



Transmitters and Receivers

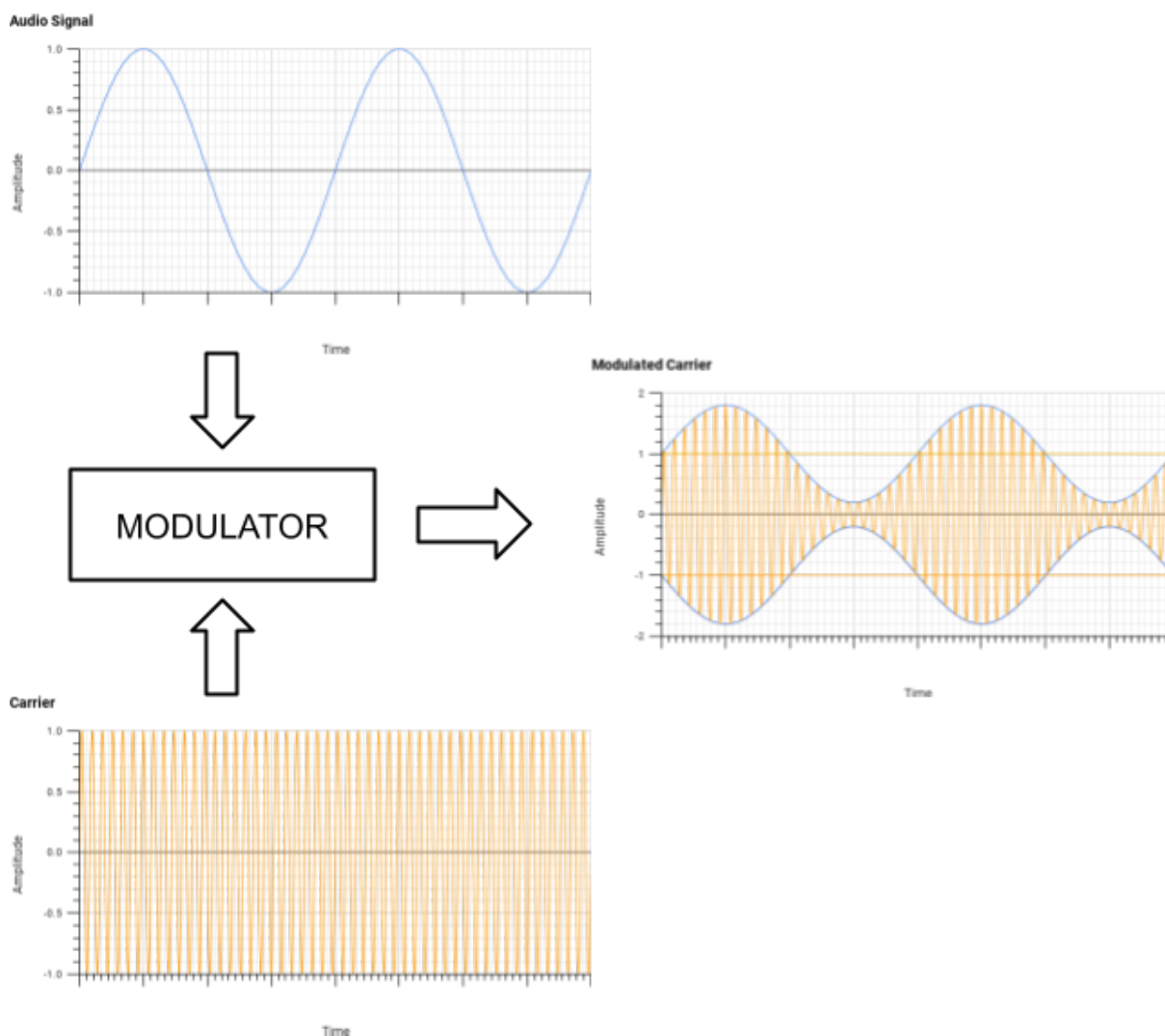
Transmitter Concepts

3A2 - Q21

Recall the meaning of depth of modulation for amplitude modulation.

Recall the meanings of wide band and narrow band frequency modulation.

Recall the meaning of the term Peak Deviation.

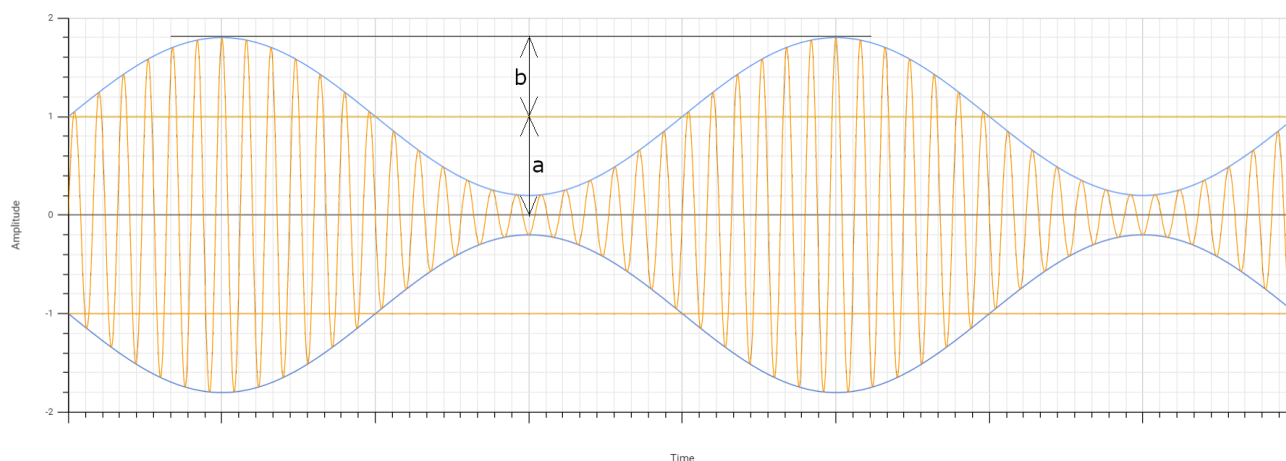




From the Foundation Course the concept of mixing an Audio Frequency (AF) signal with a Radio Frequency (RF) carrier is modulation. In the case of Amplitude Modulation (AM) the result is a wave at the frequency of the RF carrier but with its amplitude altered in response to the AF input.

Looking more closely at the diagram of the modulated waveform it can be seen that the imposition of the AF signal onto the RF carrier has caused the modulated wave to have peaks greater than the original carrier amplitude and troughs that are lower than the original carrier amplitude. In the diagram below the horizontal lines indicate the original amplitude of the RF carrier and the audio peaks and troughs are clearly visible.

Modulated Carrier



The **modulation depth** is the ratio of the height of the peak above the original amplitude to the amplitude of the original RF carrier, i.e.

$$m = \frac{b}{a}$$

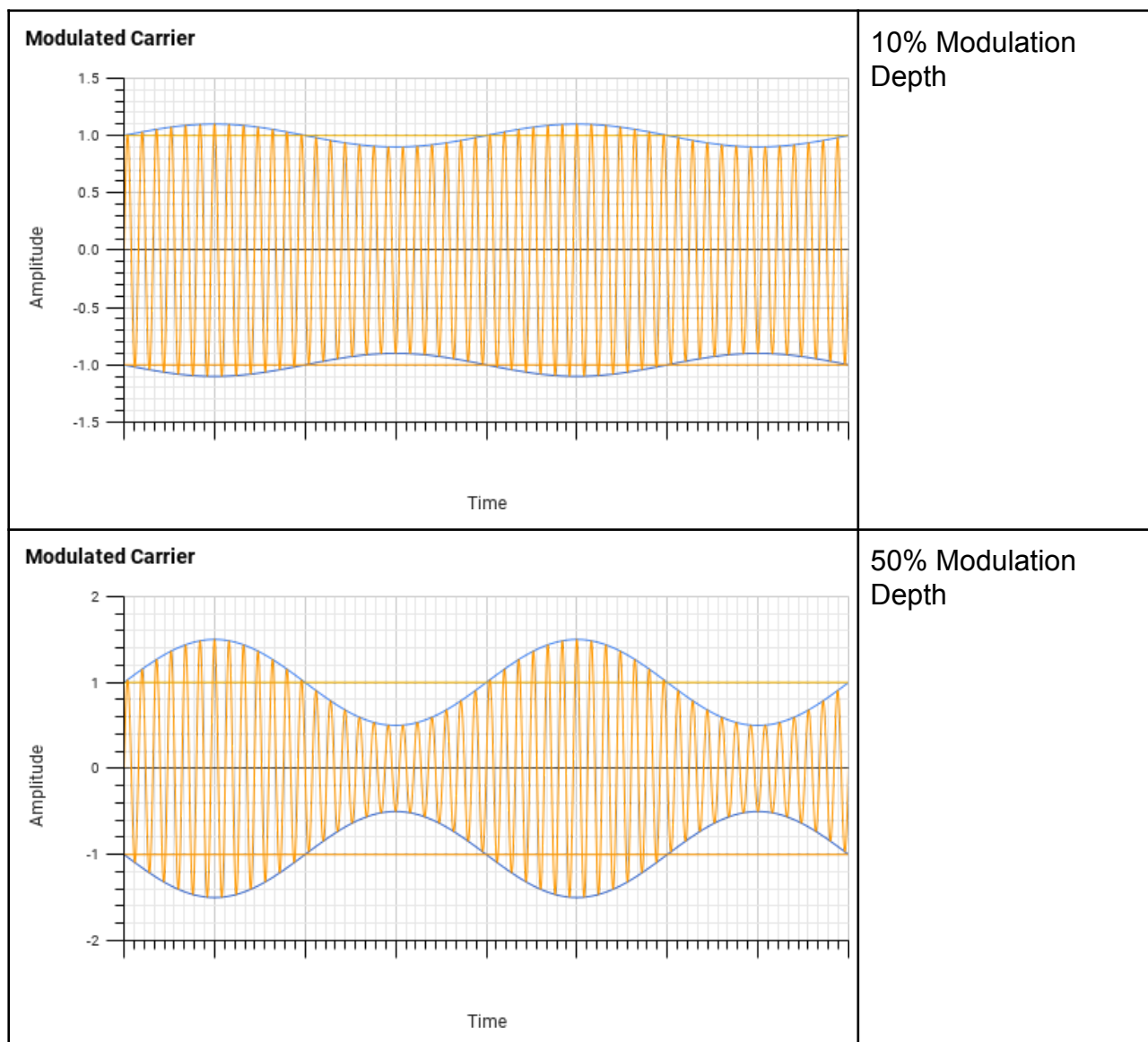
The example above is reasonably simple to work out because the amplitude of the original RF carrier "a" is 1.0 and the height of the modulated peak is 1.8 giving the height of the modulated peak above the original RF carrier "b" as 1.8 - 1.0 or 0.8. Hence the **modulation depth** is:

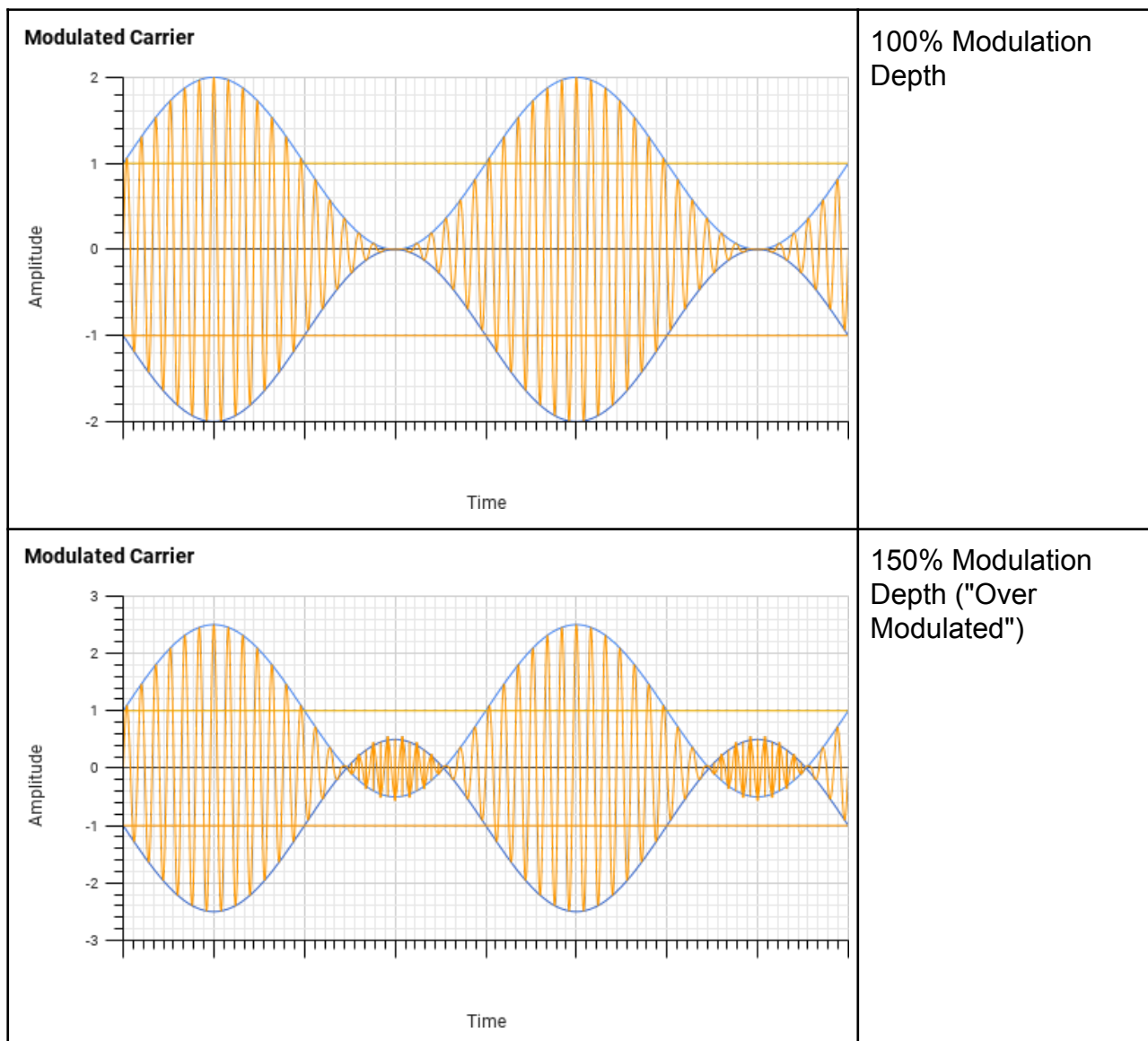
$$m = \frac{0.8}{1.0} = 0.8 \text{ or } 80\%$$

The greater the **modulation depth** the louder the signal however there are limits to this and if the **modulation depth** exceeds 1 or 100% then distortion of the transmitted signal and interference with adjacent signals will occur. Whilst 100% **modulation depth** gives



the loudest signal it is preferable to operate with about 80% - 90% **modulation depth** to leave a little headroom for excessive AF peaks.



