FM RADIO: PLAYING A BETTER TUNE

John Linsley Hood examines the beginnings of FM radio design, starting with the reasons for its adoption, a comparison between AM and FM and techniques for FM demodulation

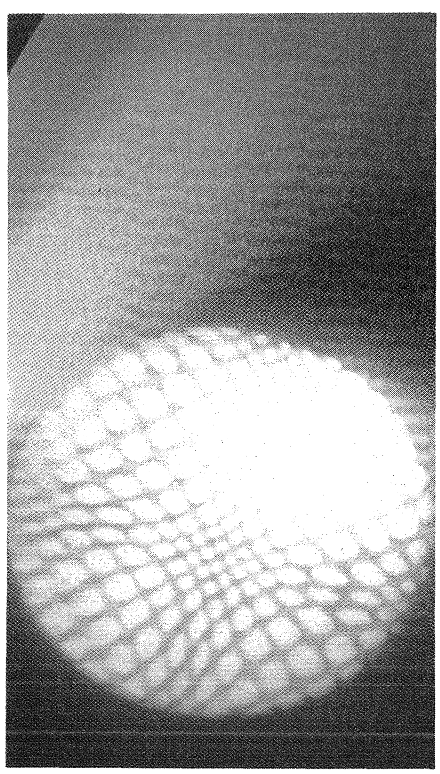
In the early years of radio transmission and reception, almost all the broadcast signals intended for domestic news and entertainment were transmitted on either the long-wave (150-285kHz) or the medium-wave (525-1605kHz) bands, because the available technology and hardware was really only suitable for use at relatively low radio frequencies.

Although each transmitted signal might occupy some 30kHz of bandwidth and the total available space in these two bands is only 1215kHz, the poor sensitivity of the average receiver and the relatively small number and low effective radiated power of the transmitters meant that there was very little likelihood of interference from broadcast transmissions on adjacent frequencies.

However, this situation could not last. National pride, and a praiseworthy desire to improve the service, led to a continuing growth in the number and power of broadcast transmitters, while the ingenuity of engineers brought forth receivers of greatly increased sensitivity. The predictable result of this was that interference from transmitters on adjacent frequencies became increasingly troublesome, particularly at night, when the reduction in density of the lower layers of the ionosphere increased the skip distance and the likely reception area of the more easily refracted LW and MW signals.

With the advent of the 1939-45 war and the desire of hostile nations to exploit the ability of radio broadcasts to encourage and inform their own populations and to discourage, mislead or simply annoy their adversaries, the situation became chaotic.

So, with the end of hostilities, the so-called Copenhagen Plan of 1948 was drawn up to try to regulate the use of the available transmission frequencies for LW and MW broadcasting in Europe. Sadly, even after this agreement, the provisions of the plan were largely ignored where they conflicted with national interests, even before the basis for the proposed frequency allocations was overtaken by events.



In particular, the military occupying powers of the three western sectors of Germany felt that the allocation of MW broadcast frequencies for their sectors was grossly inadequate and they made the sensible decision, based on experience in the USA, to build a chain of FM stations in the 88-100MHz VHF band to serve the needs of the West German population.

FM versus AM

The basic concept of modulating the frequency of a broadcast signal (FM), rather than its amplitude (AM), was due to the same remarkable Major Edwin Armstrong of the US Army Signal Corps who had conceived the idea of positive feedback (regeneration) as a way of augmenting RF signals and of providing stable RF oscillator systems, and who had also invented both the superhet and super-regenerative types of radio receiver.

FM has many advantages in use, but is much more extravagant in its need for transmission bandwidth than AM, in that while an AM transmission carrying a constant-amplitude 15kHz signal will require a transmitted bandwidth of 30kHz, (f±15kHz), an FM transmission would require a bandwidth of at least 240kHz (f±120kHz), to carry the same information without substantial constraints on modulation depth. In both cases, if the 15kHz signal is modulated in amplitude, the bandwidth requirement will be increased still further.

Clearly, FM broadcasts would be impracticable on the already overcrowded LW and MW bands, but quite feasible on the higher radio frequencies which wartime developments in components and circuitry had now made available. In this context, earlier experimental work in the USA had shown that 70-80MHz was the lowest frequency at which VHF broadcasts could be expected to be free from unwanted interference, since sporadic-E ionospheric effects brought remote transmitters within reception range.

In the light of this experience, some provisional VHF frequency allocations had been agreed, of which Band 1 (41-68MHz) had been earmarked for television use, leaving Band 2 (initially 88-100MHz, but later extended to 108MHz) available for VHF domestic radio broadcasting.

At that time, VHF broadcasting was very much a new and untried medium in Europe so, as the authority for radio broadcasting in the UK, the BBC made some preliminary trials from Alexandra Palace in 1945-46, in which the performance of AM and FM was compared at 45MHz and 90MHz, in respect of signal to background noise ratio at the receiver, and susceptibility to impulse type interference. FM transmissions were shown to be markedly superior in both of these aspects.

In spite of the results of the BBC's Alexandra Palace experiments, of which a summary was published in *Wireless World* in October 1946, there were still some misgivings among the UK engineering fraternity on the general acceptability of an FM broadcasting system, because of the need to use more elaborate and costly receiver circuitry and the practical difficulties presented by drift in domestic receiver alignment and tuning.

To help resolve these doubts, larger scale trials were made, beginning in the summer of 1949, in which the BBC Third Programme was broadcast for a few hours each evening from Wrotham in Kent, simultaneously on FM at 90.3MHz and on AM at 93.3MHz (*Wireless World*, June 1949. p. 221).

The Government's Television Advisory Committee, which had been charged with recommending a VHF sound broadcasting system, was reported in an editorial comment in *Wireless World*, December 1952, to be still divided on the question of AM vs FM. These uncertainties probably delayed the nation-wide adoption of this system although, by this time, the relative technical merits of these competing systems were quite clear.

To summarise: AM is the simpler system to use, both in transmission and reception; it uses familiar circuit technology, which makes for less expensive receivers; and it is more economical in its use of the available RF bandwidth. FM allows a significantly lower receiver background noise level, whether caused by devices and circuitry or impulse interference, since these noise sources are predominantly amplitude modulated.

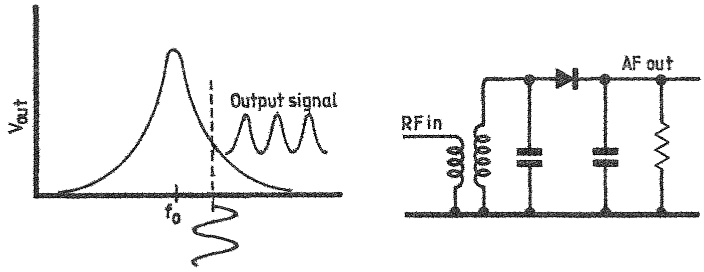


Fig. 1. Simple slope detector, in which the receiver is tuned to one side of the carrier. Considerable distortion was produced.

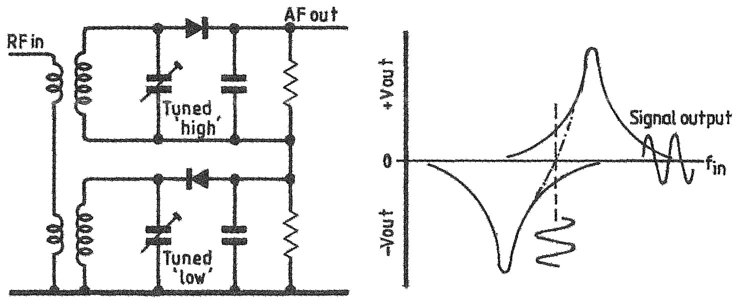


Fig. 2. Round-Travis demodulator, consisting of two of the Fig. 1 circuits connected back-to-back, to some extent cancelling distortion.

