circuit, the greater its **selectivity**.

Summary

The **series tuned circuit** has a low reactance near its resonant frequency and a high reactance at other frequencies. Series tuned circuits are often used to allow signals near the resonant frequency to pass, while blocking signals at other frequencies.
[A "Band-Pass Filter]

The **parallel tuned circuit** has a high reactance near its resonant frequency and a low reactance at other frequencies. Parallel tuned circuits are often used to block signals near the resonant frequency, while allowing signals at other frequencies to pass.
[A Band-Stop Filter]

The resonant frequency of a series or parallel tuned circuit may be calculated as

$$f = 1 / (2 \pi \sqrt{LC})$$

The **Quality Factor ("Q")** is defined as the reactance of either the inductor or the capacitor at resonance divided by the circuit resistance. A tuned circuit with a high Q is more selective than a tuned circuit with a low Q.

The **circulating current** in a parallel tuned circuit may be many times the current drawn by the tuned circuit. The ratio between the circulating current and the current flowing into the tuned circuit is the same as the Q.

Chapter 12 - Decibel Notation

In amateur radio we often deal with ratios of powers. For example, the gain of an amplifier is the ratio of its output power to its input power. These ratios can be very large or very small.

For example, the gain of a typical amateur radio receiver – the ratio between the output power into the speaker or headphones to the input power from the antenna – is in the region of 100 000 000 000 000. That's an amplification of a hundred trillion times! While we could use scientific notation to represent these large numbers (the one above is 10^14), another way of expressing the ratio of two powers is commonly used. This is the "**deciBel**", 'dB' for short.

The unit "bel" was first used by telephone engineers at Bell Laboratories (now AT&T) and was named after Alexander Graham Bell (1847-1922), the inventor of the telephone and founder of Bell Laboratories. The "decibel" is simply one tenth of a **bel**, which turned out to be a more popular size.

One decibel represents roughly the minimum discernible change in the loudness of an audio signal. The abbreviation for the decibel is "dB", which is also often used in general conversation such as "your signal is S9 plus 20 dB".

A ratio of two powers can be expressed in decibels as follows:

dB = 10 log10 (PR) ….   [dB = 10 * log10 * PowerRatio]

where PR is the ratio of two powers (e.g. PR = P1/ P2), "dB" is the same ratio expressed in decibels, and "log10" means the mathematical logarithm to the base 10.

If you are not familiar with logarithms then **don't panic** – once we have explored a couple of the properties of decibels we will see that there is a simple way to calculate many common values.

12.1 Adding Decibels

A fundamental property of decibels is that when two ratios expressed in decibels are added, it is equivalent to multiplying the original ratios. For example, a ratio of 2 times is 3 dB and a ratio of 10 times is 10 dB. If we add the decibel representations we get 3 dB + 10 dB = 13 dB, which is equivalent to a ratio of 20 times. This is the same as we get if we multiply the ratios:

2 * 10 = 20. This bit of magic is possible because of the use of the logarithm function in the definition of the decibel.

Example
In a radio receiver the radio frequency (RF) amplifier has a gain of 6 dB; the intermediate frequency (I.F.) amplifier has a gain of 110 dB and the audio frequency (A.F.) amplifier has a gain of 20 dB.

What is the total gain of the receiver?
If the gains of the amplifiers had been expressed as simple ratios (POUT/PIN) then we would have to multiply the ratios together to get the total gain. However since the gains are expressed in decibels, we can add them to get the total gain. So in this case the total gain is 6 dB + 110 dB + 20 dB = 136 dB.

12.2 Representing Losses

The decibel can also be used to represent losses, i.e. situations where a signal gets smaller. If you calculate the decibel equivalent of a ratio that is less than 1, then the formula gives a negative number. For example we can calculate the decibel equivalent of a power ratio of 0,1 as follows:

dB = 10 log10 (PR)
= 10 log10 (0,1)
= 10 * -1

= -10 dB

So, for example, an attenuator that reduces a signal to one-tenth its original power could be described as having a gain of –10 dB. Note that the minus sign indicates that it is actually making the signal smaller even though it is expressed as a “gain”. The same attenuator could also be described as having a loss of 10 dB. This time there is no minus sign because it is being described as a loss.

However if you add decibels together (which as we have seen is equivalent to multiplying the original ratios), then you should express all the ratios as either gains or losses before adding them together.

You can’t add a decibel representing a gain to one representing a loss.

Example

An **attenuator** with a loss of 6 dB is added before the RF amplifier in a receiver. Before adding the attenuator, the receiver had a gain of 136 dB. What is the total gain of the receiver with the attenuator?

Because we can’t add the 6 dB loss of the attenuator to the 136 dB gain of the receiver, we first convert express the attenuator’s gain as –6 dB. Then we can calculate the total gain of the receiver by adding the –6 dB gain of the attenuator to the 136 dB gain of the receiver to get the answer 130 dB.

Finally, a gain of exactly 1 (i.e. a signal that gets neither stronger nor weaker) can be represented as 0 dB. This makes sense, since adding 0 dB to a ratio represented in decibels will not change it; just as multiplying a ratio by 1 won’t change it either.

GAIN – An amplification or multiplication of a signal.
LOSS – A lowering or division of a signal.

Does an antenna have 'gain'? NO it always has a loss. But the focussing action provides a more sensitive/effective radiated power at points around the antenna.

12.3 Quick and Easy Decibel Conversions

Some commonly used ratios are easily converted to decibels. These are shown in the table below:

Power Ratio	Calculated	Decibels	Power Ratio	Calculated	Decibels
1000000	60.00	60 dB	0,000001	#VALUE!	-60 dB
100000	50.00	50 dB	0,00001	#VALUE!	-50 dB
10000	40.00	40 dB	0,0001	#VALUE!	-40 dB
1000	30.00	30 dB	1	0.00	-30 dB
100	20.00	20 dB	0,01	#VALUE!	-20 dB
10	10.00	10 dB	0,1	#VALUE!	-10 dB
5	6.99	7 dB	0,2	#VALUE!	-7 dB
4	6.02	6 dB	0,25	#VALUE!	-6 dB
2	3.01	3 dB	0,5	#VALUE!	-3 dB
1	0.00	0 dB			

ERK! What happened here? Just that commas are NOT ACCEPTABLE TO SPREADSHEETS!]

Possible solution You can try to use point (.) instead of (,) when you write numbers, it might consider those are different numbers.

GAIN

Power Ratio	Calculated	Decibels
1000000	60.00	60 dB
100000	50.00	50 dB
10000	40.00	40 dB
1000	30.00	30 dB
100	20.00	20 dB
10	10.00	10 dB
5	6.99	7 dB
4	6.02	6 dB
2	3.01	3 dB
1	0.00	0 dB

LOSS

Power Ratio	Calculated	Decibels
0.000001	-60.00	-60 dB
0.00001	-50.00	-50 dB
0.0001	-40.00	-40 dB
0.01	-20.00	-20 dB
0.1	-10.00	-10 dB
0.2	-6.99	-7 dB
0.25	-6.02	-6 dB
0.5	-3.01	-3 dB
0.9	-0.46	-0.5 dB
1	0.00	0 dB

You don't need to remember all the powers of ten (the numbers 10, 100, 1000 etc.). If a ratio consists of a 1 followed by any number of zeros, then it to convert it to decibels simply multiply the number of zeros by ten. For example, 1000000 has 6 zeros so it is equivalent to 60 dB (the number of zeros times ten).

Using these values it is possible to easily calculate the decibel representation of many other common ratios. For example, what is the decibel equivalent of a ratio of 20? Well 20 is not in the table, but 2 and 10 are, and 20 = 2 * 10. However we know that multiplying ratios is the same as adding their decibel equivalents, so the decibel equivalent of 20 must be the decibel equivalent of 2 plus the decibel equivalent of 10. So the answer is 3 dB + 10 dB = 13 dB, which is the decibel equivalent of 20.

Of course this works the other way round as well. Suppose we want to calculate the ratio represented by 27 dB. Although 27 dB is not in the table, we know that 27 dB = 20 dB + 7 dB, and both the values are in the table. Since adding decibels is equivalent to multiplying ratios, the ratio represented by 27 dB is the ratio represented by 20 dB multiplied by the ratio represented by 7 dB. So the answer is 100 * 5 = 500, which is the ratio represented by 27 dB.

12.4 Expressing Voltage Ratios as Decibels

Throughout this module We have stressed that decibel notation is used to express the ratio of two powers. However because there is a relationship between voltage and power, decibels are also sometimes used to express the ratio between two voltages. Now the relationship between voltage and power can be expressed as

P = V^2 / R ... That is V Squared over R.

Because power is proportional to the voltage squared, if the voltage is doubled then the power will be multiplied by 4; if the voltage is increased by a factor of 10 then the power will be multiplied by 100. (Note that this does not depend on the resistance, it will hold true for any resistance as long as the same resistance is used to calculate the power before and after the voltage is increased.)

Because of this, a modified formula is used to express a ratio of voltages in decibels:

dB = 20 log10 (VR) ... 20 log(base 10) . Voltage Ratio.

where VR is the ratio of two voltages and dB is the same ratio expressed in decibels. Note that the constant “10” in the formula used for power ratios is replaced by “20” in the formula for voltage ratios. This is to take into account the V2 factor in the formula for power. In other words, when we representing a voltage ratio in decibels, we are still representing a ratio between two powers. In this case, however, it is the notional power that would be dissipated by some (unknown) load if the voltages in question were applied across the load.

If you want to express a voltage ratio in decibels using the “quick and easy” conversions outlined above, then you should square the voltage ratio (multiply it by itself) to convert the voltage ratio to a power ratio before converting it into decibels.

[Note that expressing voltage ratios as decibels is a confusing and potentially misleading exercise.] **[BUT A lot of people do...]**

Wherever possible, deal with power ratios not voltage ratios.

For example, suppose the input voltage of an amplifier is 10 μV and the output voltage is 1 mV. The input and output resistances of the amplifier are both 50 Ω and we want to calculate the gain of the amplifier in decibels.

The input and output powers can be found from

PIN = V2 / R
= (10 * 10^-6)2 / 50
= 2 pW [pico Watts]

POUT = V2 / R
= (10^-3)2/50
= 20 nW [nano Watts]

Having calculated the powers, we can express them as a ratio and then convert it to decibels:

POUT/PIN = 20 nW / 2 pW
= 10 000
= 40 dB

An alternative way to reach the same answer would be to take the voltage ratio

VR = 1 mV / 10 µV [micro Volts]
 = 100

Then square this to find the power ratio

PR = 100^2
 = 10 000

And then convert this express this as 40 dB. (Remember, the number of zeros multiplied by ten!) However this alternative approach will only work if the input and output resistances are equal. The first method – calculating the actual input and output powers – will work whatever the input and output resistances, as long as you know what they are.

12.4 Expressing Power Levels in dBW and dBm

In the 'Radio Regulations', the power levels that apply to amateur transmissions are not expressed in Watts as before, but rather in dBW.

The unit dBW means "decibels referenced to 1 Watt". It is a way to express actual powers in decibel notation. Note that one cannot express an actual power – say 100 W – in decibels since decibels are used to express the ratio of two powers. However if you make one of the two powers a standard reference level, then by expressing the ratio of the other power to this standard reference level you can communicate an actual power level. One of the common reference levels is 1 W, and the resulting unit is given the abbreviation "dBW".

For example, the maximum power level specified for a Class A1 (ZS) license is 26 dBW. This means "26 dB over 1 W". Since 26 dB is a ratio of 400, 26 dBW means 400 W.

A related unit is decibels over 1 milliWatt. This unit is abbreviated "**dBm**". For example, the **sensitivity** of most amateur receivers is around –130 dBm, meaning "130 dB less than 1 mW". (The minus sign means that the level is less than the reference level of 1 mW). This is equivalent to the incredibly small value of 10^-16 Watts, or 0,1 femto-Watts!

[Some examples]

1 milliWatt in 50 Ohms is .001 Watt.
Power = Vrms ^ 2 / R (Ohms) Vrms ^ 2 = Power * R (Ohms)

Vrms = SQR(Power * R(Ohms))

So :- 50 * 0.001 = SQR(0.05)
 = 0.2236068 Volts rms **0.224 Volts**

1 Watt in 50 Ohms = SQR(1 * 50)
 = 7.071 Volts

Quite a lot of **double balanced mixers** require a local oscillator voltage of +17 dbm. This is usually into an impedance of 50 Ohms. So the voltage would be?

Summary

The decibel is a logarithmic unit used to express the ratio of two powers.
The ratio of two powers can be converted to decibels using the formula :-

dB = 10 log10 (PR)

Adding two ratios expressed in decibels is equivalent to multiplying the original ratios. However both of the figures added must express either a gain or a loss; you cannot add a gain to a loss. To convert a gain to a loss or vice-versa, simply put a minus sign before it. If a ratio consists of a 1 followed by any number of zeros, then it to convert it to decibels simply multiply the number of zeros by ten.

A ratio of two voltages can be expressed in decibels using the formula

dB = 20 log10 (VR)

This is equivalent to first converting the voltage ratio to a power ratio by squaring it, and then expressing the resulting power ratio in decibels. This will only give the correct result if both voltages are applied across the same resistance.

Although absolute powers cannot be expressed in decibels, they can be expressed in dBW (decibels referenced to 1 W) or dBm (decibels referenced to 1 mW).

NB

Transmitter Power Output of Amateur Radio Stations

(32) The maximum power output of the transmitter, as measured at the **antenna port**, must not exceed the levels specified in the **national radio frequency plan** for the relevant licence classes and **linearity** must be maintained.

Frequency Measuring Equipment

(36) Every amateur or experimental radio station must have frequency measuring equipment with accuracy of at least zero point one percent (0.1%), unless the frequencies of all transmitters of the station are crystal-controlled and are accurate to at least zero point one percent (0.1%).

(20) Where the amateur service allocation is on a secondary basis, frequency spectrum bands must be shared with other services subject to the following conditions:-

(a) **amateur radio stations must not interfere** with these services; and
(b) users of frequency bands **must unconditionally accept interference** from Industrial, Scientific and Medical (ISM) equipment.
(21) Radio apparatus used at an amateur radio station must not be tuned to a frequency other than a frequency for amateur services referred to in Annexure I in these regulations.

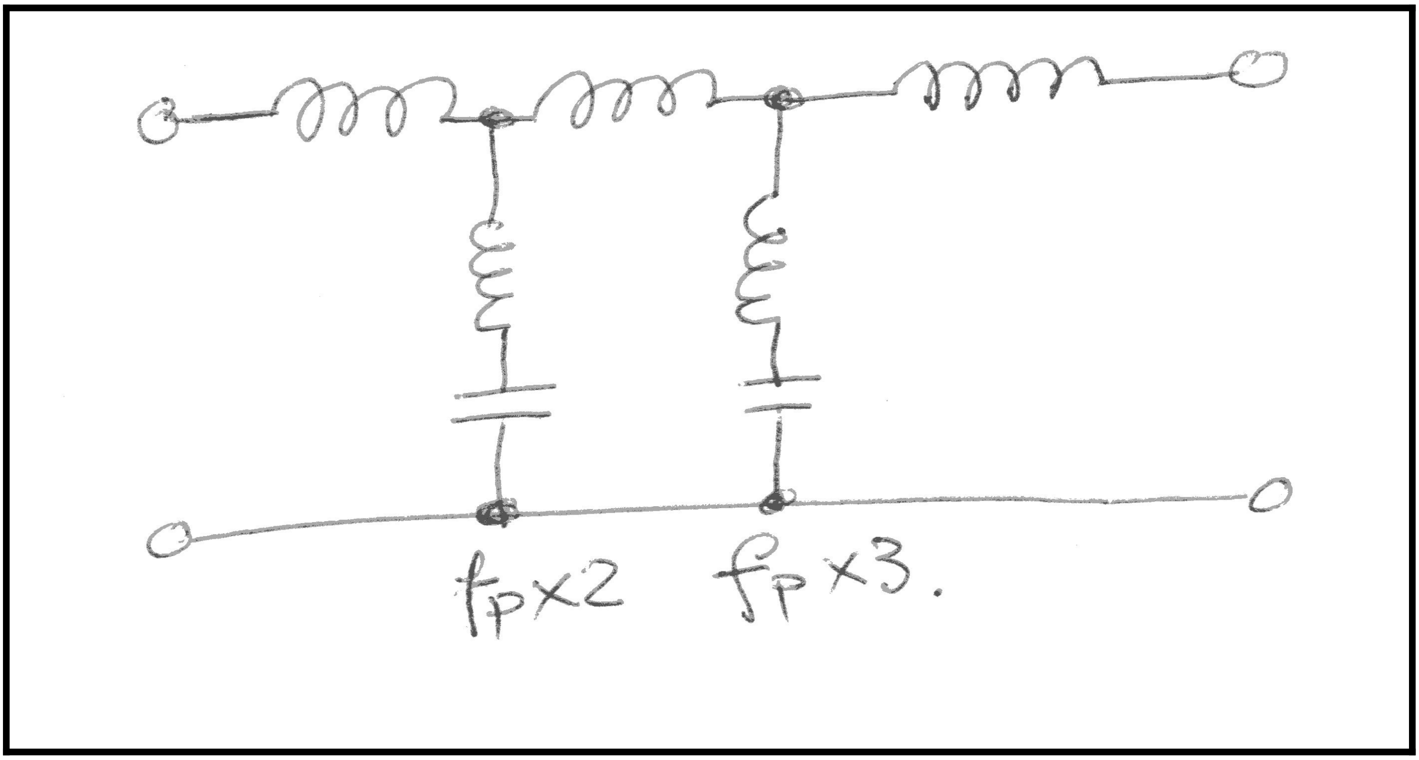
(22) Radio apparatus must only be tuned to the harmonised public protection and disaster relief frequencies for disaster relief radiocommunication purposes.
(23) The frequencies required by the licensee must be **selected in such a manner** that **no power is radiated at frequencies other than those referred to in the amateur radio frequency plan**, provided that the bandwidth of emissions on bands that have been allocated to the amateur radio service in terms of these regulations shall be **restricted to the minimum**.

Chapter 13 – Filters – prelude

- So you want to use a Raspberry Pi as a transmitter for WSPR...
- Or you want to try a DDS [Direct Digital Synthesizer] chip that can synthesize any frequency from 0.0Hz to 70 MHz...
- Or you want to make a Direct Conversion Receiver for the 40 metre band, so you can listen to the West Rand Bulletin on 7140 kHz [7.140 MHz]..

Yes both can produce a wave at these frequencies, but they all produce spurious frequencies and harmonics as well. To remove the **harmonics** you will need a **“Low Pass” Filter.**

For the DDS, you will need a **“Band Pass” Filter.** Or you will need a **“Low Pass” Filter** to remove the harmonics.

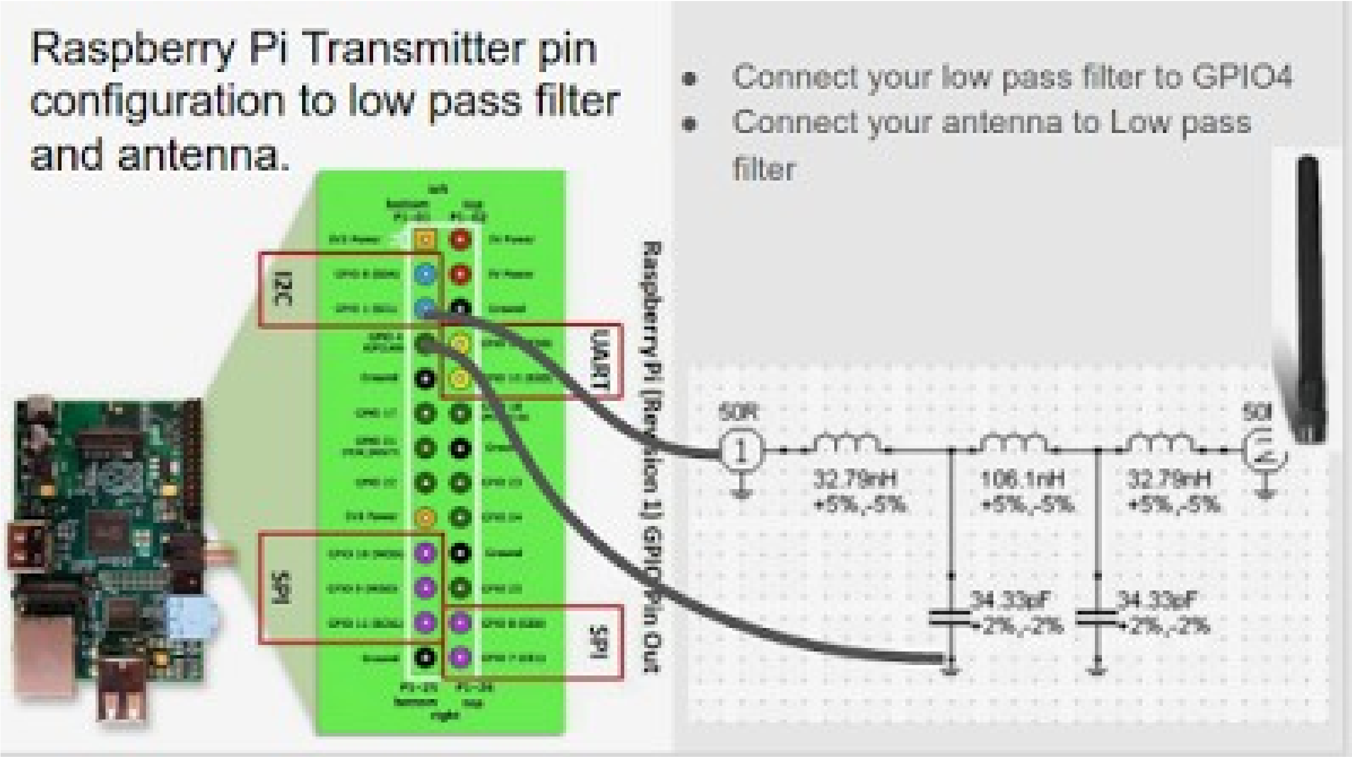


- To stop your 100 Watt H.F. transmitter from overloading a neighbour's [VHF/UHF] television set, you will need a **“High Pass” Filter**.

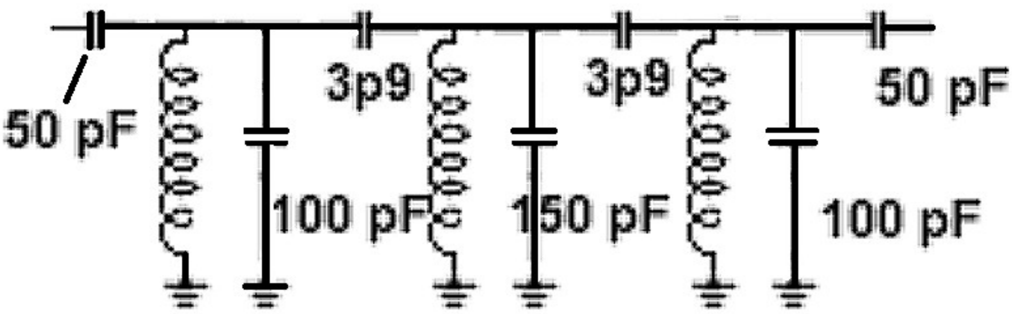
Television frequencies are VHF to UHF. That is **V**ery **H**igh

Frequencies or **U**ltra **H**igh **F**requencies. And your H.F. Transmitter is a maximum of 30 MHz. So a **“High Pass” Filter** will reduce the level of H.F. Signal to the television set.

Raspberry Pi Transmitter

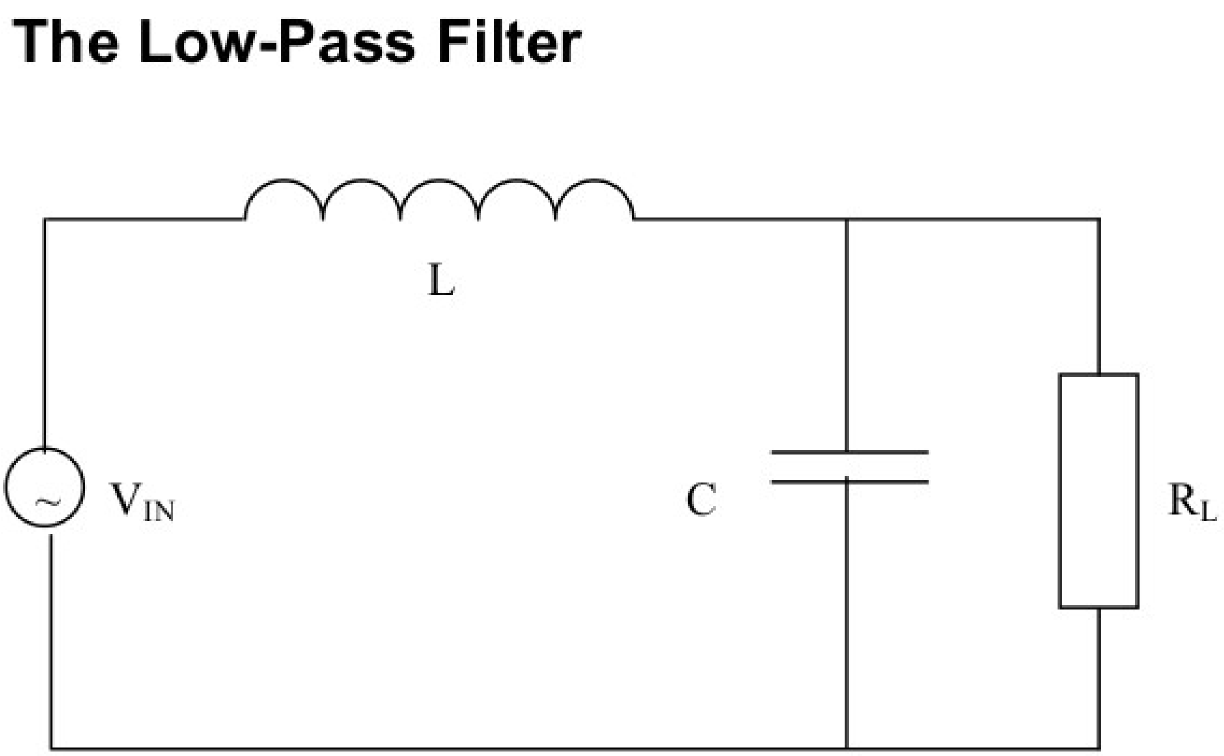


For the DC RX, you will need a **"Band Pass" Filter**, tuned to allow the 7.0 to 7.2 MHz frequencies to pass. To reject the very strong A.M. signal on 6.190 MHz from the BBC, you may also need a **"Band Stop Filter"** (tuned to 6.190 MHz).



Chapter 13 – Filters

Filters are electrical circuits that allow signals of particular frequencies to pass, while blocking signals of other frequencies. They can be used, for example, to select the signal that a radio receiver is tuned to, while blocking the signals that it is not tuned to.



An input voltage V_{IN} is applied across a voltage divider consisting of an inductor L and a capacitor C in parallel with a resistive load, RL.
[Ok, this is a very theoretical circuit. The generator (Oscillator) has a "source impedance/resistance". Usually of 50 Ohms. RL should "match" this source resistance with 50 Ohms.]

