HI-FI ON THE AIR

BACK TO BASICS

Ever wondered about the mysteries of detection, discrimination, MPX and other radio tuner terminology? This article explains the basics . . .

by DAVE BERRIMAN

Since the first experiments by Heinrich Hertz*, which proved the existence of radio waves, technology has developed the use of radio to a level which would have been unbelievable to those early pioneers. In Hertz's experiment, a powerful spark jumping across an air gap caused a smaller spark to jump across another, quite separate, gap a few feet away.

Hertz's early spark transmitter produced a broad spectrum of radio waves, what we now describe as interference (of the sort produced by electrical machinery and car engines, for instance, see Fig 1a). It was about as much use for selective communications as using a shotgun to remove a single leaf from a tree!

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The breakthrough for radio communications came with practical experimenters like Marconi, who married improved spark generators to tuned electrical circuits, which narrowed the range of frequencies which were transmitted (Fig 1b) enabling more than one station to use the air waves simultaneously, while allowing the distant selection and reception of just one of them (see Fig 1c).

AMPLITUDE MODULATION

The earliest transmissions could not transmit voices or music. They were known as continuous wave transmissions (see Figs 1b and 1c). The radio wave could only be turned on or off; the only method of conveying information was to do just that to the radio signal, ie, in Morse code (Fig 3).

Later, with the advent of the thermionic valve, it became possible to construct powerful radio transmitters without the use of spark generators. Valves were used to modulate (vary) the strength (or amplitude) of the radio wave to convey continuously varying (analogue) waveform of speech and music (as opposed to merely turning it on and off in bursts). This is amplitude modulation (Fig 4). It is a simple and elegant method still used for many radio transmissions.

In an AM receiver, a rectifying circuit or AM *detector* removes half of the alternating radio wave (say the bottom half) leaving a rectified radio wave (Fig 5). When the radio frequency half-wave pulses are filtered, their average value, which is the audio signal (plus some DC), remains. The DC is usually used to control the gain of earlier parts of the radio receiver, to compensate for changing reception conditions, which would otherwise cause fading of the signal. This is called Automatic Gain Control, or AGC for short.

Early AM sets used a small crystal of gallium with a whisker of wire in contact, known as the cat's whisker. Its use involved the laborious quest for a 'sweet spot' on the crystal where rectification was most effective. This was later replaced by the point-contact diode (essentially the same thing sealed in a miniature glass tube but using germanium or silicon), and more recently Shottky barrier diodes (which work well at very high speed).

DEFINING THE BANDWIDTH: SELECTIVITY

Achieving the desired sharpness, or 'Q', of transmission or tuning has always presented problems. A high 'Q' would seemingly be what is desired because this would allow only a narrow band of frequencies and thus filter out extraneous signals more effectively (Fig 1c). This would be fine, so long as the radio transmission contained only the radio-frequency 'carrier' wave and no modulation (Fig 1b). However, as soon as the radio wave is modulated, the range of transmitted frequencies is increased, either by Amplitude or Frequency Modulation. Amplitude Modulation produces extra signals, called sidebands, at frequencies of the carrier plus the modulating frequency and the carrier minus the modulation frequency (Fig 6).

For instance, take a carrier frequency of 600kHz. If this is amplitude modulated by a frequency of 4kHz, the result is the carrier at 600kHz and in addition, sidebands at 604kHz and 596kHz. If the complete band of frequencies between DC and 4kHz were present, the spectrum would show a multitude of sidebands ±4kHz from 600kHz.

AM SELECTIVITY

The long, medium and short-wave AM bands are so crowded that it is not practical to transmit the full audio spectrum, so it is usually limited to around 5kHz, thus curtailing the sidebands and cramming in more transmissions. The tuned response of the receiver should be broad and flat, with steep sides, if the higher audio frequencies are not to be severely reduced in level (attenuated). A very sharply tuned filter will achieve the desired selectivity, but will result in a very dull sound. This, as much as the inherent 4kHz upper limit, is why many AM radios and the AM sections of some hi-fi tuners often sound dull and lacking in sparkle.