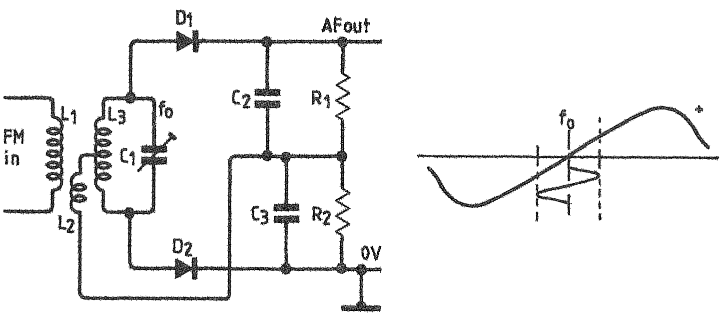


Fig. 3. Foster-Seeley phase detector, using phase shift with varying frequency in a loosely coupled tuned circuit.

FM is also largely immune to fading, particularly if an amplitude limiter circuit is used in the IF stages of the receiver, and it offers scope for lower-distortion demodulation systems. Also, with a suitable FM demodulator circuit (also known as a discriminator) a further unique benefit is conferred, in that a stronger signal can completely suppress a weaker interfering one, thereby eliminating the major existing problem in radio reception.

The necessary superiority in amplitude required from the wanted signal, to cause this to happen is called the capture ratio, which can be very small (<1dB) with a well designed demodulator system.

One must assume that the technical advantages of FM were also obvious in practical tests, since the VHF-AM broadcasts of 1949-50 were soon abandoned, while experimental broadcasts from Wrotham continued on FM only until May 1955 when work beagan on the commissioning of a national network of FM transmitters.

FM demodulator systems

Slope detection. The easiest way to convert a change in received signal frequency into a change in output voltage is simply to tune the receiver so that the received signal sits on the skirt of the response of a simple tuned circuit, as shown in **Fig. 1**. This technique, usually called slope demodulation, was widely used in early FM receivers, but the demodulated output signal would suffer from substantial, mainly even-order, harmonic distortion because of the curve of the resonance characteristic.

Round-Travis detector. If two such tuned circuits are employed, one tuned

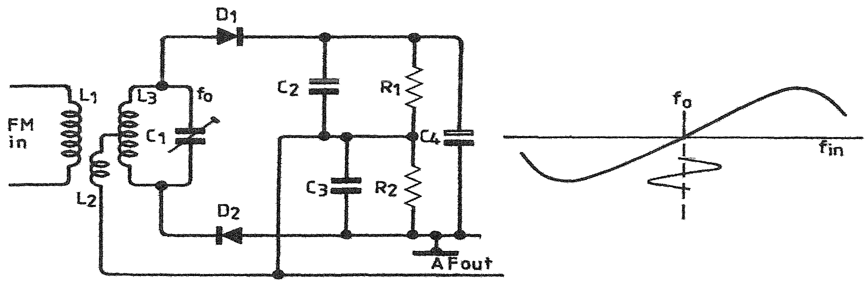


Fig. 4. Ratio detector. Since output is proportional to ratio of diode outputs, rather than their sum, AM rejection is improved

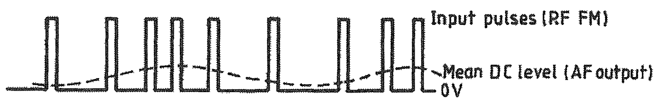


Fig. 5. Pulse-counting FM demodulator. Pulses are derived from FM signal, AF output being proportional to average level and therefore frequency.

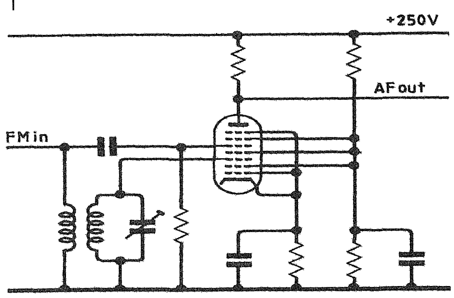


Fig. 6. Coincidence detector using the nonode–a nine-electrode valve.

above and the other below the resonant frequency, the distortion caused by the curve of the response characteristic will largely cancel out, as shown in **Fig. 2**. This arrangement is known as a Round-Travis demodulator.

Foster-Seeley discriminator. Most contemporary FM demodulator systems make use of the fact that the phase of the induced voltage on a loosely coupled tuned circuit will alter if the frequency of the induced voltage moves above or below the natural resonant frequency of the tuned circuit.

In the simple Foster-Seeley demodulator, shown in **Fig. 3**, this characteristic is combined with the further effect that the voltage developed across such a secondary tuned circuit will, when at resonance, be in quadrature (90° or 270°) to that of the input signal.

This leads to the possibility that, if the input voltage is applied to the centre tap of a tuned circuit, which is arranged to feed a pair of diode rectifiers, the input signal will either add to, or subtract from the output rectified voltage from each diode, depending on whether the input signal frequency is above or below the resonant frequency of the tuned circuit. This circuit will therefore give a voltage output which will vary with the input frequency.

However, since the output from the demodulator is the sum of the voltages developed by the diode rectifiers, such an arrangement could offer little AM rejection in the absence of some preceding amplitude limiter stage. A modification of this layout, the ratio detector shown in **Fig. 4**, in which the polarity of the diodes is reversed, was more popular in the early days of FM receiver design.

Ratio detector. In this, the output derived from the junction of C_2 and C_3 is proportional to the ratio between the two diode output voltages, which is related to the input signal frequency but not directly to its amplitude, its sensitivity to AM signals thereby being reduced.

If a large-value capacitor, C_4, is connected across C_2 and C_3, a pulse of AM interference will cause both diodes to conduct more heavily, momentarily increasing the damping on the secondary tuned circuit and lessening its output. The output voltage from the Foster-Seeley demodulator is higher, for a given input signal level, and it also has a rather lower harmonic distortion (typically 1% rather than 3%) when correctly tuned. However, because of its better AM rejection, the so-called ratio detector circuit of **Fig. 4** was more popular in early receivers.

Pulse counter systems. If the incoming signal can be converted into a stream of identical, unidirectional pulses having a narrow aspect ratio, as shown in **Fig. 5**, the average DC level of this pulse stream will be proportional to the frequency of the signal, so that a simple RC network would be able to demodulate it. Moreover, such a system should have a very linear relationship between output voltage and input frequency.

Clearly, the sensitivity of such a system based on a 90MHz carrier frequency and a ±75kHz modulation width would be far too low, at a maximum average DC level change of 0.17%. By the use of a superhet system, the centre carrier frequency could be reduced to around 500kHz and the modulation sensitivity would be increased to a usable 33%. Several receiver circuits using pulse-counter techniques were proposed, such as that due to Scroggie[1].

Gate-coincidence detector. As noted above, there is a change in the phase of the voltage induced in a loosely coupled tuned circuit, as a function of its input frequency. The direct use of this effect had been explored in a demodulator in the earlier years of FM by causing it to modulate the anode current through a multi-grid valve, but the requirement for a special type of valve, a nonode, had discouraged its use. **Fig. 6** shows a typical circuit for such a system, described by Amos[2].