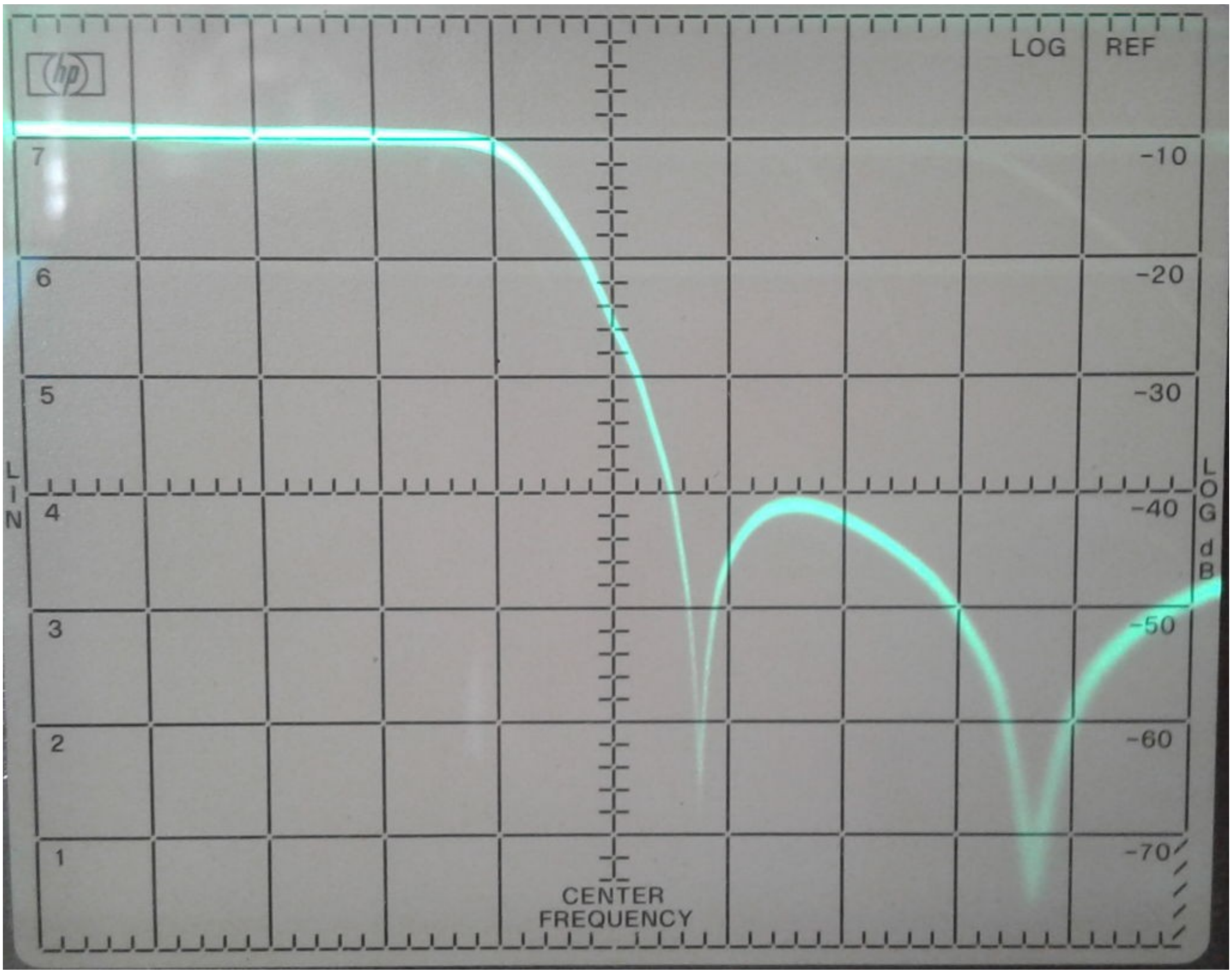
Although we are not in a position to analyse this circuit quantitatively, we can get a good qualitative idea of what happens. When the frequency of the input voltage is low, the inductor has low reactance while the capacitor has high (negative) reactance. This means there is little opposition to current flowing through L, but significant opposition to current flowing through C. As a result, most of the input voltage is applied across the load resistance RL, and power is efficiently transferred to the load.

Now consider what happens when the frequency is high. Since the reactance of an inductor is proportional to the frequency, L will have high reactance. On the other hand, the reactance of a capacitor decreases with frequency, to C will have a low impedance. This means that the inductor provides significant opposition to the flow of current; and what current is able to flow is mostly diverted through the capacitor rather than flowing through the load. As a result, little power is transferred to the load.

This circuit is called a "low-pass filter" because it allows low frequency signals to pass (in other words, to be efficiently coupled to the load), while blocking high frequency signals.

A graph can be plotted showing the frequency response of the filter – that is, its ~~gain~~ **[NO! NOT GAIN BUT LOSS! By inspection you will also realise that with just one L and one C, the suppression will not be very effective.]** at different frequencies.

To make the filter more effective, more inductors and more capacitors can be used to give higher suppression of harmonics. Like this 5 element Cauer Low-Pass Filter:-



The Frequency Response of a Low-Pass Filter

The cut-off frequency FC is the frequency at which the attenuation of the filter is 3 dB (i.e. the gain is –3 dB). At this frequency, half the input power reaches the load. For a low-pass filter, signals with frequencies lower than the cut-off frequency have relatively little attenuation; these signals are in the pass band of the filter.

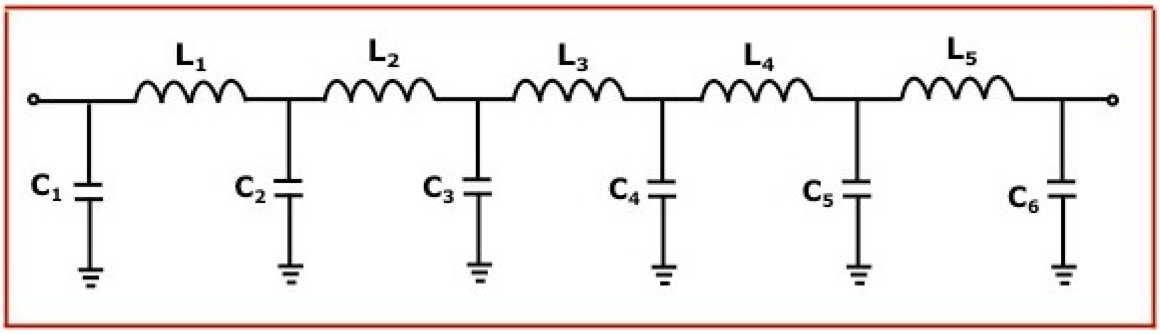
Signals with frequencies higher than FS are greatly attenuated – in this case by 60 dB or more. These signals are in the stop band of the filter. Signals with frequencies between FC and FS are somewhat attenuated. These frequencies are sometimes called the transition band of the filter since it is in transition between the pass band and the stop band.

Most amateur radio transmitters have a low-pass filter after the final power amplifier to attenuate any harmonics of the output frequency. Harmonics are multiples of the output frequency caused by distortion in the amplifier, so for example a transmitter that is transmitting on a frequency of 3.5 MHz might have harmonics on 7 MHz, 10.5 MHz, 14 MHz, 17.5 MHz, 21 MHz and so on. It is very difficult to design a power amplifier that does not generate any harmonics, and in any case such an amplifier would probably be quite inefficient. However it is easy to use a low-pass filter at the output to pass the desired frequencies and attenuate the harmonics to an acceptably low level.

Seriously – don't think you can just whip one up using a coil and a capacitor...
Who said that? These days there are many sites offering to design the filter for you...

Chebyshev Pi LC Low Pass Filter Calculator

Enter value, select unit and click on calculate. Result will be displayed.



Enter your values:

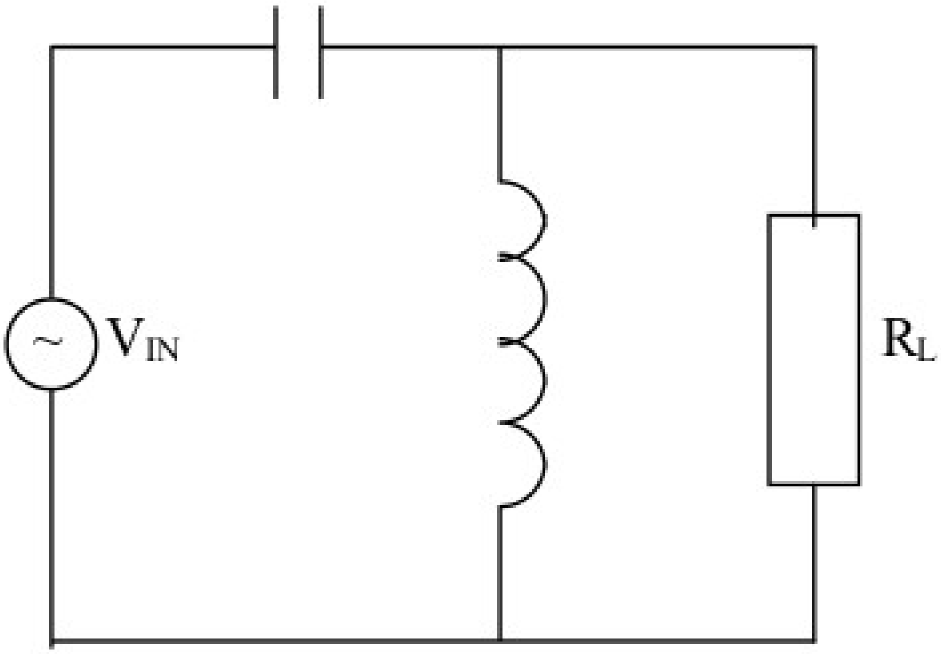
Cutoff Frequency:	50	MHz
Impedance Z0:	50	ohm
Frequency Response Ripple:	1	db
Number of Components:	11	(1-11)

Calculate   Clear

Results:

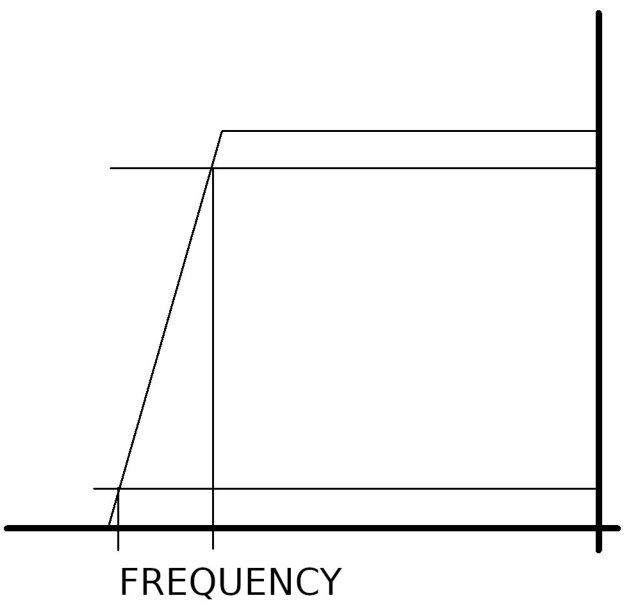
Inductance:		**Capacitance:**	
Unit :	nH	**Unit :**	pF
L_1:	178.7111	C_1:	139.1967
L_2:	190.2988	C_2:	199.5017
L_3:	191.6390	C_3:	203.5902
L_4:	190.2988	C_4:	203.5902
L_5:	178.7111	C_5:	199.5017
		C_6:	139.1967

The High-Pass Filter



Once again the input voltage VIN is applied to a voltage divider, but this time the capacitor and inductor in the voltage divider have been swapped. At low frequencies, the capacitor has high reactance and so opposes the flow of current; while the inductor has low reactance so the current that does flow is diverted through the inductor rather than flowing through the load.

At high frequencies, the capacitor has low reactance, so does little to oppose the flow of current. The inductor has high reactance, so most of the current flows through the load resistor RL rather than through the inductor. This circuit is called a "high-pass" filter because it allows high frequency signals to pass (in other words to be efficiently coupled to the load) while blocking low frequency signals. The frequency response of a high-pass filter looks something like this:

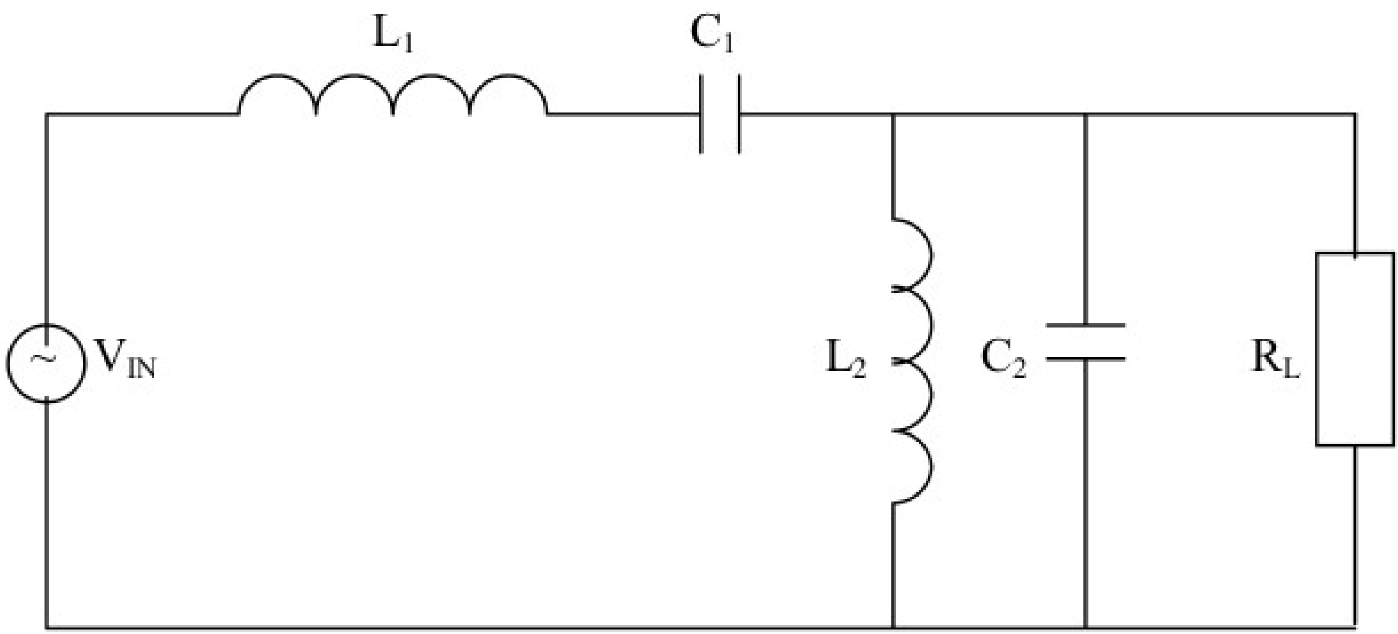


The Frequency Response of a High-Pass Filter

Once again, the cut-off frequency is the frequency at which the attenuation of the filter is 3 dB (the half-power point), while We have chosen to measure the stop-band from the point where the attenuation is 60 dB.

High-pass filters are often used in the input stages of receivers to reject the very strong radio signals found in the medium wave broadcast band from 500 kHz to 1,5 MHz so they do not overload the receiver, while allowing signals in the amateur bands starting at 1,8 MHz to pass.

The Band-Pass Filter



Band-pass filters pass signals in a certain frequency range known as the pass band and reject signals with frequencies above or below the pass band. They can be constructed using series and parallel tuned circuits. For example, consider the circuit below:

[Typical use for this type of filter is the band pass in the front-end of a receiver.]

Once again we have a circuit resembling a voltage divider, although this time it is made up of two tuned circuits – a series tuned circuit consisting of L1 and C1 in series with the source, and a parallel tuned circuit consisting of L2 and C2 across the load. Assume that the two tuned circuits have the same resonant frequency. Near this frequency, the series tuned circuit has low reactance while the parallel tuned circuit has very high reactance, so almost the entire input voltage appears across the load. This is the pass band of the filter.

However at frequencies well above or below the resonant frequency, the series tuned circuit has a high impedance while the parallel tuned circuit has a low impedance, so very little of the input voltage appears across the load. This is the stop band of the filter.

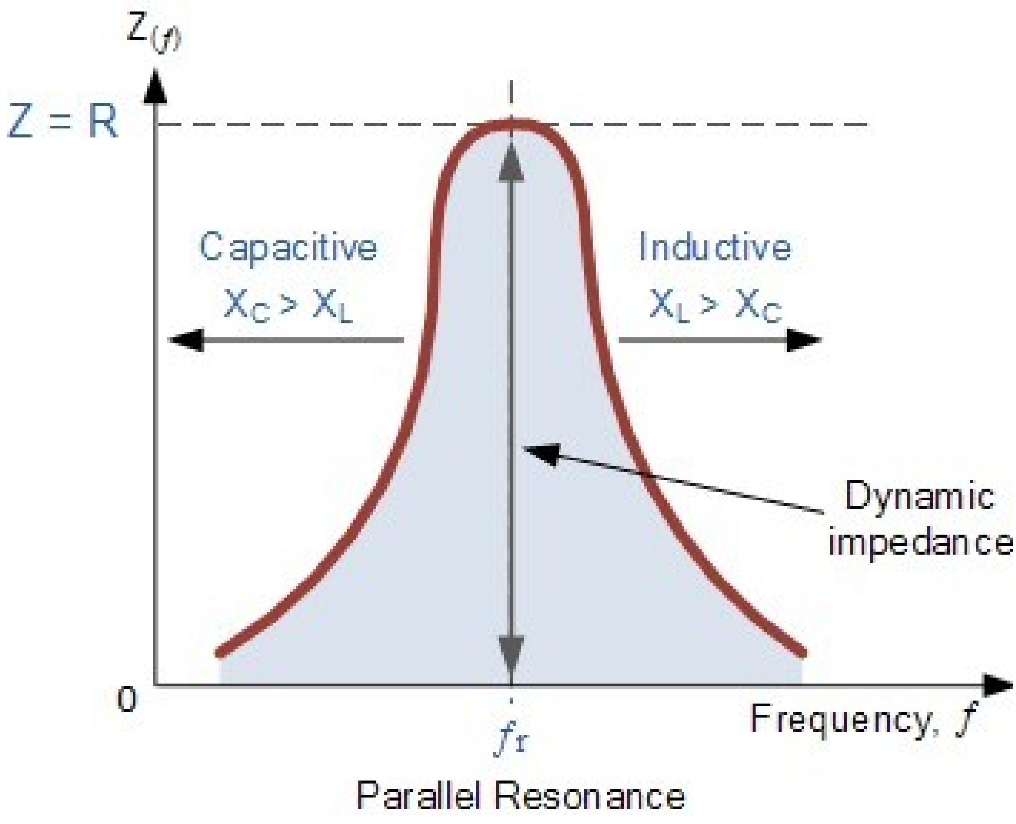
Bandwidth

[The "half power" bandwidth is -3dB down from the band centre frequency]

The Frequency Response of a Band-Pass Filter

The band-pass filter has two cut-off frequencies, a high cut-off labelled FH and a low cut-off labelled FL. Both cut-off frequencies are measured at the point where the output from the filter is 3 dB below the input to the filter (the half-power points). The bandwidth of the filter is the difference (in Hertz) between the high cut-off frequency and the low cut-off frequency. For example, if the high cut-off frequency is 2 700 Hz and the low cut-off frequency is 300 Hz then the bandwidth is 2700 – 300 = 2 400 Hz. The centre frequency of a band-pass filer is the frequency half way between the high cut-off frequency and the low cut-off frequency; in this case it would be 1500 Hz.

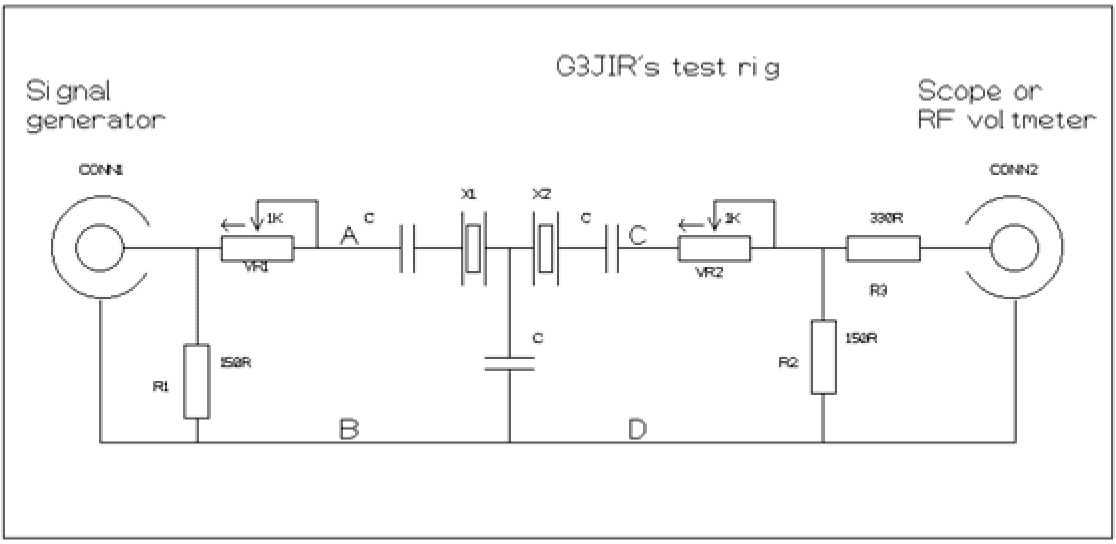
Most amateur receivers use band-pass filters to allow signals from a particular amateur band to enter the receiver while rejecting signals from other amateur bands. This is called a pre-selector.



Crystal Filters

[Band-pass filters](#) can also be implemented using quartz crystals. These have a piezoelectric property, which means that a voltage applied to the crystal causes a slight physical movement of the crystal; and physical movements of the crystal will in turn cause a voltage to appear across it. Quartz crystals have very similar properties to tuned circuits and can be used to make highly selective band-pass filters. These “crystal filters” are responsible for the **selectivity** – that is, the ability to distinguish one signal from another – of many modern amateur receivers and transceivers.

Although crystal filters are very selective – that is, their bandwidth is very narrow in comparison with the centre frequency of the filter – they have the disadvantage that they only work at a single fixed frequency. That is, a crystal filter cannot be tuned to different frequencies. When we look at the design of super-heterodyne receivers we will see how this limitation is overcome while allowing the receiver to take advantage of the exceptionally good **selectivity** of crystal filters.



Amateur receivers and transceivers often allow you to select different bandwidth crystal filters for different purposes. Some of the common bandwidths are 2,4 kHz for normal phone (SSB) operation, 1,8 kHz for phone operation under difficult conditions (often used in contests) and between 250 Hz and 500 Hz for CW (Morse Code) operation. Most transceivers come with one or two basic filters (for example, just a 2,4 kHz filter) but additional filters can often be purchased, although they can be quite expensive.

The Band-Stop Filter

A band-stop filter works in the opposite way to a band-pass filter. Frequencies in a certain range (the stop-band) are attenuated, while frequencies either above or below those frequencies are passed. Amateur receivers and transceivers often provide a manually adjustable band-stop filter that can be used to attenuate undesired signals, for example a carrier generated by someone tuning up close to the frequency that you are listening to. These are known as **“notch filters”** because they allow you to **“notch out”** undesired signals.

[One very useful aspect of this filter is to notch out a single frequency, such as 1 kHz, to allow the distortion and harmonics to be measured in the output of an amplifier.]

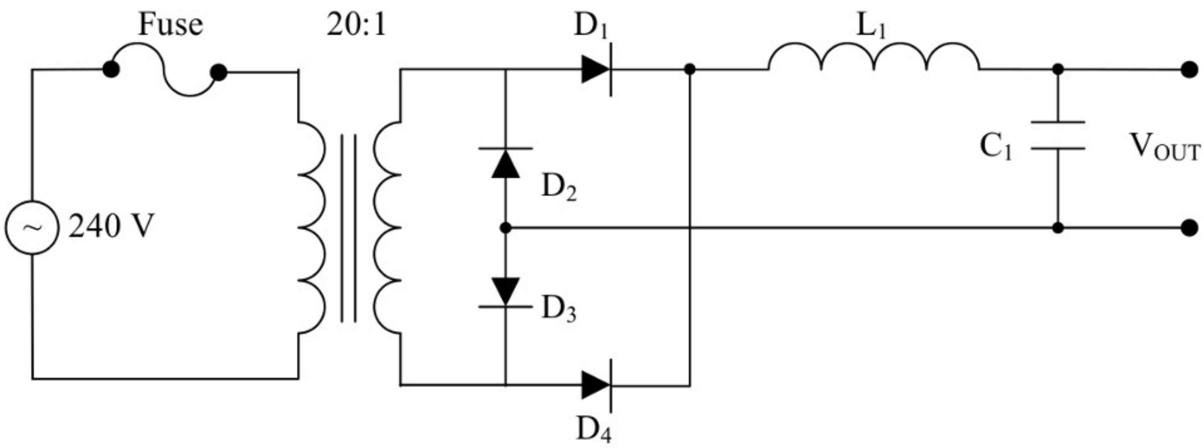
Summary

Low-pass filters allow signals with frequencies below the cut-off frequency to pass with little attenuation, while significantly attenuating signals with frequencies well above the cut-off frequency. High-pass filters allow signals with frequencies above the cut-off frequency to pass with little attenuation, while significantly attenuating signals with frequencies well below the cut-off frequency. In both cases, the cut-off frequency is measured from the point where the signal is attenuated by 3 dB; this is also known as the "half power" point.

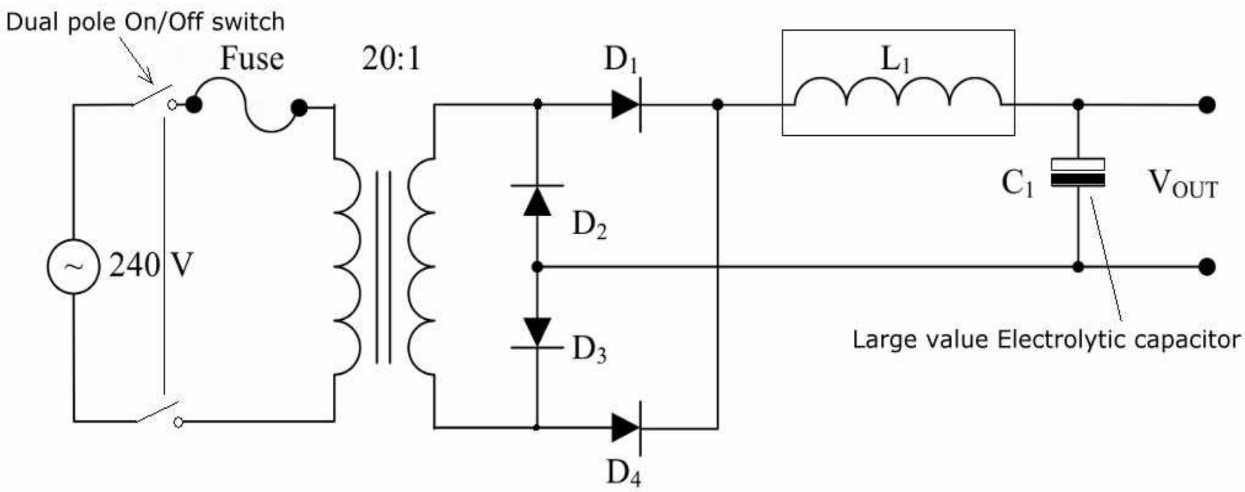
Band-pass filters allow signals with frequencies between the low and high cut-off frequencies to pass, while attenuating signals with frequencies significantly higher or lower than the passband. The bandwidth of a band-pass filter is the difference between the high cut-off and low cut-off frequencies. Crystal filters are highly selective band-pass filters. Band-stop filters attenuate signals with frequencies in a particular range, while allowing signals outside that frequency range to pass.

Chapter 16 - The Power Supply

The circuit below shows a simple unregulated D.C. power supply that is easy to construct.



<u>You should recognise the various parts of the circuit and understand their function.</u>



Same circuit with modifications [jb]

The power supply takes a 240 V A.C. mains input and applies this to the primary winding of a 20:1 transformer through a fuse. The fuse is there to protect the circuit if too much current is drawn, either by overloading the output or due to a circuit fault.
[NOTE: A fuse works by getting too hot such that the wire melts. See 'How does a fuse work'.]

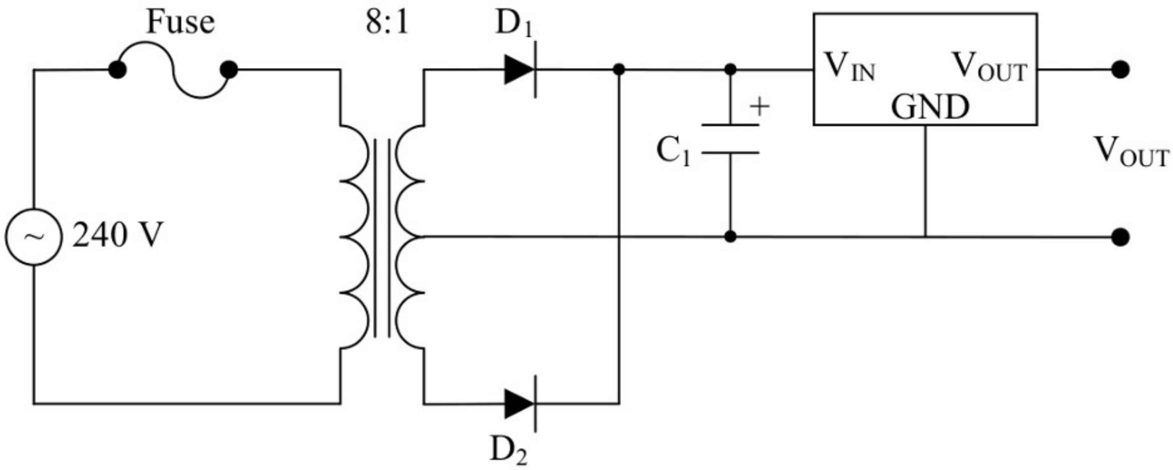
The transformer steps the voltage down to 12 V RMS. This voltage is rectified by a full-wave bridge rectifier consisting of diodes D1 – D4. The resulting waveform is passed through a low-pass filter consisting of inductor L1 and smoothing capacitor C1, which reduce the ripple to an acceptably low level.