FREQUENCY MODULATION

Hi-fi tuners are much more advanced in their operation than the early AM radios. They use an alternative form of modulation known as Frequency Modulation (or FM), in which it is the *frequency* of the radio-frequency carrier which is varied in sympathy with the audio waveform at the transmitter; the amplitude stays constant. As shown in Fig 7, the carrier frequency deviates above and below its centre frequency in direct response to the modulating (audio) signal. The degree of Frequency Modulation is thus referred to as the deviation, the amount of increase and decrease in frequency to which the carrier is subjected. For hi-fi FM, the *maximum* deviation, corresponding to the peaks of the audio signal, should be ±75kHz, but smaller modulating signals produce smaller deviations. Note, the frequency of FM deviation is *not* the frequency of the audio signal, but is dependent on its level.

FM SIDEBANDS

FM modulation produces a different pattern of sidebands from AM. With FM they spread out in pairs from the centre frequency, their frequencies and amplitudes depend both on the deviation and the audio signal frequency (see Fig 7). FM sidebands, perhaps surprisingly, extend beyond the maximum deviation of the radio frequency carrier wave. Away from the carrier they are of progressively lower level. For reception, it is not vital to include all the sidebands, but curtailing too many, or shifting their phase, will worsen the performance.

FM SELECTIVITY

Fortunately in the 87.5 to 108MHz VHF waveband used for UK FM radio, interference from other transmissions is less of a problem than on the AM wavebands because the signals don't usually travel so far as at the lower frequencies used for long

**As many readers will know, Hertz's name, appropriately, has been immortalised as the term for frequency, the abbreviation Hz replacing the old 'cycles per second' or c/s.*

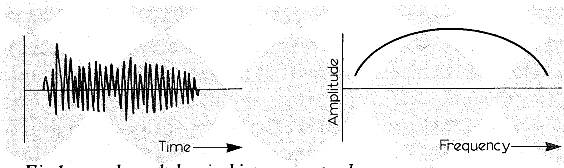

Fig 1a: sparks and electrical intererence produce a broad spectrum of radio waves, which are of little use for communications

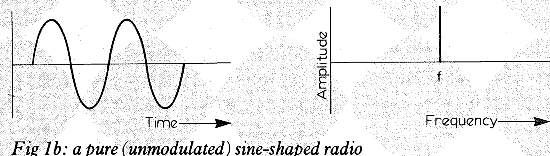

Fig 1b: a pure (unmodulated) sine-shaped radio wave contains only one frequency and this can be used for selective radio transmissions

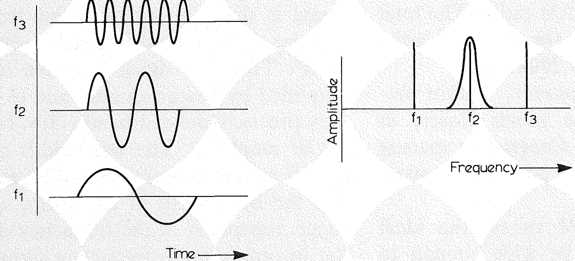

Fig 1c: though three radio waves are present at the same time in this example, a single one can be selected by tuned circuits in the receiver

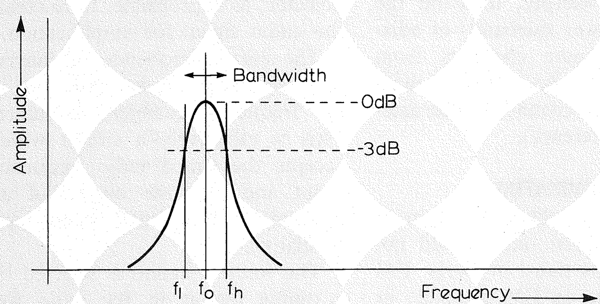

Fig 2: the bandwidth of a tuned circuit is defined as the part of the response curve which is within -3dB of the peak. Thus in this diagram the bandwidth is simply fh-fl in Hz