With the evolution of multiple transistor arrays, this type of system became much easier to employ and the gate-coincidence detector shown in outline in **Fig. 7a** is now the most widely used demodulator system in FM receivers. In this, the phase shift in the output voltage E_2 of the tuned circuit L_1C_1, on either side of its resonant frequency, is used to generate a differential voltage output by modulating the current stream through a group of transistors arranged in ladder form.

In the absence of an output voltage E_2 from the quadrature coil L_1, or if this is truly in quadrature with the incoming signal E_1, the currents through R_1 and R_2 are identical and there is no differential output voltage. If the phase of E_2 alters in relation to E_1 as a result of a change in E_1 frequency, the equal division of the currents through R_1 and R_2 is disturbed and there is a voltage output.

With the simple quadrature coil system shown in Fig. 7a, the non-linearity of the demodulator circuit can be as low as 0.5%, but figures of 1-2% for ±75kHz deviation are more common. Elaborating the quadrature coil circuit as shown in Fig. 7b is claimed to allow THD figures as low as 0.1% for the same modulation depth, although this will depend critically upon the phase/frequency linearity of the preceding frequency-selective circuitry.

Such gate-coincidence demodulator systems are normally fabricated on a single IC, along with a high-sensitivity IF amplifier/limiter circuit and features such as off-station noise muting, outputs for automatic gain control, automatic frequency control and signal-strength indication. **Figure 8** shows a typical contemporary IC system, that used in the RCA CA3189.

Phase-locked loop demodulator. A technically interesting, though seldom used, method of demodulating an FM signal is that in which a phase-locked loop, of the kind shown in **Fig. 9**, is used to force a linear voltage-controlled oscillator into frequency synchronism with the incoming signal. If the relationship between the DC control voltage for the VCO and its output frequency is truly linear, then the variations in the control voltage of this oscillator are an accurate replica of the modulation of the input RF signal.

Also, so long as the VCO remains in lock, its control voltage, which provides the AF output, is only related to the frequency of the incoming signal and is completely independent of its amplitude. This gives a very high degree of AM rejection, as well as removing a secondary source of signal distortion.

This technique has always attracted me and I described an early receiver using this demodulation method[3]. The basic problem in using a PLL demodulator is that, on the edges of the signal-capture band, the audible noise generated by a mis-tuned FM signal as it swings into and out of the PLL capture range is very high and effective out-of-band muting systems are needed. Nevertheless, within these limitations, a PLL demodulator can offer high linearity, high sensitivity and excellent S:N and capture ratios.

Early FM receiver designs

With the adoption of an entirely new broadcasting system, at least so far as the UK was concerned, engineers needed to become familiar with the circuit techniques used in RF amplifiers, oscillators and frequency changers operating at frequencies in the 88-108MHz range and also with high-gain IF circuitry operating at the now-conventional value of 10.7MHz. Achieving adequate gain and stability was, in both cases, much more difficult than at the lower frequencies with which they were familiar.

Considerable thanks are due to the BBC and those of its engineers working in this field for the circuit designs and accompanying explanatory information which they published during the early years of this period, such as those due to Spencer[4,5] and Amos and Johnson[6,7,8] of the BBC Engineering Training Department. A typical high-sensitivity FM tuner design of the mid 1950s, using a pentode RF stage, a pentode additive mixer with separate oscillator and two 10.7MHz IF stages driving a balanced ratio detector was also described by Amos and Johnstone[9,10].

I built a receiver using this circuit, at the time, for my parents. However, I had not paid sufficient attention to ventilation so that, in spite of my best efforts to establish optimum values for the temperature compensation capacitors, it continued to suffer from frequency drift in the tuning setting, first in one direction and then in the other, due to the differential warming of the various components in the circuit during use. This gave point to some of the earlier reservations which had been expressed by engineers about the likely problems with stability of tuning when VHF broadcasting was first mooted.

A circuit which was somewhat more economical of components, and was therefore perhaps rather more typical of contemporary commercial design, was also described at this time by Hampson[11] of Mullard Ltd. This used a basically similar circuit layout, though with a self-oscillating pentode mixer, and is shown in **Fig. 10.**

Another successful FM tuner circuit published at this time was the very popular Jason design, described in Radio Constructor in 1955, which used a double-triode cascode RF stage rather than a pentode, which gave an improved input s/n ratio. A later design, due to Spencer[11] of the BBC Research Dept, was noteworthy in that it showed a complete FM radio circuit, using a Foster-Seeley demodulator preceded by a dynamic limiter stage, in which the component complement was reduced to the four valves plus rectifier typical of existing domestic AM radio sets.

However, in general, with all valve-operated tuner designs, stability of tuning, particularly over the first few minutes following switch-on, always left something

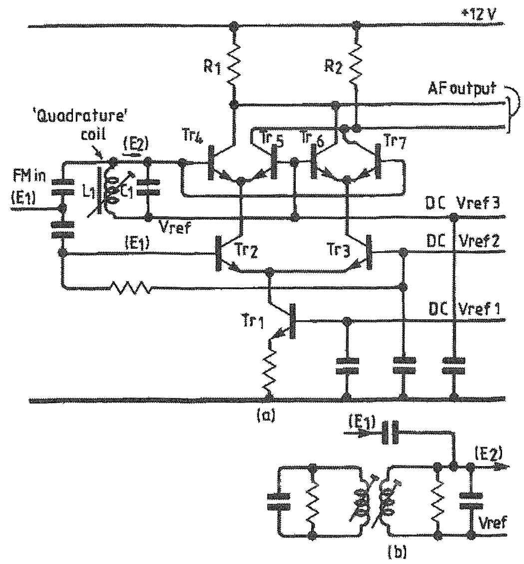
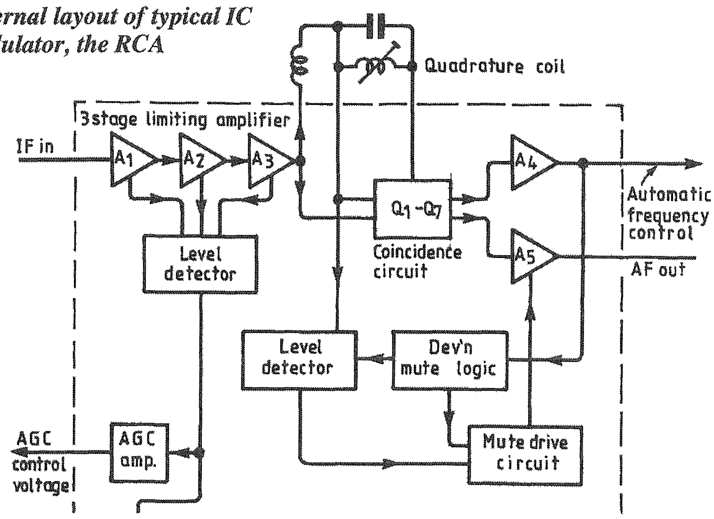


Fig. 7. Modern IC gate-coincidence demodulator, the most common type now in use. Use of a parasitic tuned circuit coupled to the quadrature coil (b) improves linearity.

Fig. 8. Internal layout of typical IC FM demodulator, the RCA CA3189.



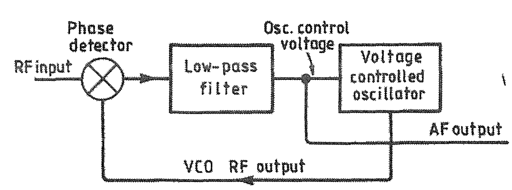


Fig. 9. Bolock diagram of simple phase-locked loop, where VCO control voltage forces oscillator into step with input signal, control voltage being AF output, assuming linear control voltage/frequency relationship.