The inductor also provides "inrush protection", preventing the transformer from being damaged by the high current that would flow when the supply was first turned on, when the capacitor is completely discharged, if there was no inductor. Instead of allowing a very high current to flow initially, the inductor opposes the change in current, allowing it to build up more gradually. In practise this inductor is often omitted in simple designs, with the selfinductance of the transformer secondary being sufficient to prevent damage.

A Regulated Power Supply

Although the simple power supply is quite practical, it has two weaknesses. First, although the ripple is significantly attenuated by the low-pass filter on the output, some ripple will remain, which may cause problems with sensitive equipment. Second, although the output voltage is nominally 12 V, in practise it will vary between 11 and about 16 V depending on the circuit load, which again may cause problems for sensitive equipment.

Both of these problems can be solved by adding a voltage regulator to the basic design.



Although a Zener diode could be used as a voltage regulator, they are only suitable for low current applications, so they are typically used to stabilize a reference voltage that is then used to control the output voltage of the voltage regulator. Although the entire regulator can be (and often is) made from discrete components like diodes, capacitors and transistors, we will use an integrated circuit that is specifically designed as a voltage regulator. An integrated circuit consists of a number of different electronic devices all fabricated (made) and interconnected together on a single wafer of silicon.

They are available to perform many common tasks, including voltage regulation.

You will notice that the rectifier design is different. Instead of using four diodes in a bridge circuit, this design only uses two diodes. However it still achieves full-wave rectification by making use of a centre tap on the secondary winding of the transformer. This is just a separate connection to the middle of the secondary winding.

This allows the secondary to function almost like two separate windings. On each half cycle either D1 or D2 will conduct, but not both. Whichever diode conducts will connect the positive side of the transformer to the positive side of C1. Current flowing back to the centre tap of the secondary completes the circuit. In effect, only half of the secondary winding is used in each half cycle.

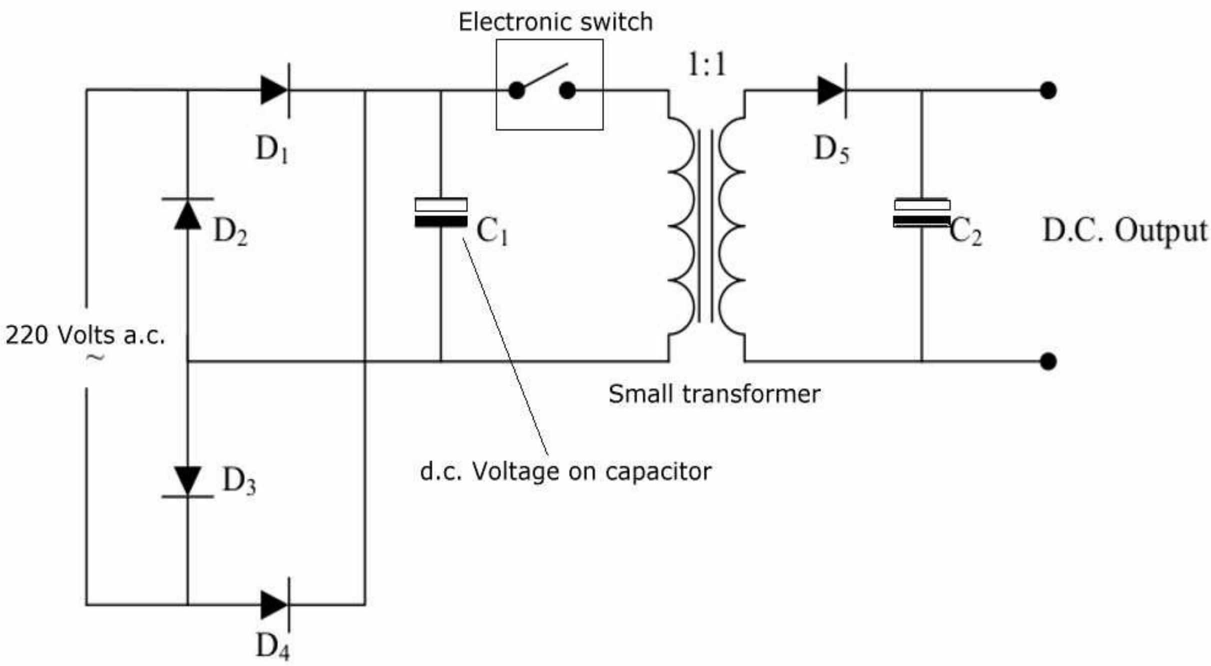
The voltage regulator has three terminals labelled VIN, VOUT and GND (for "ground").

It acts a bit like a variable resistor that is connected between the VIN and VOUT terminals and which is continually adjusted to maintain the voltage between the VOUT and GND terminals at a constant level, say 12 V. As well as regulating the voltage, the voltage regulator also substantially reduces the ripple, since it is able to change its internal resistance fast enough to counteract the voltage fluctuations due to the ripple, thus maintaining a good "clean" D.C. supply despite considerable ripple in the input voltage. For this reason We have also simplified the low-pass filter by removing the inductor, leaving only the smoothing capacitor C1, which should provide adequate ripple rejection when used with a voltage regulator integrated circuit.

Most integrated voltage regulators provide another benefit: they automatically limit the current and power dissipation of the regulator to safe limits, avoiding damage to the power supply even if the load is short circuited. Despite this the mains supply of a power supply (or any other mains powered device) should always be fused in case of a short circuit within the power supply itself.

Switching Power Supplies

Power supplies that use a transformer to reduce the voltage followed by a rectifier and a voltage regulator are called linear power supplies. An alternative design is the switching power supply. Instead of using a transformer to reduce the voltage, these supplies rectify the mains voltage to generate a high voltage D.C. voltage source. This is then switched on and off at a high frequency using a fast solid-state switch, and the resulting waveform is fed through a low-pass filter to filter out the A.C. switching components.



Simplified Circuit Diagram of a Switching Power Supply

The 240V A.C. mains supply is rectified by the full-wave bridge rectifier consisting of diodes D1 to D4 and smoothed by C1 to generate 338 V D.C. This is switched on and off at a frequency of 100 kHz or so by a high-speed electronic switch, which is shown in the circuit diagram as a switch. The resulting high frequency A.C. waveform is fed into the primary of the isolating 1:1 transformer. The purpose of this transformer is to prevent the D.C. output from being connected to the mains supply, rather than to perform any voltage conversion. The voltage on the secondary of the isolating transformer is half-wave rectified by D5 and smoothed by C2 to give a D.C. output.

The output voltage of the power supply depends on how much time the switch spends in the “on” position compared to how much time it spends in the “off” position. If the switch spends only a small percentage of its time in the “on” position, then only a little power will be transferred to the transformer, and the output voltage will be small. If it spends a lot of time in the “on” position then a lot of power will be transferred and the output voltage will be high. In actual practice, the amount of time that the switch spends in the “on” position is controlled by electronics (not shown n the diagram) that continually monitors the output voltage and adjusts the duty cycle of the switch in order to maintain a constant output at the desired voltage; this is an example of feedback.

All this is pretty complex compared to a simple linear supply, so there must be some significant advantages to make it worthwhile. The main advantages of the switching supply are that

1. It dissipates very little power. The switch is either “on”, in which case there is current flowing through it but little voltage across it; or “off” in which case there is a high voltage across it but no current flowing through it. In either case the power dissipation is minimal compared to the linear voltage regulator, which has a voltage drop across it and a current flowing through it at the same time, and so is continuously dissipating power.

2. Because the transformer in a switching supply operates at a very high frequency usually around 100 kHz instead of the 50 or 60 Hz of a standard mains supply - it can be physically very small and light. This is because the size of the core required in a transformer decreases as the frequency increases.

As a result, switching power supplies are generally smaller, lighter and more efficient than their linear counterparts, and can often run off any mains voltage without having to change a selector switch. However they also have their disadvantages. In particularly, poorly designed switching supplies can generate a lot of radio frequency interference, which is a real problem for amateur radio applications. However well designed and properly shielded switching supplies do not necessarily cause interference. Because of their high power requirements and space limitations, virtually all personal computers use switching power supplies (and in fact some of these supplies can be adapted as general purpose power supplies for amateur use).

Please note that switching supplies are very difficult to design and build and can be quite dangerous due to the high voltages they use, and the switching circuitry is often connected directly to the mains input supply. Although the linear power supplies in earlier sections make good projects for amateurs, the design and construction of switching power supplies should be left to professionals!

Summary

Linear power supplies use transformers to reduce the mains voltage, rectifiers to convert the A.C. to D.C. and an output filter including a smoothing capacitor to reduce the ripple to acceptable levels. In power supplies that use a half-wave rectifier the ripple is at the same frequency as the mains, while in power supplies that use full-wave rectifiers the ripple is at twice the mains frequency.

A voltage regulator serves two purposes: to keep the output voltage constant despite fluctuations in the input voltage or load current; and to further reduce ripple.

Integrated circuit voltage regulators may also limit current and power dissipation by the regulator to safe levels.

Switching power supplies work by rectifying the mains supply and then switching it on and off at a high frequency. The voltage output is regulated by changing the duty cycle of the switching waveform; that is, the percentage of the time that the switch is “on”. Although most switching supplies still use transformers to isolate the output from the mains supply, these transformers can be small and light because they operate at high frequency, typically around 100 kHz. The advantages of switching supplies are that they are smaller, lighter and more efficient than linear supplies.

However poorly designed switching supplies can generate a significant amount of radio frequency interference.

Chapter 19 - The Oscillator

The Electronics Curse
“Your amplifiers will oscillate, your oscillators won't!”

19.1 Question, What is an Oscillator?

A Pendulum is an Oscillator...

“Oscillators are circuits that are used to generate A.C. signals. Although mechanical methods, like alternators, can be used to generate low frequency A.C. signals, such as the 50 Hz mains, electronic circuits are the most practical way of generating signals at radio frequencies.”

Comment: Hmm, wasn't always. In the time of Marconi, generators were used to generate a frequency to transmit. We are talking about kiloWatts! e.g. Grimeton L.F. Transmitter.

Oscillators are widely used in both transmitters and receivers. In transmitters they are used to generate the radio frequency signal that will ultimately be applied to the antenna, causing it to transmit. In receivers, oscillators are widely used in conjunction with mixers (a circuit that will be covered in a later module) to change the frequency of the received radio signal.

19.2 Principal of Operation

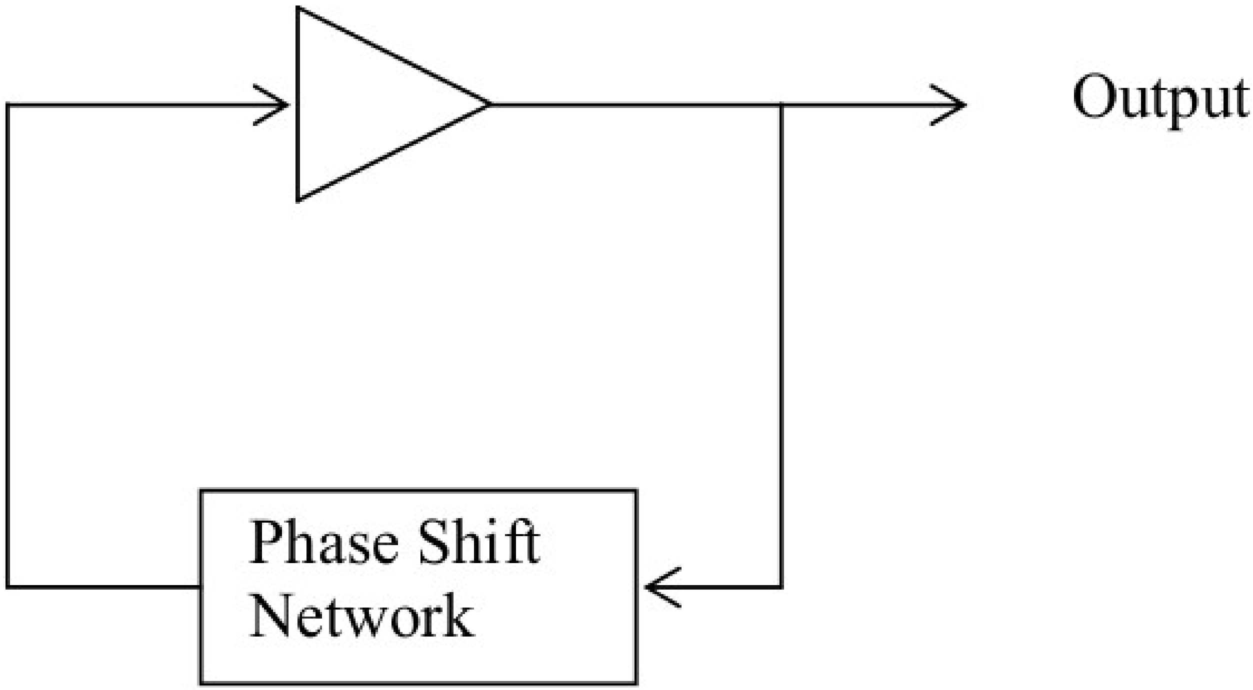
The diagram below is a block diagram showing a typical oscillator. Block diagrams differ from the circuit diagrams that we have used so far in that they do not show every component in the circuit individually. Instead they show complete functional blocks – for example, amplifiers and filters – as just one symbol in the diagram. They are useful because they allow us to get a high level overview of how a circuit or system functions without having to show every individual component.

Comment: The author forgot the simplest form a multi-vibrator. Described as “an electronic see-saw”. He also 'forgot' the Pierce Oscillator, the simplest form of crystal oscillator...

There are an awful lot of oscillators. With a great many specific applications. In the Radio Amateurs world, they mostly run as Variable Frequency Oscillators [V.F.O.].

Another way of generating a sine wave is to "shock excite" a tuned circuit.

An Oscillator [Block Diagram]



Block Diagram of an Oscillator

19.3 Block Diagram of an Oscillator

The triangular symbol at the top represents an amplifier. The input of the amplifier is the blunt side of the triangle, on the left in this diagram; the output is the pointy side of the triangle. Since this symbol always represents an amplifier, there is no need to label it. The output of the amplifier is connected to the input of the block labelled “phase shift network”, and the output of the phase shift network is connected back to the input of the amplifier.

(Since the rectangular box of the phase shift network does not indicate the input and output, you must surmise this from the directions of the arrows on the connecting lines.) The output of the oscillator is taken from some point in the circuit - in this diagram We have shown it being taken from the output of the amplifier.

The lines connecting the symbols in the block diagram represent the flow of signals from one functional block to another. In this type of diagram, a line does not necessarily represent a single wire, as it would in a circuit diagram. A signal might flow along a single wire (with respect to earth), or it might flow in two wires, with the current flowing in opposite directions in both wires. In either case, it could be represented by a single line in a block diagram. The arrows at the end of the lines show the direction that the signal flows in – in this case, from the output of the amplifier to the input of the phase shift network, and from the output of the phase shift network back to the input of the amplifier. The direction in which the signal is flowing does not in general correspond with the direction in which current is flowing – after all, most of the signals we deal with will in any case be A.C. so current flows in both directions.

So how does this circuit oscillate? When it is initially turned on, there will be some (very small) thermal noise present in the circuit. This type of noise is generated by the random motion of electrons due to heat, and exists in all conductors. Thermal noise is broadband in nature, meaning that it includes frequency components at all possible frequencies. (When you turn the volume of a Hi-Fi amp up without any input signal, the hiss you hear is the audio frequency component of the thermal noise. If you hear a hum, this is mains pick-up, not thermal noise.)

Thermal noise at the input to the amplifier will be amplified, causing a larger noise signal at the output of the amplifier, some of which is bled off to the output, and some of which is applied to the phase shift network. The phase shift network does what its name implies – it changes the phase of the input signal, so the output of the network will have a phase that either leads or lags the input signal. The phase relationship between the output and the input depends on the precise frequency of the input signal.

At most frequencies, the output of the phase shift network, which is fed back into the amplifier, will not be at precisely the same phase as the noise component that caused it in the first place. In this case, the signal that is "fed back" to the input of the amplifier from the phase shift network will partially cancel out the signal that caused it, so the noise components at these frequencies will die out. However at one frequency, the output of the phase shift network will be exactly in phase with the noise component that caused it, and so it will reinforce that particular frequency component of the noise signal at the input to the amplifier.

This reinforced signal will again be amplified by the amplifier, phase shifted by the phase shift network, and fed back to the input of the amplifier. And once again, the output from the phase shift network is precisely in phase with the input signal from the "last round" that caused it, and so the signals reinforce each other and keep on growing.

Of course the signal cannot grow larger forever. As the signal grows bigger, ultimately the gain of the amplifier will be reduced (for example, it may be limited by the power supply voltage to the amplifier) until we reach the point that the amplified signal that is passed through the phase shift network and back to the input of the amplifier is only just as strong as the input to the amplifier that caused it. At this point, the signal is no longer growing, but remains constant and we have reached a stable oscillating state. If the oscillator has been designed correctly, then the output will be a constant amplitude signal at the desired frequency.

19.4 Positive Feedback

Feeding back some of the output of the amplifier back to the input in such a way that it reinforces the original input signal is called **positive feedback**. This is the same effect that you get when the audio output of a PA system is fed back to the microphone creating "howl-round" or "feedback".

The Barkhausen Criteria for Oscillation [That's NOT 'dog box' in German!]

Comment: In electronics, the **Barkhausen stability criterion** is a mathematical condition to determine when a linear electronic circuit will oscillate.[1][2][3] It was put forth in 1921 by German physicist Heinrich Georg Barkhausen (1881–1956).[4] It is widely used in the design of electronic oscillators, and also in the design of general negative feedback circuits such as op amps, to prevent them from oscillating..

The loop gain of an oscillator is the total gain that the signal experiences starting from any point in the circuit and going around the loop until it gets back to the starting point. For example, suppose the amplifier has a gain of 10 dB, that half the power is "bled off" to the output (resulting in a loss of 3 dB), and that the phase shift network also has a loss of 3 dB.

Converting the losses into negative gains, we get the following figures:

Amplifier	10dB
Loss of output signal	-3dB
Phase shift network	-3dB
Total loop gain	4 dB

Similarly, you can calculate the total phase shift around the loop. The amplifier will contribute some phase shift, and the phase shift network will contribute some more. Even the interconnecting wires may contribute significant phase shift at high frequencies – for example, the wavelength of a 100 MHz signal is 3 metres, so every centimetre of connecting wire would contribute a phase shift of about 1.2˚!

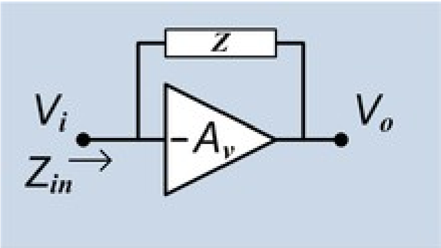
When the oscillator is oscillating stably – that is, with constant amplitude and frequency, the following criteria must be fulfilled:

The loop gain must be exactly 1. If the gain was more than 1, then the amplitude of the output would be increasing. If less than 1, then the amplitude would be decreasing.

The total phase shift around the loop must be 0 or an integer multiple of 360˚. This is necessary for the signal to reinforce itself as it goes around the loop, so it does not cancel itself out.

These requirements are known as the Barkhausen criteria for oscillation.

It is entirely possible for these criteria to be met at more than one frequency. In particular, it is easy for the phase requirement to be met, since it only specifies a phase shift of 0 or any integer multiple of 360˚, so it could be satisfied for different frequencies that had a total phase shift around the loop of say 0˚, 360˚ and 720˚. If both criteria are met for several frequencies, then oscillator will oscillate at all these frequencies simultaneously, which is usually not the desired result! Oscillations at undesired frequencies are called **parasitic oscillations**.



Effect of parasitic capacitance *Z* = *C* between the input and output of an amplifier.

MILLER Capacitance.

Instability – in Oscillators

In order to minimize the chance of this happening, the phase shift network is usually also made frequency selective, so that it will pass frequencies in the region of the desired frequency of oscillation, while attenuating frequencies that are higher or lower than this. In other words, it is made to be a band-pass filter as well as a phase shift network. The advantage of this is that even if the phase shift criterion is met for some other frequencies, as long as they are far enough away from the desired frequency, they can be attenuated sufficiently by the band-pass characteristic of the network to ensure that the loop gain remains less that 1 so oscillation will not occur at these unwanted frequencies.

Fortunately there is a simple circuit that provides both a phase shift and band-pass filter characteristics simultaneously: **the parallel tuned circuit**. At the resonant frequency the reactance of a parallel tuned circuit changes rapidly from being highly inductive just below the resonant frequency to being highly capacitive just above the resonant frequency. This sudden change in reactance results in a change in the phase relationship between the voltage across the tuned circuit and the current flowing through it (remember that for in inductive reactance, voltage leads current, while for a capacitive reactance current leads voltage). At the same time, the parallel tuned circuit can be used to provide good **band-pass filter** characteristics, minimizing the likelihood of parasitic oscillation.

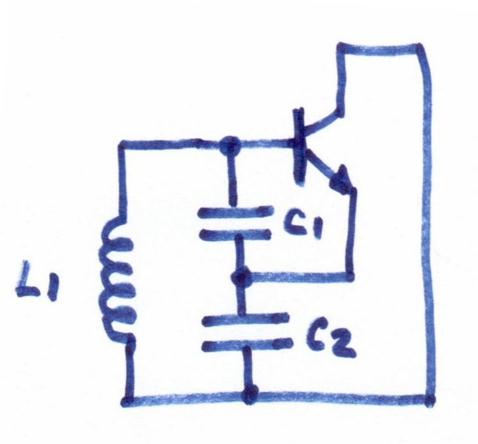
An oscillator that uses a tuned circuit as its phase shift network will oscillate at (or very close to) the resonant frequency of the tuned circuit.

Comment: It also depends on the 'QUALITY' or ‘Q’ of the tuned circuit.

The Colpitts Oscillator

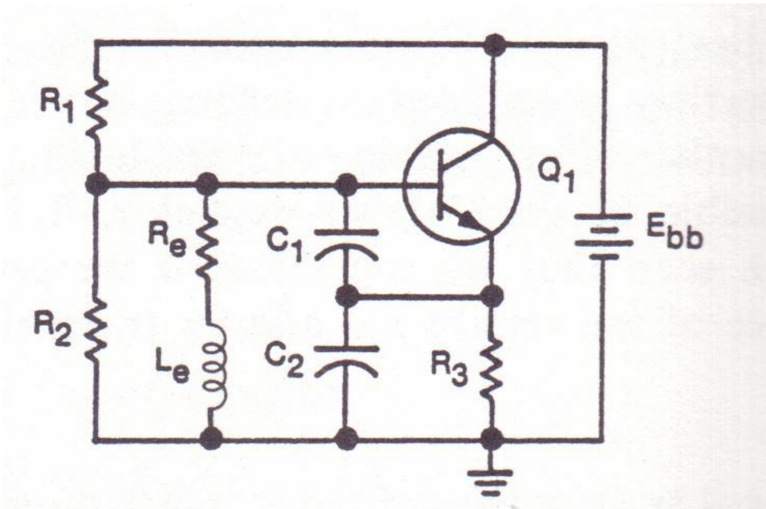
Note: A **Colpitts oscillator**, invented in 1918 by American engineer Edwin H. Colpitts

Note: This is where I diverge from the course notes a bit. I want you to see how I was taught in college about oscillators. First of all we need to see the "a.c. circuit" for a Colpitts Oscillator.

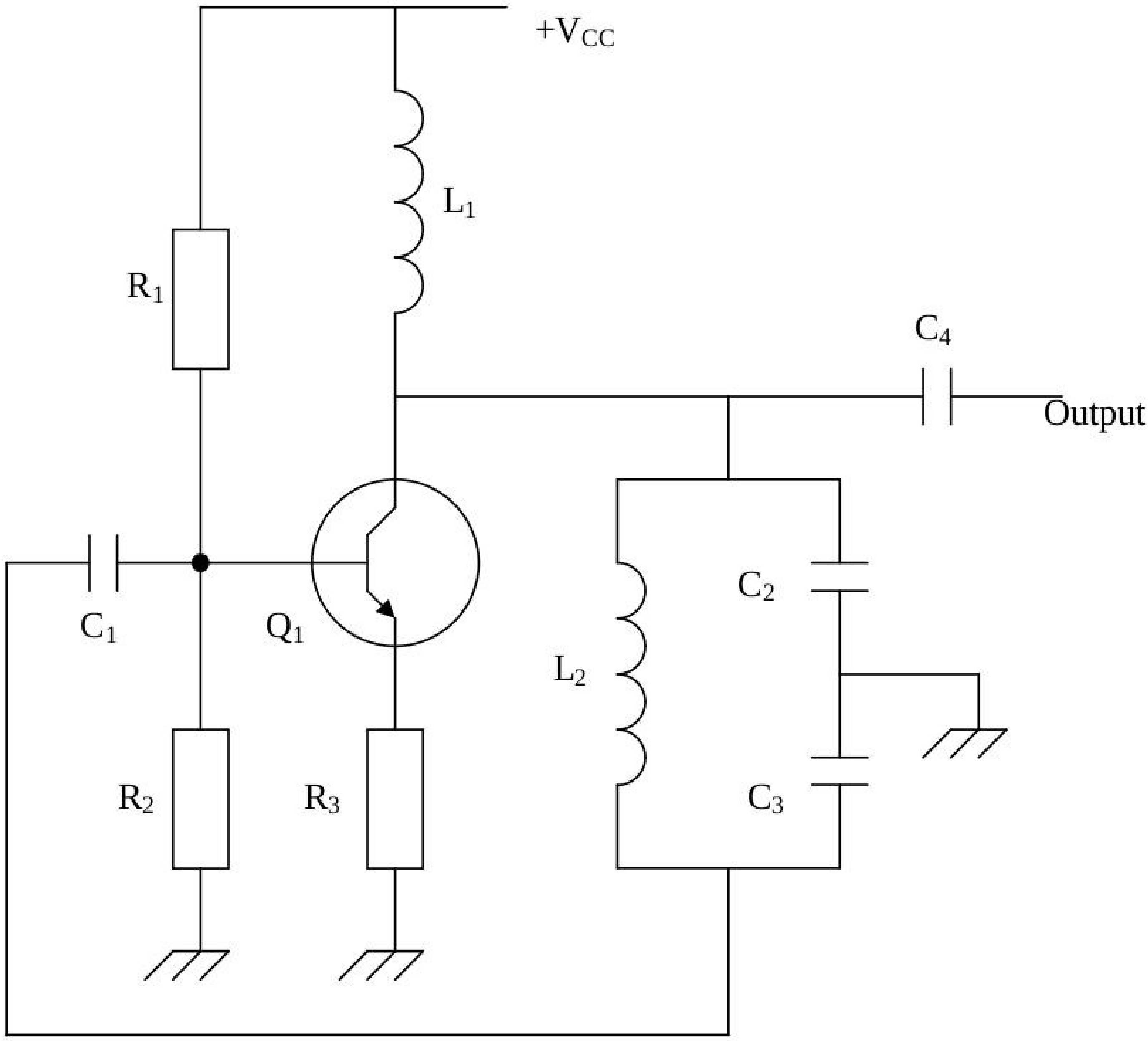


So you can quickly identify the circuit by the fact that the capacitors are split and connected across the coil.

The Colpitts oscillator is typical of how these concepts can be implemented in a practical circuit. Hmm, yes...



See the capacitors are split and connected across the coil. BUT even this one in a text book has an error!
If you connect the coil and Re from base to ground, it will short out the bias voltage for the transistor.