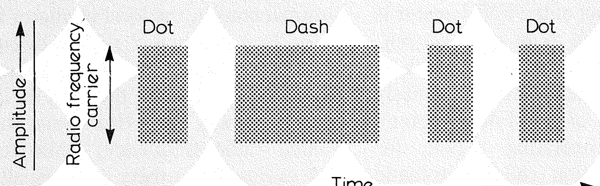

Fig 3: turning the carrier wave on and off using Morse code was the earliest method of communicating using radio waves

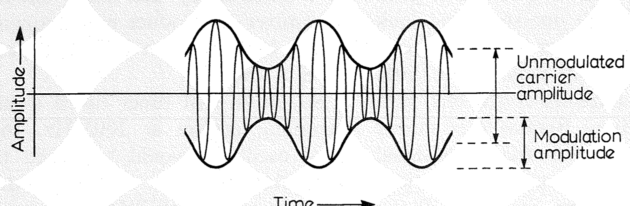

Fig 4: Varying the amplitude of the radio wave in sympathy with the audio waveform is called amplitude modulation and is usually expressed as a percentage of the carrier wave. If the peaks of the audio waveform reduce the RF carrier momentarily to zero, the modulation is 100%

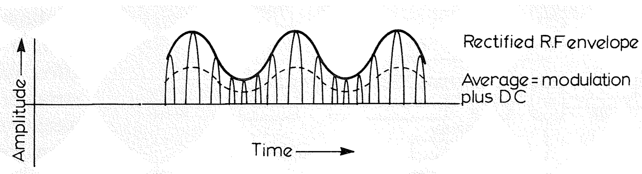

Fig 5: If the RF carrier is rectified in the receiver, the average value of the RF pulses is the original modulating wave. Some DC is also present and this can be used to regulate the gain of earlier stages in the radio to prevent signal fading or circuit overload

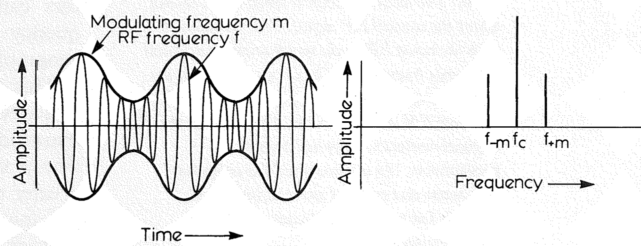

Fig 6: amplitude modulation introduces sidebands at frequencies spaced above and below the carrier, at the carrier frequency, plus the modulating frequency and also at the carrier minus the modulating frequency

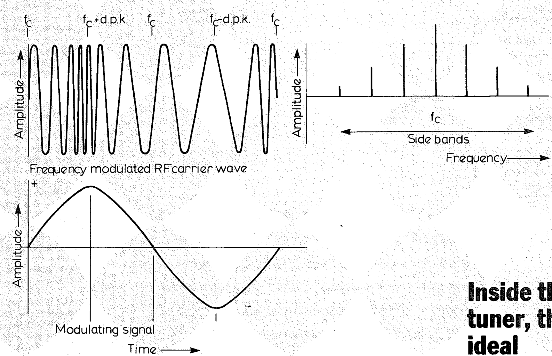

Fig 7: frequency modulation creates an infinite number of pairs of sidebands which reduce in level away from the carrier, fc. The distribution of sidebands depends on the modulation index (the maximum deviation divided by the maximum modulation frequency). It is not necessary to receive all the sidebands to recover the modulation accurately, For a deviation of 75kHz and audio modulation up to 15kHz, the bandwidth required is 2×(75+15), which equates to 180kHz