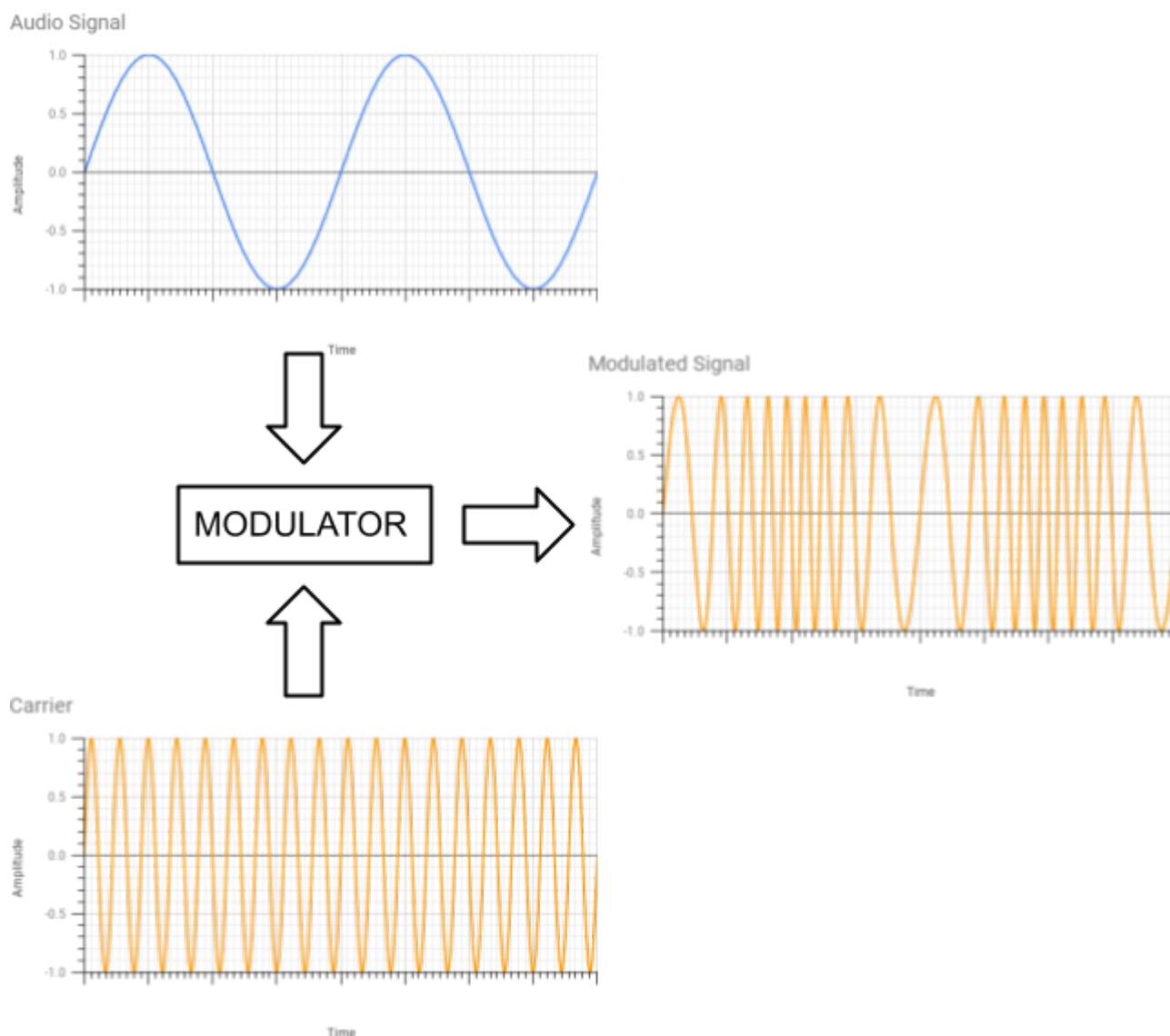


Frequency Modulation (FM) results in a modulated waveform where the frequency of the carrier is altered in response to the audio peaks and troughs but the amplitude remains constant. In the same way as AM has a **modulation depth** then FM has a **modulation index** based on the "*deviation*".

The **deviation** is proportional to the amplitude of the audio signal, that is how loudly you speak. Unlike AM there is no natural limit to the amount of deviation so limits are imposed based on the purpose of the transmission and the desired quality of the recovered audio.



FM broadcast radio uses a **peak deviation** of +/- 75kHz. This gives a good quality signal but requires a lot of bandwidth. For Amateur FM transmissions a **peak deviation** of +/- 2.5kHz (12.5kHz channel spacing) or +/- 5kHz (25kHz channel spacing).



The **modulation index** is defined as the ratio of the **peak deviation** to the **maximum audio frequency**, thus:

$$h = \frac{\text{Peak Deviation}}{\text{Maximum Audio Frequency}}$$

For Amateur transmissions the normal maximum audio frequency is 3kHz since Amateur transmissions usually only employ the 300Hz - 3000Hz (3kHz) AF range. The full range of audio hearing is typically regarded as being 20Hz - 15000Hz (15kHz) and for



transmissions that are true to the original audio (such as music) the maximum audio frequency is taken as 15kHz.

With this information it is possible to determine the **modulation index** of an FM broadcast station, an amateur station on 2m (12.5kHz channel spacing) and an amateur station on 70cm (25kHz channel spacing).

For a broadcast station:

$$h = \frac{\text{Peak Deviation}}{\text{Maximum Audio Frequency}} = \frac{75 \times 10^3}{15 \times 10^3} = 5$$

For an amateur station on 2m (12.5kHz channel spacing):

$$h = \frac{\text{Peak Deviation}}{\text{Maximum Audio Frequency}} = \frac{2.5 \times 10^3}{3 \times 10^3} = 0.833$$

For an amateur station on 70cm (25kHz channel spacing)

$$h = \frac{\text{Peak Deviation}}{\text{Maximum Audio Frequency}} = \frac{5 \times 10^3}{3 \times 10^3} = 1.667$$

If the **modulation index (h)** is greater than 1 then the transmission is considered to be **Wideband FM** whereas if the **modulation index (h)** is less than 1 the transmission is considered to be **Narrowband FM**.



3A3 - Q21

Understand that single sideband (SSB) is a form of amplitude modulation where one sideband and the carrier have been removed from the transmitted signal.

Understand that SSB is more efficient than AM or FM because power is not used to transmit the carrier and the other sideband.

Understand that a second advantage is that the transmitted signal takes up only half the bandwidth, e.g. 3kHz not 6kHz.

Recall that :

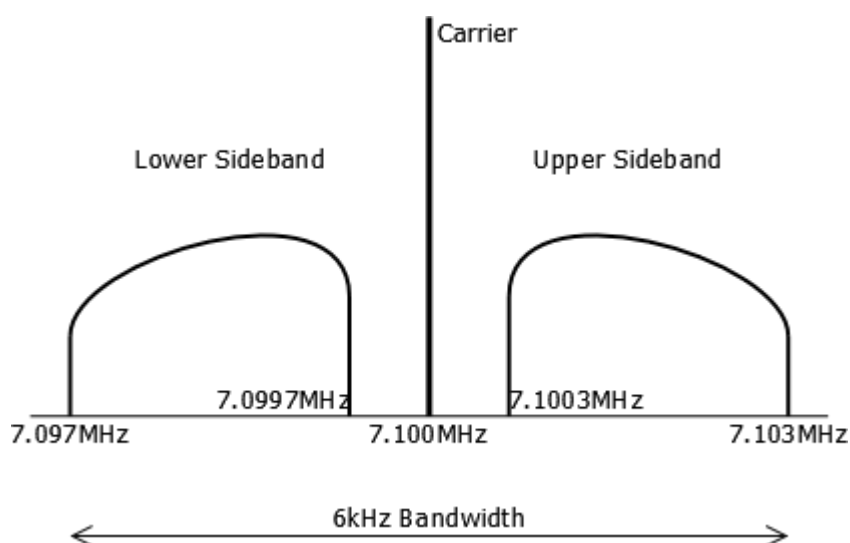
AM uses less bandwidth than FM

SSB uses less bandwidth than AM

CW uses less bandwidth than SSB.

Digital modes may use less bandwidth than any of the above.

A transmitted AM signal has 3 parts: An upper sideband, a lower sideband and the carrier. Each of these components uses some of the transmitted power, specifically each sideband takes 25% of the transmitted power and the carrier takes 50%.

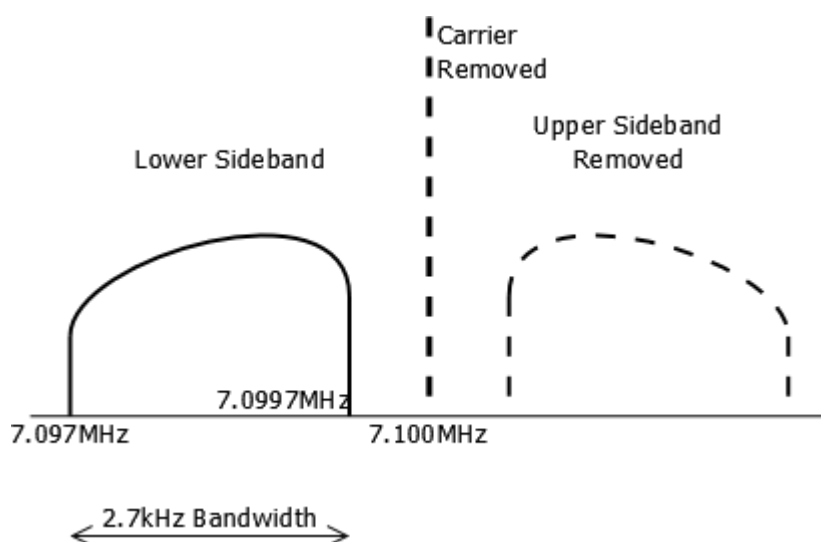




Each side band starts above or below the carrier by the lowest audio frequency (300Hz) and continues to the highest audio frequency (3000Hz) resulting in a total transmitted bandwidth of 6kHz.

By using Single Sideband (SSB) in place of AM (or indeed FM) two advantages are gained:

1. All of the transmitted power is used to send the one remaining sideband
2. The total bandwidth is reduced from 6kHz (for AM) to around 2700Hz (2.7kHz) making more efficient use of the available frequency allocation



In terms of signal bandwidth the modulation scheme with the largest bandwidth is FM and the modulation scheme with the smallest bandwidth is CW/Digital modes, the overall list from largest bandwidth to smallest bandwidth is:

1. FM has the largest bandwidth
2. AM occupies around 6kHz
3. SSB occupies around 2.7kHz
4. CW occupies 300Hz - 500Hz
5. Digi-modes have the smallest bandwidth in general

Note that whilst some digital modes such as PSK31, PSK63 and FT8 are deliberately designed to be narrow band there are modes that have been designed to increase the speed of transfer such as Olivia (anything from 125Hz - 2000Hz depending on which standard is employed) which can occupy greater bandwidths than CW.

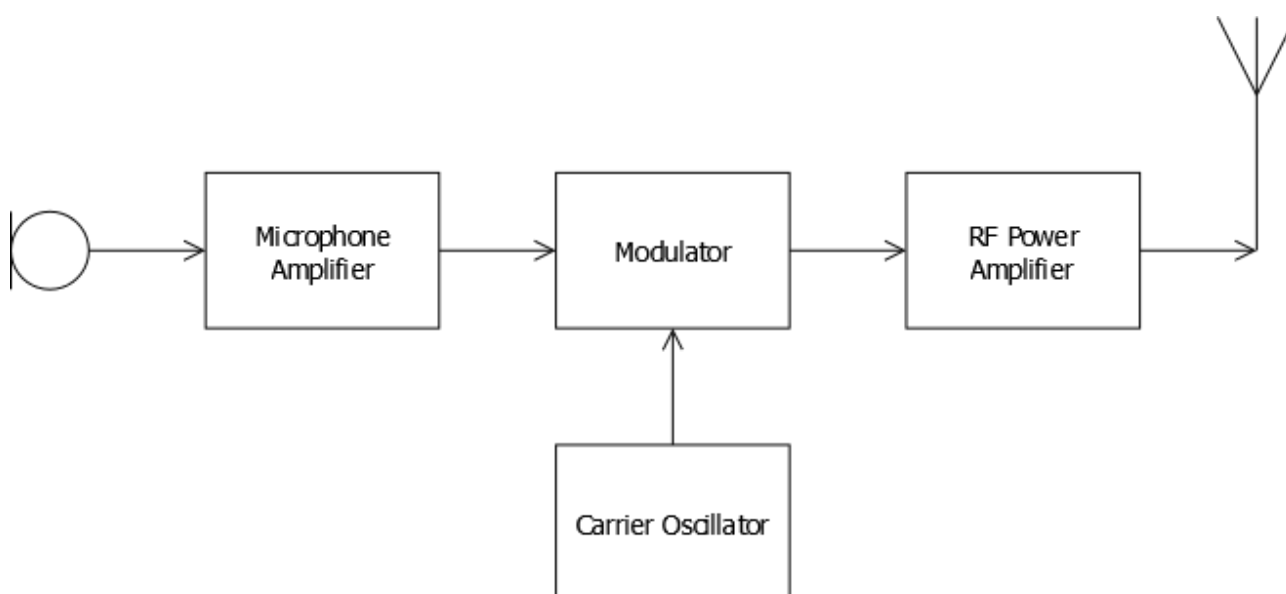


Transmitter Architecture

3B1 - Q21

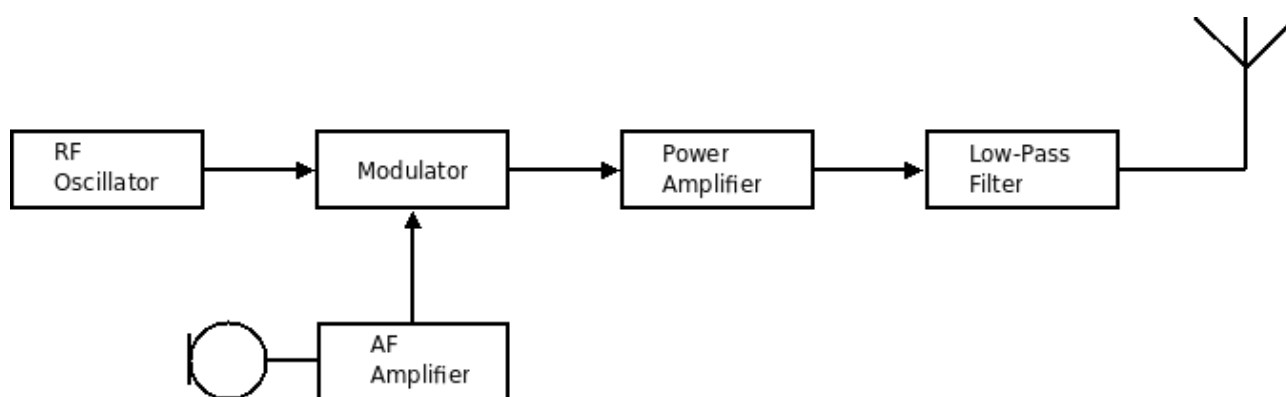
Understand the block diagrams of CW, AM, SSB and FM transmitters.