A Colpitts oscillator

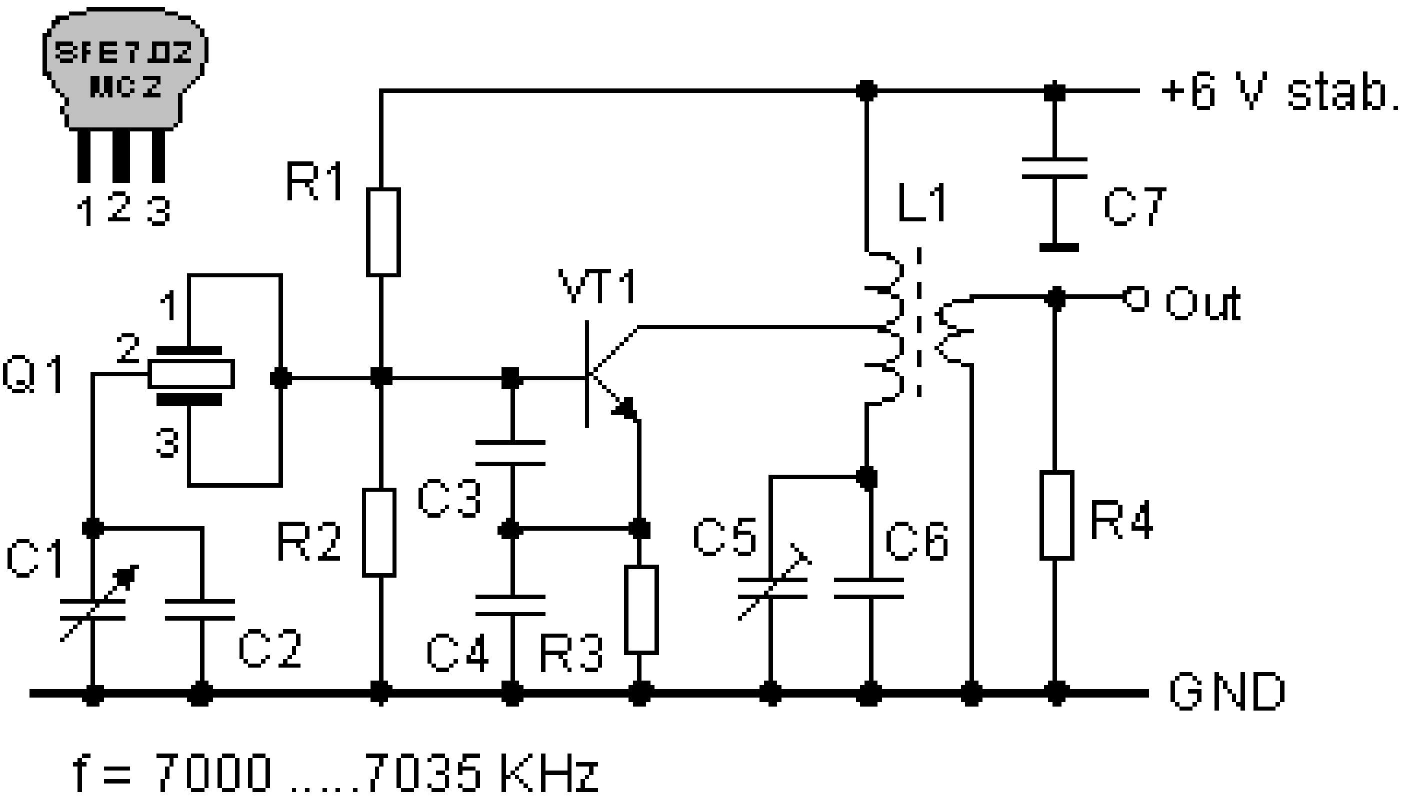
Circuit Diagram of a Colpitts Oscillator

Transistor Q1 and the associated components R1, R2, R3 and L1 form a common-emitter amplifier. The output of the amplifier, taken from the collector of Q1, is fed into a parallel tuned circuit consisting of L2, C2 and C3. The capacitor in this tuned circuit has been “split” into two capacitors, C2 and C3, to allow the output current from the collector of Q1 to flow to ground via C2. This causes a voltage across the whole parallel tuned circuit (also known as the tank circuit of the oscillator). This voltage is fed back to the input of the amplifier via C1.

The output of the oscillator is taken from the collector of the transistor via C4. The label “VCC” represents the positive power supply voltage.

The defining characteristic of the Colpitts oscillator – i.e. what makes it a Colpitts oscillator as opposed to any other type of oscillator – is the way the tank circuit (the parallel tuned circuit) uses a split capacitor to allow the output of the amplifier to be injected across one of the capacitors, while the input to the amplifier is taken from across the other capacitor.

Why is there an L1 in the collector circuit? This would not be manufactured by an electronics company this way! The circuit would be 'turned around' again and the output taken from the emitter circuit to a buffer transistor.



A 'real' Circuit – Colpitts Oscillator - very much like a Clapp Oscillator

Buffering

Because the amount of signal that is drawn off by the output of the oscillator affects the loop gain of the oscillator, it will also affect the frequency of the oscillator. For this reason it is important that the amount of signal drawn off does not change, for example in response to a Morse code (CW) transmitter being keyed, otherwise the frequency of the transmitter will change as it is keyed, a phenomenon known as “chirp”. Most transmitter designs prevent this by having a buffer amplifier between the oscillator and the keyed stages of the transmitter.

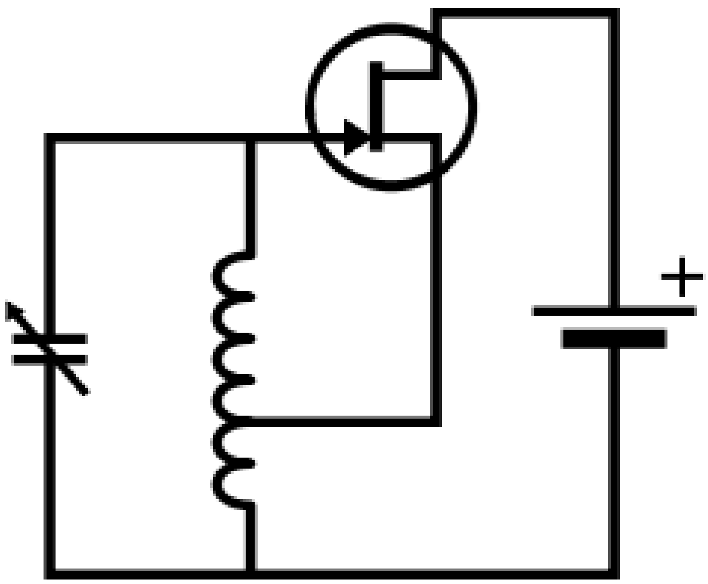
The buffer amplifier is often a common-collector (emitter follower) amplifier, which shows constant high impedance to the oscillator while having a low output impedance that can supply sufficient current to drive the stages that follow.

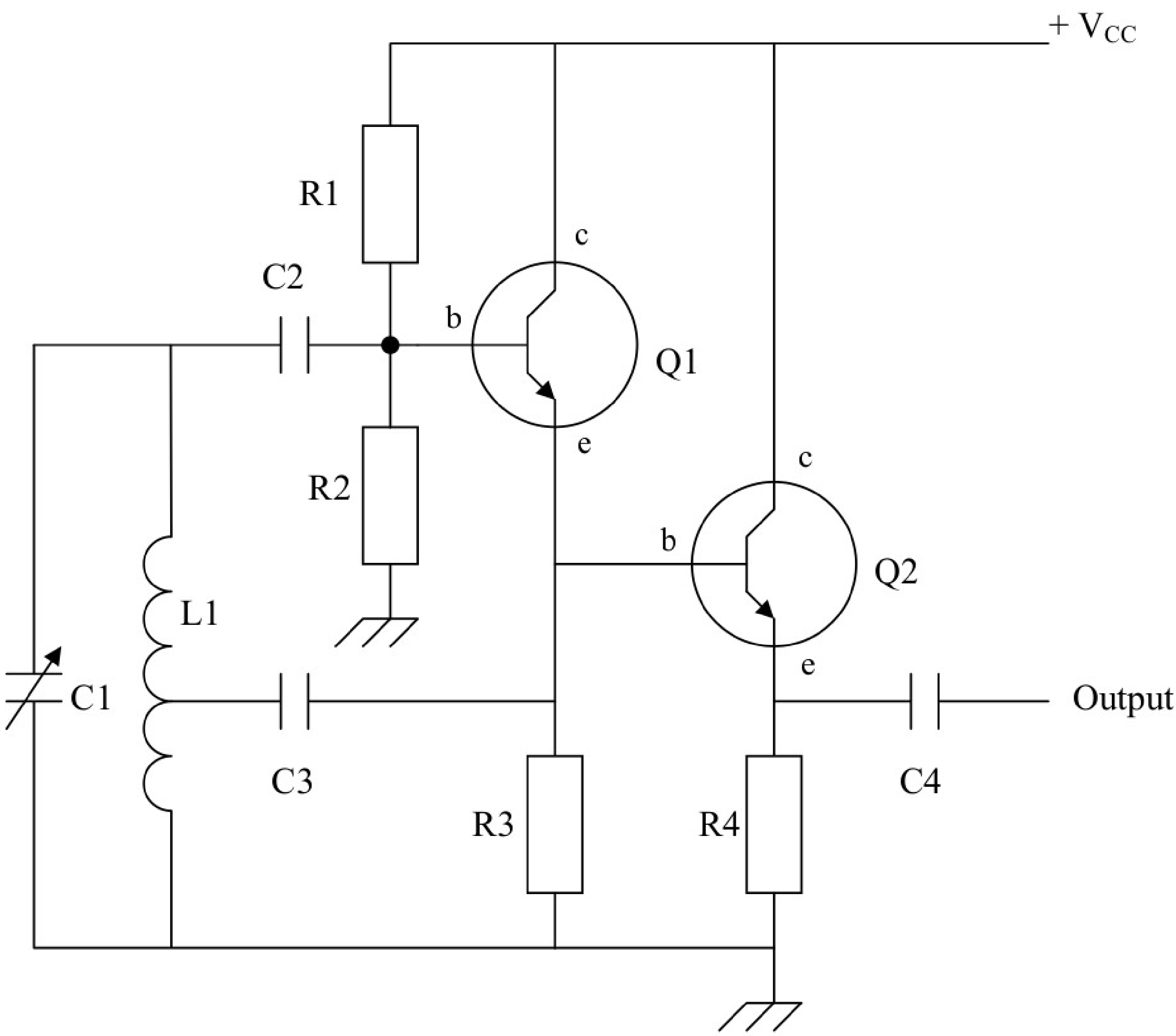
The Hartley Oscillator

Note: The circuit was invented in 1915 by American engineer Ralph Hartley.

Another way of feeding the output of the amplifier into a parallel tuned circuit, and the output of the tuned circuit back to the input of the amplifier, is to use a centre-tapped inductor in the tank (tuned) circuit. This is the principal of the Hartley oscillator.

Theoretical Circuit / a.c. circuit – Hartley Oscillator





Circuit Diagram of a Hartley Oscillator with a Buffer Amplifier

In this circuit, transistor Q1 is a common-collector (emitter follower) amplifier that is biased by R1, R2 and R3. The output of the amplifier, at the emitter of Q1, is coupled via DC blocking capacitor C3 into the parallel tuned tank circuit consisting of L1 and C2 through a tap in the inductor. The tank circuit is coupled back to the input of the amplifier via C2, which serves as another DC blocking capacitor to prevent the base of Q1 from being shorted to earth via L1. The arrow through C1 indicates that it is a variable capacitor, so the resonant frequency of the tank circuit, and hence the oscillator frequency, can be changed by varying C1. The output of the oscillator at the emitter of Q1 is fed to Q2, which is a common-collector (emitter follower) buffer amplifier. R4 sets the emitter and collector current for Q2. The output of the buffer amplifier is taken from the emitter of Q2 via DC blocking capacitor C4.

An Oscillator where the frequency to be varied, typically by turning a control knob, is known as a Variable Frequency Oscillator (VFO).

In this circuit, the centre-tapped inductor L1 acts a bit like a step-up transformer, since an AC voltage applied between the centre tap and the chassis connection (the bottom of the inductor) generates a varying magnetic field, which causes a larger voltage to be generated between the “hot” side of L1 (the top of the inductor) and the chassis. This voltage step-up allows the common-collector amplifier to provide power gain in this circuit, despite the fact that the voltage gain between the base and emitter of the transistor is unity (1). A tapped inductor like this is also called an autotransformer.

Other types of Oscillator

Clapp

It was published by James Kilton Clapp in 1948.[1] According to Vačkář,[2] oscillators of this kind were independently developed by several inventors, and one developed by Gouriet had been in operation at the BBC since 1938.

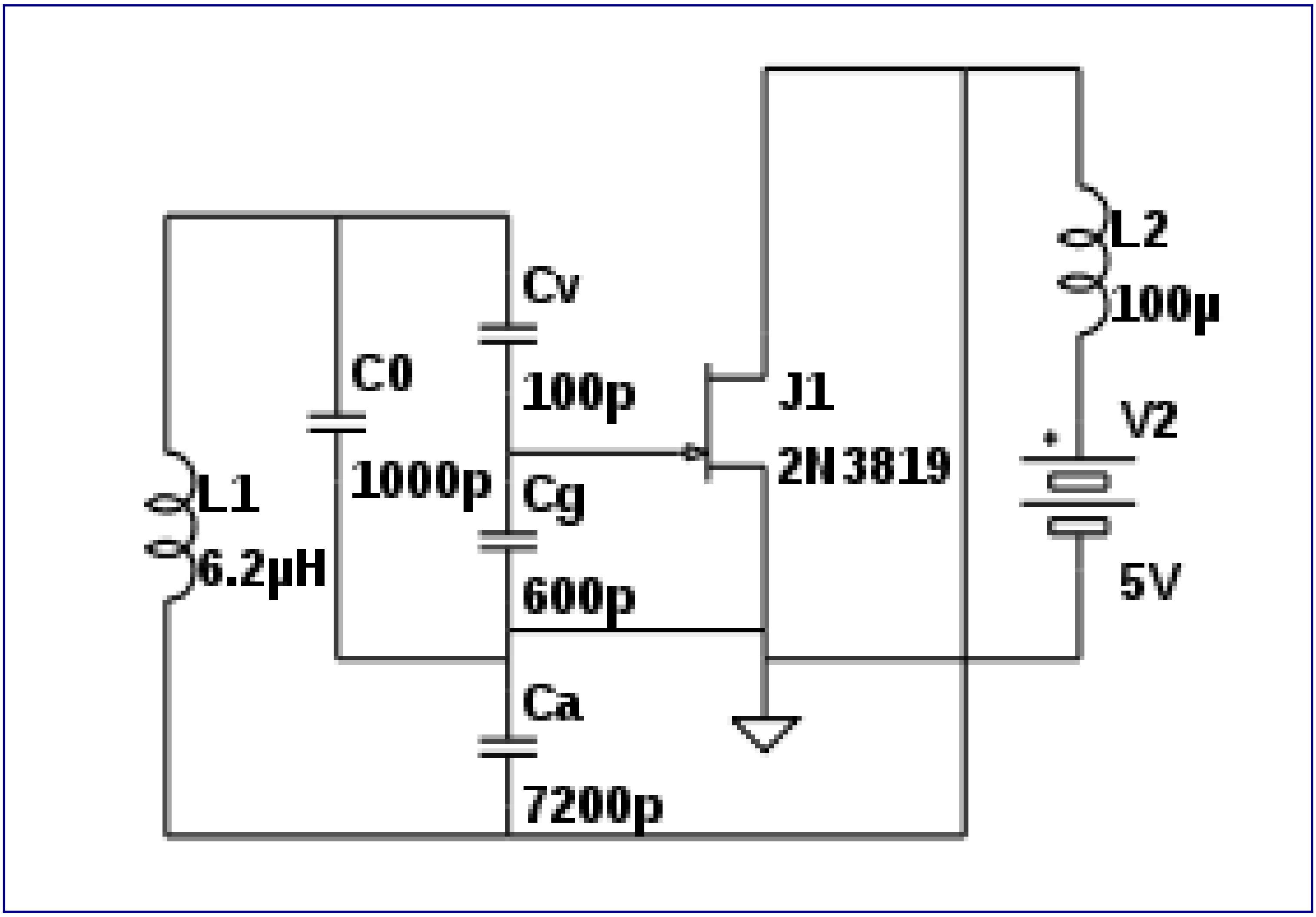
Gouriet

Geoffrey George Gouriet M.I.E.E joined the Drive Section of the Transmitters Department of the BBC in 1937,[1][2] and in 1937/38 he was the inventor of a high stability crystal-controlled variant of the Colpitts oscillator. With the outbreak of war imminent, his circuit was put to immediate use by the BBC to drive its Medium Wave broadcast transmitters, allowing the implementation of Britain's wartime single-frequency synchronised radio services from multiple transmitters. This was a technique adopted to try to prevent the Luftwaffe conducting air raids on British cities using BBC transmitters for navigation.

Due to wartime security measures, Gouriet's oscillator design was kept secret until after WWII. Meanwhile, the same circuit was independently discovered by James Kilton Clapp of the USA, and published by him in 1948. Gouriet's oscillator is usually known as the Clapp oscillator as a result, although newer books use the term *Gouriet-Clapp oscillator*.[3]

Vackář oscillator

From Wikipedia, the free encyclopaedia



Schematic of what is commonly called the Vackář oscillator. Vackář credited Radioslava with developing this circuit in 1945.[1]

A **Vackář oscillator** is a wide range variable frequency oscillator that strives for a near constant output amplitude over its frequency range. It is similar to a Colpitts oscillator or a Clapp oscillator, but those designs do not have a constant output amplitude when tuned.

Stability [Very Important!]

In 1949, the Czech engineer Jiří Vackář published a paper on the design of stable variable-frequency oscillators (VFO).[2] The paper discussed many stability issues such as variations with temperature, atmospheric pressure, component ageing, and micro-phonics. For example, Vackář describes making inductors by first heating the wire and then winding the wire on a stable ceramic coil form. The resulting inductor has a temperature coefficient of 6 to 8 parts per million per degree Celsius.[3] Vackář points out that common air variable capacitors have a stability of 2 parts per thousand; to build a VFO with a stability of 50 parts per million requires that the variable capacitor is only 1/40 of the tuning capacity (.002/40 = 50ppm). The stability requirement also implies the variable capacitor may only tune a limited range of 1:1.025.[3] Larger tuning ranges require switching stable fixed capacitors or inductors.[4]

Vackář was interested in high stability designs, so he wanted the highest *Q* for his circuits. It is possible to make wide range VFOs with stable output amplitude by heavily damping (loading) the tuned circuit. However, that tactic substantially reduces the *Q*.[5]

Vackář was also concerned with the amplitude variations of the variable-frequency oscillator as it is tuned through its range. Vackář assumes the tuned circuit has a constant *Q* over the VFO's frequency range. Vackář reviewed several existing circuits for their amplitude stability.[1] The Clapp oscillator's transconductance requirement is proportional to ω^3. If the Clapp transconductance is set to just oscillate at the lower frequency range, then the oscillator will be over-driven at its highest frequency. The Seiler and Lampkin oscillators have a transconductance requirement that is proportional to ω^{-1}.

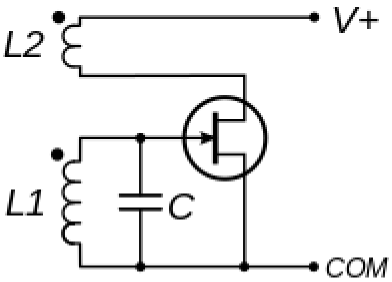
Vackář then describes an oscillator circuit due to Radioslava in 1945 that maintained "a comparatively constant amplitude over a wide frequency range."[6] Vackář reports that VFO circuit being used by the Czechoslovak Post Office since 1946. Vackář does not claim the circuit, but he analyses the circuit and explains how to get an approximately constant amplitude response. This circuit has become known as the Vackář VFO.[7] Vackář does refer to the circuit as "our circuit" and states that O. Landini independently discovered the circuit and published it (without an analysis) in Radio Rivista in 1948.[8] Vackář describes a VFO design using this circuit that covers a range of 1:1.17.[8]

Vackář then describes a variation of the Radioslavia circuit that can cover a range of 1:2.5.[9] The circuit need not assume the tuned circuit has a constant *Q*. Vackář patented this new circuit and two variations of it.[10]

Armstrong Oscillator

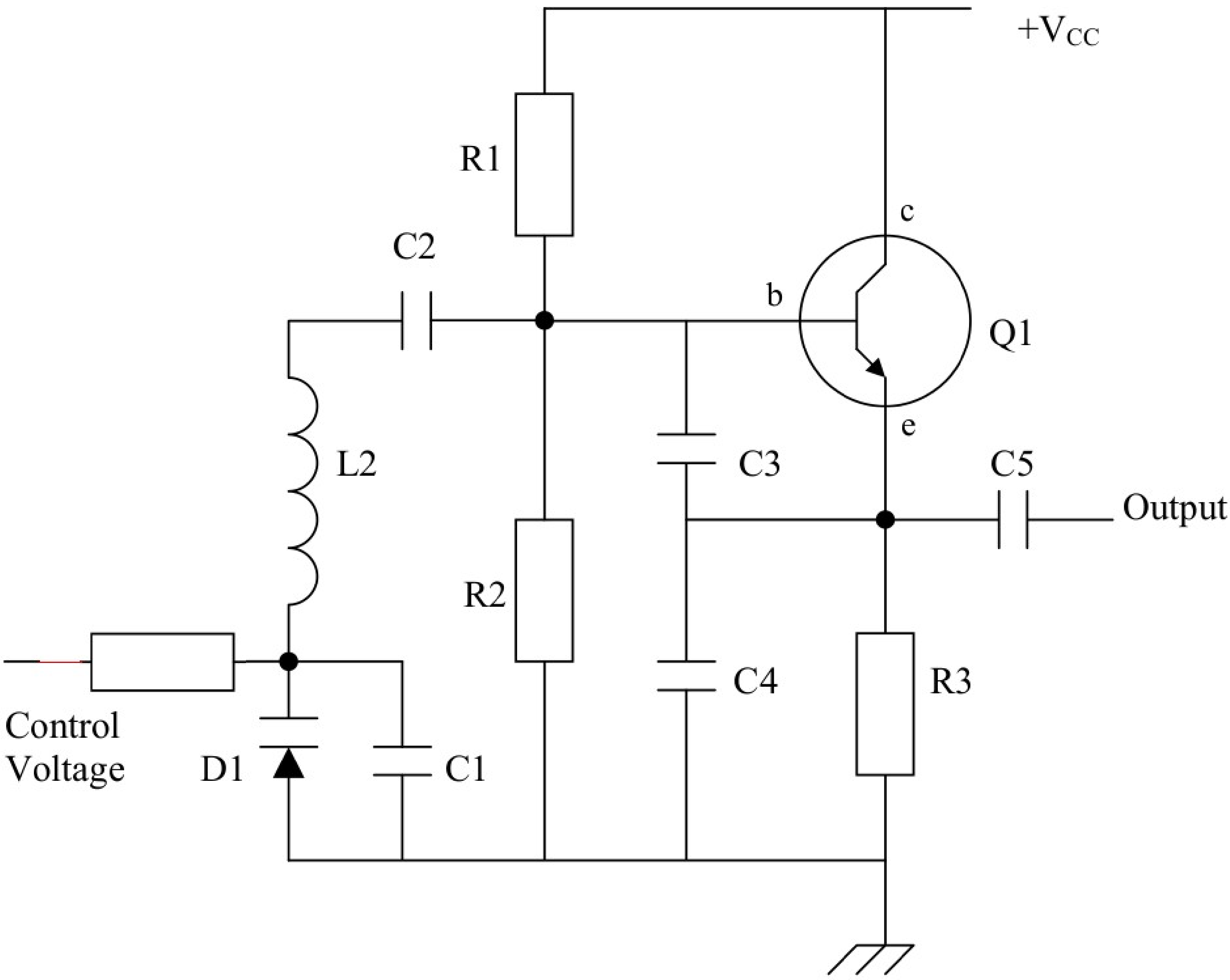
The **Armstrong oscillator**[1] (also known as the **Meissner oscillator**[2]) is an electronic oscillator circuit which uses an inductor and capacitor to determine the oscillation frequency; an LC oscillator. It is the earliest oscillator circuit, invented by US engineer Edwin Armstrong in 1912 and independently by Austrian engineer Alexander Meissner in 1913, and was used in the first vacuum tube radio transmitters. It is sometimes called a *tickler oscillator* because its distinguishing feature is that the feedback signal needed to produce oscillations is magnetically coupled into the tank inductor in the input circuit by a "tickler coil" *(L2, right)* in the output circuit. Assuming the coupling is weak, but sufficient to sustain oscillation, the oscillation frequency *f* is determined primarily by the tank circuit *(L1 and C, right)* and is approximately given by

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



The Voltage-Controlled Oscillator

If part of the capacitance forming the tuned circuit in an oscillator is made up of capacitance from a varicap diode, then the frequency of the oscillator can be varied by changing the reverse-bias voltage applied to the varicap diode. This is called a voltage-controlled oscillator (VCO). An example circuit, using a **Clapp** (series-tuned Colpitts) configuration is shown below:



Circuit Diagram of a Voltage Controlled Oscillator

The control voltage is applied through radio-frequency choke L1 to reverse-bias the varicap diode D1. This is in parallel with C1, which provides some additional capacitance (necessary since varicap diodes have fairly low capacitance). They are in series with L2, hence the name "series-tuned Colpitts" oscillator (also called a Clapp oscillator). C2 prevents the DC control voltage from interfering with the bias voltage generated by the voltage divider consisting of R1 and R2 (or vice-versa). Q1 is operated as a common collector (emitter follower) amplifier, and the output at the emitter of Q1 is fed back into the tank circuit at the junction between C3 and C4, which form the tank circuit along with C1, D1 and L2. The oscillator output is taken from the emitter of Q1 via DC blocking capacitor C5.

The Crystal Oscillator

Quartz crystals exhibit the **piezoelectric effect** – a voltage applied across the crystal causes the crystal to distort ("bend") slightly, and when the crystal returns to its undistorted shape a voltage is generated across it. As a result, the crystal appears similar to a series tuned circuit and it can be used as the frequency-determining element in an oscillator. A typical circuit is shown below.

Comment: Gas cigarette lighter crystal generates a high-voltage spark to light the gas!

Circuit Diagram of a Crystal Oscillator

Here the resonant circuit consists of crystal X1 with series capacitor C1 and capacitors C2 and C3. Q1 operates as a common-collector (emitter-follower) amplifier biased by R1, R2 and R3.

The output of the amplifier is fed back into the tank circuit at the junction between C2 and C3. This circuit also uses a "series-tuned Colpitts" or "Clapp" configuration.

Crystals have the advantage of providing very good frequency stability – that is, the frequency of a crystal controlled oscillator will remain stable with little tendency to "drift", which is a problem with oscillators using traditional inductor-capacitor tuned circuits. The disadvantage of crystal oscillators is that they cannot be tuned over any great range. The variable capacitor C1 in this circuit can varying the frequency slightly (which is known as "pulling" the crystal), but the tuning range is very limited. Crystal oscillators that allow the frequency to be varied are called "**v**ariable **c**rystal **o**scillators", abbreviated "VXO".

Comment: Crystals are EXPENSIVE. Also these days they have been replaced with CERAMIC RESONATORS and Surface Wave Acoustic (S.A.W.) devices.

Summary

Oscillators are circuits that generate AC signals. Oscillators consist of an amplifier with **positive feedback** through a phase-shift network. The phase shift network usually also serves as a band-pass filter. An oscillator will oscillate at any frequency and amplitude where the Barkhausen criteria for oscillation are met:

- The loop gain is unity.
- The sum of the phase shifts around the feedback loop is zero or an integer multiple of 360˚.
- The output of an oscillator should be buffered to prevent the frequency of the oscillator from changing as the load on the oscillator varies.

There are several different oscillator circuits, including the Colpitts, Hartley and Clapp oscillators, which differ in the precise arrangement of the tank circuit. An oscillator that allows the frequency to be varied is called a Variable Frequency Oscillator (VFO). If the frequency is varied by applying a control voltage, then it is a Voltage Controlled Oscillator (VCO).

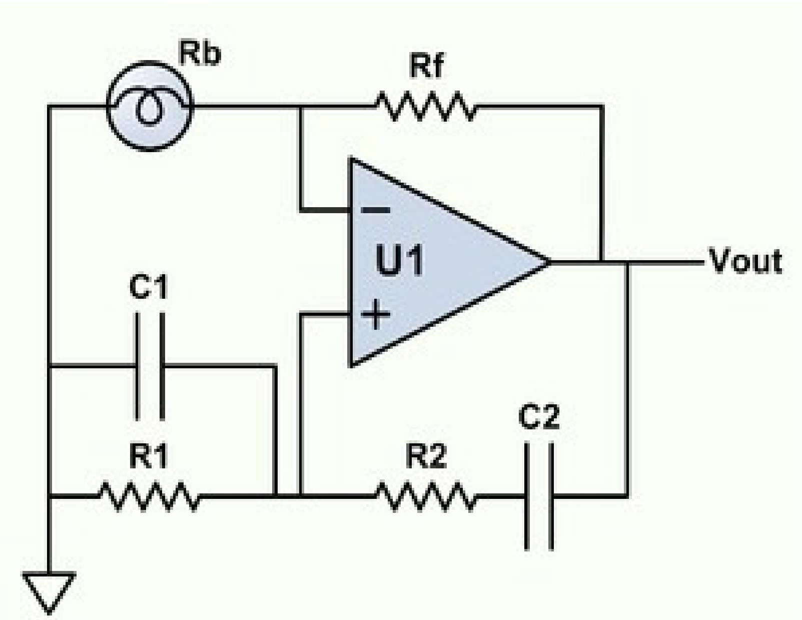
Quartz crystals exhibit the piezoelectric effect and act like series tuned circuits. They can be used to control the frequency of an oscillator. Crystal-controlled oscillators exhibit excellent frequency stability, with very little drift. However they are essentially fixed-frequency oscillators; although the frequency can be “pulled” slightly using a variable capacitor, the tuning range is not nearly as wide as for oscillators using ordinary tuned circuits. Crystal oscillators that allow the frequency to be varied are called “variable crystal oscillators”, abbreviated “VXO”.