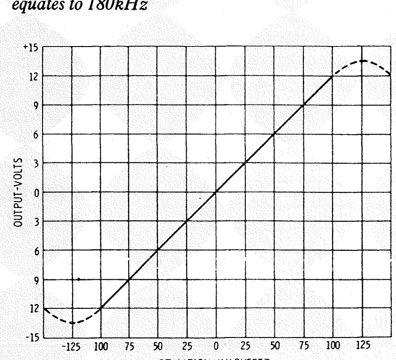

Fig 8: the FM detector provides an output which is dependent on the frequency from the I.F amplifier. Frequency modulation at audio frequencies therefore produces a corresponding audio output. If the tuner drifts off-station, the centre frequency will shift from 10.7MHz, producing either a positive or negative DC. voltage. This is fed back to earlier stages to pull the tuner more accurately back on to the station

Inside the FM tuner, the ideal bandwidth will be wide enough to include all of the information contained within the radio signal. Outside this wanted range, the response should drop sharply, to avoid the upper and lower sidebands of adjacent or alternate channels from interfering with the wanted channel (called adjacent-channel or alternate-channel interference)

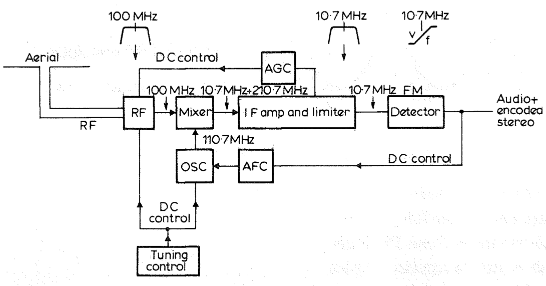

Fig 9: simplified block diagram of an FM tuner showing how the superheterodyne principle works. The local oscillator is tuned to 10.7HMz above the wanted RF signal. This is combined with the tuned RF in the mixer to create sum and difference frequencies. The IF section selects the difference of 10.7MHz, amplifies and limits it and then FM demodulates it. The output is the original modulation applied prior to transmission. (In a stereo tuner this comprises both mono and encoded stereo information, which is fed to the stereo decoder, see stereo encoding and decoding)

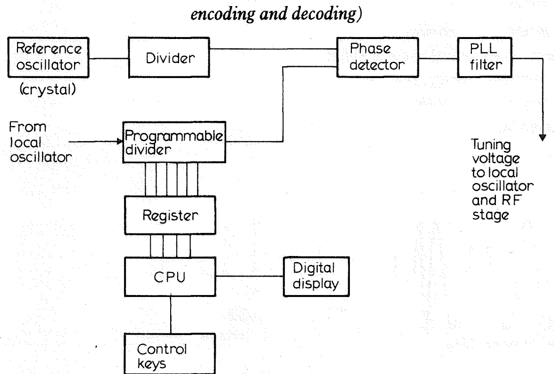

Fig 10: a synthesised tuner works in exactly the same way as a conventional superhet, except for the way the local oscillator frequency is derived. The output from a highly stable internal crystal oscillator is divided down to a more manageable frequency and fed to a phase detector. The local oscillator frequency is also divided down in frequency, but by a programmable divider which is under control of a Central Processor Unit (computer chip) which also controls a digital readout for display of frequency, etc.

The output of the programmable divider is some fraction of the desired local oscillator frequency and this is compared to the set fraction of the crystal oscillator by the phase detector. This is followed by a phase-locked loop filter which produces a DC control voltage for feeding to the local oscillator and RF stage. This seemingly complex system effectively locks the local oscillator very accurately to the desired frequency with the precision of the crystal oscillator, thus virtually eliminating drift and enabling tuning to be programmed very accurately