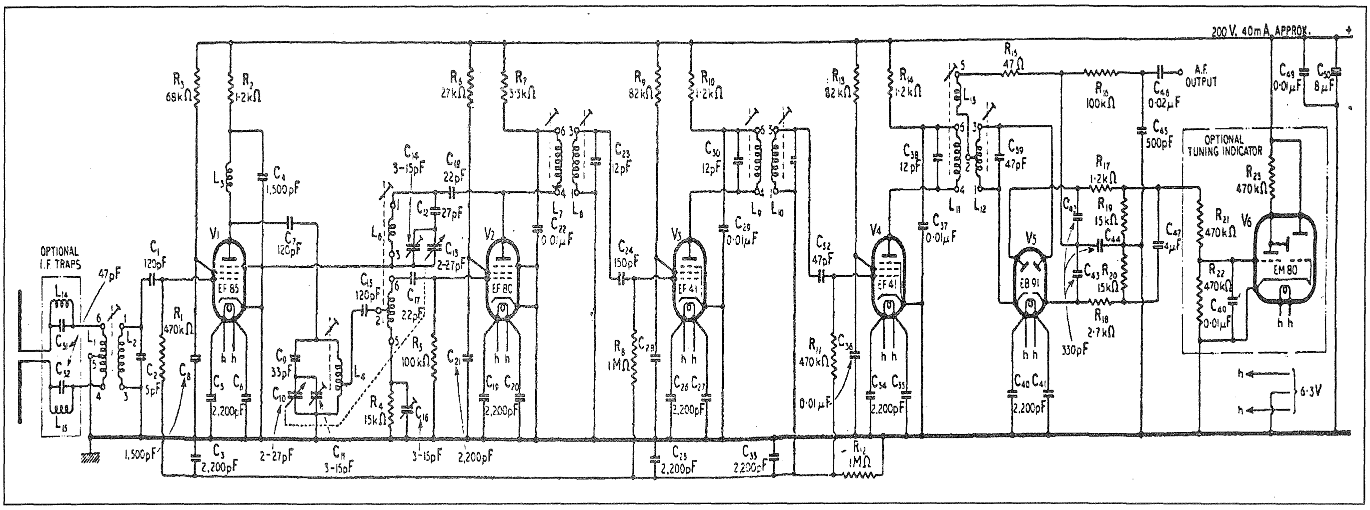


Fig. 10. FM receiver circuit by L. Hampson, published in this journal in 1955.

to be desired; for example, the quoted drift for the Spencer design, which was typical of its genre, was 38kHz during the first eight minutes. The advent of cool-running transistor circuitry offered great hopes for improved performance in this respect.

Transistor FM tuner designs

Because of the difficulty in making reliable transistors with physically thin base regions using simple diffusion systems, most early transistors had a relatively poor HF performance, so that while devices were available, such as the Mullard OC170 which could be used in 10.7MHz IF amplifier stages, there were none which could be used as RF or mixer stage components.

An early solid state design which avoided this difficulty was described in 1960 by Harvey[13,14], also of the BBC Research Dept. A balanced-diode mixer was coupled directly to the aerial circuit, as shown in **Fig. 11**, without any preceding RF amplifier stage. Oscillator radiation into the aerial circuit was minimised by the mixer design.

Phase/frequency characteristics of tuned circuits

The linearity of the demodulation of an FM signal by a gate coincidence type of detector of the kind shown in **Fig. A** depends largely on the linearity of the relationship between the phase of the input signal and the voltage developed across the loosely coupled quadrature coil which provides the other input to the circuit.

An ideal frequency/phase relationship for this application would be as I have shown in line m in **Fig. B** Unfortunately, in real life, a more typical phase/frequency curve would be as I have shown in line n, for which I have also shown the related voltage/frequency response in **Fig. B** Such a sharply tuned circuit will give a phase-shift response which is too abrupt and which also has an S-shaped characteristic, which would give rise to third-harmonic distortion in the recovered audio signal.

Adding a damping resistor across the quadrature coil, as shown in Fig. A2 will make both the voltage and phase/frequency curves less steep, as shown in line r in Figs. B and C and will thereby improve the linearity of the demodulator, but at the price of reducing its signal voltage output.

An alternative approach, frequently recommended for this application, is to add a secondary coil as in a band-pass coupled tuned circuit, as shown in Fig. A3. However, potential users are warned not to employ this technique unless they have access to adequate instrumentation, such as an FM oscillator or distortion meter, since if the secondary tuned circuit is incorrectly tuned, the effect of the added tuned circuit may disappear.

Moreover, if the bandpass coil is over-coupled, the resultant double peak in the tuned circuit response can put a kink into the phase response, as shown in line s, which would cause some very unpleasant audible effects. However, if the tuning, the degree of damping and the coupling coefficient are all correct, a much more linear response can be obtained, as shown by the response curve t.

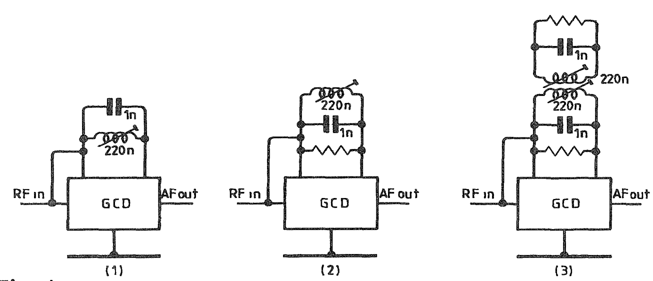


Fig. A

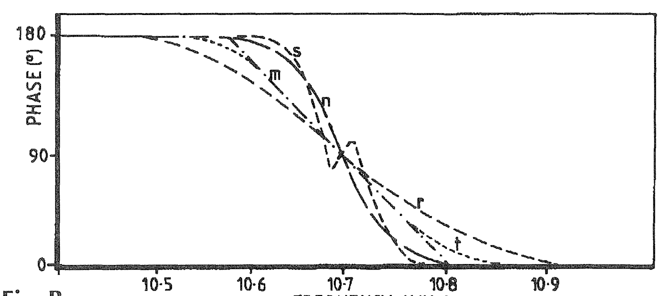


Fig. B

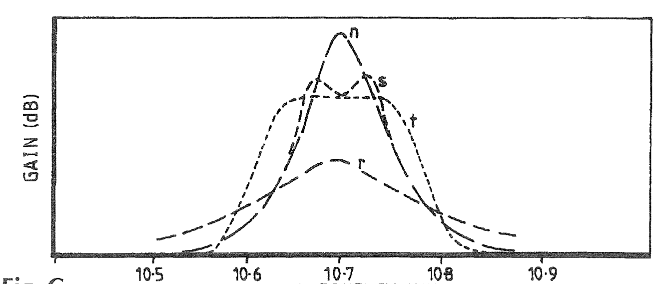
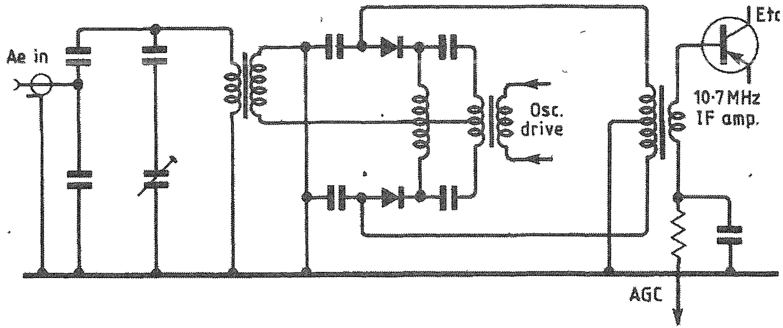


Fig. C

Fig. 11. Since, in 1960, no VHF transistors were readily obtainable, this circuit for a mixer by R.V.Harvey of the BBC was coupled directly to the aerial.