Comment: BUT
Crystals can be 'cut' for specific high temperature operation. So that they can be used as a very stable frequency source. Usually inside a temperature controlled oven.

Crystals in general these days are used as “Frequency Reference” for a “Phase Locked Loop” or a “Direct Digital Synthesiser” [DDS].

We did not even mention “tone generator/oscillators”!

Wien Bridge Oscillator



A **Wien bridge oscillator** is a type of electronic oscillator that generates sine waves. It can generate a large range of frequencies. The oscillator is based on a bridge circuit originally developed by Max Wien in 1891.[1] The bridge comprises four resistors and two capacitors. The oscillator can also be viewed as a positive gain amplifier combined with a bandpass filter that provides positive feedback.

The modern circuit is derived from William Hewlett's 1939 Stanford University master's degree thesis. Hewlett figured out how to make the oscillator with a stable output amplitude and low distortion.[2][3]

Hewlett, along with David Packard, co-founded **Hewlett-Packard**, and Hewlett-Packard's first product was the HP200A, a precision Wien bridge oscillator.

The frequency of oscillation is given by:

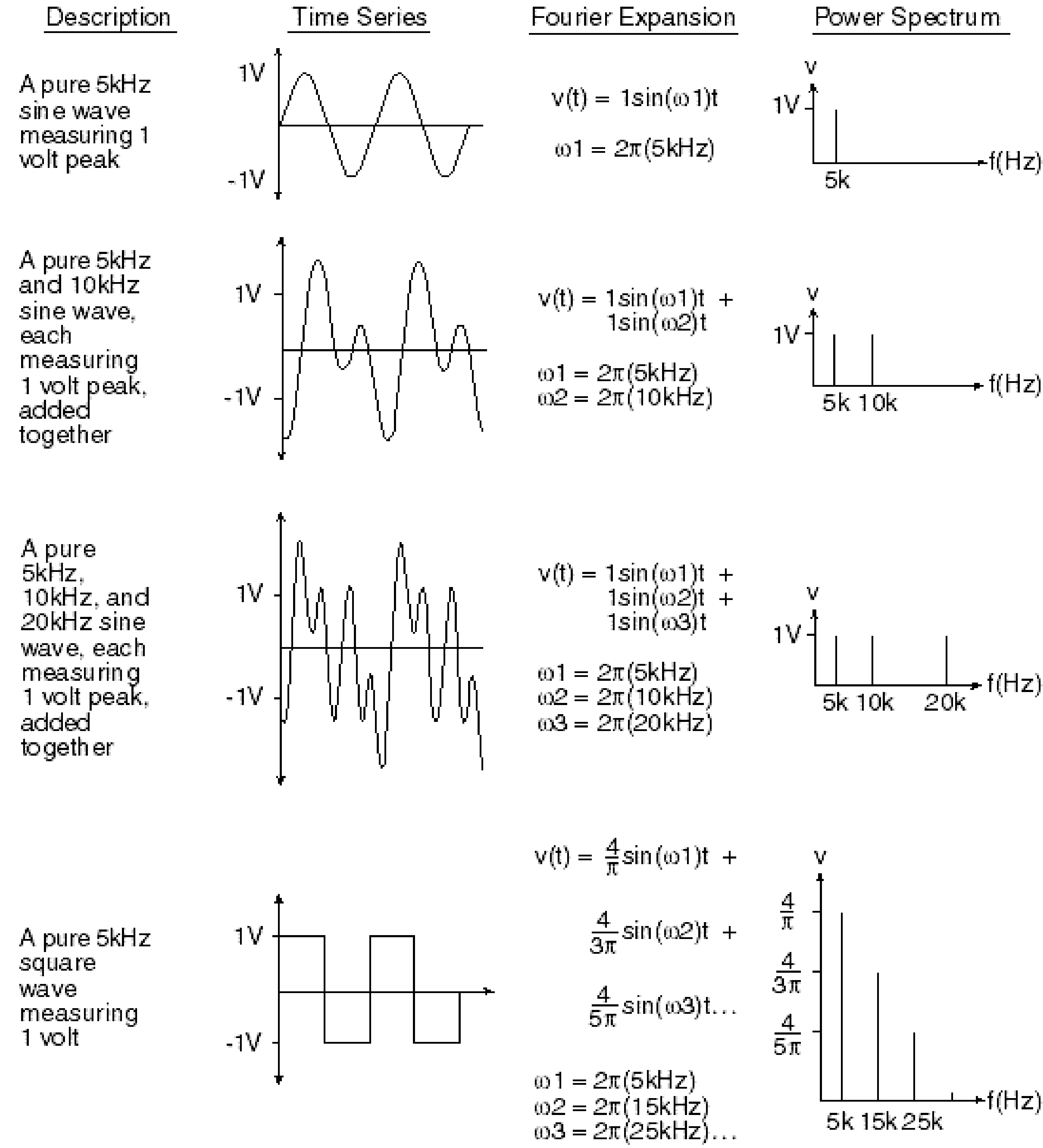
$$f = \frac{1}{2\pi RC}$$

OR Phase-Shift Oscillators
OR Multivibrators [Digital Oscillators]
OR Direct Digital Synthesizers - DDS Oscillator

Chapter 20 - Frequency Translation

Oscillators are used to generate the signals of various frequencies that are needed by in transmitters and receivers. However often it is useful to be able to create a signal of a desired frequency from signals of other frequencies. For example, it can be very beneficial to generate a signal at the precise output frequency the user has chosen from a very stable reference signal at a fixed frequency. The circuits we use to do this are frequency multipliers, frequency synthesizers and mixers.

[This is where those charts (pictures) come in handy!]
[Remember Fourier?]



20.1 The Frequency Multiplier

Any waveform other than a sine wave contains harmonics as well as the fundamental frequency. Harmonics can be found at any integral multiple of the fundamental frequency.

For example, a 10 MHz signal that was not a sine wave might have harmonics at 20, 30, 40, 50, 60 and 70 MHz, and so on.

This can be used to create a frequency multiplier. The input sine wave is intentionally distorted; creating a signal that is rich in harmonics. The desired harmonic is then selected using a band-pass filter, yielding a signal that is some integer multiple of the input signal. The most common multiples are 2 and 3. For example, a 7 MHz signal applied to the input of a x2 frequency multiplier would give a 14 MHz signal; applied to a x3 multiplier it would give a 21 MHz signal.

Different types of distortion result in different amounts of the various harmonics. When designing a frequency multiplier, the type of distortion introduced should maximize the desired harmonic. For example, a frequency doubler (a x2 multiplier) could use a full-wave rectifier to distort the input waveform, since the resulting rectified sine wave has a high second-harmonic content. A typical circuit is as follows:

The input signal is full-wave rectified by T1, D1 and D2. L1 and C1 form a parallel tuned circuit, which is resonant at the output frequency (twice the input frequency). It shorts the DC component of the full-wave rectified signal to ground and attenuates the undesired higher order harmonics. Transistor Q1 with resistors R1, R2 and R3 form a common-emitter amplifier. There is another parallel tuned circuit made up of L2 and C3 in the collector circuit of the amplifier, which further attenuates undesired high-order harmonics (3, 4, 5 times the input frequency etc.). C2 and C4 are DC blocking capacitors. The output is a sine wave at twice the frequency of the input wave.

A x3 multiplier (frequency tripler) might use a class C amplifier to introduce the necessary distortion, since the output of a class C amplifier has a high third-harmonic component. In VHF and UHF applications, varicap diodes (also known as varactor diodes) are often used as the non-linear element to distort the input waveform and generate harmonics.

Because frequency multipliers introduce distortion, they cannot be used with signals that contain a range of frequencies, such as audio signals or amplitude modulated (AM) and single-sideband RF signals. If they were, then the many different frequency components of these signals would interact with each other causing unwanted inter-modulation distortion (IMD) components, that are too close to the desired frequencies to be filtered out. However they can safely be used with un-modulated signals, or with CW (Morse code), frequency modulated (FM) and phase modulated signals.

Frequency multipliers are only useful for multiplying by fairly small numbers, such as 2, 3 or 4. They cannot be used to multiply by large numbers – say 100 – because it would be too difficult to construct a filter to separate the 100th harmonic from the 99th or 101st harmonics, and the nature of frequency multipliers means that they tend to generate at least some amount of most harmonics!

20.2 The Frequency Divider

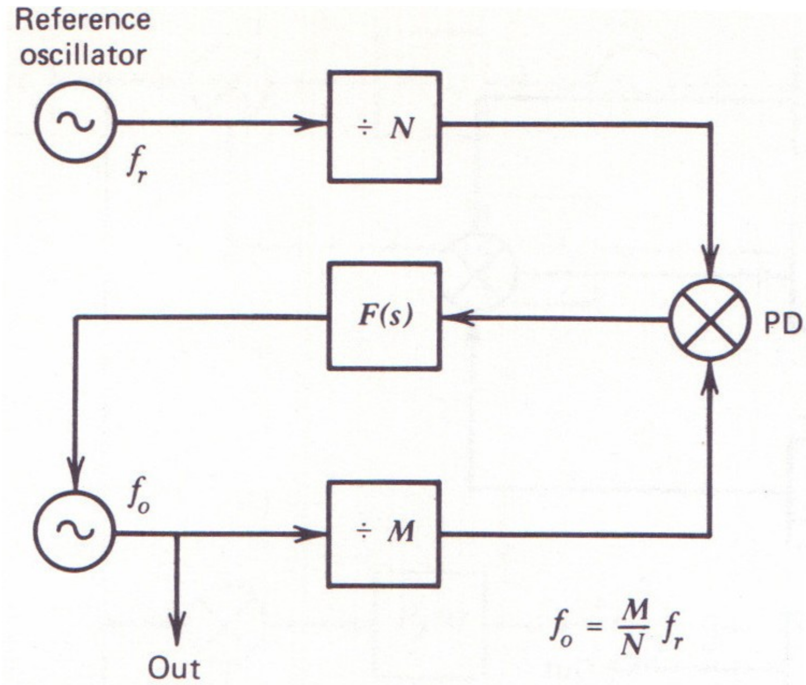
Digital integrated circuits are available that can divide the frequency of an input waveform by any integer number – either a fixed number, or one that can be programmed by a microprocessor. The output of these “digital dividers” is typically a square wave, which contains high harmonic content (especially the odd harmonics at 3 times, 5 times, 7 times the input frequency and so on). These harmonic content can be removed using a suitable low-pass or band-pass filter leaving a sine wave at the desired frequency.

20.3 The Phase Locked Loop Frequency Synthesizer

Although variable-frequency oscillators (VFO’s) can be used to generate a signal at a frequency selected by the user, they suffer the disadvantage that it is difficult to make them very stable, and their frequency tends to “drift” in response to changes in the ambient temperature, and to make more rapid excursions if bumped or otherwise maltreated. Crystal oscillators, on the other hand, are very stable in the face of temperature variations and mechanical knocks. However their very limited tuning range makes them unsuitable for use as, say, the main oscillator for a transmitter that must cover an entire amateur band.

The most common solution in modern amateur equipment is to use a frequency synthesizer. This is a circuit that can generate many programmable output frequencies based on a single reference frequency derived from a stable crystal oscillator. Although there are several different types of frequency synthesizer, this section will only cover one of these, the phase locked loop (PLL) frequency synthesizer. The block diagram of a simple PLL synthesizer is shown below.

Block Diagram of a PLL Frequency Synthesizer



The output of the frequency reference is fed into a phase comparer. This is a circuit that compares the phase of two signals and generates an output voltage that depends on the phase difference between the signals. This voltage is smoothed by a low-pass filter, and used to control the frequency of a voltage-controlled oscillator. The signal generated by the VCO is the input to a frequency divider that divides the input frequency by some (usually programmable) integer N. The output of the frequency divider is the second input to the phase comparer.

To understand how this circuit works, suppose that the frequency of the VCO is exactly N times the reference frequency. Then the phase comparer will generate a DC output voltage that is dependant on the phase difference between the two signals. This DC voltage will pass through the low-pass filter, and will affect the frequency of the VCO. Suppose the effect is to increase the frequency of the VCO slightly. As the frequency increases, the phase of the VCO output signal will begin to shift relative to the phase of the reference signal, which will change the output voltage of the phase comparer, which is the VCO control voltage.

The circuit is arranged so that if the frequency of the VCO increases slightly, then the resulting output voltage from the phase comparer will reduce the frequency of the VCO again, to bring it back to its “correct” frequency, which is N times the reference frequency.

Similarly, if the frequency of the VCO decreases slightly, then the resulting output voltage from the phase comparer will act to increase the frequency of the VCO, again returning it to a frequency N times the reference frequency. In this condition, the VCO is said to be phase locked to the reference frequency, since any change in the phase relationship between the two signals (caused, for example, by a change in the VCO frequency) will act on the VCO in a way that will return it to the correct phase relationship with the reference frequency. This is an example of negative feedback In case you were wondering, the reason for the low-pass filter is because most phase comparators actually generate a fairly complex output signal that has a DC (or low-frequency) component that reflects the phase difference between the inputs, as well as components at the different input frequencies to the phase comparer. The low-pass filter rejects the high frequency outputs, leaving only the low-frequency phase comparison voltage.

So now we have a circuit that can generate a frequency that is N times a stable reference frequency, and is phase locked to the reference frequency, so that it is almost as stable as the reference frequency itself. However by changing the value of N, we can change the output frequency, making it any integer multiple of the reference frequency. If the reference frequency is small enough - say 10 Hz – then we can generate an output frequency that is any multiple of 10 Hz. For example, if the reference frequency is 10 Hz and the divider N is 1 402 000, then the output frequency will be 10 * 1 402 000 = 14 020 000 Hz. If N is increased by 1 to 1 402 001 then the output frequency would be 14 020 010 Hz. This allows us to synthesize almost any desired frequency from a single stable reference frequency. In modern radios, the divider N that controls the output frequency is usually set by a microprocessor in response to user input, such as adjusting the tuning control.

The only remaining problem is to generate a stable 10 Hz reference frequency for our synthesizer. We can't use a crystal oscillator directly, since 10 Hz is much too low a frequency for a quartz crystal. So what we can do is run a crystal oscillator at a more suitable frequency – perhaps 100 kHz – and then use a digital divider to reduce the frequency to the desired reference frequency. In this case, dividing the 100 kHz oscillator output by a factor of 10 000 would give a 10 Hz reference frequency.

In practical PLL synthesizers it turns out that there is a trade-off between the speed at which the synthesizer can change its frequency (the "tuning rate" if you like) and the resolution of the synthesizer (its "step size"). This is because the resolution of the synthesizer is set by the reference frequency, so a high resolution requires a low reference frequency. But that requires a low cut-off frequency for the low-pass filter, which limits the speed at which the synthesizer can respond to changes in frequency. One solution is to combine the outputs of two synthesizers, one with a high reference frequency that can easily make large frequency changes but has limited resolution, and the other with a small step size that can "fill in" the missing frequencies, but which is never required to make large frequency changes (because the "coarse" synthesizer takes care of that). This is known as a multiple-loop synthesizer.

PLL synthesizers are very versatile and are the basis for most modern transceivers, allowing them to achieve very high stability combined with wide frequency coverage. However, they do have some disadvantages. In particular, early synthesizers such as those found in amateur equipment from the early 1980s suffered from significant phase noise, with the phase and frequency of the output signal varying very slightly as the loop adjusted it to keep it locked to the reference frequency. Modern synthesizer designs are much better in this respect.

20.4 The Mixer

Another circuit that is commonly used for frequency translation in both transmitters and receivers is the mixer. It is based on the interesting mathematical result that if you multiply two sine waves together, you get a waveform that consists of two components: one with a frequency that is the sum of the frequencies of the inputs, the other with a frequency that is the difference between the frequencies of the two inputs.

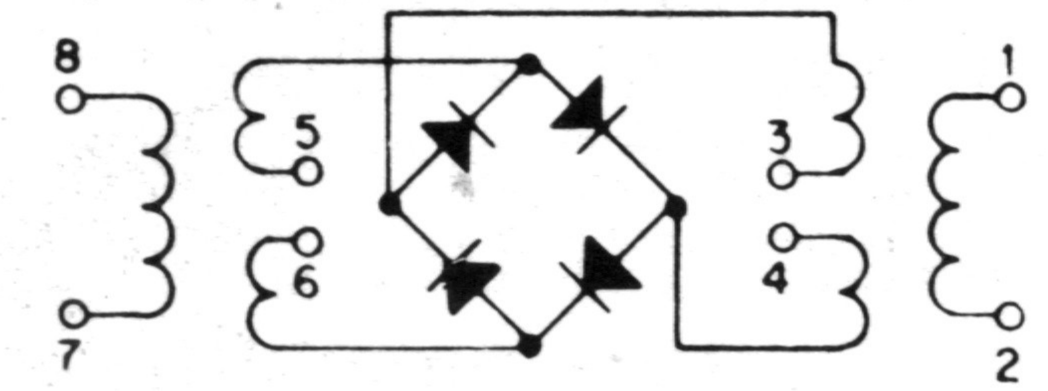
For the mathematically inclined, the relevant mathematical identity is:

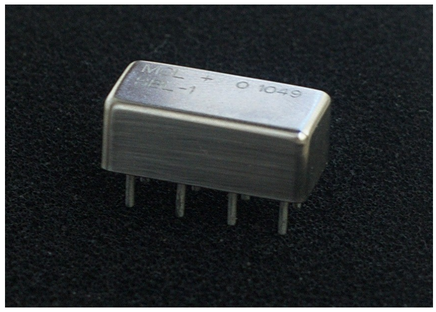
$$2 \sin(A) \cos(B) = \sin(A+B) + \sin(A-B)$$

If you set $A = 2\pi f1t$ and $B = 2\pi f2t$ then the left-hand side, "2 sin(A) cos(B)" represents two sine waves of frequency f1 and f2 with a phase shift of 90˚ multiplied together, while the right hand side "sin(A+B) + sin(A-B)" represents the superposition (adding together) of sine waves with frequencies f1+ f2 and f1-f2, the sum and difference frequencies. Slightly more complex maths shows that the precise phase difference between the input signals is not important.

For example, if you multiply a 9 MHz sine wave by a 6 MHz sine wave, you end up with two sine waves superimposed: one with a frequency of 15 MHz (the sum of the input frequencies), the other with a frequency of 3 MHz (the difference between the input frequencies). Electronic circuits that do this multiplication are called **mixers**.

Now it turns out that it is not very easy to accurately multiply two sine waves without introducing significant distortion into the output. One common solution is to use a switching mixer. Instead of actually multiplying the two signals together, it uses one of the input signals to switch the other input signal on and off, or to reverse its direction. Here is a typical circuit diagram for a switching mixer:





20.5 A Double-Balanced Diode Mixer

In this mixer, diodes are used as the switching elements. A strong input signal (generally derived from a local oscillator) is applied at the point marked LO, while a much weaker radio frequency signal is applied to the point marked RF. The output signal is taken from the point marked IF, for “intermediate frequency”. The reason for these names will become apparent once we have studied the design of radio receivers.

The strong LO signal is used to “chop” the weaker RF signal, with the output appearing at the IF port. Here is how it works. Assume that the LO signal’s polarity is such that point C is positive with respect to point D. Diodes D1 and D2 will be forward biased (turned on) while diodes D3 and D4 will be reverse biased (turned off). If the diodes are properly balanced, with identical forward bias voltages, then the point between D1 and D2 will be at the same potential as the centre-tap on the secondary winding of T2, that is at chassis (earth) potential.

This will earth point A on the secondary winding of T1. If the polarity of the signal applied to the RF port is such that point A is positive with respect to point B, then A will also be positive with respect to the output IF port, so the IF port will be negative with respect to point A, which as we have seen is earthed.

Now suppose the LO signal reverses polarity, while the RF signal remains as it was. Point D is positive with respect to C, so diodes D3 and D4 will conduct, effectively earthing point B.

Since the RF signal is making A positive with respect to B, it will also make the IF output positive with respect to B, which is earthed.

So in one half cycle of the LO (switching) input, the RF signal makes the IF output negative with respect to earth, while in the other half cycle, the RF signal makes the IF output positive with respect to earth. The result is that the LO signal is effectively switching the polarity of the RF signal as it appears at the IF output.

Hold on a moment. We started talking about multiplying two signals together, now we are talking about using one signal to switch the polarity of the other. What is the connection?

Well it turns out that using the LO signal to switch the polarity of the RF signal is equivalent to multiplying the RF signal by a square wave with the values +1 and –1. (Multiplied by +1, the polarity is unchanged; multiplied by –1 it is reversed). One effect of this is that, because a square wave contains not only the fundamental frequency, but also many harmonics, these harmonics are effectively mixed with the input signal as well. So instead of only getting the sum and difference frequencies, we also get the sum and difference frequencies of the RF signal and each harmonic of the LO signal. The unwanted mixing products can usually be filtered out by suitable filters following the mixer.

Diode mixers like this one require fairly high drive power at the LO port – typically +7 dBm (5 mW) or more. They usually exhibit a conversion loss of 6-7 dB, meaning that each of the output signals is 6 or 7 dB lower than the RF input signal. However they are widely used in amateur applications because they have good low-distortion properties.

This mixer design is “double balanced” because neither the RF input signal nor the LO input signal will be reflected in the output. An unbalanced mixer would allow both the RF and the LO signals to get into the IF output, while a “single balanced” mixer would allow only one of these signals (typically the weaker and therefore less troublesome RF signal) to make it into the output.

There are many other mixer designs using transistors, specialized integrated circuits and other components.

One big advantage of mixers over other frequency translation circuits (frequency multipliers and the like) is that properly designed mixers do not introduce significant distortion into the signals, and so they can be used with all types of signals, including amplitude modulated (AM), single sideband (SSB) and audio signals.

Summary

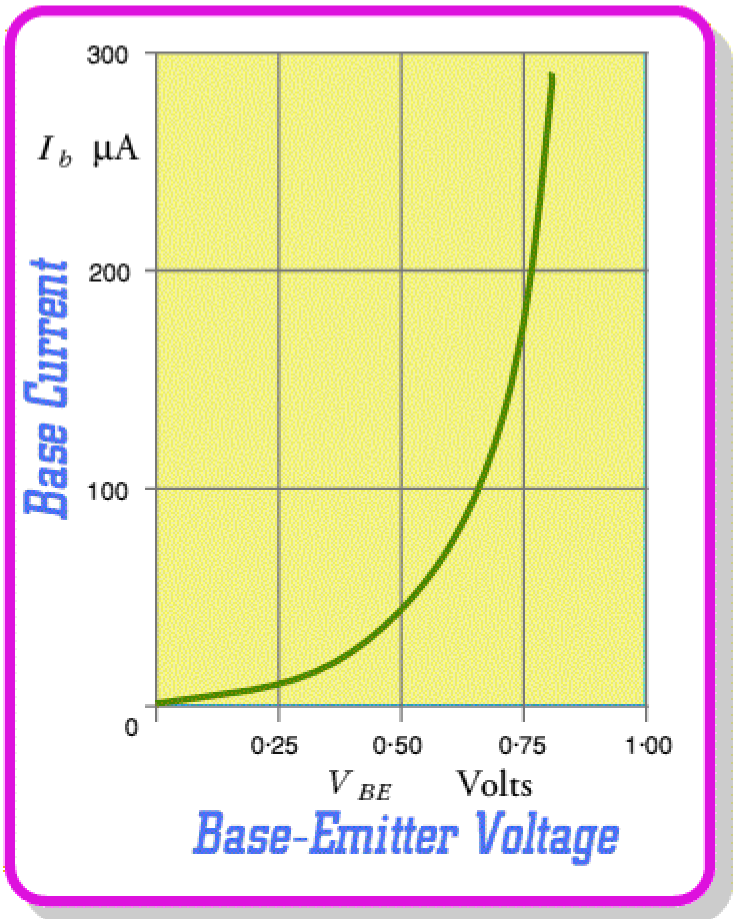
Frequency multipliers distort the input waveform to generate harmonics, and then select the desired harmonic using a band-pass filter. They can be used to multiply frequencies by small integers, typically 2 or 3. **[Whoops! The author never heard of a "comb generator"?]** Frequency multipliers cannot be used with signals that contain many frequencies, such as AM or SSB signals, as they cause too much distortion. **However they can be used with CW and FM signals.**

Digital integrated circuits are available that can divide a frequency by any integer number. **[Ok, another small mistake. It is possible to "variable divide" in a 10/11 or a 63/64 divider. This will provide a non-integer division. Most pll circuits for VHF/UHF/Microwave use them.]** In a **phase locked loop** frequency synthesizer, the output frequency is locked to an integer multiple of a **stable reference frequency**. By changing the multiple, different frequencies can be generated from a single reference frequency. The output of a PLL synthesizer has similar stability to the reference frequency, although it will have **additional phase noise**. **[DDS Direct Digital Synthesizer chips do this nowadays.]**

The output of a **mixer** will contain signals with frequencies that are the **sum** of the frequencies of the input signals and the **difference** between the frequencies of the input signals.

Depending on the mixer type, it may also contain signals at the same frequency as one or both of the input frequencies – if both input frequencies are suppressed then the mixer is "double balanced" while if only one input signal is suppressed it is "single balanced". Switching mixers will also typically contain mixing products caused by mixing various harmonics of the switching (LO) input with the low-level (RF) input. **Unwanted mixing products must be removed by suitable filters at the output**.

How it is done



Chapter 21 - Modulation Methods

[How far can you shout?]

Carrier based transmission. Wireless Microphones...

Radio is based on the fact that electromagnetic waves of certain frequencies can travel great distances and still be strong enough to be detected by a radio receiver. However in order for this to be useful, we need a way of sending information with, or imprinted upon, the radio waves. The sort of information that we wish to send – human speech, images or perhaps digital information – is not generally of the correct frequency to benefit directly from the ability of radio to span great distances. For example, the human voice has frequencies that range from approximately **300 Hz to 3 kHz.** These frequencies are much too low to be effectively propagated as radio waves.

Modulation is the process of imprinting information on radio waves, so we can take advantage of the propagation of radio waves to transmit the information to a distant receiver.

(1) What do the following abbreviations stand for?

AM
SSB
CW
FM
NBFM

(2) What frequency components would be present on an **AM signal** with a **carrier frequency of 7050 kHz** in the 40m band?

(3) What frequency components would be present on an **LSB signal** with a **carrier frequency of 7050 kHz** in the 40m band?

(4) Would it be legal to transmit a **USB signal** on a frequency of **7050 kHz**?

(5) Name **three advantages** of an **SSB** signal over an **AM** signal.

(6) What **deviation** is typically employed in amateur **FM signals**?

(7) Name **three advantages** of an **FM signal** over an **AM signal**.

(8) Would it be **legal** for an amateur to transmit music using commercial **deviation** FM signals in the **80m band**?

(9) Name at least two **digital modulation methods**, and provide the **official emission modes** as employed in the **regulations**.

Amplitude Modulation (AM)

One of the earliest methods of modulation is amplitude modulation, or A.M. Although not widely used in the amateur service any more, it still lives on in the A.M. transmissions of commercial radio stations in the **medium frequency** (or **"medium wave"**) broadcast band. In amplitude modulation, the **amplitude** (strength) of a radio frequency signal, called the **carrier** is varied according to the amplitude (strength) of the modulating signal.

The plot below shows a low frequency sine wave, and the result when this is used to amplitude-modulate a higher frequency carrier.

A modulating signal and amplitude-modulated carrier

See how the amplitude of the high frequency carrier wave varies in step with the amplitude of the low frequency modulating signal. When the amplitude of the modulating wave is zero, the amplitude-modulated wave is at its "average" output level. When the amplitude of the modulating waveform is positive, the amplitude-modulated signal is above this "average" amplitude, and when the modulating wave is below zero, the output is below this "average" level.

The modulation depth of an amplitude-modulated signal is the percentage by which the carrier signals varies above and below its average level in response to the modulating signal.

In this example, the carrier is 80% modulated because the peak in the carrier amplitude is 80% above its average level, and the minimum carrier amplitude is 80% below its average level. The maximum possible modulation depth is 100% modulation. In a 100% modulated A.M. signal, the carrier amplitude decreases to zero when the modulating signal is at its most negative. Any attempt to modulate at more that 100% would result in the carrier "bottoming out" at zero amplitude and distorting the modulation signal. This is known as over modulation which introduces a great deal of distortion and should be avoided. An example of an over modulated signal is shown below:

An over-modulated A.M. signal

Amplitude modulation has the advantage that it is very simple to recover the modulating signal from the amplitude-modulated signal in the receiver. A simple half-wave rectifier followed by a low-pass filter will recover the modulating signal, which is typically an audio signal. The plot below shows a half-wave rectified A.M. signal, and the result of passing this through a low-pass filter.