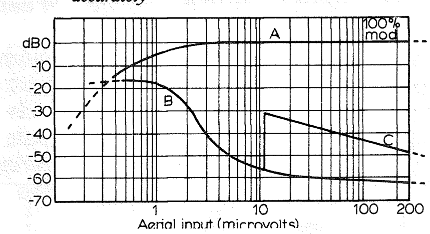

Fig 11: as the radio-frequency input of a FM tuner is increased, the output increases as shown by curve A. Simultaneously, the noise drops as shown by curve B, which indicates the noise in mono. At some pre-determined threshold, the stereo decoder will switch on automatically. In our example this occurs at just over 10μV, above which the stereo noise decreases as shown by curve C

medium and short waves. In addition, a phenomenon called *capture effect* locks an FM tuner on to the stronger of two signals, rejecting the weaker one even if it is on exactly the same frequency. The minimum ratio between the wanted and unwanted signals, for the larger one to dominate, is known as the capture ratio and is normally expressed in dB. The figure can be as low as 1 or 2dB in top tuners. Consequently, FM transmitters can operate at the same frequency as others, provided they are geographically distant.

This is fortunate because, with an FM deviation of ±75kHz, the total bandwidth taken up by the transmission is in excess of 150kHz, – much greater than for AM radio. The total spacing between the so-called adjacent channels is 200kHz.

In reality, however, station frequencies can be much closer, at 100kHz, or less. Geographic spacing and capture effect combine to avoid interference.

Inside the FM tuner, the ideal bandwidth will be wide enough to include all of the information contained within the radio signal. Outside this wanted range, the response should drop sharply, to avoid the upper and lower sidebands of adjacent or alternate channels from interfering with the wanted channel (called adjacent-channel or alternate-channel interference).

THE FM DISCRIMINATOR

With FM the amplitude is not modulated, so an AM radio would not respond to this signal properly. FM tuners include an FM detector, or *discriminator*, the output of which depends on the frequency of the radio frequency from the IF amplifier signal (see Fig 8).

A by-product of the FM detector is that, in addition to an audio output, it provides a DC voltage which is used to help tune the station precisely to the centre of the FM detector's most linear operating region, for minimal distortion (called Automatic Frequency Control, or AFC).

HOW RECEIVERS WORK

Having got this far, this is a good place to retrace our steps and look again at just how radio receivers work. The early radios were little more than amplified crystal sets, with poor selectivity. Because they simply tuned into the radio frequency signal prior to detection and amplification, they were known as Tuned Radio Frequency radios. Their selectivity was often enhanced by feeding part of the radio-frequency output back to and in phase with the aerial input (a form of positive feedback called regeneration). If taken too far, the radio would burst into uncontrolled oscillation (and effectively become a transmitter). Just below that point, however, the amplification was enhanced, the 'Q' increased and tuning sharpened to a significant degree.

FREQUENCY CONVERSION: THE SUPERHET

A significant improvement over the early TRF receiver was the Superheterodyne, or Superhet for short. The system is so effective that it is still in use today in just about every radio and hi-fi stereo FM tuner.

The basics are simple. The radio frequencies are selected by tuned circuits, just as for the TRF receiver, but the frequency of the incoming signal is then shifted to a new frequency, the Intermediate Frequency, or IF. This new signal can then be amplified by a large amount and to a very precisely defined bandwidth, by an IF amplifier. The bandwidth of the IF amplifier therefore effectively sets the bandwidth of the radio, or tuner, regardless of the frequency of the incoming radio frequency signal (see Fig 9, which shows the basic layout of an FM tuner). Finally, the signal is passed to an AM or FM detector, as appropriate, to re-create the audio signal for amplification.

The shift in frequency, or conversion as it is called, is carried out in the frequency changer, or mixer. This is effectively a circuit which accepts the tuned radio frequency input and also the output of an internal oscillator (known as the local oscillator).

The wanted station is selected by a tuning circuit in the radio frequency section. This part of the radio is designed to be as linear as possible to avoid intermodulation between powerful transmissions. Simultaneously, the local oscillator is tuned to a frequency *slightly to one side* of the wanted radio frequency signal. In FM tuners, the local oscillator is always 10.7MHz *above* the wanted radio frequency, while for AM radios and tuners it is usually around 450kHz above.

The incoming radio frequency and local oscillator signals mix together in a deliberately non-linear circuit (the mixer) to produce new frequencies at both the sum and the difference of the two.