This circuit used the same low-distortion dynamic-limiter/Foster-Seeley discriminator combination used by Spencer. No figures were quoted for frequency drift, but the author described it as "quite small".

Predictably, the evolution of transistor and integrated-circuit technology led to a period of rapid change in FM receiver design during the later 1960s and 1970s before this type of circuitry began to settle down to a fairly standard layout, based on well proven component types and application systems. I will explore this period in a further article. ■

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FM RADIO: PLAYING A BETTER TUNE

In the last part of his series, John Linsley Hood describes the evolution of stereo broadcasting and the influence of higher-quality sound on requirements for studio-transmitter links. He also examines frequency synthesis in local oscillators and RDS

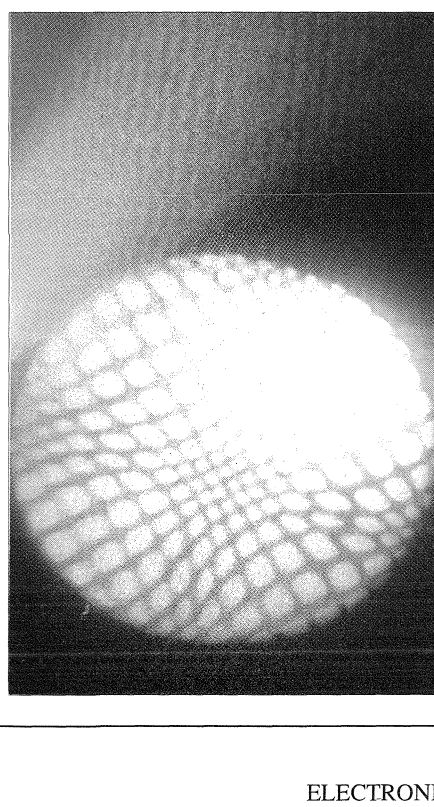
Historically, the BBC has shown a continuing interest in the possibility of transmitting stereo radio signals since before 1957 and broadcast experimental programmes on a fortnightly basis from 1958, one channel being transmitted on VHF FM and the other using a standard medium-wave AM broadcast transmitter.

Meanwhile, the search for a stereo broadcast system which could be employed with existing VHF FM transmissions was being pursued with enthusiasm in the USA and a number of alternatives had been proposed. Six of these possible competing methods were tested by the Federal Communications Commission in the USA in 1961. The final choice was for an amalgamation of pilot-tone systems proposed by Zenith and GE, now defined as CCIR recommendation 450/1970, and broadcasting in the USA, using this system, was authorised by the FCC in June 1961. (See the *WW* editorial comment of the same date).

System requirements

The basic need was for a method of transmission on a single carrier of a stereo signal, having a +0-3dB audio bandwidth of 40Hz-15kHz, which would be received by an existing monophonic receiver as a normal L+R mono signal, but as a stereo signal by a receiver equipped with a decoder. It should not significantly degrade the existing FM transmissions received by normal mono FM sets.

In the Zenith/GE pilot-tone system[1], this was accomplished by the method shown in **Fig. 1**. The existing L+R signal was broadcast to a maximum modulation level of 90% of the permitted 75kHz deviation, together with a L-R signal, which was transmitted as a double sideband AM signal, with an equivalent 45% peak modulation level. The transmission was centred on a suppressed 38kHz sub-carrier which could be regenerated from a continuously present 19kHz pilot tone, broadcast at a 10% modulation level.



This composite signal had the equivalent modulation depth of the existing 40Hz-15kHz mono transmission, where 100% was equivalent to a modulation of 75kHz.

Stereo decoding methods

In principle, all that is needed to receive the stereo signal is to use the 19kHz pilot tone to generate a suitable-amplitude 38kHz carrier waveform, from which the double-sideband L-R supersonic signal, which occupies the 23-53kHz part of the audio spectrum, can be resolved into the required L-R audio output. The L+R and L-R signals can then be converted into the left and right audio channels by the simple matrix system, also referred to as a frequency-division multiplex decoder, shown in **Fig. 2**.

A practical decoder of this type was shown by Browne[2], and illustrated in **Fig. 3**, in which both the 38kHz carrier regeneration and the matrix addition were accomplished simultaneously.

Unfortunately, this simple circuit does not offer reverse compatibility in which a mono signal can be received in the absence of the stereo pilot tone. This drawback could be removed by the simple expedient of making V_2 self-oscillatory, although this would slightly degrade the mono s/n ratio.

This method of stereo signal decoding was the major technique used during the 1960s, and formed the basis for the bulk of the stereo decoder ICs designed during this period, such as the Motorola MC1304, MC1305 and MC1307 and similar devices from the other major semiconductor manufacturers.

A survey of the various possible techniques for decoding the Zenith/GE pilot-tone stereo signal was made by *Wireless World* in September 1966, (pp. 445-448), shortly before the BBC began making test transmissions using this system. This survey drew attention to the possibility of using a simple 38kHz channel-switching (time-multiplex) system, of the kind shown in **Fig. 4**, as an effective method of decoding the composite signal. There were some inherent problems in this technique, as discussed later.

A channel-switching decoder, using the circuit shown in **Fig. 5**, was described by Waddington[3] in 1967. He used a pair of silicon planar transistors in a shunt-mode chopper circuit, driven by a 38kHz signal derived from the 19kHz pilot tone by way of a frequency multiplier and phase-splitter circuit.

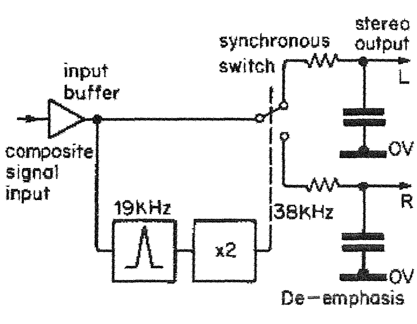
Although Waddington's circuit provided automatic (stereo-mono) reverse compatibility, several difficulties still remained with the 38kHz switching technique, of which the most immediately obvious was a 6dB fall in mean signal level when a stereo signal was received, due to the chopping action of the switching circuit.

A further problem was that the correct matrix addition of the L+R and L-R components of the composite signal in the time-multiplex mode would only be obtained if the reconstructed 38kHz switching signal had the correct phase relationship to the other modulation components. This was critically dependent on the adjustment of the tuned circuits in the frequency multiplier chain; if these were incorrectly tuned, the stereo separation would be greatly impaired.

Both of these problems were solved by the very elegant phase-locked-loop stereo decoder shown in **Fig.6**, described by Portus and Haywood[4]. The PLL circuit regenerated an accurate 38kHz square wave, locked in phase to the 19kHz pilot tone, so that the maximum practicable stereo separation could be ensured without the need for very accurate alignment of a tuned-circuit frequency-multiplier chain.

Equality of gain between mono and stereo operation and true reverse compatibility was automatically ensured by the

Fig. 4. Basic channel-switching, time-division multiplex decoder, switching at 38kHz.