A half-wave rectified A.M. Signal and the recovered modulation

The low-pass filter has removed what remains of the carrier, leaving the modulation "envelope" and a D.C. offset (which is indicated by the fact that the recovered signal is not symmetrical about the X axis). The D.C. offset can be simply removed by a D.C. Blocking capacitor to obtain the original modulating signal. The process of recovering the modulation signal from a modulated signal is known as demodulation.

Another way of looking at amplitude modulation is that it consists of multiplying the carrier by the modulating signal plus a D.C. offset. The value of the D.C. offset would be chosen to ensure that the sum of the modulating signal and the offset always remains positive, in order to prevent over-modulation. This means that amplitude modulation consists of mixing the carrier and modulation signals. Of course we know that mixing two signals results in an output that contains the sum and difference of the input frequencies, and possibly other components. In this case, the output also includes the carrier wave. This is because the DC offset that we added to the modulating signal has a frequency of zero (because it's D.C.) which also mixes with the carrier, creating a sum frequency (the carrier frequency plus zero) and a difference frequency (the carrier frequency minus zero) that are both the same frequency as the carrier.

So if the carrier frequency is FC and the modulating frequency FM, then the amplitude modulated signal will have frequency components of FC, FC-FM and FC+FM. These components can be plotted on a graph that shows frequency on the X-axis, and the relative amplitude of different components of the signal on the Y-axis. This is called the frequency spectrum of the signal.

The vertical line above the carrier frequency FC represents the carrier, while the lines above frequencies FC+FM and FC-FM represent the sum and difference frequencies respectively. Note that the carrier is much stronger than either of the other components. In an amplitude modulated signal, two thirds of the power is contained in the carrier; the sum and difference frequencies together make up only one third of the total power of the modulated signal.

The frequency spectrum of a carrier amplitude-modulated by a sine wave

So far we have only considered a carrier that has been modulated by a single sine wave.

However speech consists of a whole range of frequencies, with many different frequency components present simultaneously in a speech waveform.
Fortunately it is quite simple to figure out what happens if we amplitude-modulate a carrier with a speech signal that contains many different frequency components. Each of the different frequency components in the speech will create two output signals, one at the sum of the carrier frequency and this component of the modulating signal and one at the difference frequency. The following plot shows the frequency spectrum of some modulating signal (it is on the left of the graph, at a low frequency) and the corresponding amplitude-modulated signal.

Spectrum of a modulating signal and the corresponding amplitude-modulated signal

See how each component of the modulating signal corresponds to two components of the resulting amplitude-modulated signal, on above the carrier (the sum) and one below the carrier (the difference).

The total of all the “sum” components of the modulated signal – that is, all the components of the modulated signal that are higher in frequency than the carrier – is called the upper sideband of the A.M. signal. The total of all the “difference” components – that is, all the components of the modulated signal that are lower in frequency than the carrier – is called the lower sideband of the A.M. Signal.

In order for speech to be reproduced intelligibly, frequencies from about 300 Hz to 3 kHz are required. This means that for a communications grade A.M. signal, such as is used in the amateur service, the upper sideband will extend from 300 Hz above the carrier to about 3 kHz above the carrier, while the lower sideband will extend from about 300 Hz below the carrier to about 3 kHz below the carrier. So the total bandwidth of the signal is 6 kHz, from 3 kHz below the carrier frequency to 3 kHz above the carrier frequency.

This analysis of the frequency spectrum of an A.M. signal shows the two greatest disadvantages of amplitude modulation.

1. The component of the signal at the carrier frequency conveys no information (it is an unvarying carrier), and yet it consumes two thirds of the power of the signal. This makes amplitude modulation quite inefficient power wise.
2. An A.M. signal transmits two copies of the modulating information, one in the upper sideband and one in the lower sideband, while only one of these would be sufficient to recover the original modulation. This is why the bandwidth of an amplitude modulated signal is twice the bandwidth of the modulating signal, and so A.M. Is quite inefficient in terms of the amount of spectrum (frequencies) required. This is particularly important on the crowded amateur bands.

Double-Sideband Suppressed-Carrier Modulation

We could overcome the first of these problems – the power wasted by the carrier – if we could generate a signal without a carrier. This can be done by using a balanced modulator, which outputs only the sum and difference components, but not the carrier itself.

Mathematically this is equivalent to simply multiplying the carrier signal by the modulating signal, without adding any D.C. offset. The plot below shows a low frequency sine wave modulating signal, and the resulting double-sideband suppressed-carrier modulated signal.

A sine wave and double-sideband suppressed-carrier modulated signal

This time, because there is no D.C. offset on the modulating signal, the resulting double sideband modulated signal is zero when the modulating signal is zero. When the modulating signal goes from being positive to being negative or vice-versa, the phase of the modulated signal is inverted, indicating that the modulating signal has crossed the axis. Note that you could not use a simple half-wave rectifier and low-pass filter to recover the modulation.

The frequency spectrum of a double-sideband, suppressed carrier signal is shown below, using the same multi-frequency modulating signal as in the last plot.

Modulating signal and the corresponding double-sideband suppressed-carrier signal

Not surprisingly, it looks exactly like the frequency spectrum of the amplitude-modulated signal, but without the carrier. Double-sideband suppressed-carrier signals are more powerefficient than amplitude-modulated signals, since they do not waste any power on the carrier.

However they still occupy twice the bandwidth as the original modulating signal, making them wasteful of spectrum. For this reason, double-sideband suppressed-carrier signals are rarely used in practice.

Single-Sideband (SSB)

In order to avoid wasting bandwidth, we could simply take a double-sideband suppressedcarrier signal and remove one of the sidebands, leaving only a single sideband remaining.

This type of modulation is formally known as “single sideband suppressed-carrier modulation”, but is usually called just “single sideband” or “SSB”. If we remove the lower sideband, then the result would be an upper-sideband (USB) signal. If we remove the upper sideband, then the result would be a lower-sideband (LSB) signal. The two plots below show the frequency spectra have lower-sideband and upper-sideband signals. The carrier frequency is shown as a dotted line so you can see where the frequency spectrum is in relation to where the carrier would have been if it had not been suppressed; but of course the carrier is not actually transmitted.

The frequency spectrum of a modulating signal and the corresponding lower-sideband signal

The frequency spectrum of a modulating signal and the corresponding upper-sideband signal

Note that in the lower-sideband signal, the frequency spectrum of the modulating signal has been inverted (low frequencies in the modulating signal correspond to high frequencies in the lower-sideband signal and vice-versa), while in the upper sideband signal the spectrum in the modulated signal is the same way around as it was in the modulating signal. In fact, an upper sideband signal has an identical frequency spectrum to the original modulating signal, it has just been translated to a higher frequency.
Single sideband is the most commonly used means of transmitting speech in the amateur service. Both upper- and lower-sideband are used. By convention, lower-sideband is used on frequencies below 10 MHz, while upper-sideband is used on frequencies above 10 MHz.

Because SSB signals do not have a carrier, the receiver frequency must be accurately adjusted to properly recover the original audio. Any maladjustment of the receiver frequency will result in the pitch of the audio being slightly too high or too low. This is not important for speech, as it is easy to adjust the receive frequency sufficiently accurately to make speech intelligible, but it is the reason why AM or FM are usually preferred for music transmissions, where even a slight frequency shift in the received audio would be problematic.

Continuous Wave (CW)

Continuous Wave (CW) consists of turning the carrier on and off in order to convey information in Morse code. The name comes from the fact that the first transmitters used sparks, and were not capable of transmitting a continuous signal. Their transmitted signals would consist of an initial strong oscillation when the spark sparked that rapidly died down, known as “damped waves”. So when the first valve-based transmitters became available that were capable of transmitting continuously, they were called “continuous-wave” or “CW” transmitters, despite the fact that information was transmitted by turning the carrier on and off with the resulting dots and dashes standing for letters in Morse code.

It might seem at first as thought he frequency spectrum of a CW transmission should contain only the carrier, since the transmission consists of turning the carrier on and off. However turning a carrier on and off is the same as amplitude-modulating it with a waveform that is at some fixed D.C. level when the carrier is to be turned on, or at zero when it is to be turned off, and so we should expect some sidebands in the keyed signal. These are called “key clicks” because they can be heard as clicks in a receiver when it is tuned close to, but not actually on the same frequency as, as CW transmission.

The shape of the envelope of the CW waveform – that is, the way it is turned on and off – has a big influence on the strength of the key clicks and how far they extend away from the carrier frequency. If the carrier reaches full amplitude as soon as it is turned “on”, and zero amplitude as soon as it is turned “off” then a lot of key clicks will be generated, causing noticeable interference to stations several kilohertz away. To avoid this, the carrier should be allowed to “ramp up” to full volume relatively slowly, and to decay back to zero over a while when it is turned off. The optimum ramp-up and decay period for a CW signal is around 5 ms.

This can be achieved using a capacitor that is charged and discharged through a resistor to determine the keying envelope. This acts as a simple low-pass filter, attenuating the high frequency harmonics of the keying waveform that would otherwise cause key clicks.

Although it may appear archaic, CW is still in widespread use. One of its advantages is that it is intelligible with much lower signal strengths than any voice signal. Practical listening tests have shown that CW requires about 13 dB less power for the same intelligibility as an SSB signal. So a 100 Watt CW transmitter will “get out” as well as a 2 kW SSB transceiver!

Frequency Modulation (FM)

Instead of varying the amplitude of the carrier depending on the amplitude of the modulating signal, frequency modulation (F.M.) varies the frequency of the carrier in response to changes in the amplitude of the modulating signal. For example, when the amplitude of the modulating signal is positive, the frequency might be increased slightly from the original carrier frequency, and when the modulating signal is negative, the frequency of the carrier might be reduced slightly. The following plot shows a frequency-modulated signal

A sine wave and corresponding frequency-modulated signal

Note that the amplitude of the signal remains constant, while the frequency varies according to the amplitude of modulating signal. (The amount of frequency change has been exaggerated to make it easier to see.)

The amount that the frequency of the carrier increases or decreases in response to the modulation is called the deviation of the signal. The frequency of the carrier is both increased and decreased by the deviation, so for a signal with a deviation of 2,5 kHz, the frequency of the modulated signal will range from 2,5 kHz below the centre frequency to 2,5 kHz above the centre frequency. The centre frequency is the frequency with no modulation applied.

The deviation ratio is the maximum deviation divided by the highest modulating frequency.

For example, if the deviation is 2,5 kHz and the maximum modulating frequency is 3 kHz then the deviation ratio would be 2500/3000 = 0,83.

The voice-grade FM transmissions typically used by amateurs are referred to as narrow-band frequency modulation (NBFM). In NBFM the deviation is kept to about 2,5 kHz and the resulting signal has a bandwidth of 5-6 kHz, comparable to that of a communications-grade AM signal.

Commercial FM broadcast stations, by comparison, have a deviation of 75 kHz and a correspondingly much wider bandwidth.

FM signals have the advantage of better audio quality when the strength of the radio signal being received is fairly strong. This is because when an F.M signal is well above the atmospheric noise level, the amplitude variations due to noise have little effect on the receiver, which is only sensitive to variations in the signal frequency and not its amplitude.

However the quality of the recovered audio drops rapidly as the signal strength weakens and gets closer to the level of atmospheric noise. For this reason, amateurs mostly use F.M. For local communications in very high frequency (VHF) bands like the 2 m band (144-146 MHz) and ultra high frequency bands like the 70 cm band (430-440 MHz) where signals are usually strong and atmospheric noise is slight. For long-range communications in the high frequency (HF) bands between 3 and 30 MHz, where signals are often weak and atmospheric noise fairly strong, SSB is preferred.

Frequency-Shift Keying (FSK)

So far we have concentrated on “human readable” signals, like the various phone (voice) modes and CW. However an increasing role is being played by digital communications, where radio is used to transmit digital information between two computers. In this case, the information that is being transmitted consists of binary bits (ones and zeros).

A simple modulation method for digital information is frequency-shift keying, where the transmitter transmits one of two possible frequencies depending on whether it is sending a zero or a one. The two frequencies are called the “mark” and “space” frequencies, with the “mark” frequency corresponding to a logic “1” and the “space” frequency corresponding to logic “0”.

FSK is used by modes such as RTTY (radio teletype), which allows interactive communication between two computers and Packet Radio, which provides electronic mail and file transfers over radio links.

Phase-Shift Keying (PSK)

Instead of shifting the frequency of the carrier, it is possible instead to shift the phase of the carrier depending on whether a one or a zero is being transmitted. The resulting modulation method is called phase-shift keying (PSK). PSK is preferred over FSK in most modern applications because it is more efficient in terms of bandwidth usage.

PSK comes in several different forms. In binary phase-shift keying (BPSK), the transmitted signal has one of two different phases, say 0˚ or 180˚, allowing one binary bit (a one or a zero) to be transmitted at a time. In quad phase-shift keying (QPSK), the transmitted signal can have one of four different phases (0˚, 90˚, 180˚ or 270˚), allowing two binary bits to be transmitted at a time.

The most popular amateur mode to use phase- shift keying is PSK-31, which is an interactive digital mode that allows two operators to “chat” to each other in real time over the radio.

Everything that either operator types on his or her keyboard is immediately transmitted and displayed on the computer screen of the other operator (and anyone else who is listening).

PSK-31 can use either BPSK or QPSK. When using QPSK the increased throughput is used to provide error detection and correction.

Summary

In amplitude modulation (AM), the amplitude of an RF carrier is varied according to the amplitude of the modulating signal. The resulting AM signal consists of the carrier, the upper sideband (at a higher frequency than the carrier) and the lower sideband (at a lower frequency than the carrier). The carrier takes two thirds of the power of an AM signal, with the remaining one-third of the power being shared equally between the upper and lower sidebands. Although AM signals are easy to demodulate using a half-wave rectifier and lowpass filter, they are inefficient both in terms of power (because the carrier conveys no information but takes 2/3 of the power) and bandwidth (since the modulating information is replicated in both sidebands).

A double-sideband suppressed-carrier signal is similar to an AM signal but without the carrier. It can be generated using a balanced modulator. The resulting signal is more powerefficient than an AM signal, but still uses twice the bandwidth of the modulating signal.

In a single-sideband suppressed-carrier (single sideband, or SSB) signal both the carrier and one of the sidebands has been removed, leaving only a single sideband. SSB signals may be upper sideband (USB) or lower sideband (LSB). In LSB signals the spectrum of the modulating signal is inverted in the modulated signal; in USB, the spectrum is simply translated to a different frequency but is not inverted. SSB is one of the most efficient means of voice communications, especially when signal strengths are low.

Continuous Wave (CW) transmission consists of turning the carrier frequency on or off, and is used to send information in Morse code. CW is effectively a type of amplitude modulation, and the keying sidebands are known as “key clicks”. Their extent and strength can be reduced by turning the carrier on and off fairly gradually, over a period of about 5 ms.

In frequency modulation (FM) the frequency of the carrier is varied according to the amplitude of the modulating signal while the amplitude remains constant. FM signals are capable of very good audio quality provided the received signal is fairly strong, but quality deteriorates rapidly as the received signal strength weakens. Narrowband FM transmissions by amateurs usually have a deviation of 2,5 kHz, resulting in a bandwidth of 5-6 kHz, which is similar to an AM transmission.

Frequency-shift keying (FSK) and phase-shift keying (PSK) are used to transmit digital information. In FSK, one of two frequencies is transmitted depending on whether a one or a zero is being sent; while in PSK the phase of the transmitted signal is varied to indicate that a one or a zero is being sent. FSK is used by modes like RTTY and Packet, while PSK is used by PSK-31.

Some thinking required...

(10) What do the following abbreviations stand for?

AM	Amplitude Modulation
SSB	Single Sideband
CW	Continuous Wave
FM	Frequency Modulation
NBFM	Narrow Band Frequency Modulation

(11) What frequency components would be present on an **AM signal** with a carrier frequency of 7050 kHz in the 40m band?

Carrier at 7050 kHz
Lower sideband at 7050 kHz – audio frequency
Upper sideband at 7050 kHz + audio frequency

(12) What frequency components would be present on an LSB signal with a carrier frequency of 7050 kHz in the 40m band?

Only lower sideband at 7050 kHz – audio frequency

(13) Would it be legal to transmit a USB signal on a frequency of 7050 kHz?

Yes – There is no restriction in the regulations.
BUT, you may have difficulty in making a contact as amateurs normally expect to find only LSB signals in the 40m band and may not recognize the USB signal.

(14) Name three advantages of an SSB signal over an AM signal.

- Bandwidth: An SSB signal occupies only half the bandwidth of an equivalent AM signal.
- Power: The transmitter for an SSB signal does not produce a carrier, so saving 2/3 of the power in comparison with an AM signal.
- Power: The transmitter for an SSB signal only produces one sideband, so saving another 1/6 of the power in comparison with an AM signal.

(15) What deviation is typically employed in amateur FM signals?

2,5 kHz

(16) Name three advantages of an FM signal over an AM signal.

- Better immunity to noise and interference
- Better audio quality, where the signal strength is fairly good
- Better audio bandwidth for the same effective bandwidth (typical deviation in NBFM would be 2,5 kHz, with a signal bandwidth of 5-6 kHz. This would typically support audio frequencies up to twice the normal commercial communications bandwidth of approx. 3 kHz.)

(17) Would it be legal for an amateur to transmit music using commercial deviation FM signals in the 80m band?

No.
Commercial FM deviation is typically 75 kHz. While music is permitted in the 80m band, subject to some restrictions, the regulations also restrict the allowable bandwidth for F3E and G3E transmissions to 10 kHz in all bands below 30 MHz.

(18) Name at least two digital modulation methods, and provide the official emission modes as employed in the regulations.

F1B	-	Telegraphy, including RTTY and Data, by means of frequency shift keying
F1D	-	Data transmissions by means of frequency shift keying
J2D	-	Data transmission with the use of a modulating audio frequency – **Includes PSK-31**
J2E	-	Digital telephony with the use of a modulating audio frequency

Chapter 23: Receiver Fundamentals

A radio receiver is the heart of any amateur radio installation, whether it is a stand-alone receiver or combined with a transmitter as a transceiver. It is relatively easy to build a good transmitter. All you really need is good frequency stability, adequate power and a clean output signal (no harmonics, key clicks or inter-modulation distortion). It is much harder to build a good receiver, and consequently there is more variation in receiver capability amongst both commercial and homebuilt designs.

23.1 Noise in Receivers

The main limitation to a receiver's ability to demodulate a signal is the signal to noise ratio (SNR). This ratio is normally expressed as the ratio (in dB) of the total signal entering the antenna terminals (including noise) to the noise itself. So:

$$SNR = (S + N) \div N$$

S is the desired signal; N is the noise, including anything except the desired signal. Other signals are also noise, for our purposes.

The following sources of noise contribute to the term N in the above equation:

- Receiver thermal noise: Semiconductors produce noise due to the semi-random movement of electrons in the semiconductor material. This noise is temperature dependent. The intensity of the noise is expressed as $P_N = kTB$, with P_N being the noise power, k being Boltzmann's constant, T being the noise temperature in K and B being the bandwidth in Hz. The higher the temperature, the more noise. The more bandwidth, the more noise. In some specialised applications, receiver components are actually cooled down to reduce receiver noise. However, specialised semiconductors can produce lower noise temperatures in the receiver. Good design and construction practices can reduce the noise in the receiver, leading to a lower effective temperature.

- Other receiver noise: Early synthesisers created lots of phase noise. This noise was due to phase jitter in the PLL's VCO. This noise also contributes to receiver noise, and is bandwidth dependent, just like thermal noise.
- Band noise:
  - Atmospheric noise: Distant thunderstorms contribute to a general noise level on the band, which masks weak signals. Atmospheric noise is mostly a problem on the low bands, except when there is heavy weather close by.
  - Electrical noise: Powerlines, switching gear, automotive ignition systems and electric motors may produce noise that enters the antenna.
  - Ground noise: The ground around the antenna radiates noise, which is temperature dependent, much like semiconductor noise.
  - Galactic noise: Certain galaxies and regions in the sky radiate lots of noise. The sun is the most dominant of these sources. In general, galactic noise is only a problem at VHF and above, as the noise otherwise does not penetrate the ionosphere below the critical frequency.
  - Other signals: Any signal except the one that we specifically want to receive contributes to noise.

At HF and below, the band noise is normally well above the receiver noise. However, at VHF and above, the receiver noise starts becoming the limiting factor. Designers have to go to great lengths to reduce the receiver noise to be able to hear weak signals. At these frequencies, low noise preamplifiers, low-loss cables and even cryogenic cooling come into play.

The noise figure of a receiver is specified in dB. The same information can be stated as a noise temperature. The advantage of this approach is that the simple formula $P_N = kTB$ can be used to calculate the actual noise power in the receiver, which can then be simply compared to the incoming signal to determine SNR. The temperature is stated in kelvin (K), which has the same magnitude as °C but starts at a different point. At 0 K, there is no kinetic energy due to temperature. It is also known as absolute zero. At absolute zero, there is no thermal noise. 0°C = 273 K, and 100°C = 373 K. Normal room temperature is therefore around 300 K.

23.2 Receiver Characteristics

Selectivity
When propagation conditions are good (i.e. strong radio signals are propagating long distances) the amateur bands can be a very crowded place. If you listen during any CW contest, for instance, you will hear signals spaced 100 to 200 Hz apart over the entire CW section of a band. So the first attribute a good receiver must have is selectivity, the ability to distinguish between close-spaced signals and receive only the one that the listener is interested in. The selectivity is mostly determined by bandwidth, specified in Hz.

Sensitivity
Many of the signals on amateur bands are very weak, having come from low-powered transmitters a long distance away, so the second attribute an amateur receiver needs is sensitivity, the ability to "hear" very weak signals. Sensitivity can be expressed in terms of the voltage (in μV) or power (in dBm) to produce a specific SNR.

Dynamic range
Since these weak signals may be adjacent to strong signals, perhaps from other amateurs in your town, amateur receivers need another attribute: dynamic range. Dynamic range is the ability to receive a weak signal despite the presence of other much stronger signals on nearby frequencies, and is expressed as a ratio in dB, along with the separation between the signals. The further apart the signals are, the more interfering signal the receiver can tolerate.

To get an idea of the challenges faced by receiver designers, a typical weak signal on an amateur band might deliver a power of –120 dBm from the antenna—that's one thousand-millionth of a microwatt. A strong signal might deliver –30 dBm, or one microwatt. So a strong signal could be 90 dB (a thousand million times) as strong as a weak signal—and yet the receiver might need to select and amplify the weak signal to a usable level, without being affected by the strong signal a few kilohertz away!

This module introduces two simple receiver designs—the tuned radio frequency receiver and the direct-conversion receiver—and considers how well they meet these requirements.

It also introduces many of the concepts that you will need for the next module, which covers the superheterodyne receiver.

23.3 The Tuned Radio Frequency (TRF) Receiver

One of the simplest receiver designs, which has been with us almost since the dawn of radio, is the tuned radio frequency receiver. The principle is simple: you use a bandpass filter to select the signal you want, amplify the weak radio signals, demodulate the signal (to recover the audio modulating frequency) and then amplify the recovered audio sufficiently to make it audible in headphones or a loudspeaker. The block diagram below shows the layout of a TRF receiver. The block labelled "detector" is a half-wave rectifier to demodulate AM signals,

A Tuned Radio Frequency Receiver with Regeneration

The arrows through the bandpass filter indicate that they are tunable, so they can be used to select the desired signal. The dotted line joining the arrows on the two bandpass filters mean that they tune together, so a single control will change the tuning of both filters.

Many TRF receivers use regeneration, which means feeding some of the signal from the output of the RF amplifier back to its input, in such a way as to reinforce the signal at the input of the RF amplifier. This technique is a form of positive feedback. It has the benefit of increasing the amplification of the RF amplifier (because some of the signal "circulates" through it many times, being amplified each time) and also increasing the selectivity, since the signal also passes through the bandpass filter at the output of the RF amplifier many times. Of course an amplifier with positive feedback is an oscillator, so if too much regeneration is applied, the circuit will oscillate. Regenerative receivers (a name for TRF receivers that use regeneration) usually have a control to adjust the amount of regeneration, which is adjusted to get the maximum possible sensitivity and selectivity without oscillation.

The advantage of TRF receivers is that they are simple to construct and require relatively few components—typically just two or three transistors and a handful of other parts. This simplicity made them attractive in the days before transistors, when thermionic tubes were used for amplification in radio receivers, as tubes were relatively expensive, so the fewer the better!

Their big disadvantage is that they have very poor selectivity and dynamic range. Tunable bandpass filters just aren't capable of rejecting an unwanted signal that is only a couple of kilohertz away from the signal you are listening to, so unwanted signals will also get through to the detector and be recovered as audio or cause inter-modulation distortion. TRF receivers are also best suited for receiving AM signals. Although regenerative receivers can be used with CW and SSB signals, by adjusting the regeneration control so the circuit just barely oscillates, adjustment is tricky and the quality of reception poor. For these reasons, TRF receivers are no longer widely used.

23.4 The Direct-Conversion Receiver

A design that is used in quite a few homebuilt receivers is the Direct Conversion (DC) receiver. In a DC receiver, the radio-frequency signal from the antenna is mixed with a
locally generated oscillator signal, producing the usual sum and difference mixing products.

The frequency of the oscillator that generates this local mixing signal—it is known as the local oscillator (LO) or beat frequency oscillator (BFO)—is set so the difference mixing product is at audio frequency. In this way, the DC receiver "directly converts" the desired radio-frequency signal to audio, where it can be filtered and amplified. Let's look at the circuit in a little more detail.

A Direct-Conversion Receiver

The signal from the antenna first passes through a bandpass filter. Unlike in the Tuned Radio Frequency receiver, this bandpass filter is not responsible for the overall selectivity of the receiver. Its role is simply to reject interference from strong local commercial broadcast stations and the like. It does not have to be tunable—usually a fixed-tuned filter covering an entire amateur band will suffice.

The signal is then amplified by an RF amplifier and fed into the product detector, which is represented on the diagram using the symbol for a mixer—the circle with a cross in it.

"Mixer", "Modulator" and "Product Detector" are different names for essentially the same circuit, depending on the exact role it plays. The product detector mixes the amplified RF signal with a signal generated by the tunable local oscillator, generating the usual sum and difference mixing product.

Suppose we want to receive an upper-sideband signal on 14,200 MHz. By convention, we refer to the frequency of a single-sideband signal as the frequency where the carrier would have been if it had not been suppressed. This frequency is known as the pseudo-carrier frequency. The upper sideband of this USB signal will span a frequency range from 14,200 3 MHz to 14,203 0 MHz, or 300 Hz to 3 kHz above the pseudo-carrier frequency. If the local oscillator is set to exactly 14,200 MHz, the difference mixing products will range in frequency between 300 Hz and 3 kHz. What we have done is to translate the USB signal from its frequency of 14,200 MHz back to the baseband.

Receive mixer output for a USB signal

This graph shows how mixing the 14,200 MHz USB signal with a 14,200 MHz signal from the local oscillator generated a difference mixing product (signal frequency – local oscillator frequency) in the audio range and a sum product (signal frequency + local oscillator frequency) up above 28,400 MHz.

Although the example used a USB signal, the same process would work equally well using an LSB signal, and the local oscillator frequency would still be 14,200 MHz, the pseudo-carrier frequency. The following graph shows the same process with a lower-sideband signal.

Receive mixer output for an LSB signal

Once again the difference product is back in the audio frequency range, while the sum product is at around twice the signal frequency, at 28,400 MHz. Also note how for the lower sideband signal, the mixing process has inverted the sideband (so the recovered audio is the mirror image of the sideband), which makes up for the sideband inversion that would have occurred when the LSB signal was generated.

So whether the signal is USB or LSB, mixing it with a local oscillator at the pseudo-carrier frequency will demodulate it and recover the audio to the baseband.

To complete the hat trick, suppose we have a CW signal at 14,200 MHz. All we need to do is set the local oscillator just below it—say at 14,199 4 MHz, which is 600 Hz below the CW signal—and the difference mixing product will be a 600 Hz tone, just right for listening to CW. So we can also use the product detector to receive a CW signal.

Setting the local oscillator 600 Hz above the CW signal would work just as well.

We now pass the recovered audio through a lowpass filter. The main purpose of the filter is to remove the difference mixing product from signals near to the one that we are listening to. For example, suppose there is a CW signal at 14,205 MHz while we are listening to our 14,200 MHz USB signal. The difference mixing product of the 14,205 MHz CW signal and the 14,200 MHz local oscillator is 5 kHz—in other words, we have translated the unwanted CW signal downwards in frequency to the audio range just as we have translated the wanted USB signal to audio. However, a lowpass filter with a cutoff frequency of around 3 kHz or so should be able to remove the unwanted CW signal without affecting the desired USB signal.

Because it is quite easy for a strong signal to overload a mixer, causing inter-modulation distortion, the gain ahead of the mixer (i.e. the gain of the RF amplifier) is usually kept low so as not to amplify unwanted strong signals and overload the mixer. Most of the gain in a Direct Conversion receiver is at audio frequencies, in the amplifiers following the lowpass filter.