So, in an FM tuner tuned to an incoming station at 100MHz, the local oscillator would be tuned to 110.7MHz and the output of the mixer would be 210.7MHz (the sum) and 10.7MHz (the difference). The 210.7MHz is not required, but the 10.7MHz signal is the Intermediate Frequency, containing the sidebands (and hence wanted information) of the original but at a new, carefully defined range of frequencies. Unfortunately, there is another radio frequency which can produce an output of 10.7 MHz from the frequency changer and that is 121.4MHz in our example (121.4-110.7 = 10.7). This is the image frequency, which must be rejected in the RF stage. Manufacturers' specifications often quote an *image rejection* figure.

SYNTHESIZED TUNING
Many tuners have a digital display which is simply a digital readout of the local oscillator minus 10.7MHz to provide the frequency of the tuned station. These are therefore nothing more than ordinary tuners with a digital readout. A *synthesizer* tuner, on the other hand, effectively 'locks' the local oscillator to a very accurately set frequency derived from an internal crystal-controlled oscillator (see Fig 10). Automatic frequency control may still also be used to align the tuner even more precisely to the station. The benefits are very stable drift-free tuning.

STEREO MULTIPLEXING
Stereo multiplexing is a very effective way of squeezing two channels of information into one and is an example of encoding by time division multiplexing (or MXP). At the transmitter, an encoder is used which very rapidly switches between left and right channels of the audio input (Fig 12a). The frequency of switching is at 38kHz, well above the audio band, inaudible to the listener. To a normal mono tuner or radio, it is to all intents and purposes a mono signal, because though it is switched it contains both the left and right channels. A mono tuner therefore processes it as a mono signal.

For a stereo tuner to be able to recover the separate left and right channels, the tuner must switch the combined L+R signal to left and right channels at 38kHz and in exact step with the transmitter. Rather than transmit a 38kHz signal, which would take up valuable modulation space, a 19kHz signal, known as the pilot tone, is transmitted instead (see Fig 12b). This is recovered in the tuner and used to regenerate the 38kHz switching signal. Provided the original left and right signals contain no information above 19kHz, the stereo composite signal (Fig 13) will contain all the information in the original left and right channels.

PRE- AND DE-EMPHASIS
Prior to stereo encoding for radio transmission, the high-frequency range is boosted slightly by 6dB per octave above about 3kHz. This pre-emphasis improves the signal-to-noise ratio because, after decoding in the tuner, a corresponding de-emphasis is incorporated which corrects the frequency response to flat, while reducing noise in the process. The pre- and de-emphasis are equivalent to a resistor/capacitor filter having a time constant (resistance multiplied by capacitance) of 50μs (75μs in the USA).

STEREO DECODERS
Decoding involves recovering the 19kHz pilot tone, doubling it to generate 38kHz and locking an oscillator accurately to this frequency, switching to recover the original left and right channels (Fig 14).

However, if signals above 53kHz, such as pilot-tone harmonics from adjacent channels, enter the stereo decoder, they can cause 'birdie' interference (a kind of fluctuating high-frequency noise). To prevent this, a low-pass 53kHz filter is incorporated prior to the stereo decoder. After decoding and amplification, de-emphasis restores the original flat frequency response and a low-pass MPX filter removes any pilot-tone or harmonics which could interfere with tape recorder operation.

The exact method of re-creating the 38kHz switching frequency varies from tuner to tuner, but often involves a phase-locked loop which compares the phase of the 38kHz oscillator to the doubled pilot-tone and locks the oscillator to it in phase and frequency: hence the term phase-locked-loop stereo decoder.

A different way of looking at stereo decoding is to consider the frequency domain instead of the time domain. While the baseband information, up to 15kHz, represents L+R, the sub-carrier information from 23kHz to 53kHz represents L-R. If the 38kHz sub-carrier is re-inserted (after being regenerated from the 19kHz pilot tone), the 23kHz to 53kHz signal can be demodulated by an AM detector to recover L-R. This signal can then be added to and subtracted from L+R to produce 2L and 2R, which are the original Left and Right channels. It is possible to build decoders which work in exactly this fashion, as opposed to time-switching (Fig 15).

