It is a 'question of scale'

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

A 'real' meter.

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.

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]

Chapter 11 - 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 * PR]**

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.

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.

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.

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.

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.

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.

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!

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).

Revision Questions

1 An **increase** in power from 0.25 Watts to 1.25 Watts is equal to a **gain** of:
a. 3.0 dB.
b. 7.0 dB.
c. 10.0 dB.
d. 1.0 dB.

2 A transmitter has a power output of 100 Watts. This is connected to an antenna with 11 dB gain by means of a coax cable with a loss of 1 dB. The ERP (effective radiated power) of the transmitter, coax and antenna combined is:
a. 11.0 Watts.
b. 111.0 Watts.
c. 1000.0 Watts.
d. 2000.0 Watts.

3 A (-)20 dB **attenuator** is placed in line with a 40 Volt RMS signal. Assuming the impedances all remain constant what will the reduced signal level be?
a. 2.0 V.
b. 10.0 V.
c. 20.0 V.
d. 4.0 V.

4 A power **gain** of 4 is equivalent to:
a. 3.0 dB.
b. 6.0 dB.
c. 10.0 dB.
d. 16.0 dB.

5 A signal with a power of 1 mW is applied to the input of an amplifier that has a **gain** of 13 dB. The power of the output signal will be:

a. 5.0 mW.
b. 10.0 mW.
c. 20.0 mW.
d. 100.0 mW.

http://www.engineeringtoolbox.com/adding-decibel-d_63.html

Note! Adding of two identical sources [of power] will increase the total [sound] power level with 3 dB (10 log(2)).

FROM: http://www.dogstar.dantimax.dk/tubestuf/thdconv.htm

A power gain of 4 is equivalent to:

a) 3 dB.

b) 6 dB.

c) 10 dB.

d) 16 dB.

49 'n Drywingswins van 4 is gelykstaande aan:

a) 3 dB.

b) 6 dB.

c) 10 dB.

d) 16 dB.

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.

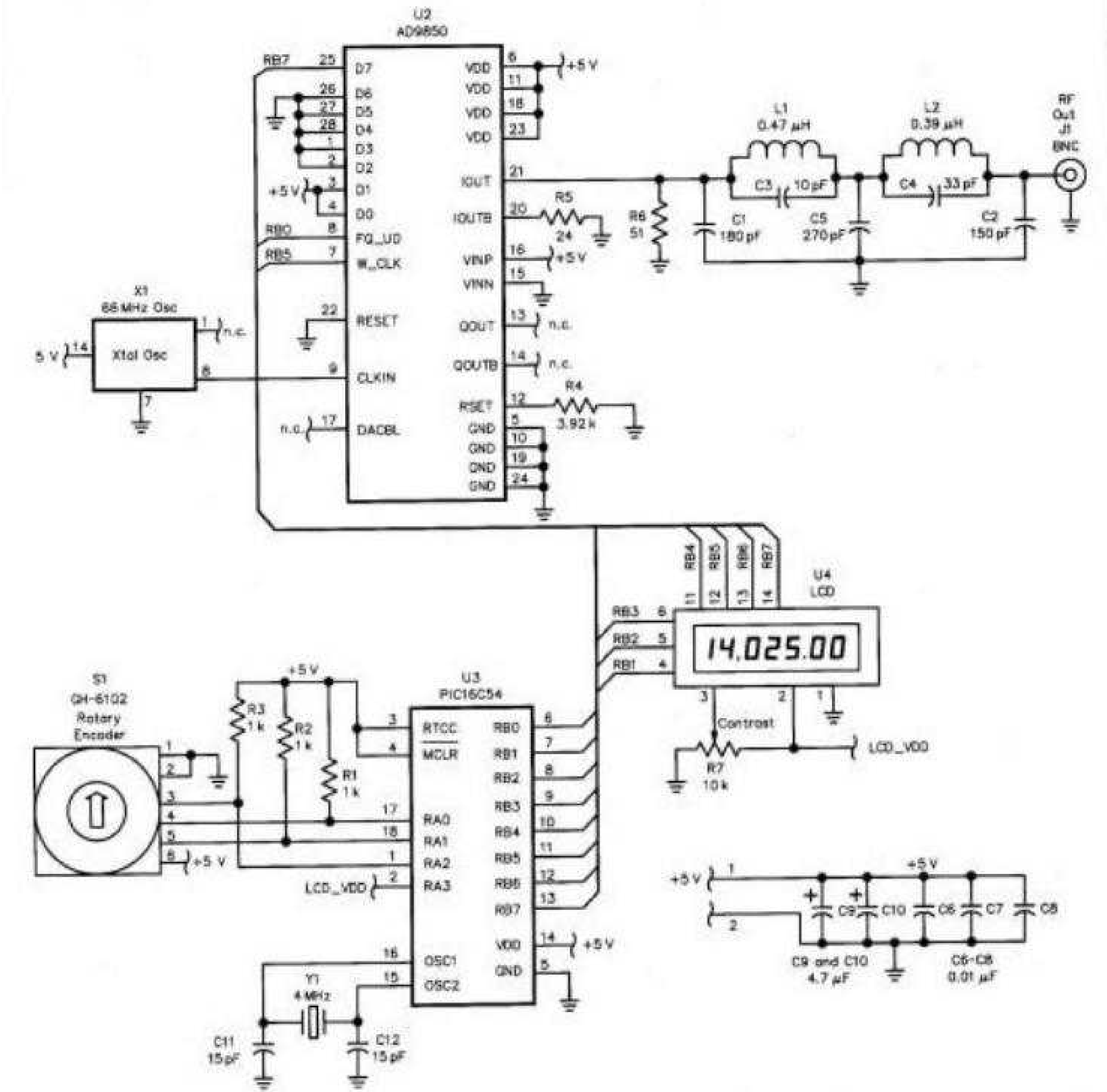
(23) The frequencies required by the licensee must be **selected in such a manner** that **no power is radiated at frequencies other than those referred to in the amateur radio frequency plan**, provided that the bandwidth of emissions on bands that have been allocated to the amateur radio service in terms of these regulations shall be **restricted to the minimum**.