The only remaining part of the circuit is the Automatic Gain Control (AGC) system.

Because there is such a wide range of signal strengths on the air, and because the strength of a particular signal can vary with propagation changes, it is useful to have some way of automatically controlling the gain of the receiver. There must be a lot of gain to amplify weak signals, which must be reduced to avoid overloading when strong signals are present.

While this effect could be achieved with a manually operated gain control, but the poor operator would have to constantly work at it, and may occasionally be punished by a painfully loud signal when the gain is not reduced quickly enough. Also, when tuning from a strong signal (with the gain turned right down) to a weak signal, a weak signal can be missed altogether unless the gain is turned up first.

The solution is AGC. The AGC detector samples the audio signal after the first audio amplifier, and automatically adjusts the gain of the RF amplifier and the audio amplifier to keep the output signal level fairly constant. The output signal is then amplified by a final audio power amplifier and used to drive headphones or a speaker.

The AGC control voltage is often also used to drive a signal strength meter, known as an “S meter”, that indicates the strength of the received signal using a fairly arbitrary scale calibrated from S1 (a very weak signal) to S9 (a very strong signal). S meters are generally not well calibrated, but S9 is often taken as being 100 μV, with every S unit below that level representing about 6 dB.

The DC receiver has several advantages over a TRF receiver. Most importantly, its selectivity is very good, because unwanted nearby signals are easily filtered out by the audio lowpass filter that follows the product detector. It is more stable, having no tendency to oscillate like regenerative TRF receivers do. And it is easy to receive single sideband and CW signals with a DC receiver—you just tune the signal in, without having to fiddle with the regeneration control.

However, the DC receiver does have one significant disadvantage. Since the same local oscillator frequency can be used to tune either a upper sideband or a lower sideband signal, if you are listening to say an upper sideband signal and there is a different signal occupying the frequencies on the other side of the local oscillator where the lower sideband would have been, the other signal will also be shifted to audio frequencies and will interfere with the station you are trying to listen to.

For example, suppose you are listening to an USB signal at 14,200 MHz as before, but there is also a CW signal at a frequency of 14,199 MHz. Mixing the 14,200 MHz local oscillator signal with the 14,199 MHz CW signal results in a 1 kHz audio tone. Since this tone falls within the same 300 Hz to 3 kHz audio range as the desired USB signal, you cannot filter it out using the lowpass filter. And because the unwanted signal is so close in frequency to the desired signal, you can't use the RF bandpass filter to reject it either.

The unwanted signal on the other side of the local oscillator signal is called an "image", so the principal disadvantage of the Direct Conversion receiver can be described as its inability to reject images, or lack of "image rejection". There are more sophisticated variations of the basic Direct Conversion design that are able to reject images, but these are quite complex and fall outside the scope of this course.

Finally, a DC receiver must be designed carefully to ensure that the LO is isolated from the antenna port. Some simple DC receivers radiate some of the LO through the antenna, causing interference to listeners near the reception frequency.

Summary
The key attributes of a receiver are sensitivity, selectivity and dynamic range.

Sensitivity is the ability to receive weak signals; selectivity is the ability to distinguish between adjacent signals; and dynamic range is the ability to receive weak signals despite the presence of strong signals nearby.

In the tuned radio frequency (TRF) receiver all signal filtering is done at radio frequencies.

As a result they have poor selectivity. Regeneration, which consists of feeding some of the output signal back to the input of the RF amplifier, can increase both the sensitivity and selectivity of the TRF receiver, but makes it prone to oscillation. The oscillation, if well-controlled, can be used to facilitate CW and SSB reception.

In the direct-conversion (DC) receiver, the incoming RF signal is mixed down to audio frequency using a product detector and local oscillator. Most of the selectivity of a DC receiver is contributed by audio filters following the product detector. DC receivers have much better selectivity than TRF receivers, but they suffer from an image response to the opposite sideband that can only be eliminated with complex designs.

A bad DC design may also radiate some of the local oscillator, causing interference to other users.

Signal to noise ratio (SNR) determines whether a signal is readable or not. Noise can originate within the receiver or on the band. The receiver has a noise figure (in dB), which can also be expressed as a noise temperature (in K). At HF and below, band noise normally limits the SNR. At VHF and above, receiver noise is normally the limiting factor. Special semiconductors, feedlines and techniques are required to minimise receiver noise at these frequencies.

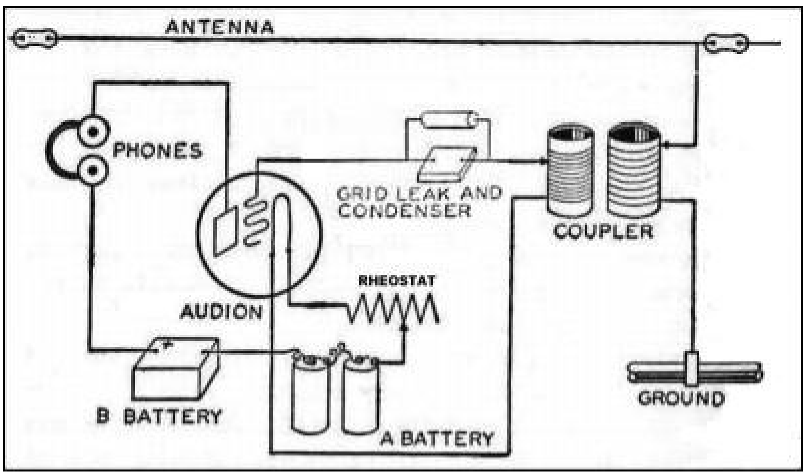
Noise 'Floor'

One aspect not mentioned in the main notes is the 'noise floor'. This is a level of random noise that will obscure a signal's modulation. Making it unreadable. At H.F. this is not an issue until the frequency is above 10 MHz or so. It varies according to daylight on the atmosphere and temperature of operation.

This module introduces two simple receiver designs – the tuned radio frequency receiver and the direct-conversion receiver, and considers how well they meet these requirements. It also introduces many of the concepts that you will need for the next module, which covers the **super-heterodyne receiver**.

The Tuned Radio Frequency (TRF) Receiver

One of the simplest receiver designs, which has been with us almost since the dawn of radio, is the tuned radio frequency receiver. The principle is simple: you use a band-pass filter to select the signal you want, amplify the weak radio signal, demodulate the signal (to recover the audio modulating frequency) and then amplify the recovered audio sufficiently to make it audible in headphones or a loudspeaker. The block diagram below shows the layout of a TRF receiver. The block labelled “detector” is a half-wave rectifier to demodulate AM signals.



A Tuned Radio Frequency Receiver with Regeneration

The arrows through the bandpass filter indicate that they are tunable, so they can be used to select the desired signal. The dotted line joining the arrows on the two bandpass filters mean that they tune together, so a single control will change the tuning of both filters together.

Many TRF receivers use **regeneration**, which means feeding some of the signal from the output of the RF amplifier back to its input, in such a way as to reinforce the signal at the input of the RF amplifier. This is a form of **positive feedback**. It has the benefit of increasing the amplification of the RF amplifier (because some of the signal "circulates" through it many times, being amplified each time) and also increasing the **selectivity**, since the signal also passes through the band-pass filter at the output of the RF amplifier many times. Of course an amplifier with positive feedback is an oscillator, so if too much regeneration is applied then the circuit will oscillate. Regenerative receivers (a name for TRF receivers that use regeneration) usually have a control to adjust the amount of regeneration, which is adjusted to get the maximum possible **sensitivity** and **selectivity** without oscillation.

The advantage of TRF receivers is that they are simple to construct and require relatively few components – typically just two or three valves or transistors and a handful of other parts.

This made them attractive in the days before transistors, when thermionic valves were used for amplification in radio receivers, as valves were relatively expensive so the fewer the better!

Their big disadvantage is that they have very poor **selectivity** and dynamic range. Tunable bandpass filters just aren't capable of rejecting an unwanted signal that is only a couple of kilohertz away from the signal you are listening to, so unwanted signals will also get through to the detector and be recovered as audio or cause **inter-modulation distortion**. TRF receivers are also best suited for receiving AM signals. Although **regenerative receivers** can be used with CW and SSB signals, by adjusting the regeneration control so the circuit just oscillates, adjustment is tricky and the quality of reception poor. For these reasons TRF receivers are not widely used any more.

The Direct-Conversion Receiver

A design that is used in quite a few homebuilt receivers **[and nowadays commercial designs]** is the Direct Conversion receiver. In a Direct Conversion receiver, the radio-frequency signal from the antenna is mixed with a locally generated oscillator signal, producing the usual sum and difference mixing products.

The frequency of the oscillator that generates this local mixing signal – it is known as the local oscillator (LO) or beat frequency oscillator (BFO) – is set so the difference mixing product is at audio frequency. In this way the Direct Conversion receiver "directly converts" the desired radio-frequency signal to audio, where it can be filtered and amplified. Let's look at the circuit in a little more detail.

A Direct-Conversion Receiver

The signal from the antenna first passes through a bandpass filter. Unlike in the Tuned Radio Frequency receiver, this bandpass filter is not responsible for the overall **selectivity** of the receiver – its role is simply to reject interference from strong local commercial broadcast stations and the like. It does not have to be tunable – usually a fixed-tuned filter covering an entire amateur band will suffice.

The signal is then amplified by an RF amplifier and fed into the product detector, which We have represented on the diagram using the symbol for a mixer – the circle with a cross in it. ("Mixer", "Modulator" and "Product Detector" are different names for essentially the same circuit, depending on the exact role it plays.) The product detector mixes the amplified RF signal with a signal generated by the tunable local oscillator, generating the usual sum and difference mixing product.

Suppose we want to receive an upper-sideband signal on 14,200 MHz. By convention, we refer to the frequency of a single-sideband signal as the frequency where the carrier would have been if it had not been suppressed. So the upper sideband of this USB signal (i.e. all that is left of it after the carrier and lower sideband were removed) will range in frequency from 14,200 3 MHz to 14,203 0 MHz, 300 Hz to 3 kHz above the (suppressed) carrier. If the local oscillator is set to exactly 14,200 MHz – the frequency where the carrier would have been – then the difference mixing products will range in frequency between 300 Hz and 3 kHz. What we have done is to translate the USB signal from its frequency of 14,200 MHz back to the audio frequency range.

This graph shows how mixing the 14,200 MHz USB signal with a 14,200 MHz signal from the local oscillator generated a difference mixing product (signal frequency – local oscillator frequency) in the audio range and a sum product (signal frequency + local oscillator frequency) up above 28,400 MHz.

Although the example used an upper sideband signal, the same process would work equally well using a lower sideband signal, and the local oscillator frequency would still be 14,200 MHz, the frequency where the carrier would have been. The following graph shows the same process with a lower-sideband signal.

Once again the difference product is back at audio frequency, while the sum product is at around twice the signal frequency, 28,400 MHz. Also note how for the lower side-band signal, the mixing process has inverted the sideband (so the recovered audio is the mirror image of the sideband), which makes up for the sideband inversion that would have occurred when the LSB signal was generated.

So whether the signal is USB or LSB, mixing it with a local oscillator with the same frequency that the carrier would have had will demodulate it and recover the audio.

To complete the hat trick, suppose we have a CW signal at 14.200 MHz. All we need to do is set the local oscillator just below it – say at 14.1994 MHz, which is 600 Hz below the CW signal – and the difference mixing product will be a 600 Hz tone, just right for listening to CW. So we can also use the product detector to receive a CW signal. (Setting the local oscillator 600 Hz above the CW signal would work just as well.)

We now pass the recovered audio through a low-pass filter. The main purpose of the filter is to remove the difference mixing product from signals near to the one that we are listening to.

For example, suppose there is a CW signal at 14.205 MHz while we are listening to our 14.200 MHz USB signal. The difference mixing product of the 14.205 MHz CW signal and the 14.200 MHz local oscillator is 5 kHz – in other words, we have translated the (unwanted) CW signal downwards in frequency to the audio range just as we have translated the (wanted) USB signal to audio. However a low-pass filter with a cutoff frequency of around 3 kHz or so should be able to remove the unwanted CW signal without affecting the desired USB signal.

Because it is quite easy for a strong signal to overload a mixer, causing inter-modulation distortion, the gain ahead of the mixer (i.e. the gain of the RF amplifier) is usually kept quite low so as not to amplify unwanted strong signals and overload the mixer. This means that most of the gain in a Direct Conversion receiver is at audio frequencies, in the amplifiers following the low-pass filter.

The only remaining part of the circuit is the **Automatic Gain Control (AGC)** system. Because there is such a wide range of signal strengths on the amateur (and other) bands, it is useful to have some way of automatically controlling the gain of the receiver, so it can have a lot of gain to amplify weak signals, but reduce this gain to avoid overload when amplifying strong signals. While this could be achieved with a manually operated gain control, this is not very operator friendly because when tuning from a weak signal (with the gain set on full) to a strong signal, the strong signal can be painfully loud. And when tuning from a strong signal (with the gain turned right down) to a weak signal, you might miss the weak signal altogether unless you remembered to turn the gain up.

The solution is automatic gain control. The AGC detector samples the audio signal after the first audio amplifier, and automatically adjusts the gain of the RF amplifier and the audio amplifier to keep the output signal level fairly constant. The output signal is then amplified by a final audio power amplifier and used to drive headphones or a speaker. The AGC control voltage is often also used to drive a signal strength meter, known as an “S meter”, that indicates the strength of the received signal using a fairly arbitrary scale calibrated from S1 (a very weak signal) to S9 (a very strong signal).

The Direct Conversion receiver has several advantages over a TRF receiver. Most importantly, its **selectivity** is very good, because unwanted nearby signals are easily filtered out by the audio low-pass filter that follows the product detector. It is more stable, having no tendency to oscillate like regenerative TRF receivers do. And it is easy to receive single sideband and CW signals with a Direct Conversion receiver – you just tune the signal in, without having to fiddle with the regeneration control.

However the Direct Conversion receiver does have one significant disadvantage. Since the same local oscillator frequency can be used to tune either a upper sideband or a lower sideband signal, if you are listening to say an upper sideband signal and there is a different signal occupying the frequencies on the other side of the local oscillator where the lower sideband would have been, then the other signal will also be shifted to audio frequencies and will interfere with the station you are trying to listen to.

For example, suppose you are listening to an USB signal at 14,200 MHz as before, but there is also a CW signal at a frequency of 14,199 MHz. Mixing the 14,200 MHz local oscillator signal with the 14,199 MHz CW signal will generate a 1 kHz audio tone. Since this falls within the same 300 Hz – 3 kHz audio range as the desired USB signal, you cannot filter it out using the low-pass filter. And because the unwanted signal is so close in frequency to the desired signal, you can't use the RF bandpass filter to reject it either.

The unwanted signal on the other side of the local oscillator signal is called an "image", so the principal disadvantage of the Direct Conversion receiver can be described as its inability to reject images, or lack of "image rejection". There are more sophisticated variations of the basic Direct Conversion design that are able to reject images, but these are quite complex and fall outside the scope of this course.

Summary

The key attributes of a receiver are **sensitivity**, **selectivity** **and dynamic range**.

- **Sensitivity** is the ability to receive weak signals; **[Specific to BANDWIDTH of signal]**
- **Selectivity** is the ability to distinguish between nearby signals; **[Rejection of unwanted signals]**
- **Dynamic range** is the ability of the receiver to receive signals of widely different signal strengths. **[Rejection of strong signals at the same time as receiving a weak signal]**

In the tuned radio frequency receiver all signal filtering is done at radio frequencies. As a result they have poor **selectivity**. Regeneration, which consists of feeding some of the output signal back to the input of the RF amplifier, can increase both the **sensitivity** and **selectivity** of the TRF receiver, but makes it prone to oscillation. **[Interferes with nearby receivers – emc. Chapter 27]**

In the direct-conversion receiver, the incoming RF signal is mixed down to audio frequency using a product detector **[mixer]** and local oscillator. Most of the **selectivity** of a direct conversion receiver is contributed by audio filters following the product detector. Direct conversion receivers have much better **selectivity** than TRF receivers, but they suffer from an **image response** to the opposite sideband that can only be eliminated with complex designs. **[NOTE – In years gone by, complex designs meant expensive. Nowadays it can be easily and cheaply fabricated using I.C.'s and multiple transistors. Or in software digital signal processing. D.S.P.]**

http://en.wikipedia.org/wiki/Radio_receiver_design

Regenerative and Reflex Receivers - This file contributed by Kim Smith and The Radio Electronique.
http://pe2bz.philpem.me.uk/Comm/-%20Receivers/-%20Regenerative/Info-900-Regen-Misc/regenrx.htm

Chapter 23 - The Super-heterodyne Receiver

The Single-Conversion Superhet

The super-heterodyne receiver or “Superhet” as it is commonly known is the most widely used receiver design in amateur radio. It overcomes the lack of **image rejection** of the Direct Conversion receiver by converting the incoming RF signal to one or more **intermediate frequencies** **[I.F.]** before demodulating it. The block diagram of a typical single-conversion superhet (one with only a single intermediate frequency) is shown below.

A Single-Conversion Superhet Receiver

The RF signal from the antenna is first filtered by a band-pass filter. As in the Direct Conversion receiver this can be a fixed-tuned filter covering an entire amateur band, since the receiver does not rely on this filter (known as the pre-selector) for its **selectivity**. As we shall see, the main purpose of the pre-selector is to reject the image frequency. The signal is then amplified in an RF amplifier – once again, not too much amplification, to avoid overloading the mixer that follows (in some designs the RF amplifier may be omitted entirely).

In the first mixer, the RF signal is mixed with the signal from the tunable local oscillator. But instead of mixing it down to audio, this converts it to an intermediate frequency (IF).

Common intermediate frequencies for single-conversion superhets are 455 kHz, 9 MHz and 10,7 MHz.

Suppose for example we want to receive a signal on 14,200 MHz again, and the intermediate frequency is 9 MHz. Then we could use a local oscillator frequency of either 5,200 MHz (because the difference between 5,200 MHz and 14,200 MHz gives the IF frequency of 9 MHz) or 23,200 MHz (because the difference between 14,200 MHz and 23,200 MHz is also the IF frequency of 9 MHz). For this example, we will assume that we chose a local oscillator frequency of 5,200 MHz, since this is within the range that can easily be generated by a VFO.

The resulting 9 MHz IF signal is then filtered by the IF filter, which is a very narrowband bandpass filter. Modern designs typically use crystal filters, so for this example we shall assume a crystal filter with a pass-band of 9.0003 MHz (300 Hz above 9 MHz) to 9.0030 MHz (3 kHz above 9 MHz). Signals within the pass-band will be passed with little attenuation, while signals that fall outside the pass-band will be blocked. So what components of our original RF signal will fall within the filter pass-band? Well an RF signal at 14.2003 MHz would be mixed down to 9.0003 MHz by the 5.2 MHz local oscillator signal; and a signal at 14.2030 MHz would be mixed down to 9.0030 MHz. So the signals that originated at these frequencies – from 14.200 3 to 14.203 MHz – will make it through the IF filter. This corresponds to an USB signal at a frequency of 14.200 MHz.

What about signals on the “other side” of 14.200 MHz, from 14.1970 to 14.1997 MHz, i.e. the frequencies that would have caused an image in a Direct Conversion receiver? Well, they will be mixed down to between 8.9970 MHz and 8.9997 MHz, and will be rejected by the IF filter, so they do not cause a problem.

There is still an image, but in this case it is from 3.8003 MHz to 3.8030 MHz. A 3.8003 MHz signal mixed with our 5,2 MHz local oscillator will generate an additive (sum) product at 9.0003 MHz, and a 3.8030 MHz signal will generate a mixing product at 9.0030 MHz. So signals within the frequency range 3.8003 MHz to 3.8030 MHz when combined with the 5.2 MHz local oscillator signal will also generate products in the IF range from 9.0003 to 9.0030 MHz that will be passed by our IF filter.

However this time the image is far away from the desired signal at 14.200 MHz, so it can easily be filtered out before the mixer, and this is the main purpose of the pre-selector. It must pass the desired frequencies, around 14.2 MHz, while rejecting the image frequencies, around 3.8 MHz.

Fortunately because these frequencies are so far apart, it is fairly easy to get good “image rejection” from a simple bandpass filter made of inductors and capacitors.

To find the image frequency, just find the sum of, and difference between, twice the IF frequency and the desired receive frequency. So for the example above, with an IF of 9 MHz, twice the IF is 18 MHz. The sum of 18 MHz and the desired receive frequency of 14.2 MHz is 32.2 MHz. This is where the image would be if the design used a local oscillator with a frequency higher than the desired signal. The difference between twice the LO frequency, 18

MHz, and the desired receive frequency, 14.2 MHz, is 3.8 MHz, and this is where the image frequency will be with the local oscillator running at a lower frequency than the desired receive frequency, as it is in the example above.

Note that by varying the frequency of the local oscillator we can change what frequency RF signal will be mixed down to the 9 MHz IF. For example, a local oscillator frequency of 5.3 MHz would mix an RF signal of 14.300 MHz down to the 9 MHz IF, while our original reception frequency of 14.200 MHz would now be mixed down to 8.900 MHz and would be blocked by the IF filter. So can you tune a superhet receiver by varying the frequency of its local oscillator (the same as for a Direct Conversion receiver).

The circuitry after the IF filter is virtually identical to that of a Direct Conversion receiver. The IF signal is amplified, and then mixed with another locally generated oscillator signal – this time called the "Beat Frequency Oscillator" or BFO – to recover the audio signal, which is then amplified by an audio amplifier. Since the IF signal is at a fixed frequency – 9 MHz – the BFO does not have to be tunable so we can use a stable fixed-frequency 9 MHz crystal oscillator for the BFO.

The Automatic Gain Control (AGC) also works similarly to that of a direct conversion receiver, although in this case the AGC control voltage is derived from the intermediate frequency, rather than the audio frequency output. This gives us "IF-derived AGC" as opposed to the "audio-derived AGC" that we had in the direct-conversion design. IF-derived AGC is superior to audio-derived AGC as it is able to respond more rapidly to sudden changes in signal strength.

The same design can be used to receive CW signals as well. For example, to receive a CW signal with a frequency of 14.200 MHz, the local oscillator would be set to 5.1994 MHz, generating an IF signal at the difference between these frequencies, 9.0006 MHz, which is within the pass-band of the crystal filter. After being amplified it will be mixed with the 9.000 MHz BFO signal in the product detector, generating an audio tone of 600 Hz.

So how about lower sideband signals? Well the simplest approach would be to have a second IF filter with a pass-band from 8.9970 (3 kHz below 9 MHz) to 8.9977 MHz (300 Hz below 9 MHz) that can be selected in place of the 9.0003 to 9.0030 MHz filter when we want to receive an LSB signal. Then when switching from USB to LSB all you have to do is switch filters, the local oscillator and BFO frequencies remain the same. Since crystal filters are quite expensive, an alternative approach is to use the same IF filter for LSB and USB reception, and just change the frequencies of the local oscillator and BFO. For example, to receive a LSB signal at 14.200 MHz using the 9.0003 – 9.0030 MHz IF filter we could set the local oscillator to 5.1967 MHz and the BFO to 9.0033 MHz. We leave it to the reader to fill in the details.

Since we can receive USB, LSB and CW signals using this design, how about AM signals? Well there are two options. The simplest is just to leave the receiver design exactly as it is, and receive AM signals as though they were single-sideband signals, ignoring the carrier and the other sideband, which will be filtered out by the IF filter. A better approach would be to provide another selectable IF filter, this time with a pass-band from 8.997 to 9.003 MHz to accommodate the 6 kHz bandwidth of an AM signal. The product detector would then be designed so that in the absence of any signal from the BFO, it would act as a half-wave rectifier and would detect AM by rectifying the IF signal (an "envelope detector"). This would give us the benefits of "proper" AM demodulation, notably accurate reproduction of the frequencies of the original audio signal even if the receiver is not perfectly tuned.

Multiple-Conversion Superhet Receivers

When choosing the IF frequency for a single-conversion superhet, there is a trade-off between image rejection and **selectivity**. It is easier to make highly selective filters at a low IF – say 455 kHz. However a low I.F. means that the **image frequency** is close to the desired frequency, making it difficult to effectively reject the image. Conversely, a high IF makes a large separation between the image frequency and the desired signal, making it easy to reject the image while passing the desired signal. However a high IF makes it harder to achieve the desired **selectivity**.

The classical solution to this dilemma has been to use a superhet design with two intermediate frequencies – a high first IF for good image rejection, followed by a low second IF for good **selectivity**. However modern crystal filters generally make this unnecessary in HF receivers, since very good **selectivity** is available from crystal filters at intermediate frequencies in the H.F. region, which is a high enough IF to attain good image rejection as well. Of course in VHF and UHF receivers, a higher first IF may be required to prevent unwanted image responses.

Despite this, the multiple-conversion superhet is still the most common approach for commercial HF receivers, but for a slightly different reason. Most commercial receivers and transceivers today offer “general coverage receive”, meaning that they can receive on any frequency in the MF and HF bands, typically from 500 kHz to 30 MHz. Unfortunately this gives them a problem with IF leak-through, which occurs when the first mixer is not exactly balanced, allowing some of the original RF signal to appear at the IF output. If the RF signal is at the same frequency as the IF, then it will be passed by the IF filter, causing the radio to respond to a frequency that it shouldn’t, a phenomenon known as a “spurious response”. This would not be a big problem for an amateur-bands-only receiver, because an IF frequency like 8,5 MHz could be chosen that is not close to any amateur band. Then the pre-selector, possibly assisted by a dedicated notch filter at the IF frequency, will be able to reject incoming RF signals at the IF frequency, so there are no signals in the RF input that could “leak through” into the IF stages.

However the designer of a general-coverage receiver is not so fortunate. If the chosen IF frequency is anywhere in the receiver’s frequency range, then it will be impossible to reject RF signals at the IF frequency, since these might include the frequency the receiver is tuned to! The solution is to choose an IF frequency that is either above or below the receiver’s frequency range. However now the **selectivity** versus image rejection tradeoff comes back with a vengeance because a filter that is above the frequency range of a typical general coverage HF receiver – that is, above 30 MHz – will not have the necessary **selectivity**; while a filter at an IF that is below the receiver’s coverage – say 455 kHz – will not allow adequate image rejection.

The usual solution is a multiple-conversion superhet where the first IF is above the receiver coverage range, allowing good image rejection and IF leak-through rejection, while the second IF is at a lower frequency where better **selectivity** can be obtained. This is known as an “up-conversion” design, since the incoming signal is first converted up to a higher frequency. The IF filter at the high first IF is often referred to as a roofing filter and is generally wide enough to permit signals of all modes through, up to 12 or 15 kHz in the case of a receiver that supports FM as well as other modes. Much narrower filters are provided for the different modes (e.g. a 6 kHz filter for AM and a 2,4 kHz filter for SSB) at the lower second IF. The block diagram below shows the “front end” (the circuitry from the antenna to the IF filter) of a typical general-coverage dual-conversion superhet.

Front-End of a General Coverage Dual-Conversion Superhet

The design includes a bank of switched bandpass filters in the pre-selector, to allow coverage of the range 0.5 – 30 MHz with good image and IF leak-through rejection. The first local oscillator is a frequency synthesizer running from 60.5 to 90 MHz, which up-converts the RF signal to the first IF of 60 MHz. Here it is filtered by the roofing filter, which would typically have a bandwidth of 12 kHz or so. The purpose of the roofing filter is to reject signals which are close enough to the desired frequency to be passed by the pre-selector, but which might cause either inter-modulation distortion or an image response in the second mixer. The IF signal is then amplified and converted back down to the second IF frequency of 9 MHz. From here on the circuitry would be similar to the single-conversion design featured earlier.

Noise Limiters and Blankers

Many common sources of amplitude-modulated noise generate amplitude “spikes” of short duration but high amplitude, which extend over a wide range of frequencies. These may contain substantial energy due to their large amplitude, even though their duration is short. Such noise is generated both by natural sources, such as thunderstorms, and by man-made ones, like inadequately suppressed ignition systems. Interference from these noise sources can be reduced by noise limiters and blankers, which are available on almost all modern amateur transceivers.

A noise limiter is a very simple circuit that limits the maximum amplitude of the received signal.

Circuit Diagram of a Noise Limiter

Assume the input signal has a maximum amplitude of 0.5 V peak under normal circumstances. This is less than the 0.6 V forward bias voltage of the diodes, so they do not conduct, and the input signal will be passed to the output unchanged. Then suppose a noise pulse generates a signal amplitude of 5 V. As soon as the amplitude exceeds 0.6

V, the diodes will conduct, effectively limiting the maximum output to 0.6 V peak and substantially reducing the energy of the noise signal.

The noise blanker is a more sophisticated variation on this idea. It detects the large amplitude of the incoming noise signal, and then immediately mutes (turns off) the audio output of the receiver completely for a predetermined time, typically a few milliseconds. Although this blocks the desired signal as well as the noise, this usually goes unnoticed by the listener as the human ear is quite insensitive to very short gaps in sounds, and the resulting signal degradation is much less than would have been caused by the high amplitude noise spike.