RDS
A recent broadcasting innovation is the addition of RDS (Radio Data System) signals on a modulated sub-carrier, providing station identification and other information. These signals are taken from the IF output, prior to the FM detector, for processing by dedicated circuitry.

The next *'Back to Basics' article will appear in the August issue.*

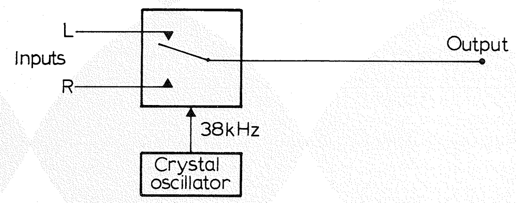

Fig 12a: the left and right channels are time-division multiplexed by switching at 38kHz

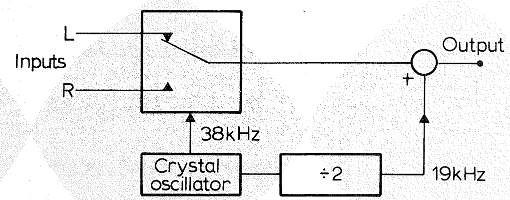

Fig 12b: before transmission, the 38kHz is divided by 2 to create a 19kHz pilot tone, which is transmitted along with the multiplexed left and right channels

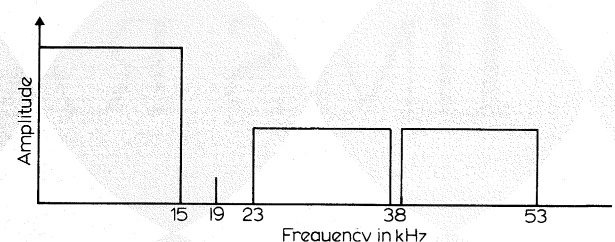

Fig 13: the spectrum of the stereo multiplexed signal looks like this. The base-band information to 15kHz is the L+R, or mono information, whereas the 23 to 53kHz information is the L-R signal. The 38kHz sub-carrier is suppressed so that the L-R signal can be transmitted at a slightly higher level for improved signal-to-noise ratio

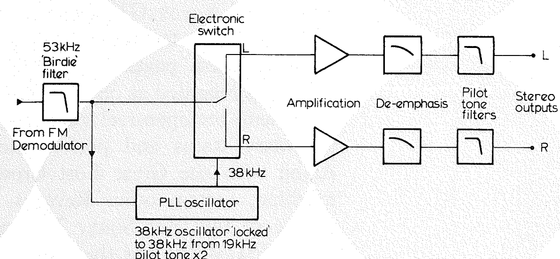

Fig 14: this form of stereo decoder works in the time domain. It takes the signal from the FM demodulator, (after signals above 53kHz have been filtered out to avoid 'birdies') and switches it at 38kHz to restore the separate left and right channels. De-emphasis restores the frequency response to flat and low-pass filters remove any 19kHz pilot tone, 38kHz or harmonics

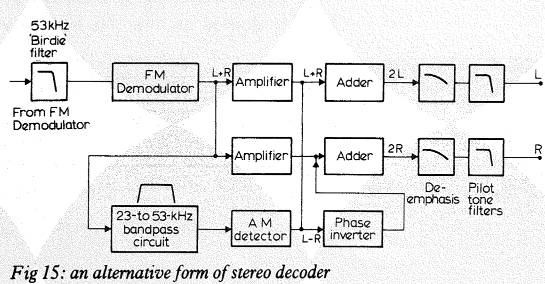

Fig 15: an alternative form of stereo decoder works in the frequency domain by selecting the 23kHz to 53kHz range, re-inserting the 38kHz sub-carrier and AM detecting the L-R signals. The L-R can then be added to (and subtracted from) the L+R baseband signal to derive 2L and 2R signals – the original stereo channels