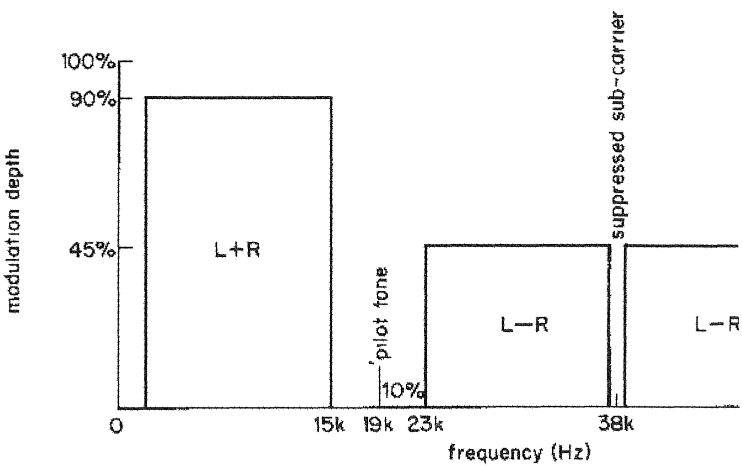


Fig. 1. Zenith-GE composite stereo modulation characteristic, producing ty with existing mono receivers.

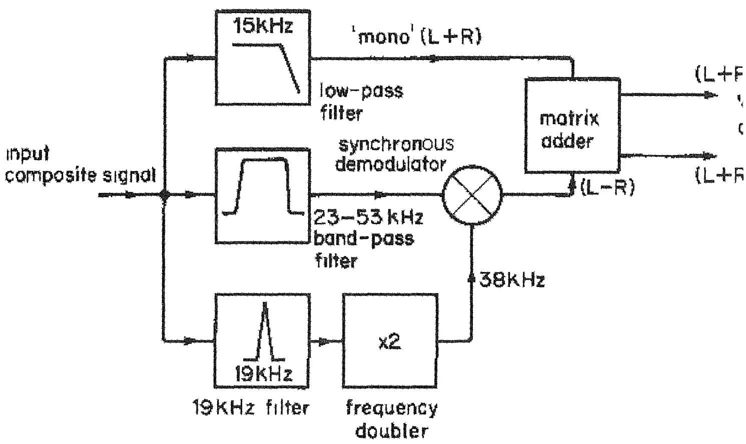


Fig. 2. Stereo decoder using frequency-division multiplex system. Chann was critically dependent on tuned-circuit alignment.

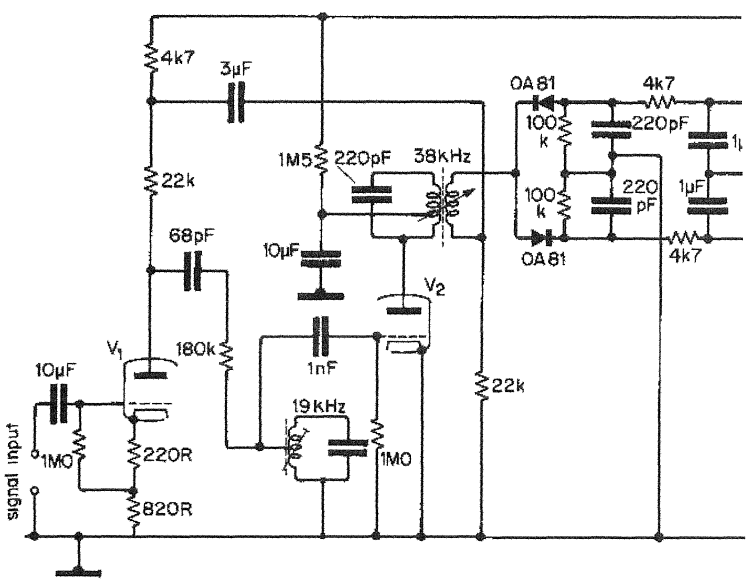


Fig. 3. Practical decoder of the Fig.2 type from 1962, in which carrier reg and matrixing were simultaneous. Mono was not received without pilot to

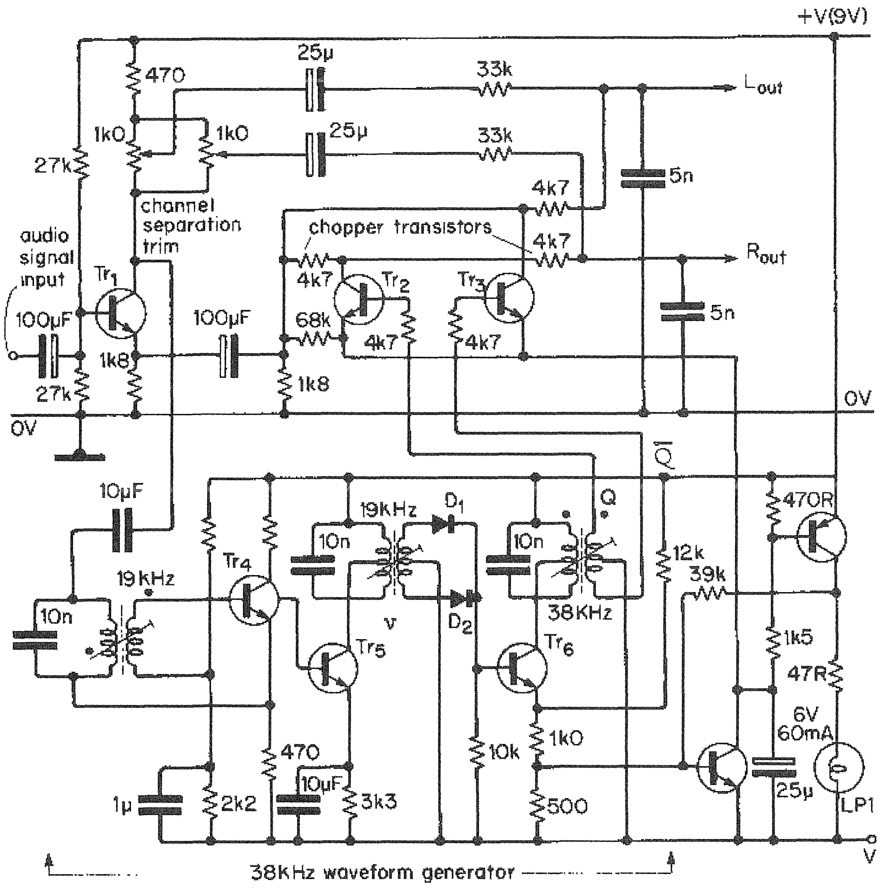


Fig. 5. Channel-switching decoder by Damer Waddington, published in 1967, which was compatible with mono signals. Phase of switching signal was critical.

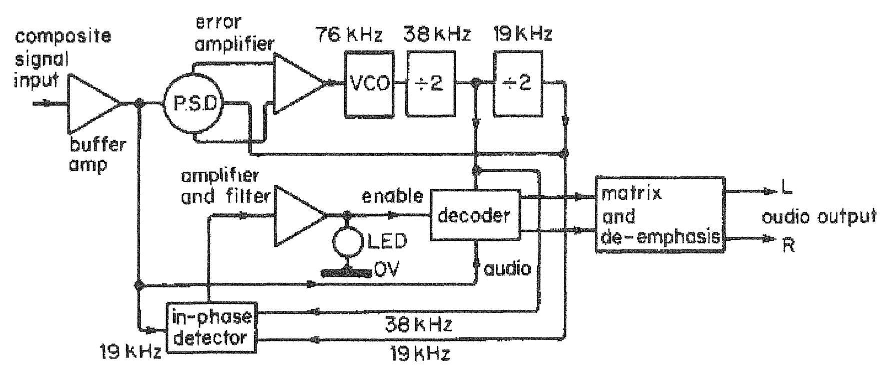


Fig. 6. Portus and Hayward phase-locked-loop decoder of 1970, which overcame both phasing problem and inequality of stereo/mono gain, suffered by circuit of Fig. 5.

use of the long-tailed pair switching circuit, shown in **Fig. 7**. A major tribute to the technical superiority and innovative quality of this design was paid by the semiconductor manufacturers, who promptly copied the design philosophy and introduced monolithic ICs, starting with the CA3090 in 1971[6] and, as a more exact replica, in the MC1310, LM1310, CA1310 and so on in 1972[6].