At the foundation level all transmitters were shown with the same architecture, which is repeated below as a recap.



Each of the different modes does in fact have a slightly different architecture and it is important to be able to differentiate between and identify the specific types.

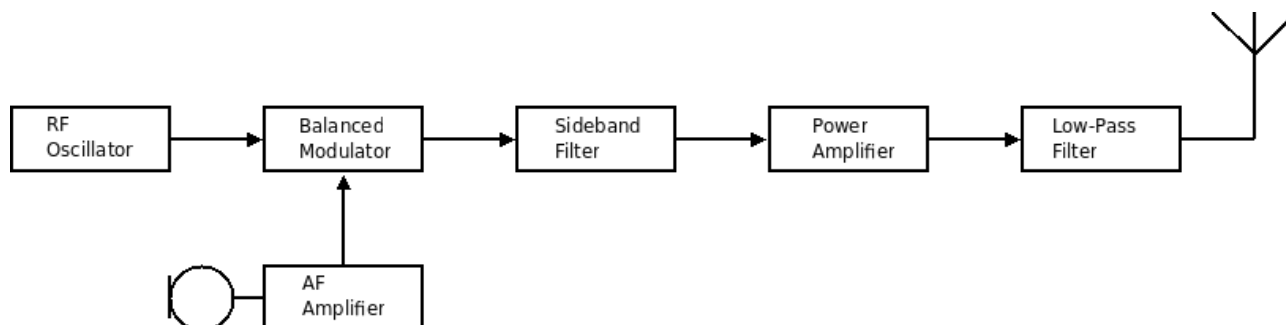
The AM transmitter most closely resembles the simple generic transmitter from the Foundation Syllabus.





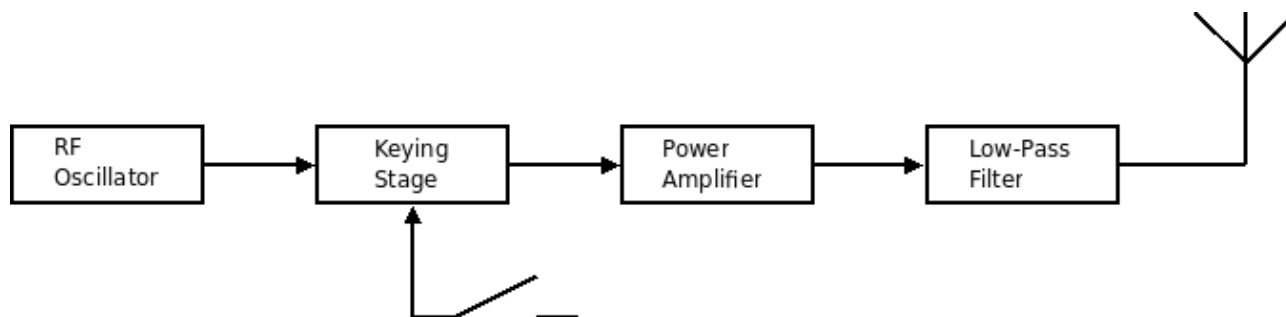
The low pass filter at the end of the transmitter chain is there to suppress harmonics of the fundamental frequency and prevent interference.

The SSB transmitter is an extension of the AM transmitter.



The Balanced Modulator is different to the Modulator in the AM transmitter, it performs a similar function but also suppresses the carrier passing only the 2 sidebands to the next stage. The next stage is a Sideband Filter which is a sharp bandpass filter designed to attenuate 1 of the sidebands and only pass the wanted upper or lower sideband to the power amplifier.

The CW transmitter is straightforward to identify, as it depicts a CW key in place of a microphone.

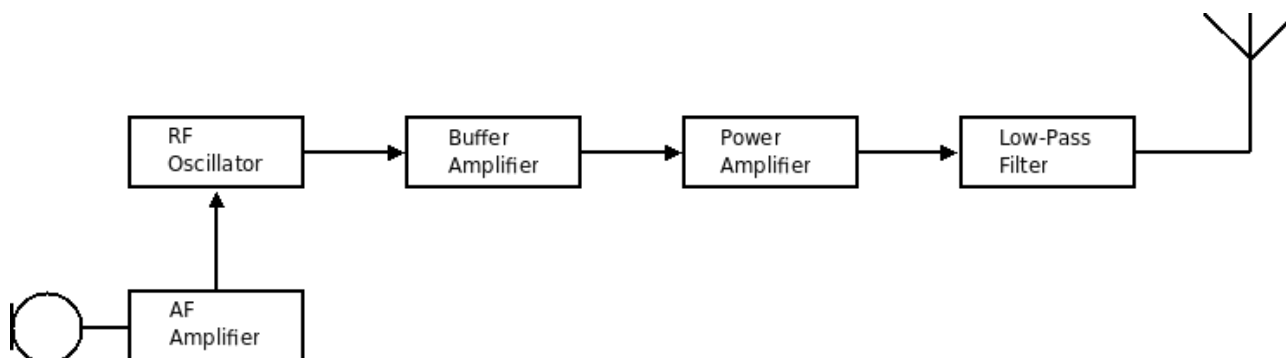


A simple CW transmitter can be constructed by simply switching the oscillator on and off but this is not good practice as it fails to provide any buffering between the oscillator and the antenna. The result of this is that the transmitted sound can be "*chripy*" as the oscillator is continually switched on and off and may result in oscillator drift.

A better approach is to allow the oscillator to run continuously and have a separate keying stage that simply interrupts the path from the oscillator to the power amplifier which prevents both "*chirp*" from switching the oscillator on and off and "*key-clicks*" from switching the power amplifier on and off.



The FM transmitter is unique in having the microphone amplifier stage directed to the RF Oscillator.



This transmitter requires that the frequency of the RF Oscillator is adjusted in response to the audio, hence why the AF Amplifier is fed into the RF Oscillator. A component, covered in the 'Technical Aspects' section, known as a "*Varicap Diode*" provides this.

A varicap diode is like an ordinary diode with a depletion layer which can act like the plates on a capacitor. The varicap diode is operated in reverse bias and as the reverse bias is increased the depletion layer gets larger and the capacitance decreases so by varying the amount of reverse bias the capacitance of the varicap diode is controlled.

The capacitance presented by the varicap diode interacts with the capacitance in the tuned circuit at the core of the RF Oscillator altering the resonant frequency of the tuned circuit and hence altering the frequency generated by the RF Oscillator.

Oscillators

3C1 - Q21

Recall and understand the relative advantages and disadvantages of a crystal oscillator and a VFO.

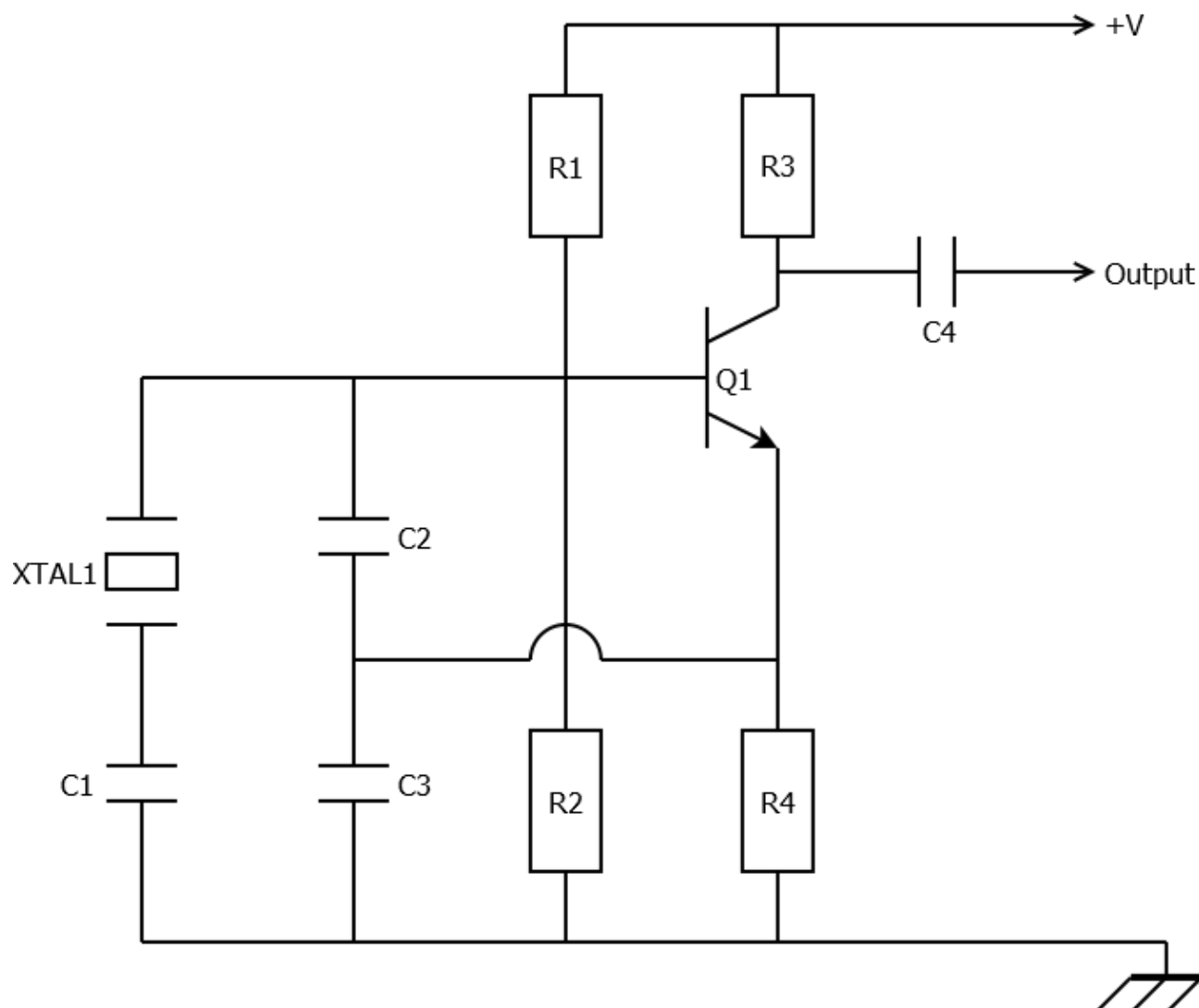
Recall that the resonant frequency of the tuned circuit in a VFO determines the frequency of oscillation.

RF Oscillators are the basis for all transmitters and many receivers, although there are a variety of designs they all need to produce clean RF energy on a fixed frequency without drift or interruption.



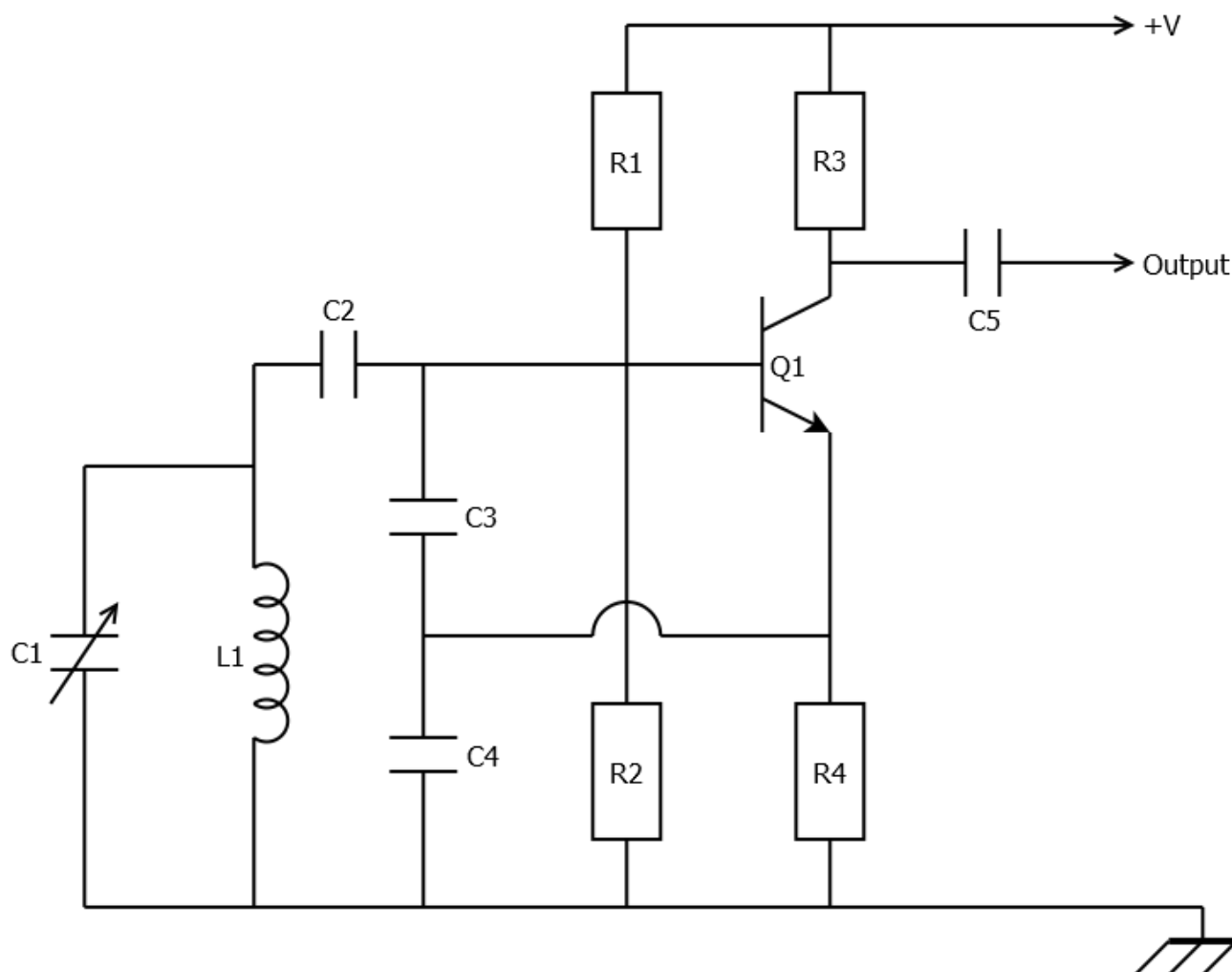
Although there are different types the most common are: crystal oscillators, LC oscillators, DDS and phase locked loops.