Frequency Modulation (FM) Reception

The basic superhet design can also be used to receive frequency modulated (FM) signals. However in this case, the product detector is replaced by a **Foster-Seeley discriminator** or a **ratio detector**. **[OK, This is no longer the case. Most modern FM discriminators use a simple 90 degree phase shifting circuit.]** These are circuits that convert frequency variations into a varying output voltage, so recovering the modulation from an FM signal.

The **discriminator** works by positioning the FM signal on the slope of a selective filter, so that variations in the frequency of the FM signal will result in variations in its amplitude. This converts the frequency modulation into a combined amplitude and frequency modulation, and a diode detector is used to recover the modulation from the AM component.

The graph shows how the slope of a high-pass filter could be used to convert frequency modulation into amplitude modulation. As the signal frequency increases from FC, the centre frequency, to FHIGH, the amplitude of the output increases from AC to AHIGH. If the frequency decreases from FC to FLOW, then the amplitude of the output will also decrease, from AC to ALOW.

Because the discriminator is also sensitive to changes in the amplitude of the incoming signal, it should be preceded by a limiter. This is a circuit that limits the amplitude of the signal, so that amplitude variations are not passed on to the discriminator or ratio detector that follows. The limiter circuit is identical to the noise limiter discussed earlier, except that in an FM receiver the circuit would be driven at a much higher input level, causing the diodes to conduct and clamp the output signal to 0,6 V peak. In this way the output of the limiter will always be at the same level (0,6 V peak), irrespective of the amplitude of the input signal. The block diagram below shows the final IF stage of a typical FM receiver

Final IF Stage of an FM Receiver

When the received signal is very weak the limiter is ineffective and the discriminator will respond to amplitude variations, which cause hiss in the audio. As the signal gets stronger and the limiter takes effect, the hiss decreases, a process called "**quieting**". In order to prevent the hiss from bothering the listener when there is no received signal, most FM receivers incorporate a **squelch** feature, which mutes (turns off) the audio output when the received signal is below a minimum level known as the squelch threshold. The squelch threshold may be fixed or it may be adjustable using a squelch control.

Summary

The 'superhet' receiver converts the incoming RF signal to one or more **intermediate frequencies** before demodulating it. Superhet receivers have an **image frequency** that when mixed with the local oscillator will also generate the same **I.F.** as the desired receive signal.

The **image frequency** will be either the sum of, or the difference between, twice the IF frequency and the desired receive frequency. The role of the **pre-selector** is to reject incoming RF signals at the image frequency, preventing them from causing a spurious (unwanted) response in the receiver. The choice of intermediate frequency is a trade-off between **selectivity** (better at low frequencies) and image rejection (better with a higher frequency I.F.).

If a single IF cannot give adequate **selectivity** and image rejection, then a **dual conversion** design may be employed, with a higher first I.F. to give good image rejection, and a lower second I.F. to give good **selectivity**.

Noise limiters limit the amplitude of pulse noise, reducing the effect on the receiver. Noise blankers mute the audio output for a short time (a few milliseconds) when the higher amplitude associated with pulse noise is detected.

FM signals are detected using a **Foster-Seeley discriminator or ratio detector**. The discriminator should be preceded by a limiter to prevent it from being affected by variations in the amplitude of the signal. Weak FM signals have a characteristic hiss on them, and as the signal strength increases and the limiter becomes effective the hiss goes away, a process known as quieting. Most FM receivers incorporate a **squelch function, which mutes the audio output** when there is no received signal to avoid the annoying hiss.

http://users.tpg.com.au/users/ldbutler/Superhet.htm

http://zpostbox.narod.ru/tuned_radio_frequency_receiver_e.html

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Chapter 28: Electromagnetic Compatibility

28.1 Definition of Electromagnetic Compatibility

Electromagnetic compatibility (EMC) is the process of ensuring that equipment that radiates electromagnetic radiation, such as an amateur transmitter, **does not interfere** with equipment that may be sensitive to **electromagnetic radiation**, such as television and radio receivers, pacemakers and computers.

As more and more electrical gadgets come into operation, the problem of mutual interference becomes worse and worse. The **electromagnetic spectrum is increasingly polluted** to an extent that makes radio communications harder and harder. In large cities, the problem may become so bad that some have referred to electromagnetic smog as a way of describing the noise produced by millions of discrete devices in a city. EMC seeks to alleviate this pollution by reducing the amount of noise generated and by addressing the immunity of devices to that noise.

There are two considerations when dealing with interference problems. The first consideration is technical: the causes of interference, and how they can be eliminated. The harder consideration is a legal and social one: Who is responsible for solving the interference problem? The problem is both legal and social because often an attempt to use a legal approach will generate undesirable social results.

Interference can be caused by **intentional** or **unintentional radiators**, and can take place to a device that is a receiver (that is designed to receive radio signals) or is not a receiver (that is not intended to receive radio frequencies, but is experiencing interference nevertheless).

28.2 Intentional and Unintentional Radiators

An intentional radiator is a device that is intended by virtue of its function to radiate, such as an amateur transmitter or a garage opening remote control.

An unintentional radiator is a device that does not need to radiate in order to perform its intended function, such as a motor vehicle ignition system or an electric fence.

There are strict limits to the maximum permitted radiation from unintentional radiators. If a system that does not include a radio transmitter of some kind is causing interference, that is generally because the system is radiating more than permitted, and it should be repaired or replaced at the owner's expense.

For example, if you receive interference from a neighbour's electric fence, that probably indicates that the electric fence is radiating more than is permitted, and the neighbour is responsible for having the defect rectified, and must turn the electric fence off until it complies with requirements. Of course convincing your neighbour of this obligation may not be so easy!

In general, any switch generates some radio noise while switching high currents. As the contact is made or broken, some sparking may occur, leading to the transmission of a noise burst. This noise may interfere with radio communications at a considerable distance. In fact, millions of such switches in an urban environment contribute to a gradual raising of the noise level, until all radio communication within the city becomes hampered.

28.3 Interference to non-receiving equipment

The converse applies when the equipment being interfered with is not intended to receive radio signals. For example, suppose your neighbour reports that your radio transmissions are "breaking through" on their stereo system when they are listening to CDs. Because the stereo system when listening to CDs is not supposed to receive radio signals, the problem lies with the stereo, not with the radio transmitter.

Often the root cause is that the affected equipment was not designed for, and has not been tested in, environments with strong RF signals present. Unfortunately it is quite legal for such equipment to be sold, and it will work fine for 99% of the time, since in most locations it will encounter only weak signals from distant transmitters. Then an amateur moves in next door, sets up equipment that is operating within the limits of their licence, and all of a sudden the neighbour's CD player receives interference. It is quite natural for the neighbour to think that this is the amateur's fault, and that they must fix the problem or stop transmitting. However, the fault lies with the manufacturer of the equipment for not designing it to withstand the levels of electromagnetic signals that may result from a nearby transmitter.

In this case, even though it is the neighbour's responsibility to solve the problem, it would be diplomatic for the amateur concerned to make his or her technical skills available to the neighbour to help diagnose the problem and suggest solutions. Apart from good neighbourliness, the same neighbour may have the opportunity to comment on your application to erect a tower, and is more likely to be kindly disposed to such a request if you have helped them to solve any problems that appear to have been caused by your transmissions in the past!

28.4 Intentional Radiators interfering with Receivers

The situation is slightly more complex if an intentional radiator (such as your amateur transmitter) interferes with a device that is intended to receive radio signals (such as your neighbour's television set). In this case, the key question is the nature of the interfering signal.

If the interfering signal is in all respects a legal licenced transmission—that is, it is within an amateur band, does not exceed the power permitted for the band and licence holder, and is a **clean signal**—then the problem is being caused by the receiving equipment being affected by an out of band signal, and it is the receiving equipment that is defective and must be repaired.

On the other hand, if the transmitted signal in any way does not conform with the requirements of your licence, you should first correct the problem with the transmitted signal before suggesting to your neighbour that they have their TV fixed! If interference is reported to ICASA (the regulator), their first course of action will probably be to inspect the transmitting equipment. If it is found to be out of order in any way, you may be held responsible for the interference and, even if you are not, the transmitting equipment can be confiscated if it does not comply with your **licence requirements**.

Once again, as a matter of diplomacy, it is a good idea to assist your neighbour if possible to solve the interference problem, even if you have determined that your transmitter is operating quite legally. As well as maintaining peace in the neighbourhood, this course of action will help to maintain the good reputation of amateur radio. However, if this is not possible—for example, if your neighbour refuses your assistance and insists that you just stop operating—then as long as you are certain that your equipment is operating legally, you are entitled to continue to operate despite the interference to your neighbour's television or other equipment.

When offering technical assistance to resolve interference problems, remember that you may be held liable for any changes you make to the neighbour's installation that may later lead to problems. If you solder a filter into your neighbour's speaker leads and the sound system suddenly stops working a few months later, you are likely to have fingers pointed at you.

28.5 Shared Bands

One exception to our classification is that some amateur bands are shared between different users, with one of the users being declared the "primary" user and the other as "secondary" users. For example, amateur radio has been allocated the 13 cm band (2,3 to 2,45 GHz) on a secondary basis; the primary use is industrial, scientific and medical.

Simply put, secondary users may not cause interference to primary users (and must stop operating if this is the only way to prevent interference), while they must accept interference from primary users. So if you live next door to a hospital and receive interference from medical equipment that is intentionally radiating in the 2,4 GHz band, there is nothing you can do about it.

Of course all amateur bands are shared with other amateurs, and it is important that we take steps to avoid interfering with our fellow amateurs. These steps should include operating courtesy and ensuring that your transmitter is radiating a clean signal.

28.6 Causes of Interference

There are three possible causes of interference.
1. The transmitter may be radiating on a frequency that it should not be radiating on.
2. The receiver might be receiving signals that it should not be.
3. The transmitter and receiver may both be working correctly, but something else is translating the transmitted signal to the frequency of the receiver. For example, corrosion in a gutter can cause the metal to operate like a rectifier, re-radiating harmonics of signals transmitted from a nearby transmitter.
4. Other sources of noise, such as high-power electrical switches or motors, or even natural phenomena such as lightning, can cause noise that will influence receivers, both intentional and non-intentional.

Since the third mode is quite uncommon and usually requires specialised equipment and significant expertise to resolve, we will only look at the first two possibilities.

28.7 Transmitter Defects

The most common problems in transmitters are **frequency instability**, **harmonic radiation**, **spurious oscillations**, and “**wide**” **signals**.

28.7.1 Frequency instability

Frequency instability is usually the result of LC (inductor/capacitor) oscillators that have not been adequately compensated for temperature variations or protected against mechanical shock. It is most likely to impact on other amateurs, unless the instability is sufficient to take the transmitter out of the amateur band and cause interference to other services. Fixing frequency instability usually requires design modifications or improved construction methods (for example, more solid construction that is less sensitive to mechanical knocks). It is quite uncommon with modern crystal-controlled synthesised radios, although it may occur if a PLL frequency synthesiser gets unlocked from the reference frequency.

28.7.2 'chirp'

Another type of frequency instability is '**chirp**', which occurs when the oscillator frequency is affected by the loading of subsequent stages or by fluctuations in the power supply voltage when a CW transmitter is keyed. It can be prevented by using a high-impedance buffer amplifier after the oscillator; and by regulating the oscillator voltage supply.

28.7.3 Harmonic radiation

Harmonic radiation occurs on multiples of the transmitter output frequency. For example, a transmitter operating at 144 MHz may interfere with a television receiver operating at 720 MHz (144 MHz x 5). It can be caused by overdriving an amplifier stage (for example by having the microphone gain or CW drive level set too high) or by inadequate attenuation of harmonics by the transmitter’s output lowpass filter (e.g. when the output controls on the amplifier are improperly adjusted).

If the problem is caused by **overdriving the transmitter**, the solution is to reduce the drive level by adjusting the microphone gain or CW drive correctly. However, if the problem persists even when the transmitter is not being overdriven, the best solution is to add an additional lowpass filter between the transmitter and the antenna. **Lowpass filters** for the HF bands (up to 30 MHz) are available at reasonable cost and provide substantial attenuation at higher frequencies, typically 50 dB or better at 50 MHz.

Another solution sometimes recommended is to use an antenna tuning unit (**ATU**) even when it is not required to match the antenna, as the ATU may attenuate out-of-band signals.

When doing so, ensure that the **ATU is a lowpass filter**, and not a highpass filter. A Pi network with a single inductor or a T network with two inductors should do the trick.

28.7.4 Spurious oscillations

Spurious oscillations may either be **self-oscillation**, at or near the intended frequency of operation of an amplifier or mixer, or **parasitic oscillations**, which usually occur at VHF or UHF frequencies far away from the intended frequency of operation. **Self-oscillation** is caused by unintended feedback from the output of an amplifier or mixer that includes tuned circuits to its input, causing oscillation at the resonant frequency of the tuned circuit. It can be suppressed either by reducing the coupling (for example by shortening component leads) or by introducing **negative feedback** to reduce the loop gain and prevent oscillation.

Parasitics are VHF or UHF oscillations that occur due to unwanted "hidden" resonances in oscillators and amplifiers—for example, between RF chokes and decoupling capacitors, or due to the inductance of capacitor leads at high frequencies. They can be eliminated by using **low-Q (lossy) RF chokes**, which are less likely to cause oscillations, or by using **ferrite beads** to add sufficient inductance to component leads or wires to dampen out unwanted VHF or UHF oscillations.

28.7.5 "Wide" Signals

"**Wide**" signals are signals where the bandwidth exceeds the minimum required due to **inter-modulation distortion**. The cause is usually that some amplifier stage is being overdriven, and while this may result from a design defect it is more often caused by an incorrectly adjusted microphone gain control or CW drive level. On most modern transmitters the **ALC** (**automatic level control**) voltage can be monitored on the transmitter's meter during transmissions. The microphone gain or CW drive level should always be adjusted so the voltage remains within the acceptable **ALC** levels at all times.

These levels are usually marked on the meter.

28.7.6 Excessive Bandwidth

Another cause of wide signals is amateurs intentionally "opening up" the audio paths on their transmitters to allow the broadcast of wideband audio signals that exceed the **3 kHz bandwidth** required for communications quality in the pursuit of "fidelity", but at the cost of causing interference to other operators.

28.7.7 Key clicks

A **CW** transmitter may generate key clicks if the carrier is switched on or off too rapidly when keying. The carrier should be turned on or off gently over a period of about 5 ms to avoid generating key clicks. Unfortunately, even some very well-regarded modern transceivers like the original FT1000 MP have a problem with key clicks and may need to be modified to reduce clicks to acceptable levels.

If a high-power stage is keyed directly, arching of the key contacts may result. The solution is to key a lower-power stage and then feed the resulting keyed signal to amplifier stages, or to use an intermediate switch like a transistor or relay to do the actual keying, keeping the high power away from the key itself.

28.7.8 Mains Hum

Mains hum may be heard on transmitted signals if the **power supply** is **inadequately filtered**.

The addition of a **voltage regulator** or additional **smoothing capacitors** should solve the problem.

If a transmitter is using an antenna like a long wire that is driven against earth, it is important to have a good RF earth system that is independent of the mains earth. The earth lead must be as short as possible and must be routed as directly as possible. The mains earth wire usually travels in close proximity to the other mains wires for some distance before being physically earthed, so RF signals in the mains earth are likely to be inductively coupled to the live and neutral wires and may travel through them to neighbouring buildings, causing interference, especially to mains-operated equipment.

28.7.9 Mains earth

The mains earth also often has high impedance at RF frequencies, so an independent earth system is necessary to remove RF voltages from equipment and antenna feed-lines. Of course, even if you cannot provide a good RF earth, a mains ground is still required to prevent the case from having a potentially lethal voltage in the case of a fault.

28.8 Receiver Defects

The most common defect in radio and television receivers that results in interference from amateur transmissions is **receiver overload**. Signals stronger that the receiver was designed to handle are present at the receiver input, and **inter-modulation distortion** in the first mixer causes spurious products that interfere with reception.

One common cause of over is inexpensive masthead RF preamplifiers that are sometimes used to improve television reception in marginal areas. While preamplifiers with decent signal-handling capabilities are available, they are generally more expensive, and the inexpensive ones that are widely available are very prone to overloading.

A solution to **receiver overload** is to add additional filtering before the receiver that removes the strong out-of-band signals that are overloading the receiver. What type of filter is required will depend on what frequency transmissions are causing interference. If transmissions in the HF bands are causing the problem, a **highpass filter** between the TV antenna and the TV might solve the problem, since the TV transmissions are on higher frequencies in the VHF and UHF region, so these frequencies can be passed while blocking HF frequencies. Obviously, if a masthead preamplifier is in use, the filter must be on the mast too, between the antenna and the preamplifier.

If amateur VHF transmissions are interfering with UHF television reception, a highpass filter with a cutoff frequency of 470 MHz might solve the problem. However, if VHF transmissions are interfering with VHF television reception, a **bandstop filter** for the particular interfering amateur transmission band might be required. These bandstop filters are also called "**traps**". A **quarter-wavelength transmission-line "stub"** connected across the feed-line and open at the far end, may also serve as a trap. It presents a low impedance at the frequency on which it is exactly a quarter wavelength, effectively shorting the two conductors in the feed-line together at that frequency, while presenting a high impedance at most other frequencies. Signals move **more slowly in coaxial cable** than in free space, so a quarter-wavelength stub is shorter than an actual quarter wavelength.

However, note that if the problem is being caused by overloading a masthead RF preamplifier, no amount of filtering of the signal between the amplifier and the television will help, as in-band spurious products may already have been generated by the amplifier.

In this case, replacing the amplifier with one that is more resistant to overload (or removing it altogether if reception conditions permit) may be the only option.

28.8.1 Image Rejection

Interference to receivers may also result from **image signals**, also known as **second-channel interference**, if the image frequency of a receiver coincides with the frequency on which a strong amateur signal is present and the receiver has insufficient image rejection.

Assessing Interference Sources

When hearing interference from a nearby transmitter, the operator must decide whether the interference is caused by the transmitter or by the receiver. Because transmitters and receivers use very similar techniques to generate and demodulate the signal, they suffer from very similar types of interference.

The key in determining whether it is a transmitter or a receiver problem is the fact that most interference is caused by saturation of some kind. If the transmitter is driven too hard, the signal will be distorted because the final amplifier cannot handle the signal amplitude, resulting in adjacent-channel interference. Likewise, a receiver that is overloaded with a very loud nearby signal will saturate, causing very similar interference.

The key is in reducing the incoming signal strength in the receiver. Most communications-grade receivers include a switchable attenuator for exactly this purpose. In other cases, receivers feature a preamplifier that is used for weak-signal work, which can be turned off to reduce signal levels. Finally, the receiver's RF gain can be reduced. Either way, the reduced signal strength will probably solve the problem if it is receiver-generated.

When hearing a "wide" signal, simply engage the attenuator and observe the effects. Most attenuators provide something like 20 dB of attenuation. If the adjacent interference disappears or decreases by 40 dB or more when the attenuator is engaged, the problem is in the receiver. However, if the adjacent interference decreases by only 20 dB when the attenuator is engaged, the problem is in the transmitter. A friendly request to the offending operator may be in order.

Remember that any of these measures to reduce adjacent interference will also reduce the **sensitivity** of the receiver. When you insert attenuation, turn off the preamplifier or reduce the RF gain, you are sacrificing **sensitivity** in exchange for a reduction in interference. Even though the incoming signal becomes weaker, the resulting SNR has improved.

28.9 Common-Mode Chokes [Braid Breaker]

Interference usually "gets into" the equipment being interfered with through the wires attached to it. These wires include antennas, speaker leads, interconnections between audio components and mains power leads. In common-mode interference, the signal is transmitted in phase by both the conductors in the connection—for example by both the live and neutral wires in the mains, or both conductors in the speaker cable, or both the inner conductor and the earth in a coax cable.

Common-mode interference can be effectively eliminated by a **common-mode choke**, also known as a "**braid breaker**". Although it does not involve physically breaking the braid in a coax cable, it effectively blocks the flow of common-mode signals that travel along the braid as well as in the inner conductor, which is where the name comes from.

The **choke** consists of several turns of the cable—which could be a mains power lead, a speaker cable, or a coax cable—wound around a suitable core to form an inductor. Ferrite toroidal cores are the best, and are available for the purpose from local suppliers. The idea is that common-mode currents will generate a magnetic field in the core, and so the choke will act as an inductor to common-mode signals. If the inductor has sufficiently high impedance at the frequency causing the interference, this signal can be rejected.

However, **differential signals**—that is, signals where currents flow in opposite directions in the two conductors, for example the signal from the antenna in a TV antenna lead—will not generate a magnetic field since the fields generated by the two currents flowing in opposite directions cancel out; and so the **common-mode choke** does not act as an inductor for differential signals, which pass through unaffected.

Common-mode chokes can be used both with receiving equipment, such as television receivers, and with non-receiving equipment such as audio amplifiers that are suffering interference from strong radio signals.

28.10 Direct Radiation and Shielding

In a few cases, electronics may be directly influenced by strong electric fields. Currents can be induced into circuits without going through connecting cables.

The problem normally manifests when very strong electric and magnetic fields are present.

The most likely situation is for equipment situated near an antenna connected to a high-power transmitter. In such cases, interference can be coupled directly into the IF stages.

28.10.1 IF 'Breakthrough'

Such **IF interference** is characterised by it being present on all channels or frequencies that the device is capable of receiving.

Remember that an electromagnetic signal is composed of E and H fields. The E field is measured in V/m and the H field in A/m. These fields can be measured using a **field strength meter**.

The problem is normally solved by good design practices—using decoupling components such as capacitors within the circuit itself—and by shielding. Consumer devices are normally made very cheaply, and often do not comply with good design practices.

Manufacturers have a duty to solve problems that are due to design inadequacies, but local distributors are not always willing and capable to do so. The problem is therefore more likely to be solved by good shielding.

28.10.2 Shielding

Shielding consists of conductive enclosures that completely surround the circuitry, known as a Faraday Cage. Shielding should be solid or have only small holes. Holes that have a circumference of more than a fraction of a wavelength of the offending signal will be penetrated by the signal, leaving the equipment vulnerable to the strong field. Shielding also serves to reduce radiation of objectionable interference by the electronic devices.

Shielding against electric fields is relatively easy, as most metals are conductive to electricity and can be used to make Faraday Cage enclosures. If the coupling mechanism is predominantly magnetic, the problem is much harder to solve. Specific materials such as Permalloy or Mu Metal must be used, and the shielding must be relatively thick. Magnetic fields are normally not the dominant problem at higher frequencies (HF, VHF and up).

Transmitters can also be prone to direct radiation, this time outbound. Some transmitters can radiate energy that does not go through the antenna connector, through so-called cabinet radiations. The cause is normally stray currents inside the cabinet. As with **susceptibility** problems, the solution is not easy, as the bad equipment design is probably not easy to fix. The problem must be solved by improving screening. The transmitter must be inside a **Faraday Cage**. If the existing enclosure is not good enough, more work may be required, analogous to the suggestions for susceptibility given above.

28.11 Sensible Measures against Interference

Many types of interference can be alleviated by simple courtesy. Mount your antennas as high and as far from potential interference as you can. Use the minimum power required to facilitate the communications of the moment. Listen before you transmit. Much of the interference that results in practice is due to a violation of one or more of these simple rules.

Summary

EMC should be looked at from two perspectives: the technical (how to solve the problem) and the legal (who is responsible for solving the problem). If the interfering signal is being generated by equipment that does not need to transmit in order to function, it is this unintentional radiator that is usually at fault since there are strict limits as to how much electromagnetic energy can be radiated by unintentional radiators. If the equipment being affected is not intended to receive radio signals of some kind, the affected equipment is at fault. If a signal from an intentional radiator is affecting equipment that is designed to receive radio signals, the key question is whether the transmitter is operating within the frequency and power limits specified by its licence. If the transmitter is not radiating legally, the exceedances must be fixed. However, if the transmitter is operating correctly and within licence requirements, the problem is being caused by the affected equipment responding to an out-of-band signal, and ultimately it is up to the owner of the affected equipment to have the problem repaired at his or her expense.

However, it is advisable for an amateur whose transmissions are causing interference to assist as much as possible in diagnosing the cause of the problem and suggesting solutions.

This is both to maintain a good relation with neighbours and to maintain the good image of amateur radio. Just be wary of making changes to the neighbour's installation, as subsequent problems with the equipment may well be blamed on the helpful radio amateur.

The most common transmitter problems are frequency instability, harmonic radiation, "wide" signals and key clicks. Frequency instability requires due attention in design and construction to temperature compensation, mechanical rigidity and suitable buffering of oscillators to avoid chirp. Harmonic radiation can be attenuated by a suitable lowpass filter.

Wide signals are usually caused by setting the microphone gain level too high. Key clicks are the result of turning the carrier on or off too rapidly.

Receiver problems can be caused by common-mode or differential signals. Common-mode signals can be attenuated by a suitable common-mode choke (also called a "braid breaker").

Differential-mode signals require the use of suitable highpass or bandstop filters between the antenna and the receiver. Mast-head TV amplifiers are often subject to overloading. The amplifier may need to be removed or replaced with one that is less subject to overloading.

An attenuator can help to diagnose interference being received. If the attenuator attenuates the interference just as much as the signal causing it, the problem is in the transmitter. If the attenuator completely cures the interference or reduces it by much more than the offending signal, the problem is in the receiver.

Strong electromagnetic fields can couple directly into electronic equipment. The solution is good design of the target electronics and thorough shielding (a Faraday Cage). Shielding should not have large holes, failing which the radio signals will still penetrate the enclosure.

For coupling that is predominantly magnetic, special enclosures of special materials will be required.

Transmitters can also suffer from cabinet radiations. Fixing these problems is similar to the suggestions for shielding given above.

Simple courtesy requires that you operate in a way that minimises the risk of interference.

Very often, that's all that is required.

It is a 'question of scale'

In the communications world, we have signals that range from the Megawatts to Picowatts.

Measuring them is a difficult business...

Path Loss to the Moon and Back: 250 dB loss

https://en.wikipedia.org/wiki/Free-space_path_loss

$$\begin{aligned}\mathrm{FSPL} &= \left(\frac{4\pi d}{\lambda}\right)^2 \\ &= \left(\frac{4\pi d f}{c}\right)^2\end{aligned}$$

where:

λ lambda is the signal wavelength (in metres),
f is the signal frequency (in hertz),
d is the distance from the transmitter (in metres),
c is the speed of light in a vacuum, 2.99792458 × 10^8 metres per second.

This equation is only accurate in the far field where spherical spreading can be assumed; it does not hold close to the transmitter.

$$\begin{aligned}\mathrm{FSPL(dB)} &= 10\log_{10}\left(\left(\frac{4\pi d f}{c}\right)^2\right) \\ &= 20\log_{10}\left(\frac{4\pi d f}{c}\right) \\ &= 20\log_{10}(d) + 20\log_{10}(f) + 20\log_{10}\left(\frac{4\pi}{c}\right) \\ &= 20\log_{10}(d) + 20\log_{10}(f) - 147.55\end{aligned}$$

where the units are as before.

For typical radio applications, it is common to find f measured in units of GHz and d in km, in which case the FSPL equation becomes

FSPL(dB) = 20 * log10(d) + 20 * log10(f) + 92.45

Transmitter signal levels - TV transmitter [Aukland Park]
F. M. Radio transmitter -
Amateur Radio transmitter -
Medium Wave A.M. transmitter [BBC]

How "sensitive" is your receiver?
How many microvolts for a good signal to noise?
What is an 'S' meter reading? S9 is ?

If you were measuring an a.c. signal from the softest voice to a fog-horn, you would not be able to change the scale quickly enough. Add a "logarithmic amplifier" [log amp – for short] to your meter and you can measure both very low level signals and very high signals. Without having to change the range.[]

Your ear can hear differences in signal level from the very quietest to the very loud. [e.g. a Disco Club] . What's more you can 'discern' differences in level... [1 deci Bel - dB for short.]

Curves and Functions
Any curve [I do mean any!] can be described by a 'function'.
The curve of a wire on an overhead cable from tower to tower. [cosecant]
The current / voltage transfer of a diode. [approx square law]
How about working out what 1 milliWatt is in 50 Ohms?
Then work out what 1 micro Watt is in 50 Ohms... 1 uW?

If you use a moving coil meter to measure current or voltage, the chances are that you will "pin the meter".

A 'real' meter. [NOT A Metre!]

Say you were measuring a 0.7 Volt d.c. source. And instead of turning the scale to 1 Volt scale, you turned it to the 0.1 Volt scale. The meter will bang hard against the end stop...

There is a 'simple way' of stopping the meter from being damaged. Simply put a germanium diode across the meter coil, to limit the voltage to 0.15 Volts.