Channel separation and distortion specifications for BBC and IBA transmitters, and for various decoder systems, fed from an ideal signal input, were quoted by Brook[7,8].

Although some minor improvements have been made in the circuitry used in this IC system, it is undoubtedly true that the majority of contemporary stereo FM tuners use the 1310 type of decoder. Its only drawback is that the 38kHz square-wave switching waveform, which is rich in odd-order harmonics, will also demodulate wide-band noise or adjacent-channel signal components based on its 3rd and 5th harmonic frequencies if they are present in its input signal; this may degrade the overall stereo s/n ratio.

The problem of the unwanted demodulation of signals near the third harmonics of both the 19kHz pilot tone and the regenerated 38kHz sub-carrier has been addressed in the LM4500A. The basic PLL oscillator operates at 228kHz and the 19kHz and 38kHz waveforms derived from this by way of a three-stage Johnson counter are free of any second or third harmonics, or their multiples. Sansui, in its TU-D99X receiver, uses a Walsh function generator to synthesise a pseudo-sinusoidal 38kHz demodulator waveform to reduce harmonic-related interference from ultrasonic and adjacent-channel signals. An explanation of this technique is given by Thomas[9].

Typical channel separation levels in excess of 45dB are claimed for most of the commercial PLL decoder ICs — a performance which could only be obtained using the frequency-multiplex systems of the type shown in Fig. 2 if their tuned circuits were very carefully aligned.

However, it should also be remembered that for this order of channel separation to be obtained in practice, with any of these decoder types, it is necessary that the relative amplitude and phase of the L+R and L-R modulation components should be correct in the input signal. This requirement places considerable demands on the gain and phase characteristics of the audio stages preceding the decoder.

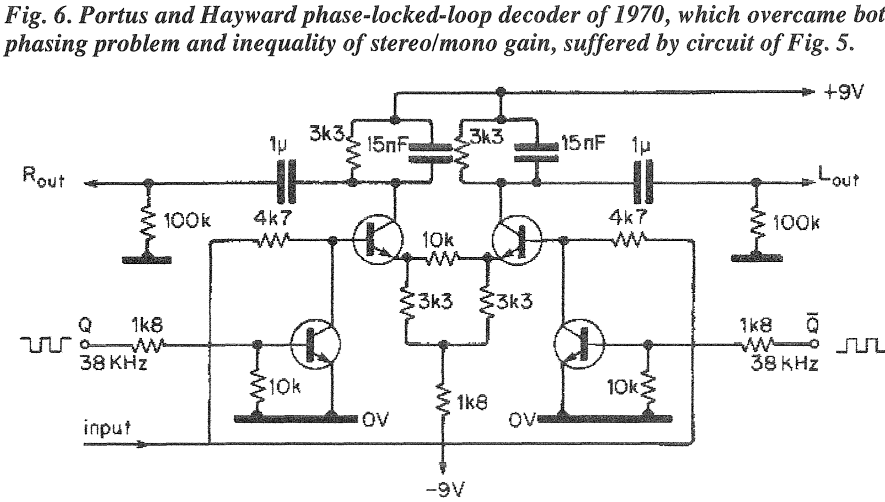


Fig. 7. Long-tailed pair used in Portus and Hayward design was responsible for elimination of gain inequality.

PCM distribution

From the inception of the BBC broadcasting service it had been customary for the BBC to rely on high-quality telephone-line links to carry the programme signals from the studio to the transmitter; in these early days, the transmitter was likely to be fairly close to the studios and a close relationship had grown between the BBC and the GPO for the maintenance of this service.

However, the bandwidth of even these high-quality links was only some 50Hz-10kHz and the proposed audio bandwidth of 40Hz - 15kHz for the new FM service could not be guaranteed, particularly for lines serving some of the projected, more remote transmitter locations. Moreover, with the new stereo broadcasting service, precise time coherence between the L and R signal channels would be essential, since any relative time delays would alter the apparent stereo image position.

The BBC decided to make use of the existing 6.5MHz-bandwidth television transmission network and to encode the audio signals in digital form. This approach was similar to that subsequently adopted by Philips in their Compact Disc recording system, but at a 13-bit rather than 16-bit resolution level, and with a 32kHz sampling rate instead of 44.1kHz.

This sampling rate imposed an absolute upper limit of 16kHz on the audio passband so, to allow a practicable low-pass filter slope, the broadcast signal bandwidth was amended to 40Hz-14.5kHz (0.2dB). The BBC 13-channel PCM encoding system is shown schematically in **Fig. 8**.

The way this works can be explained most easily by considering the path of a single signal channel, in which the signal first passes through a 15kHz low-pass filter with a very high attenuation rate, followed by the HF pre-emphasis network — 50µs in the UK and Europe.

Effective low-pass input filtering is essential in any digital encoding system because the presence of any signal components at a frequency above half the 32kHz sampling rate would create problems of aliasing, in that signals above 16kHz would be reproduced identically to those at the same frequency interval below 16kHz.

An intrinsic characteristic of the PCM system is that, after the final digital-to-analogue decoding process, the recovered waveform has a staircase-type structure in which the relative size of each individual step is determined by the sample resolution. To reduce the extent of this granularity distortion, which becomes more significant as the amplitude of the encoded signal becomes smaller, the overall input signal level to the encoder should be as high as possible.

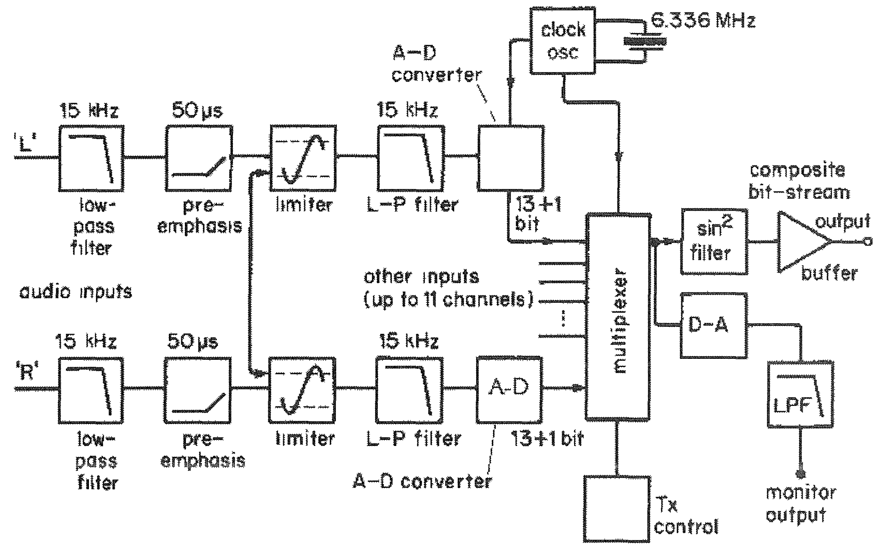


Fig. 8. BBC 13-channel pulse-code modulation system for programme distribution.