Crystal oscillators and LC Oscillators share many circuit characteristics, their principal difference being the method of producing the source signal.



A crystal oscillator relies on a piece of piezo-electric crystal, typically quartz, which oscillates at a frequency determined by the size and shape of the crystal when a potential difference is applied to it.

As the frequency is fixed by the physical properties of the crystal, changing the frequency is not possible, a crystal oscillator operates on a single frequency. Channelised crystal controlled transceivers would be fitted with crystals for each channel.



The Variable Frequency Oscillator essentially replaces the piezo-electric crystal with a tuned circuit where the range of frequencies available depends on the range of the variable capacitor. In a VFO the resonant frequency of the tuned circuit defines the carrier frequency.

Both types of oscillator have their own advantages and disadvantages.

For the crystal oscillator, the advantages are that it is: stable and free from drift, unaffected by changes in temperature, humidity etc and resilient to physical shock. On the negative side is the limited frequency range that it offers requiring multiple crystals for each spot frequency.

On the other hand the Variable Frequency Oscillator has the big advantage of being able to be set for any frequency in its coverage range, however this leads to potential problems if the VFO isn't calibrated and covers frequencies outside of the Amateur bands then it is



possible to unwittingly transmit out of band. VFOs also have to be well built to avoid drift resulting from changes in temperature, humidity and to remain unaffected by physical shock. The construction of a VFO is usually more demanding than the construction of a crystal controlled oscillator.

3G3 - Q23

Recall that the frequency stability of an oscillator can be improved by rigid mechanical construction, screening the oscillator enclosure, a regulated DC supply and a buffer amplifier immediately after the oscillator circuit.

Understand that a lack of stability (drift) may result in operation outside the amateur bands.

Recall that most modern oscillators are digital synthesisers, which are very stable and are based on a crystal reference.

As mentioned above VFOs in particular can suffer from drift and a risk that the oscillator will move the carrier out of the amateur bands.

To avoid some of the stability issues a VFO should be:

- well constructed,
- placed within its own screened enclosure,
- supplied with a well regulated DC supply
- fed into a Buffer Amplifier immediately after the Oscillator circuit

"Drift" is a phenomena more associated with older equipment, it describes the tendency for a VFO to change its frequency without any operator intervention as the equipment comes to a stable operating temperature. Usually once the equipment has "warmed-up" the frequency of the VFO stabilises.

Note that if the "drift" is excessive then it is possible for the VFO to adjust itself to a frequency outside the amateur bands.

If a VFO is constructed then it is important to calibrate the VFO so that as a minimum the Amateur Band edges that are covered are marked as it is possible for a VFO to cover a wider frequency range than Amateurs are permitted to operate on.

Modern equipment relies on digital synthesisers with crystal references which are very stable and do not suffer from "drift".



3C3 - Q21

Recall that digital signals can be used to generate audio and RF signals by Direct Digital Synthesis (DDS).

Recall the meaning of DDS.

Recall that a Direct Digital Synthesiser generates audio and RF signals from pre-set digital values held in a memory, or Lookup Table.

Direct Digital Synthesis is a combination of digital and analogue methods that provides a reliable and stable signal at any frequency from audio to radio frequencies.

It consists of a "look-up table" of preset values. The output of this "lookup-table" is connected to a Digital to Analogue Converter (DAC) which creates an analogue voltage based on the digital value.

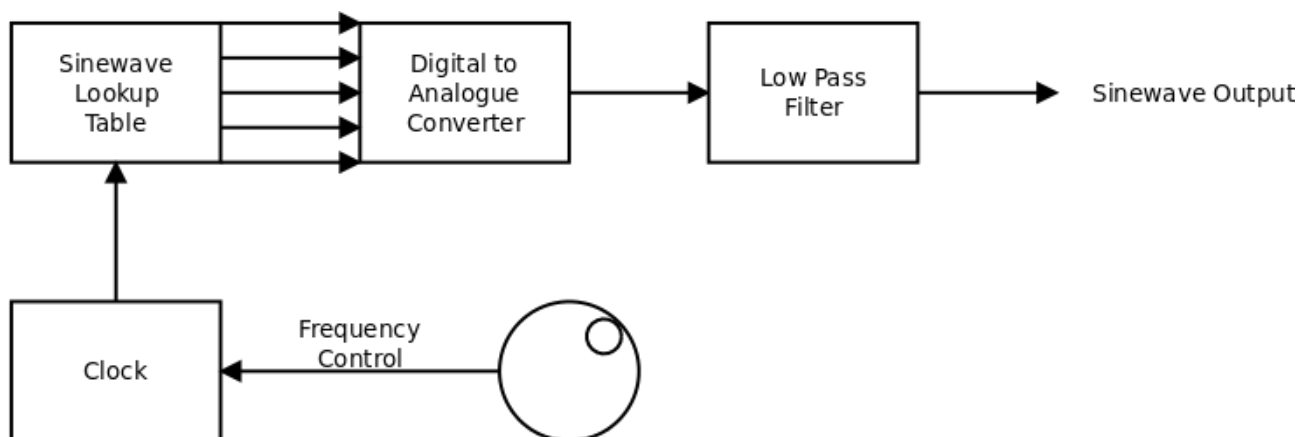
The speed that the data is fed to the DAC is controlled by a digital counter which effectively sets the frequency of the RF carrier.

In order to ensure stability the output is compared to a fixed frequency crystal controlled oscillator, or in some cases a phase locked loop.

A DDS is often combined onto a single "chip" or integrated circuit.

The accuracy of the DDS signal is dependent on the number of points and the number of bits or levels that is represented.

Usually a DDS will be followed by some filtering to remove any spurious signals contained within the DDS process.

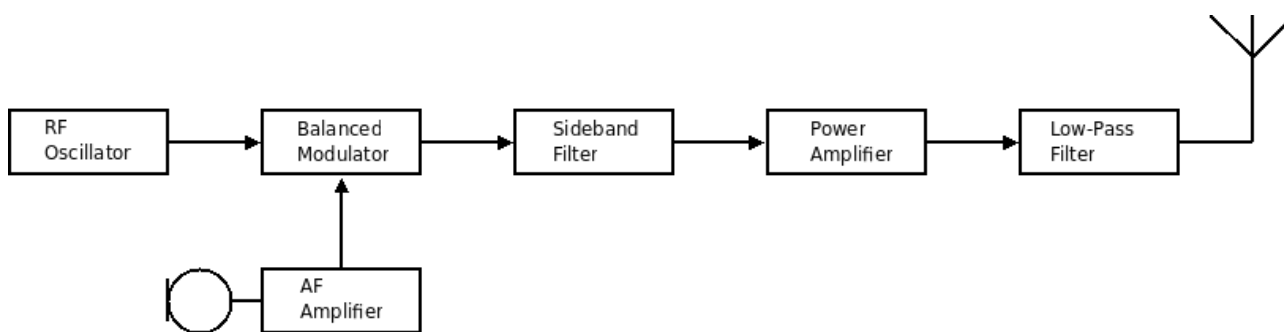


Microphone Amplifiers and Modulators

3E1 - Q22

Recall that a **Balanced Modulator** is used to produce two sidebands whilst suppressing the carrier.

A Balanced Modulator is part of the SSB transmission train and mixes the AF signal and the RF signal in the same way as the modulator in an AM transmitter does, but also suppresses the carrier passing only the 2 sidebands to the next stage.



**3E2 - Q22**

Understand that an SSB filter is a Band Pass Filter that will only allow one sideband to pass to the Power Amplifier.

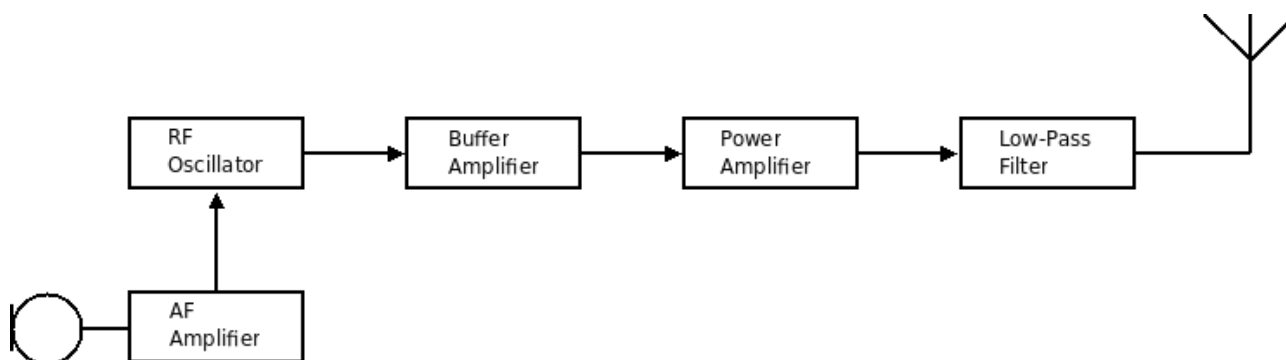
Recall that in an analogue transmitter, SSB filters are normally constructed from a number of quartz crystals or other resonators.

A Sideband Filter is a sharp or narrow bandpass filter designed to attenuate 1 of the sidebands and only pass the wanted upper or lower sideband to the power amplifier.

In order to achieve the necessary sharpness, bear in mind that the other sideband is only 600Hz away from the wanted sideband, the sideband filter is usually made from several quartz crystals or other resonators.

3E3 - Q22

Recall that a variable capacitance diode can be used in an oscillator to produce frequency modulation (FM).



A varicap diode is like an ordinary diode with a depletion layer which can act like the plates on a capacitor. The varicap diode is operated in reverse bias and as the reverse bias is increased the depletion layer gets larger and the capacitance decreases so by varying the amount of reverse bias the capacitance of the varicap diode is controlled.

The capacitance presented by the varicap diode interacts with the capacitance in the tuned circuit at the core of the RF Oscillator altering the resonant frequency of the tuned circuit and hence altering the frequency generated by the RF Oscillator.



RF Power Amplifiers

3F1 - Q22

Understand the concept of the efficiency of an amplifier stage and estimate expected RF output power for a given DC input power, given the stage's efficiency.

There are different types of amplifier depending on which modes are being transmitted. So called "linear amplifiers" are required where the amplitude of the signal is important, so AM and SSB require linear amplification which preserves the ratio between the peaks and troughs of the signal. FM on the other hand can have the amplitude of the signal corrupted by the amplification process and thus non-linear amplification is a viable method.

Linear amplifiers are less efficient than non-linear amplifiers and therefore need a greater input power to produce the same output power as a non-linear amplifier.

Amplifiers are grouped into classes, the rationale for this is beyond the current syllabus but Classes A and B are linear amplifiers and Class C is a non-linear amplifier.

Although not important for the Intermediate examination, a class A amplifier is biased so that it conducts over the whole of the cycle of the waveform. It conducts all of the time, even for very small signals, or when no signal is present. A class B amplifier is biased so that it conducts over half the waveform. By using two amplifiers, each conducting our half the waveform, the complete signal can be covered. A Class C amplifier is biased so that it conducts over much less than half a cycle. This gives rise to very high levels of distortion, but also it enables very high efficiency levels to be achieved. This type of amplifier can be used for RF amplifiers that carry a signal with no amplitude modulation - it can be used for frequency modulation with no issues. The harmonics created by the amplifier effectively running in saturation can be removed by filters on the output. These amplifiers are not used for audio applications in view of the level of distortion.



Table of Amplifier Classes and Typical Efficiencies

Amplifier Class	Efficiency (%)
Class A	35%
Class B	50%
Class C	67%

Consider an amplifier powered by a bench power supply at 13.8V delivering 20A.

The power supplied to the amplifier from the power supply would be:

$$P = V \times I = 13.8 \times 20 = 276W$$

The power leaving the amplifier will be the input power multiplied by the efficiency which is dependant on the amplifier class as listed above, so:

$$\text{Class A Amplifier Output: } 35\% \times 276 = 0.35 \times 276 = 96.6W$$

$$\text{Class B Amplifier Output: } 50\% \times 276 = 0.5 \times 276 = 138W$$

$$\text{Class C Amplifier Output: } 67\% \times 276 = 0.67 \times 276 = 184.92W$$

Clearly the more efficient Class C Amplifier has the highest output of the group. An alternative way to look at this is to consider the input current required to produce 100W from each of the amplifier classes when supplied with 13.8V.

Class A Amplifier producing 100W output:

$$\text{Input power} = 100 \div 35\% = 100 \div 0.35 = 285.7W$$

$$\text{Input Current} = P \div V = 285.7 \div 13.8 = 20.7A$$

Class B Amplifier producing 100W output:

$$\text{Input power} = 100 \div 50\% = 100 \div 0.50 = 200.0W$$

$$\text{Input Current} = P \div V = 200.0 \div 13.8 = 14.5A$$

Class C Amplifier producing 100W output:

$$\text{Input power} = 100 \div 67\% = 100 \div 0.67 = 149.3W$$



$$\text{Input Current} = P \div V = 149.3 \div 13.8 = 10.8A$$

This should demonstrate why 100W multi-mode HF transceivers require such high currents since these rigs have to use linear amplifiers to accommodate AM and SSB.

Transmitter Interference

3G2 - Q23

Recall that oscillators, mixers and amplifiers can produce harmonics which are multiples of the fundamental frequency.

Recall that harmonics can cause interference to other amateur bands and other radio users.

Harmonics are multiples of the carrier or **fundamental** frequency, there is no first harmonic and the term sub-harmonic is, at best, misleading.

Harmonics occur because of the mixing and amplification processes that take place within a transceiver and if not adequately filtered out can be the cause of interference to other radio hams or other radio users.

Consider a signal on 3.55MHz, this will give rise to the potential for harmonics on even and odd multiples of the fundamental or carrier frequency as shown below.