Chapter 12 – Filters – prelude

- So you want to use a Raspberry Pi as a transmitter for WSPR...
- Or you want to try a DDS 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.

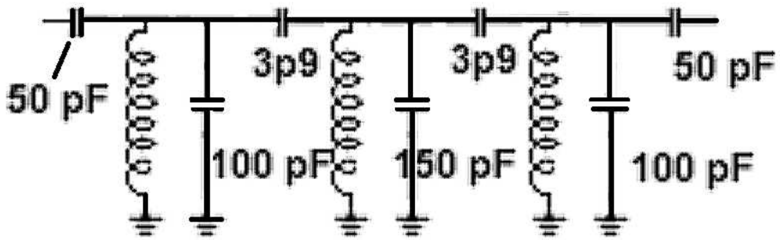


Top right is a low pass filter.

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

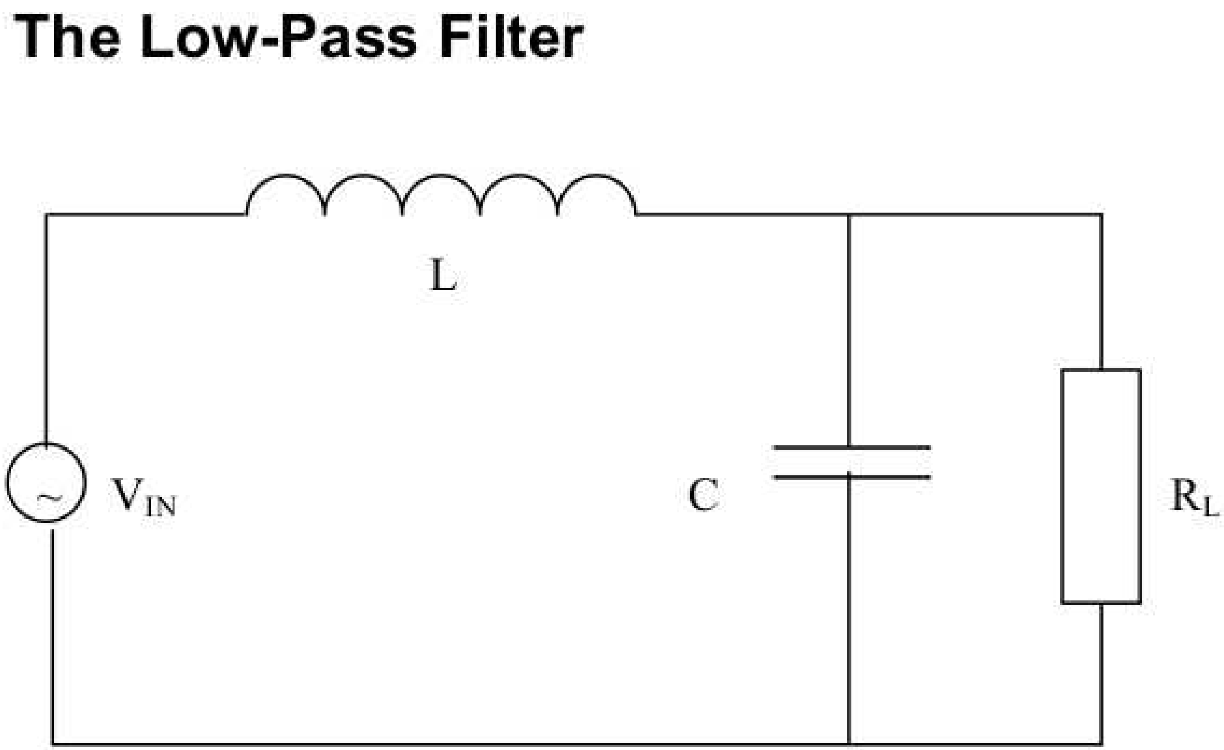
Television frequencies are VHF to UHF. That is **V**ery **H**igh **F**requencies or **U**ltra **H**igh **F**requencies. And your H.F. Transmitter is a maximum of 30 MHz. So a **“High Pass” Filter** will reduce the level of H.F. Signal to the television set.

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.



Chapter 12 – 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 VIN is applied across a voltage divider consisting of an inductor L and a capacitor C in parallel with a resistive load, RL.

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!]** at different frequencies.

[Chart the filter with a spreadsheet program]

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.

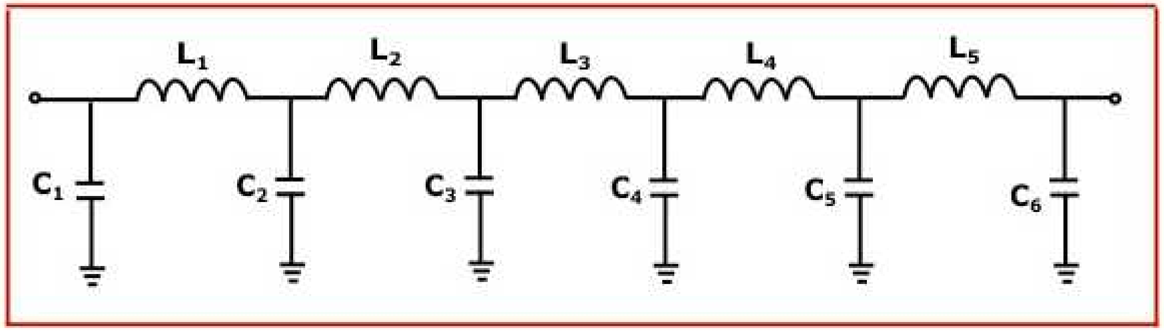
Signals with frequencies higher than FS are greatly attenuated – in this case by 60 dB or more. These signals are in the stop band of the filter. Signals with frequencies between FC and FS are somewhat attenuated. These frequencies are sometimes called the transition band of the filter since it is in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

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 Z_0: 50 ohm

Frequency Response Ripple: 1 db

Number of Components: 11 (1-11)

Calculate Clear

Results:

Inductance:

Unit : nH

L_1: 178.7111

L_2: 190.2988

L_3: 191.6390

L_4: 190.2988

Capacitance:

Unit : pF

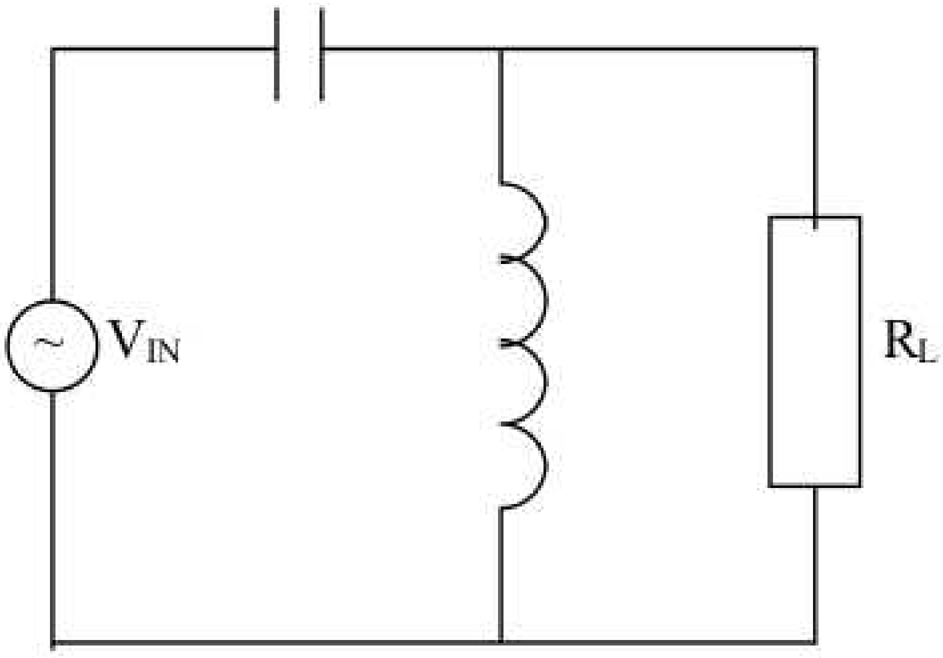
C_1: 139.1967

C_2: 199.5017

C_3: 203.5902

C_4: 203.5902

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.

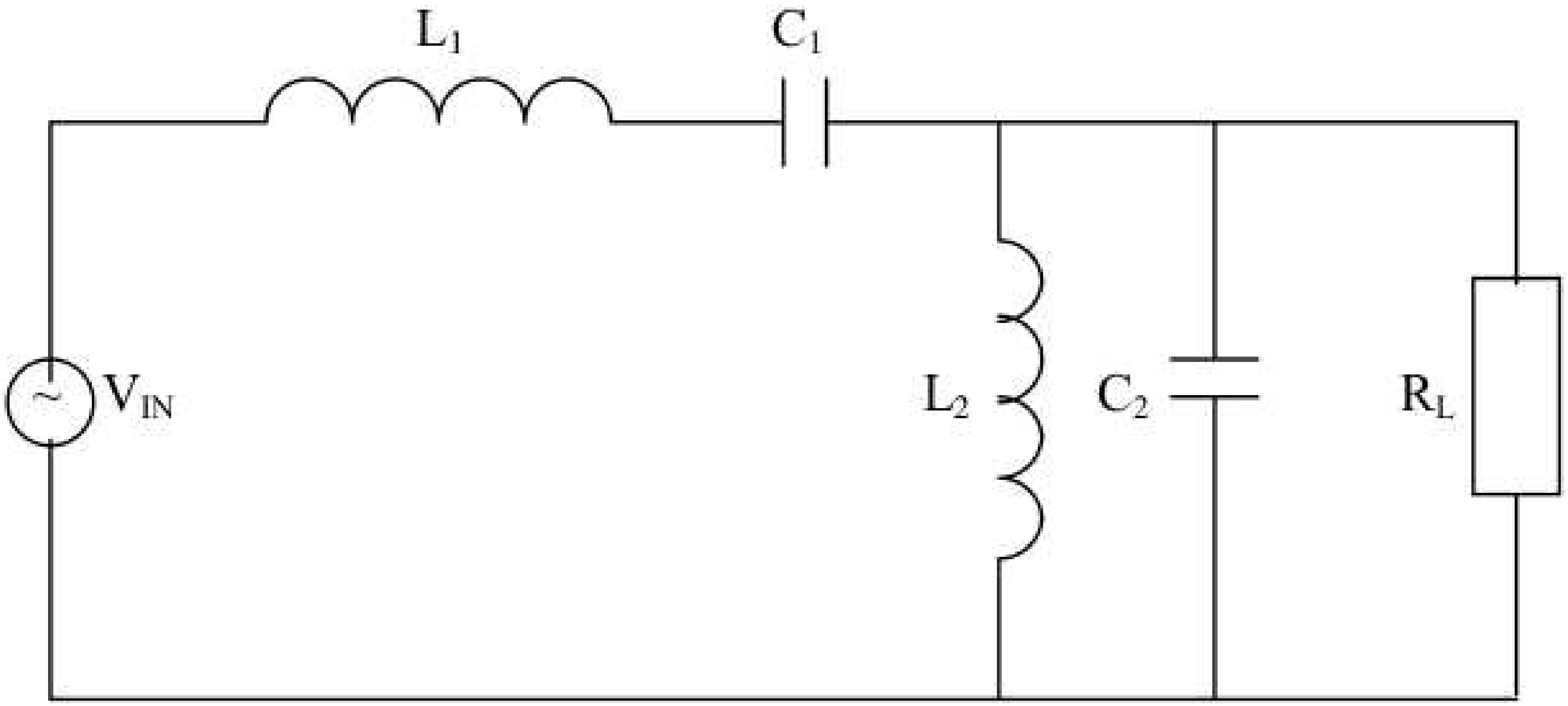
At high frequencies, the capacitor has low reactance, so does little to oppose the flow of current. The inductor has high reactance, so most of the current flows through the load resistor RL rather than through the inductor. This circuit is called a “high-pass” filter because it allows high frequency 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:

[ ]

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 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.

[ ]
The Frequency Response of a Band-Stop Filter

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.

50 The purpose of a high pass filter is to:

a) Attenuate all frequencies apart from a specific one.
b) Pass all frequencies apart from a specific one.
c) Pass all signals below a specified frequency but attenuate frequencies above it.
d) Attenuate all signals below a specified frequency but pass frequencies above it.

50 Die doel van 'n hoë deurlaatfilter is om:

a) Alle frekwensies behalwe 'n spesifieke een te verswak.
b) Alle frekwensies behalwe 'n spesifieke een deur te laat.
c) Alle seine onder 'n spesifieke frekwensie deur te laat maar frekwensies bo dit te verswak.
d) Om alle seine onder 'n spesifieke een ter verswak maar frekwensies bo dit deur te laat.

---

51 A band stop filter:

a) Allows all frequencies to pass.
b) Attenuates all frequencies.
c) Decreases bandwidth of a receiver.
d) Attenuates signals between two frequencies.

51 'n Bandstop filter:

a) Laat alle frekwensies deur.
b) Verswak alle frekwensies.
c) Verminder bandwydte van 'n ontvanger.
d) Verswak seine tussen twee frekwensies.

Revision Questions

1 A band pass filter:

A) Allows all frequencies to pass.
B) Attenuates all frequencies.
C) Allows signals between two frequencies to pass.
D) increases bandwidth of a receiver.

2 A band stop filter :

A) Allows all frequencies to pass.
B) Attenuates all frequencies.
C) decreases bandwidth of a receiver.
D) Attenuates signals between two frequencies.

3 A Low pass filter:

A) Attenuates all signals above a known cut-off frequency.
B) Introduces harmonics.
C) Removes RF signals from an input signal.
D) Requires the use of high gain amplifiers.

4 A high pass filter:

A) Introduces harmonics.
B) Removes RF signals from an input signal.
C) Requires the use of high gain amplifiers.
D) Attenuates all signals below a known cut-off frequency.

5 What is a circuit called which passes electrical energy above a certain frequency, but blocks electrical energy below that frequency?

A) An input filter.
B) A low-pass filter.
C) A high-pass filter.
D) A band-pass filter.

6 The purpose of a low pass filter is to:

A) attenuate all frequencies apart from a specific one.
B) pass all frequencies apart from a specific one.
C) pass all signals below a specified frequency but attenuate frequencies above it.
D) attenuate all signals below a specified frequency but pass frequencies above it.

7 The purpose of a high pass filter is to:

A) attenuate all frequencies apart from a specific one.
B) pass all frequencies apart from a specific one.
C) pass all signals below a specified frequency but attenuate frequencies above it.
D) attenuate all signals below a specified frequency but pass frequencies above it.