Table of Harmonics for 3.55MHz Fundamental Frequency

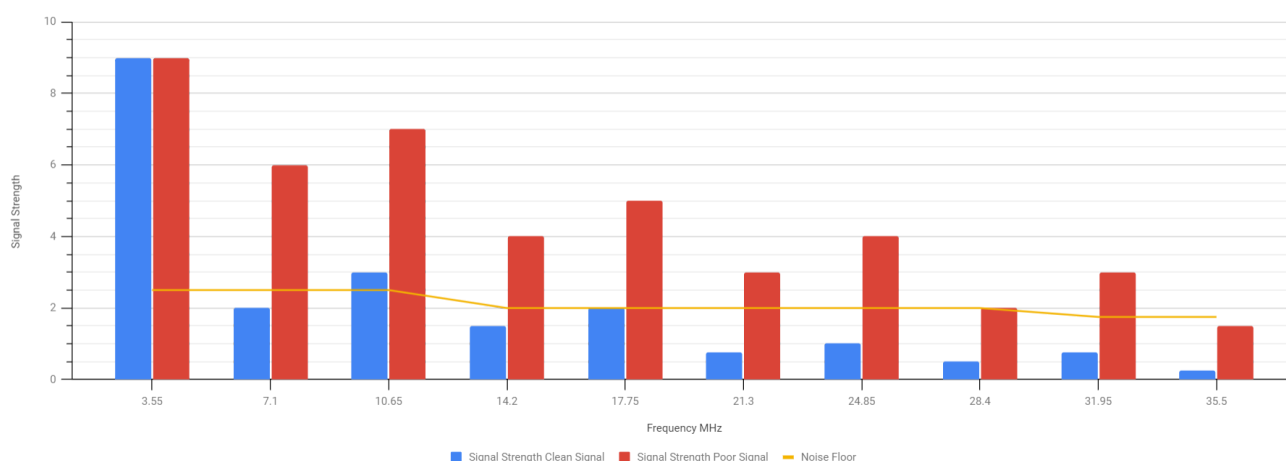
Name	Frequency (MHz)	Comments
Fundamental Frequency	3.55	80m Amateur Band (3.5MHz - 3.8MHz)
Second Harmonic	7.10	40m Amateur Band (7.0MHz - 7.1MHz)
Third Harmonic	10.65	Non Amateur Allocation
Fourth Harmonic	14.20	20m Amateur Band (14.00MHz - 14.35MHz)
Fifth Harmonic	17.75	Non Amateur Allocation
Sixth Harmonic	21.30	15m Amateur Band (21.00MHz - 21.45MHz)
Seventh Harmonic	24.85	Non Amateur Allocation



Name	Frequency (MHz)	Comments
Eighth Harmonic	28.40	10m Amateur Band (28.0Mhz - 29.7Mhz)
Ninth Harmonic	31.95	Non Amateur Allocation
Tenth Harmonic	35.50	Non Amateur Allocation

As the harmonics increase in frequency so the strength decreases so the ninth harmonic will be weaker than the third harmonic. Generally the odd harmonics tend to be stronger than the preceding even harmonics so it is a safe assumption that if the third harmonic is suitably suppressed then all of the harmonics will be suitably suppressed.

Harmonics





3G3 - Q23

Recall that a filter is a device that blocks some frequencies and passes others.

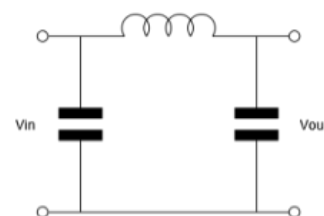
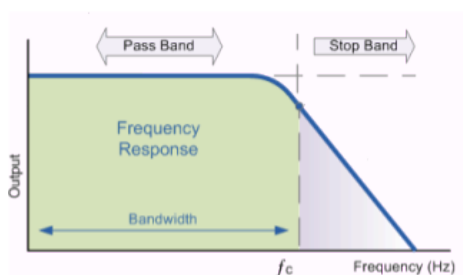
Understand the effects of low-pass, bandpass and high-pass filters.

Understand that a low-pass filter, a band-pass filter and a band stop (notch) filter can minimise the radiation of harmonics.

Recall that RF power amplifiers can produce harmonics of the wanted signals and that suitable filtering is required to avoid harmonic radiation.

The circuitry and theory of filters is covered in Section 2 'Technical Aspects' but the application of these filters is worth reconsidering here in the context of suppression of harmonic signals.

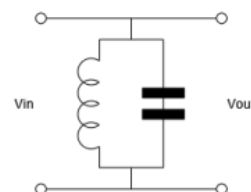
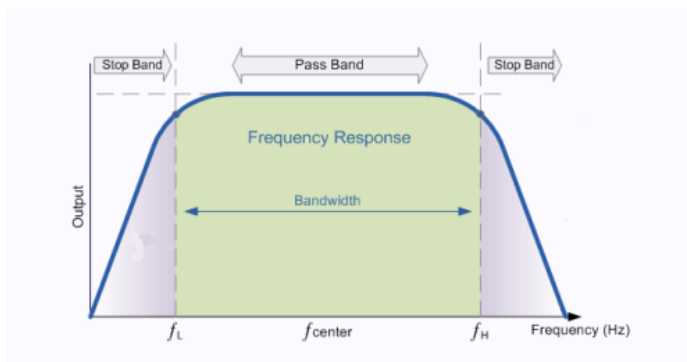
Low Pass Filter



Designed to pass all frequencies below a certain point and attenuate anything above the **cut-off frequency**. Typical uses would be in the feeder from a HF transceiver to prevent higher frequency harmonics from being included in the transmission.

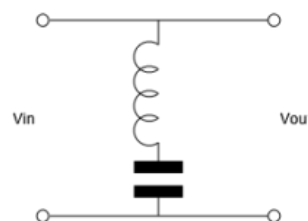
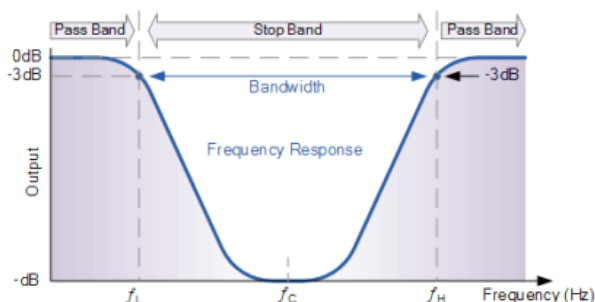


Band Pass Filter



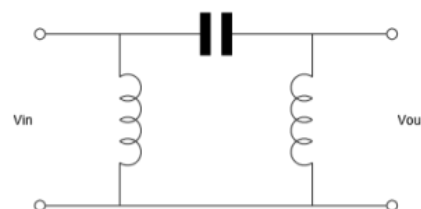
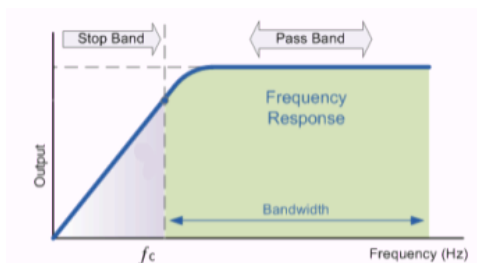
A filter designed to pass frequencies between two cut-off points.

Band Stop (Notch) Filter



A filter designed to stop signals with frequencies between two cut-off points but allow anything below the lower cut-off or above the upper cut-off to pass.

High Pass Filter



Designed to pass all frequencies above a certain point and attenuate anything below the **cut-off frequency**. Typical uses would be in the download to a TV or other high frequency receiver to prevent lower frequency signals from being passed to the receiver.



As previously described the principal sources of harmonics in a transceiver come from the mixing/modulation processes and the RF Amplifier, particularly if the latter item is not operating in its linear range.

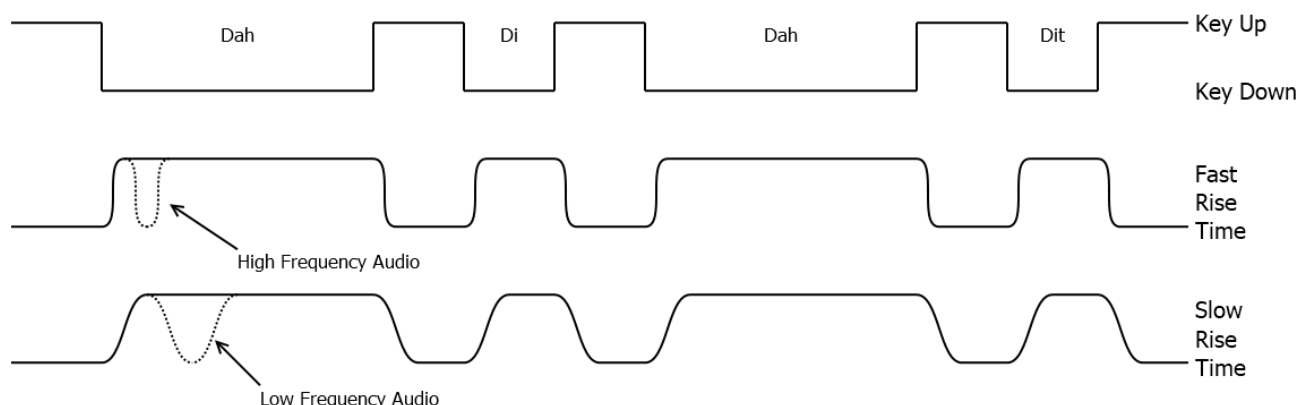
3G4 - Q23

Understand that too fast a rise and fall time of the transmitted RF envelope of a CW transmitter may cause excessive bandwidth (key clicks) and that this can be minimised by suitable filters in the keying stage. Recognise a diagrammatic representation of rise and fall time.

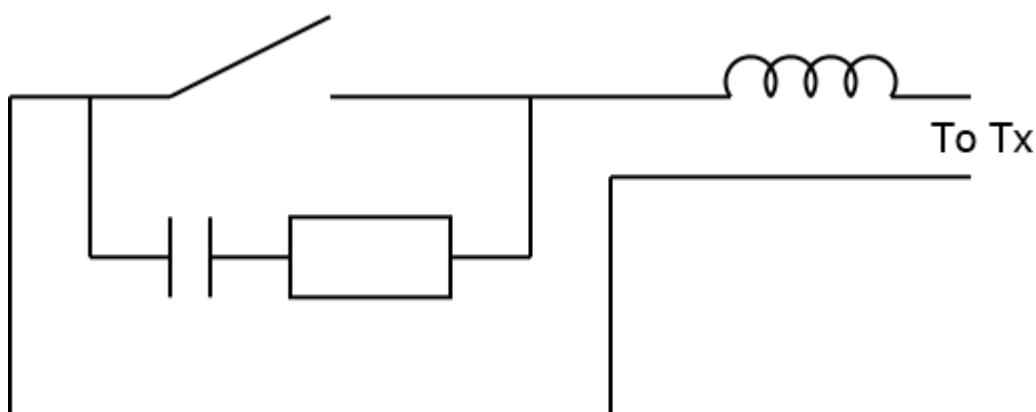
In the same way as AM transmissions can suffer if the modulation depth is too high CW transmission have their own foibles that have to be avoided.

One of these is the phenomenon of "*key clicks*". Key clicks are not related to the sending speed, they can occur just as readily with slow CW as with high speed transmissions.

Key clicks are related to the rise (and decay) time. Too fast a rise (or decay) time effectively creates high frequency audio at the point of key down or key release which generates large sidebands which are heard as clicking in time with the CW on many frequencies right across the band.



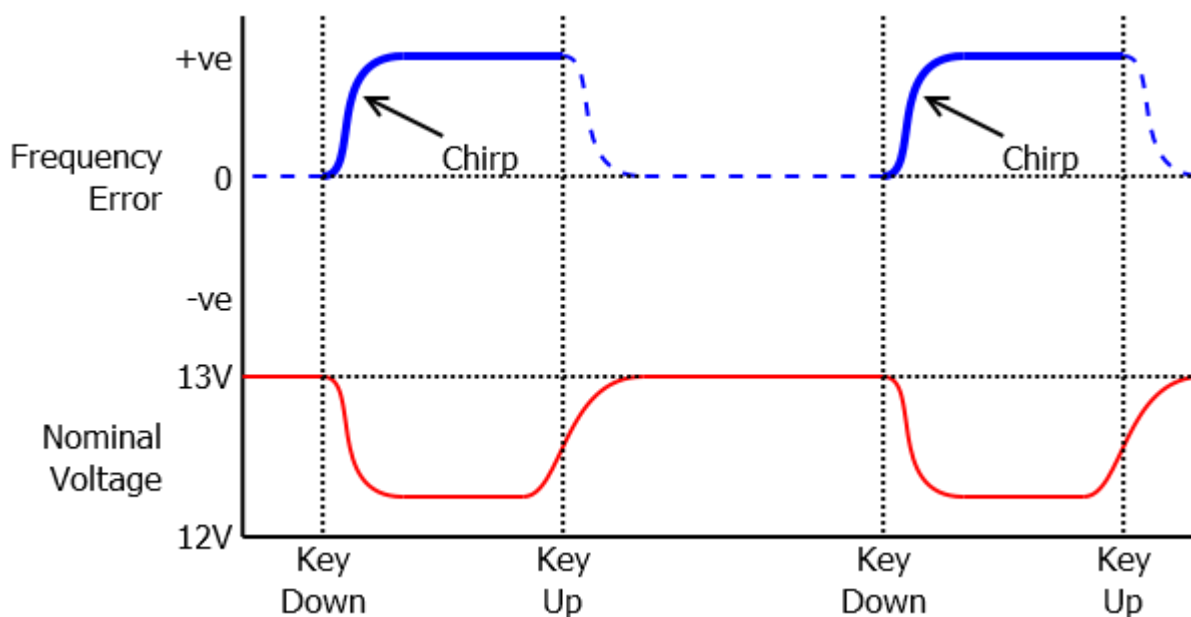
Key clicks can be avoided by the introduction of a simple key-click filter.



3G5 - Q23

Recall the cause and effect of 'chirp' and identify suitable remedies.

Chirp is caused when the action of key down causes the voltage to drop momentarily which results in a frequency drift in the RF Oscillator.



To address this ensure that any transmitter has a well regulated power supply with headroom in its current supply so that a constant voltage can be maintained.



Receiver Concepts

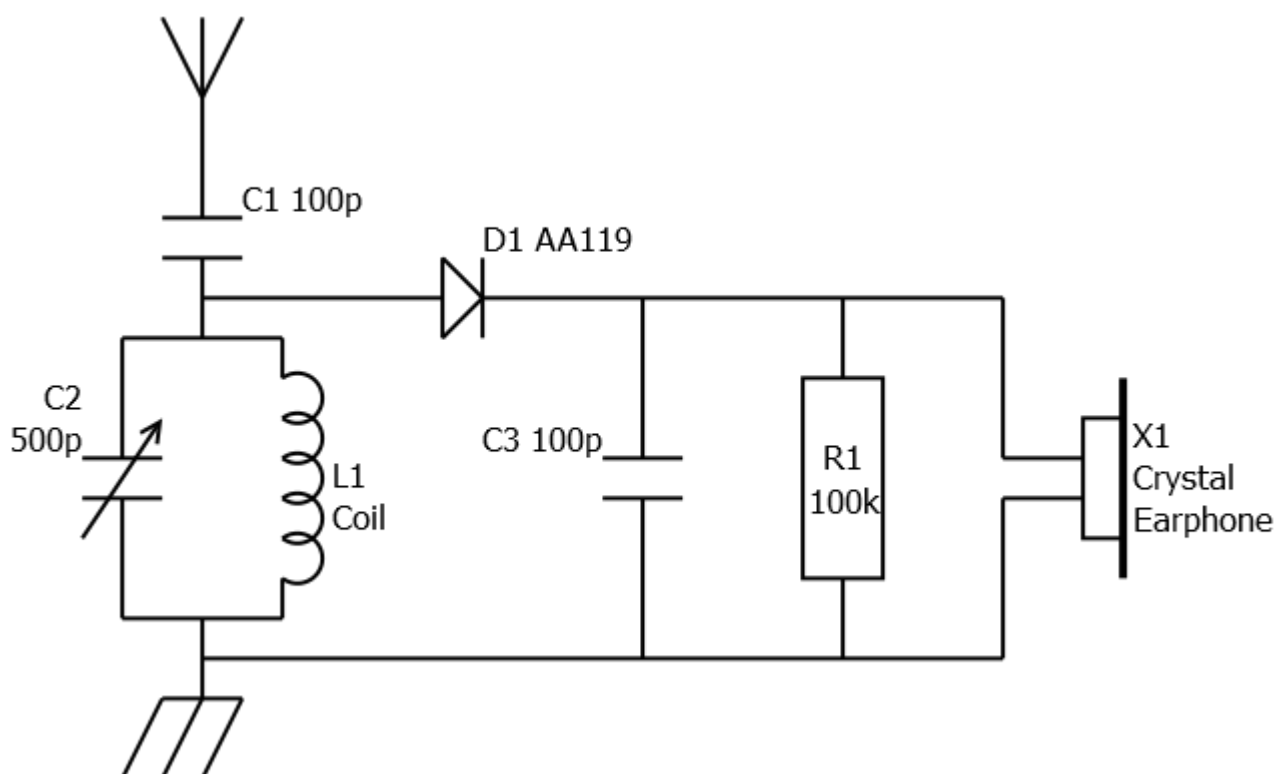
3H2 - Q24

Understand the block diagrams of the crystal diode receiver, and direct conversion receiver.

Understand the functions of the RF amplifier, demodulator (detector), and audio amplifier as used in an analogue receiver.

Crystal Set

Probably the simplest form of receiver is the crystal set or "diode receiver" which essentially comprises a tuned circuit and a germanium diode to act as a demodulator. Having no audio amplification the crystal set doesn't require power but does lack both selectivity and sensitivity.



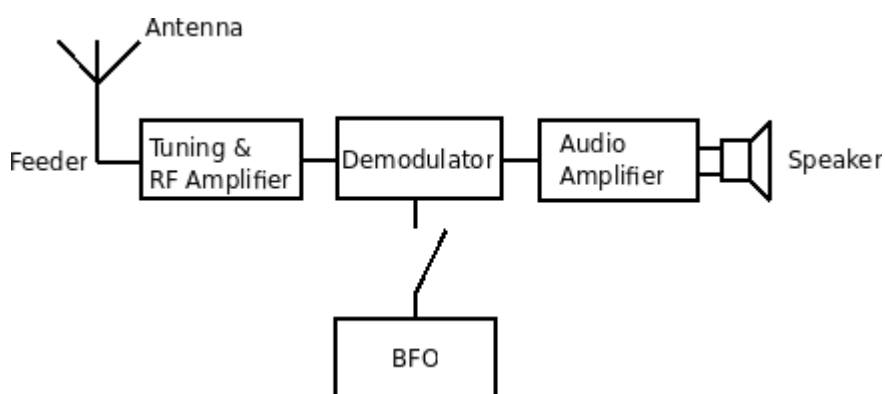


The germanium diode is able to rectify (or demodulate) an AM, and only an AM, signal by rectifying the wave to remove the negative component and recover the audio waveform, which will be covered later.

Tuned RF Receiver

The tuned RF receiver sought to improve on the crystal set in a couple of areas: firstly in terms of selectivity it included a tuned RF Amplifier in the first stage which was better able to isolate a wanted signal and secondly by providing a RF Amplifier first stage weaker signals become resolvable improving the sensitivity.

In addition the tuned RF receiver also included an AF Amplifier which allowed speaker output to be used in addition to headphones.



The block diagram includes a switchable Beat Frequency Oscillator which is one method for resolving SSB and CW signals which will be covered later. Tuned RF receivers also included the same diode demodulator as a crystal set.

The RF Amplifier is employed to amplify the signals coming into the receiver, the absence of this facility in the Crystal Set is why it is only good at receiving strong signals.

The Demodulator is the stage that recovers the original audio information from the modulated AM, SSB, FM or CW signal. Demodulation is specific to the mode in use and will be covered in more detail later. However, the process of demodulation is present in all analogue receivers.

The audio amplifier, which is also not present in the Crystal set, amplifies the recovered audio signal to allow playback through either headphones or a speaker.

**3H3 - Q24**

Recall that a receiver's ability to detect weak signals is known as its sensitivity.

Recall that very strong signals can overload a receiver and cause distortion to the audio output.

For a signal to be discernible a receiver has to be able to detect it. The weaker the signals that can be reliably detected the greater the sensitivity of the receiver. It should be noted that the power levels employed by radio amateurs, even those in countries that allow 1000W and 1500W, is still very small compared to the power levels operated by commercial broadcasters who typically employ 100's of kilowatts (Radio 4 in Droitwich used 2 x 250kW transmitters in 1985). It follows then that even mediocre amateur radio receivers have sensitivity figures well above those of even the best domestic commercial broadcast receivers.

The problem with having a receiver that is capable of discerning very weak signals and amplifying these signals to a usable level is that when these receivers encounter a strong signal they can become overloaded and this is where the selection of attenuators becomes important to prevent distortion of the output audio.

3H4 - Q24

Recall that a receiver's ability to reject frequencies outside the wanted signal bandwidth is known as its selectivity.

Understand the limitations of tuned circuits in selecting wanted frequencies and the effect of the Q factor of tuned circuits.

See also 2H4 in 'Technical Aspects'

Selectivity in contrast to sensitivity relates to the ability of a receiver to isolate one signal from among many and only demodulate the wanted signal without interference from adjacent signals. This can be quite difficult to achieve, particularly in a Crystal Set which only employs a tuned circuit.

In the 'Technical Aspects' section the concept of "Q" factors to describe the sharpness of a filter or tuned circuit was introduced. As a recap:

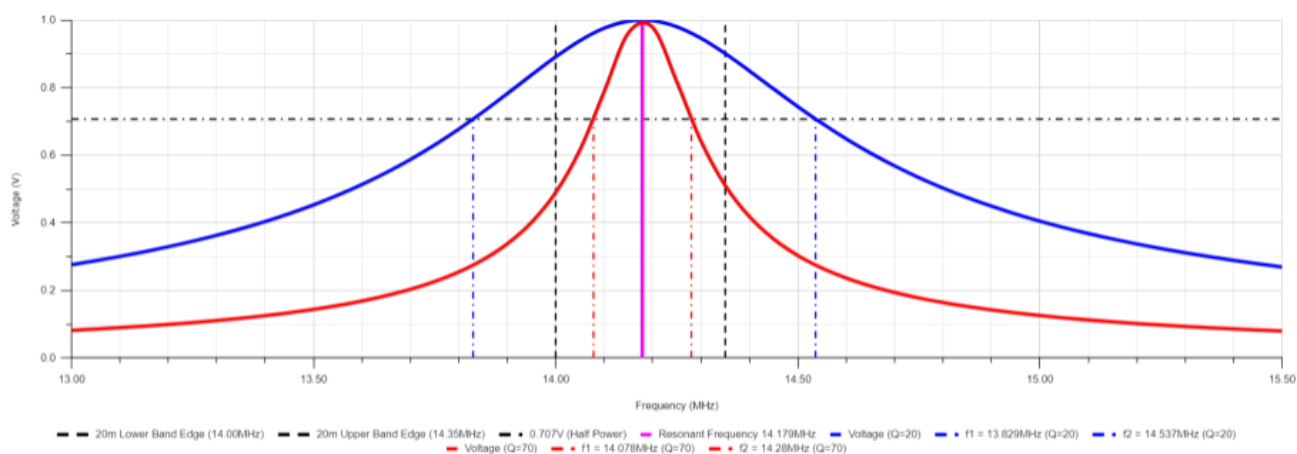


$$Q = \frac{f_0}{\text{bandwidth}} = \frac{f_0}{f_2 - f_1}$$

where :

- f_0 = wanted frequency
- f_1 = lower frequency at the half power point
- f_2 = upper frequency at the half power point

Bandwidth and Q Factor



In the UK on the MW band (531kHz – 1,602kHz) where AM prevailed stations are kept in 9kHz channel spacing. Note that this limits the audio bandwidth to 4.5kHz maximum, insufficient for faithful music reproduction but suitable for news and talk based programming. In fact most UK MW AM stations used 6.3kHz as the upper audio frequency meaning that where there were adjacent transmissions there would have been some overlap.

However, looking at a crystal set with a tuned circuit to provide selectivity then to ensure that the bandwidth was limited to 9kHz to limit acceptance of adjacent stations then the Q factor of the tuned circuit would need to be:

$$Q = \frac{f_0}{\text{bandwidth}} = \frac{531}{9} = 59 \text{ at the lower end of the medium wave band}$$

$$Q = \frac{f_0}{\text{bandwidth}} = \frac{1602}{9} = 178 \text{ at the upper end of the medium wave band}$$

Obtaining Q factors over 70 becomes difficult with a single tuned circuit, so whilst a well made crystal set can operate in the medium wave band with high powered broadcast stations when the AM signals move into the HF spectrum on frequencies of 1800kHz



upwards and reduced bandwidths of 6kHz it is clear that the crystal set isn't going to be able to offer the selectivity required to isolate a single signal for demodulation.

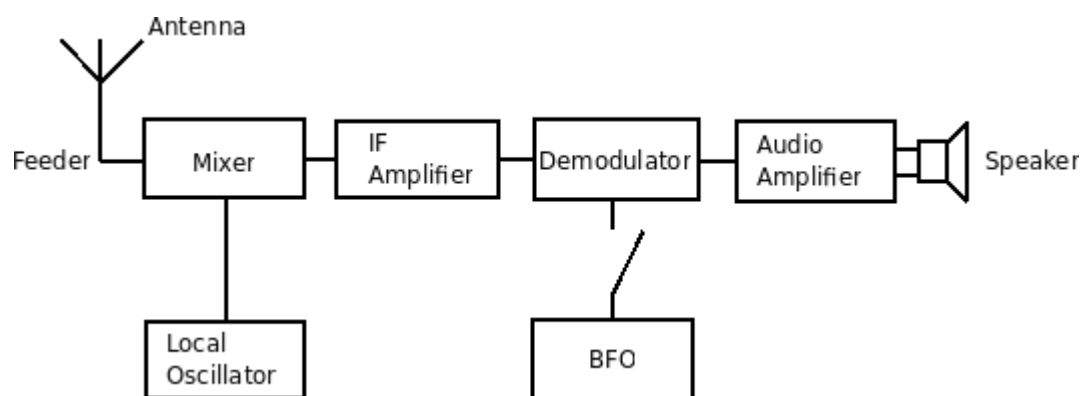
Superheterodyne concepts

211 - Q25

Understand the need for and advantages of the superheterodyne architecture.

Up until the ready availability of software defined radios the "superhet" was the de-facto standard architecture for a receiver with any intention of giving a good combination of both selectivity and sensitivity.

In fact many software defined radios are often hybrids involving aspects of both designs.



Comparing this diagram with the tuned RF receiver and working from right to left then:

- Both receivers have a speaker
- Both receivers have a demodulator
- Both receivers have an optional BFO or Beat Frequency Oscillator to resolve CW and SSB

However, the front end of the superhet (that is everything before or upstream of the demodulator) is different from the tuned RF receiver.

The three new processes: the mixer, local oscillator and IF amplifier are what allows the superhet to improve the selectivity of the tuned RF receiver.

The biggest problem with building a selective receiver is the requirement to be able to narrowly filter the wanted frequency and avoid any adjacent signals. Without knowing the



frequency of the wanted signal, as shown with the crystal set, getting a tight filter is difficult to achieve and any amplification of the signal has to utilise a broadband amplifier because the frequency of the wanted signal is also not known exactly.

The superhet overcomes this issue by mixing the incoming RF signal with a local signal to produce a fixed **Intermediate Frequency** irrespective of the frequency of the incoming RF signal.

3I2 - Q25

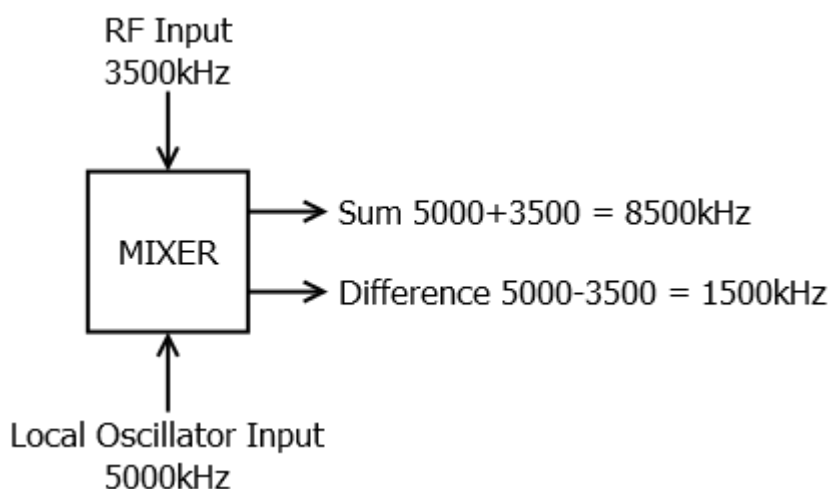
Recall that the intermediate frequency is the sum of or difference between the RF and local oscillator frequencies, and is produced by a mixer.

The process of mixing two signals is very powerful and finds many uses in radio including in the demodulation process and in transverters. Modulation is actually a mixing process except that in the case of modulation the mix is an AF signal with a RF signal, whereas in the superhet receiver the mixing is two RF signals one received from the antenna and one locally generated by an oscillator.

When two signals are mixed the result is that there are two outputs from the mixer:

1. The sum of the two input frequencies
2. The difference between the two input frequencies

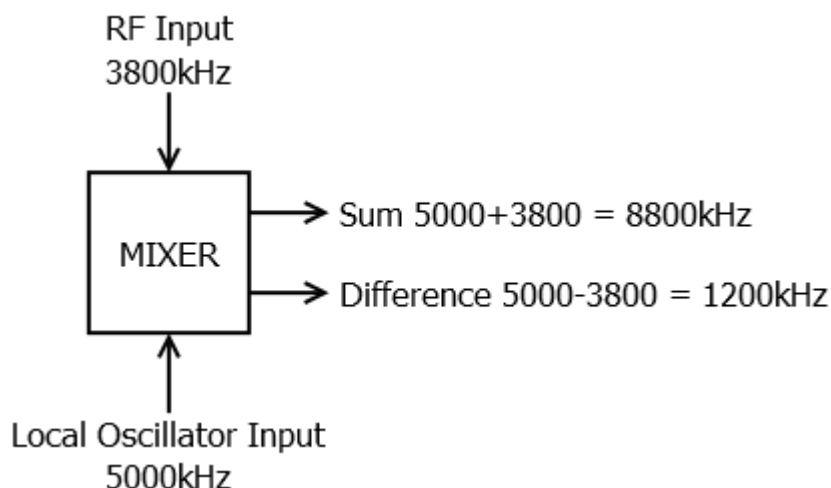
Consider a received RF signal with a frequency of 3.5MHz (i.e. at the bottom of the 80m band) mixed with a locally generated signal on 5MHz.





Mixing these two signals will result in two outputs one at the sum of the inputs (8.5MHz) and the second at the difference between the inputs (1.5MHz).

If the RF input is shifted up to 3.8MHz (i.e. the top of the 80m band) and the Local Oscillator is left at 5MHz then two different outputs are produced.



Mixing these two signals will result in two outputs one at the sum of the inputs (8.8MHz) and the second at the difference between the inputs (1.2MHz).

This doesn't help in limiting the frequencies that have to be accepted by the following stages, yes the frequencies are different from the input but they are simply following the RF signal, the design of the filters at the output of the mixer still have to be broadband enough to deal with the band edges and no improvement to the selectivity has been achieved.

The clever part of a superhet receiver is that the local oscillator isn't set to a fixed output but tracks the input RF so as to maintain a constant offset from the RF input so that either the sum or the difference output from the mixer remains at a fixed frequency.

If the decision is made that the difference frequency is the one to be passed down the receiver train for demodulation then at the bottom of the band the difference is 1.5MHz, so at the top of the band the difference also has to be 1.5MHz meaning that as the receiver is tuned up by 300kHz (0.3MHz) then the local oscillator has to be tuned up by the same 300kHz (0.3MHz) so that the difference output remains at 1.5MHz.



INTERMEDIATE
Transmitters and Receivers
Syllabus 1.5

Local Oscillator VFO



Tuning/RF Amplification



Difference



This means that irrespective of the input RF frequency the output difference signal will always be at 1.5MHz.

3I3 - Q25

Recall that a superheterodyne receiver uses a fixed IF stage to enable good selectivity and that mixing ahead of the IF enables multiband reception.

Understand that tuned circuits in RF and IF amplifiers select the wanted signal.

Identify the tuned circuits in the circuit of an IF amplifier.

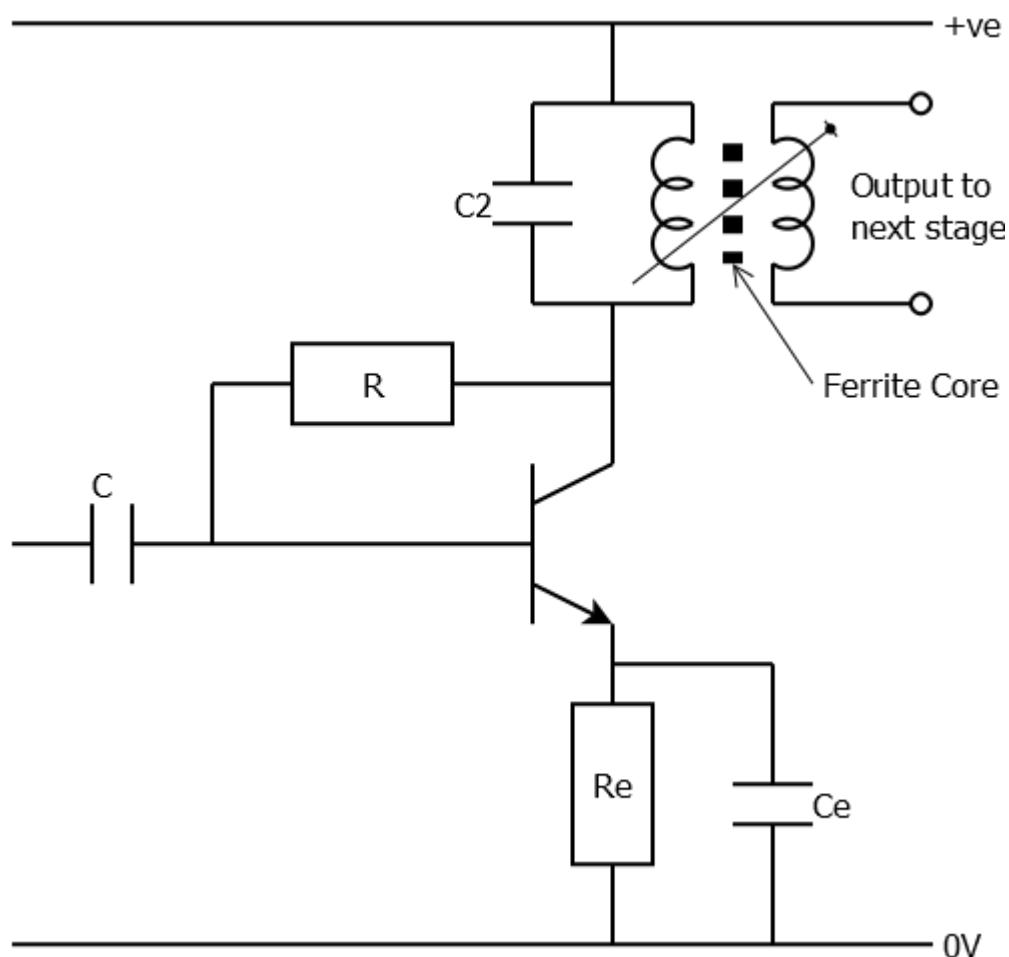
By fixing the output from the mixer the IF amplifier does not have to be broadband, it only has to be designed to operate at 1.5MHz, this allows it to have much tighter (higher Q) filtering that only has to be designed for a single frequency making it far easier to block adjacent off frequency signals and only amplifying the wanted signal.

If multiband reception is required then clearly the local oscillator has to be adjusted to ensure that each band when mixed with the local oscillator signal produces one output on the fixed intermediate frequency.



Tuned circuits and filters in both the RF reception line and the IF amplifier limit the input and select or isolate the wanted signal.

The Intermediate Amplifier contains tuned circuits as shown below.

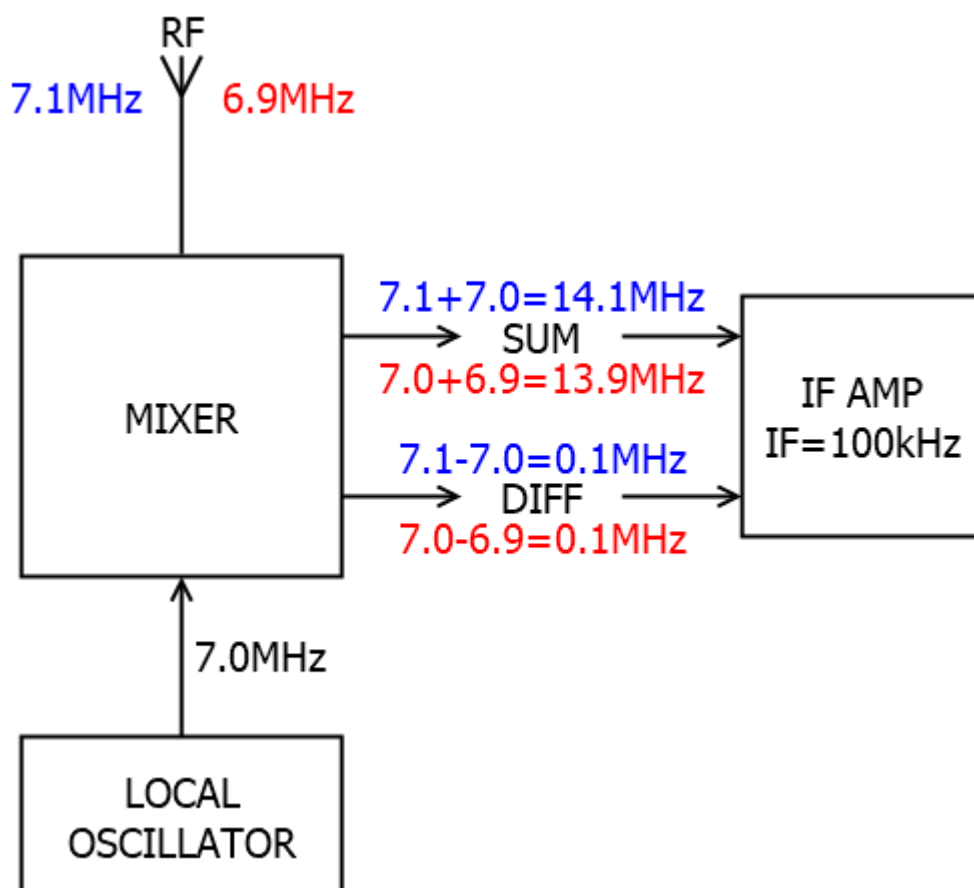


The output from the IF amplifier is passed to the downstream stage (in this single conversion superhet) via the IF transformer. The IF transformer is usually encased in a metal enclosure and an iron or ferrite "slug" is used to fine tune the resonant frequency.

The receiver train demonstrated here is a single conversion superhet, most modern receivers are double conversion superhets that use two different intermediate frequencies (or in some cases triple conversion). The reason for this is because the selection of the intermediate frequency is something of a compromise. For demodulating the signal a small IF is useful around 455kHz (based on Yaesu FT817ND 2nd IF specification) whereas for easy separation of the mixer outputs and importantly to avoid problems with image signals a high IF is preferable, typically around 68.33MHz (based on Yaesu FT817ND 1st



IF specification). A lower IF also allows a higher Q to be achieved in the tuned circuit. In a single conversion design some compromise has to be made between the requirements but with a double conversion design a large first IF can be adopted to deal with the image rejection and separation of the mixer outputs and then mixed down a second time to a low 2nd IF for final selection and demodulation.





Demodulation

3K1 - Q26

Understand how a diode detector will recover the audio from amplitude modulated signals.

Understand that to generate the audio from CW signals a Beat Frequency Oscillator (BFO) is used; for the recovery of single sideband audio a carrier insertion oscillator (CIO) and product detector are used and for the recovery of FM audio a discriminator is used.

Identify the waveforms produced in a diode AM detector.

A diode detector is probably the simplest form of detector. It is only capable of demodulating AM and it does this by removing the negative section of the modulated waveform. Around the remaining part of the wave there is a clear smooth envelope which can be resolved in headphones as audio.

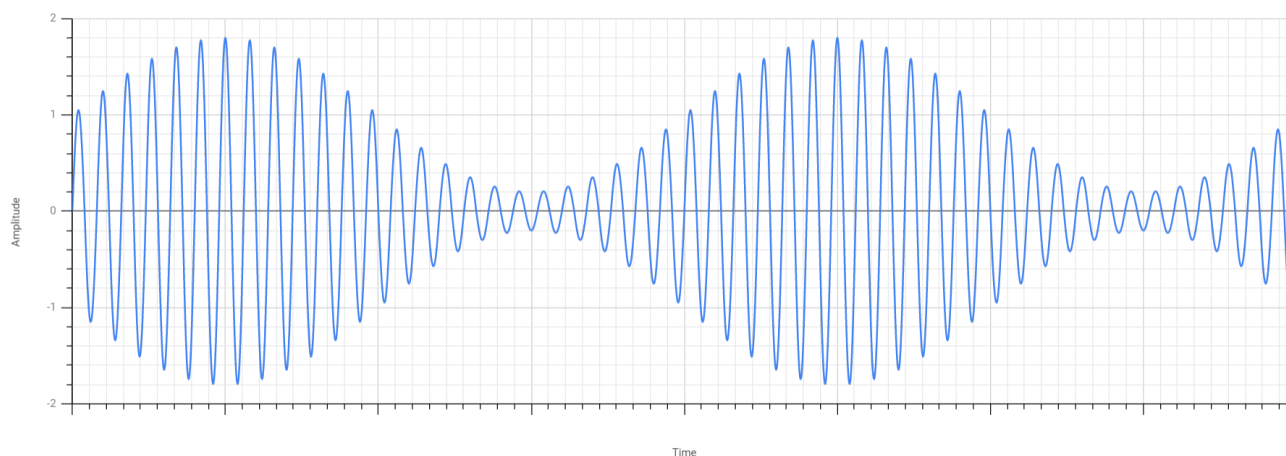


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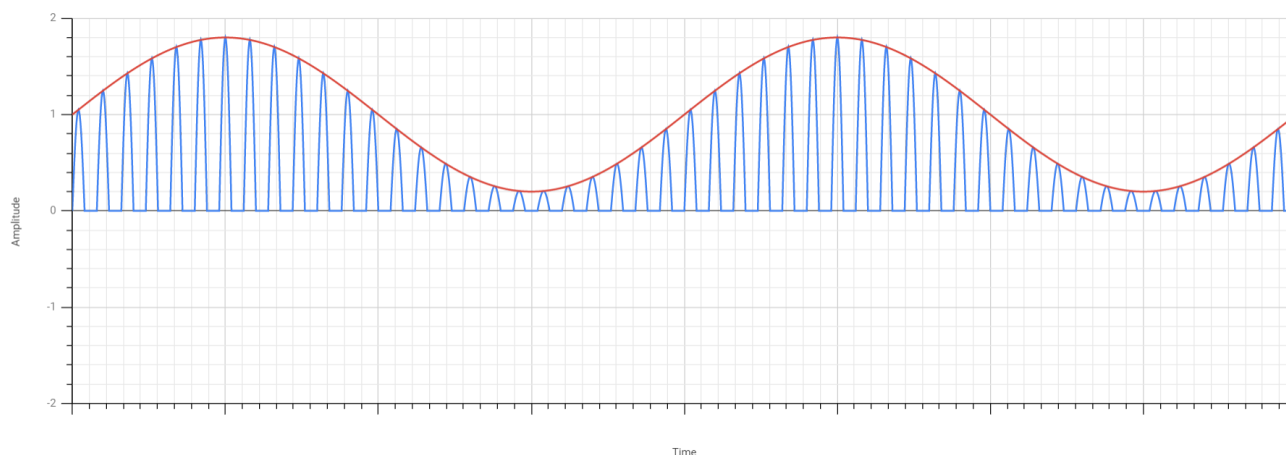
Transmitters and Receivers

Syllabus 1.5

Modulated Carrier

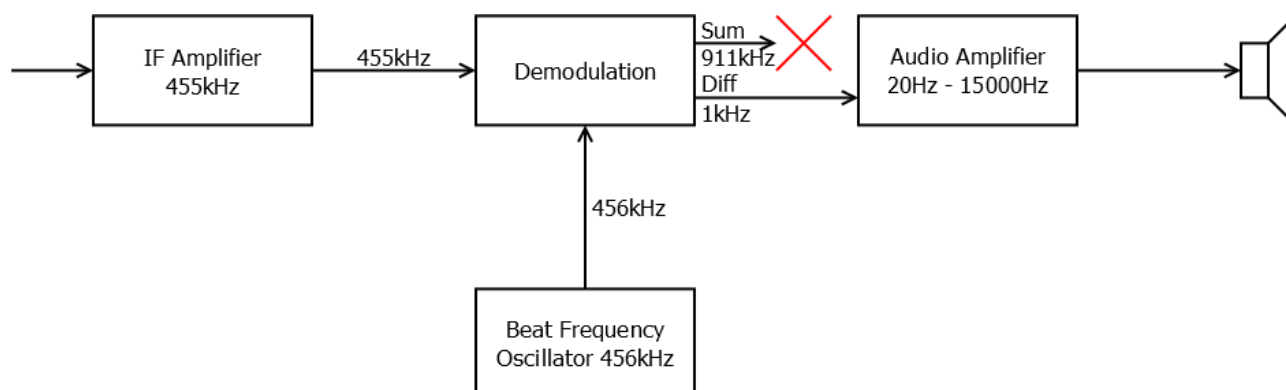


Demodulated Carrier



Recovery of other forms of transmission such as CW and SSB require more sophisticated methods.

To demodulate CW a **Beat Frequency Oscillator** (BFO) is employed. A beat frequency oscillator runs about 1kHz higher than the IF of a superhet (in a tuned RF receiver it runs about 1kHz above the RF frequency). This oscillator's signal is mixed with the IF signal and a sum and difference are created. The difference will be audible whilst the sum will be well beyond the input of a typical audio amplifier.



To demodulate a SSB signal requires a **product detector** coupled with a **carrier insertion oscillator** (CIO). Similar to the concept of a BFO, the CIO runs at a frequency slightly lower than the IF when handling a USB signal and slightly higher than the IF when handling a LSB signal. In effect the CIO replaces the carrier that was removed by the balanced modulator in the transmitter chain.

To demodulate a FM signal requires a **frequency discriminator** which resolves the audio from the small changes in the IF.

Automatic Gain Control (AGC)

3L1 - Q26

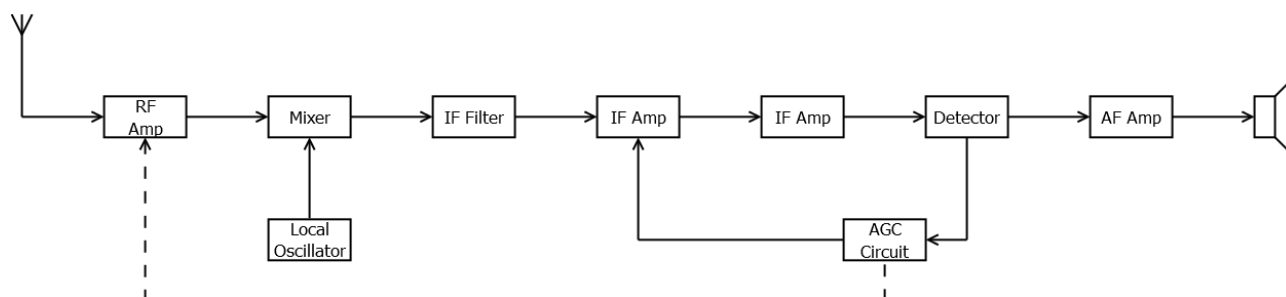
Understand that the automatic gain control (AGC) of a receiver operates by sensing the strength of the received signals at the detector and adjusting the gain of the IF and sometimes the RF amplifiers to keep the audio output level fairly constant. Recall that the AGC signal can also drive a signal strength meter (S-meter).

The purpose of the AGC, which is present on most modern receivers, is to try to maintain a constant audio output in the face of a received signal of varying signal strength. It is not uncommon, particularly on HF, for signals to vary in strength during a QSO due to fluctuations in the transmission path resulting from changes in the ionosphere and signals arriving by different paths which reinforce or partially cancel the signal.

The AGC works by monitoring the strength of the signal leaving the detector/demodulator stage and increasing or decreasing either the IF amplifier gain or the RF amplifier gain. As the signal level leaving the detector/demodulator reduces so the AGC loop will increase the gain of either the IF amplifier or the RF amplifier to compensate. Similarly if the signal



level leaving the detector/demodulator increases the AGC loop will decrease the amount of gain in either the IF amplifier or the RF amplifier.



SDR Transmitters and Receivers

3M1 - Q27

Recall that SDR software uses a mathematical function called a Fourier transform which sifts the composite signal into its constituent independent frequencies for processing.

Recall that this can also be used to provide a spectrum or waterfall display.

Recall that digital filters can be much more selective than analogue filters.

Software Designed Radio (SDR) is becoming the common technology replacing older analogue equipment. The basic principle of the receive side is that all signals received by the antenna are passed through an **Analogue to Digital Converter** (ADC) and all selection, filtering and demodulation are undertaken using software.

The 'Technical Aspects' section has already described the concepts of **sample rate**, **resolution** (no. of bits or reference levels) and importantly the **Nyquist Rate** which states that the sample rate should be at least twice the highest frequency present in the input signals.

The **Fourier transformation** is the key building block to software defined radios, it is essentially capable of sifting through a complex composite signal and deriving the individual signal components for subsequent processing.

Essentially the Fourier transform can be likened to a machine into which you pour a fruit smoothie and which gives you a banana, some cherries and an apple or whatever fruits went into producing the smoothie!



Unlike the fictitious smoothie deconstructor the results of the Fourier transform is a spectrum scope showing the relative signal strengths over a frequency range and can be adapted to produce a waterfall showing the presence of signals.

In a traditional analogue superhet (i.e. one without the benefit of DSP) any installed filters have to be constructed from discrete components, either mechanical filters which tend to be expensive or combinations of inductors and capacitors. This limits the options within a traditional analogue typically to a SSB filter and a CW filter.

By contrast DSP functionality allows filters of any width often with variable sharpness that can be adjusted "on the fly". What's more, digital filters can be more sensitive than their analogue counterparts due to a complete lack of resistive losses.

3M2 - Q27

Recall the meaning of the time domain and the frequency domain.

Understand how signals in the time domain may also be viewed in the frequency domain.

Identify for some simple harmonic waves, the spectrum obtained using the Fourier transform.

(Waves composed of one and two Harmonics will be examined).

Key to the SDR process is the Fourier transform and the concepts of **frequency domain** and **time domain**. The differences between these two domains is simply how to look at a composite signal made up of many individual signals such as might be received on an active band.

Consider three separate harmonically related signals with differing amplitudes and phases say: 7.075MHz, 14.150MHz and 21.225MHz. In the time domain, when considered separately, the three signals appear as follows:

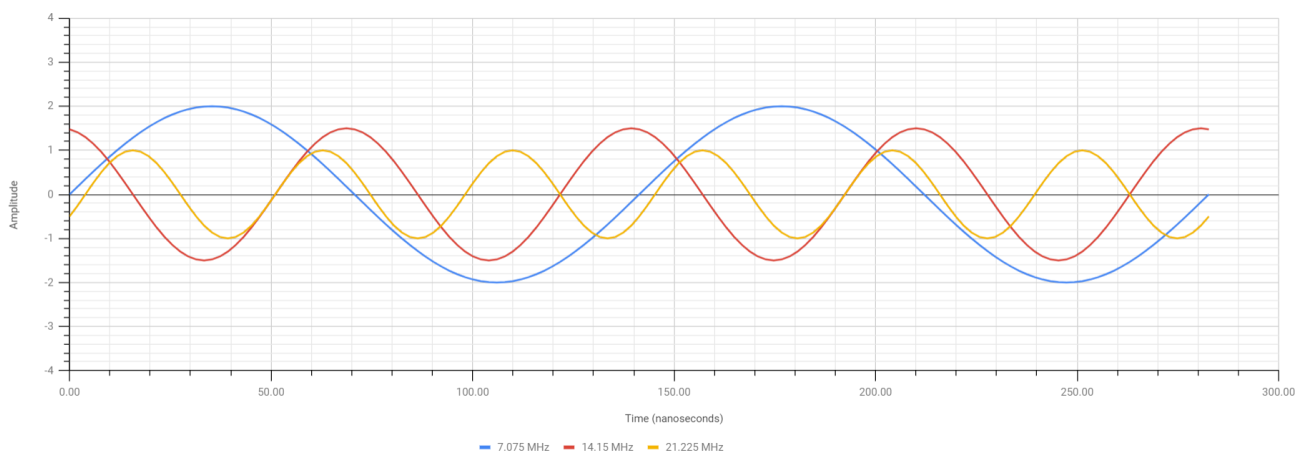


INTERMEDIATE

Transmitters and Receivers

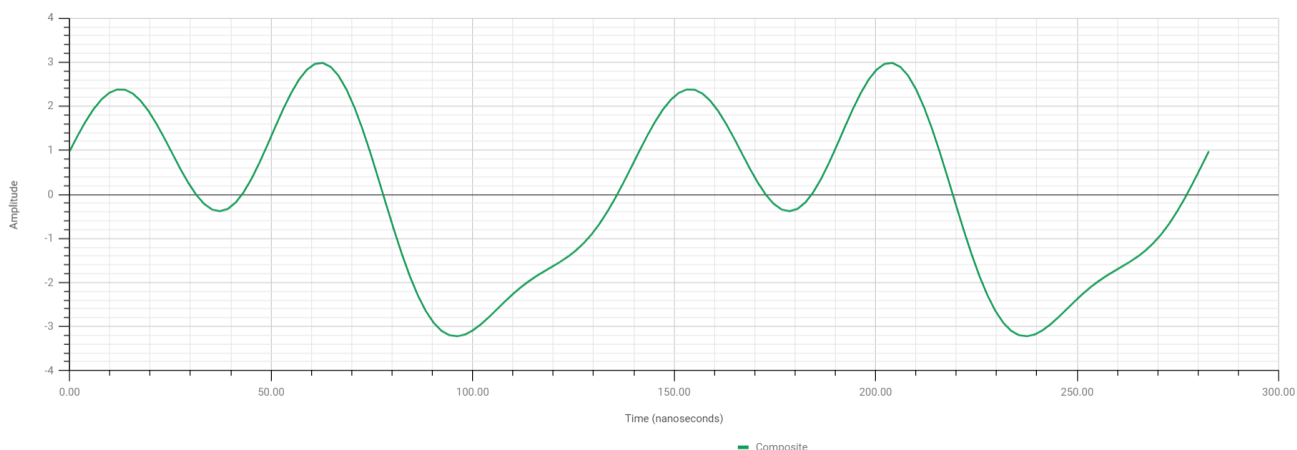
Syllabus 1.5

Signals in Time Domain



Looked at this way, the signals are easy to visually separate, however the receiver doesn't see 3 discrete signals like this, what the receiver sees is a composite signal in the time domain as shown below.

Signals in Time Domain

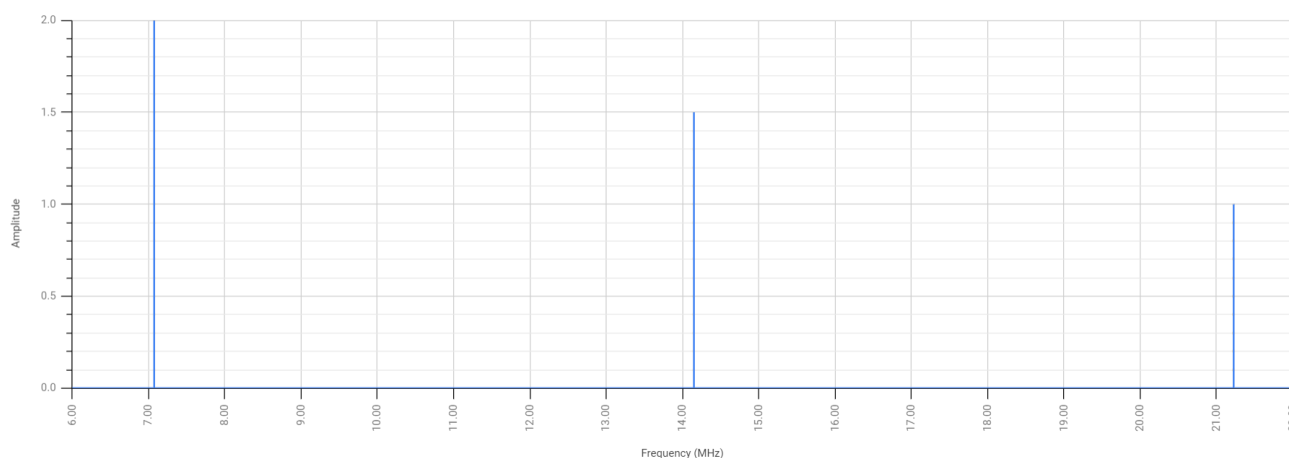


This is clearly a complex signal, but even so some patterns should be discernible, there is an obvious repeating pattern.

The Fourier transform can take this composite signal and reduce it to the component signals, and in doing so converts the information from the **time domain** to the **frequency domain** so that the resulting output looks like this:



Signals in Frequency Domain



This produces a spectrum scope showing signals with their relative amplitude spread out by the frequency of the signal and can also be used to develop a waterfall display.

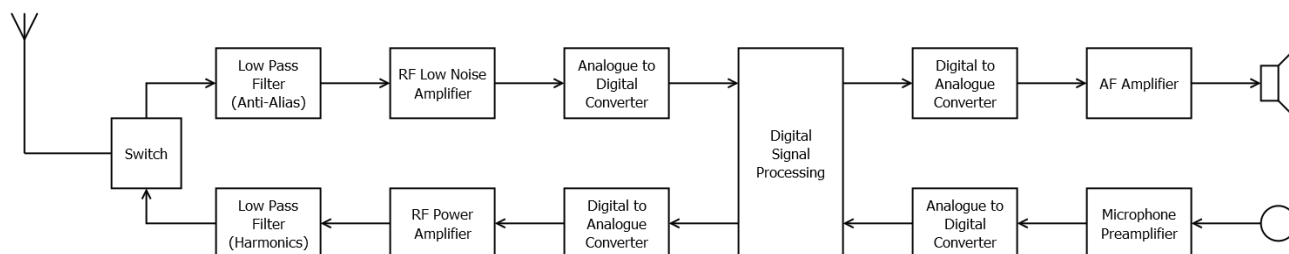
This process can also apply to non-harmonically related signals and even close-in signals within the same band can be distinguished by this method.

The Fourier transform is also able to identify signals within a composite signal which in the time domain looks full of noise with no obvious pattern.

Once the signals have been identified in the frequency domain the wanted signal can be selected using digital filtering. Digital filtering is found on many non-SDR receivers and transceivers through the application of DSP techniques; however their application in Software Defined equipment is one of the strengths of the digital evolution of equipment that is currently taking place.

3M3 - Q27

Recall the different elements that make up the functions of an SDR (block diagram).





INTERMEDIATE
Transmitters and Receivers
Syllabus 1.5