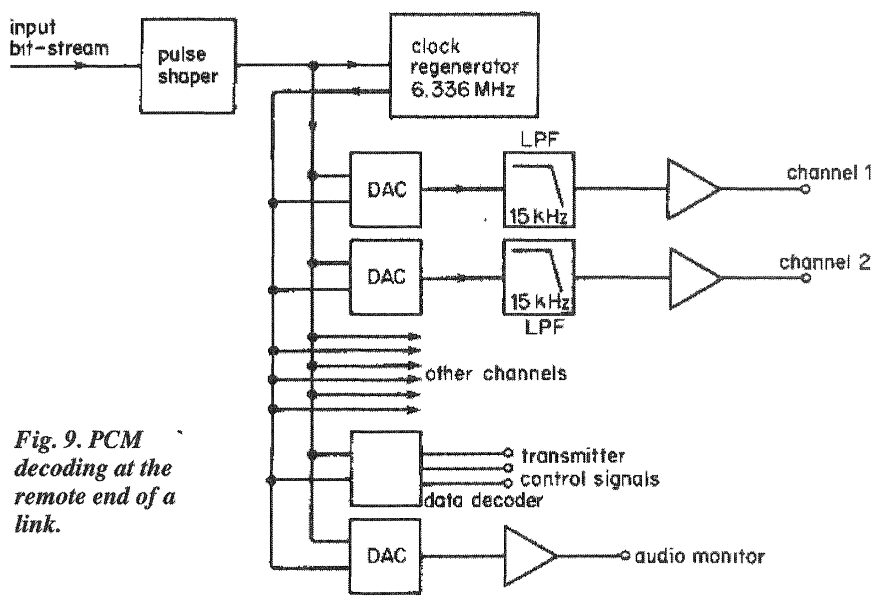


Fig. 9. PCM decoding at the remote end of a link.

On the other hand, it is essential that this A-to-D encoder should not be overloaded, and also that the signal should not be subjected to hard clipping, since both of these would produce audibly unpleasant effects. The BBC therefore use delay-line type limiter circuits, in which the signal is delayed for long enough for an appropriate and gradual reduction in gain level to be applied. These limiters are arranged to have an absolute maximum output level of 2dB above the nominal peak programme level.

An ingenious feature of these limiters is that their actions are linked, so that if the peak output level is exceeded in one half of a stereo channel, the other channel is also limited to prevent any sudden shift in the position of the stereo image.

Since the action of peak limiting can itself introduce harmonic components, the signal encounters a further 15kHz low-pass filter before passing to the A-to-D encoder. A clocked, double-ramp-type converter transforms the amplitude and bandwidth-limited signal into a digitally encoded pulse train which is fed, along with the bit streams from up to twelve other channels, to a time-domain multiplexing circuit.

The output pulses from this are fed through a sine-squared filter, which greatly reduces harmonics beyond the second, giving a final output of a rounded-off pulse of 158ns duration at 158ns intervals. Since there is little harmonic output above the second, there is a negligible energy

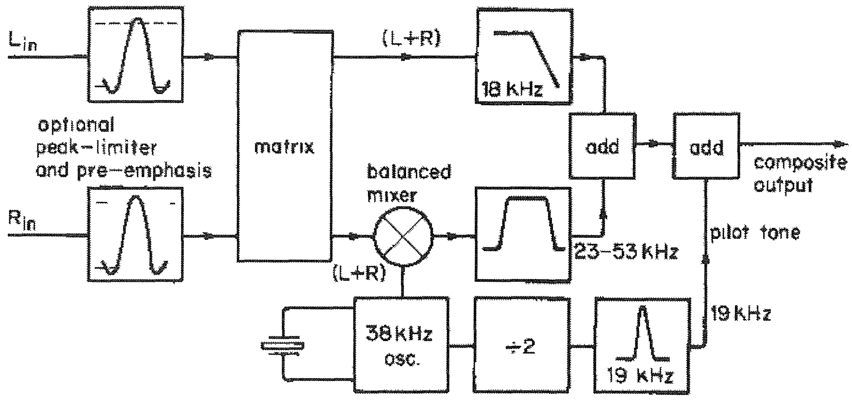


Fig. 10. Zenith-GE encoding system, one of which is required for each programme at each transmitter. 40dB carrier suppression is obtained.

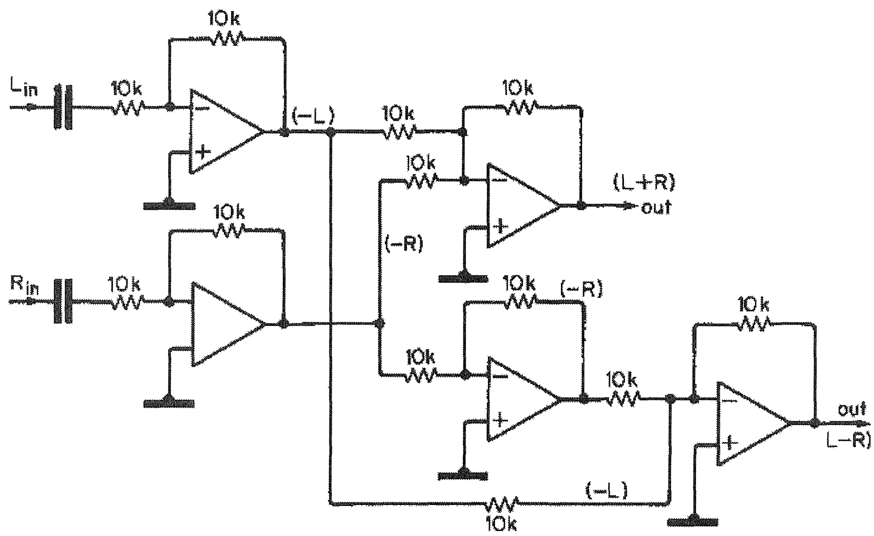


Fig. 11. Matrix circuit used in arrangement of Fig.10 to generate L+R and L-R signals.

content above 6.33MHz, which allows the composite signal to be carried by a 625-line TV channel.

As in the Philips Compact Disc system, a parity bit is added to each preceding 13-bit word. This provides a check on the accuracy of encoding of the preceding five most significant digits, so that if an error is detected, the faulty word can be rejected, and the preceding 13-bit word substituted.

If the error persists, the signal output is muted until a correctly encoded signal is again received. This technique gives a high degree of immunity to noise and transmission-line errors.

With a 32kHz sampling rate, the 6.336MHz channel bandwidth allows a group of 198 bits in each sample period. This contains 13 multiplexed 14-bit channels (182 bits) and 16 spare bits. Of these, 11 are used to control the multiplexing matrix and four are employed to carry transmitter remote-control instructions.

Decoding system. At the receiving end, the incoming bit stream is cleaned up, decoded using the circuit layout sketched in **Fig. 9**, and fed to the stereo encoding system, described below.

Stereo encoding systems. The technique employed in the Zenith/GE stereo encoding system is shown in **Fig. 10**. In this, the input left and right stereo channels are passed through a matrix circuit, such as that shown in **Fig. 11**, to generate a pair of L+R and L-R signals. In principle, an encoder of this type is required for each programme channel at each transmitter location.

The L+R (mono) signal, occupying the 40Hz-15kHz part of the audio spectrum, is then filtered and added to the double-sideband L-R modulated output of a balanced mixer, fed from a 38kHz crystal controlled oscillator, which gives rise to the 23-53kHz part of the modulation spectrum. If the mixer is accurately balanced, the 38kHz carrier component will be largely absent from the mixer output — 40dB suppression is typical.

Signal from the 38kHz quartz oscillator is then divided in frequency by two, filtered and phase corrected to provide the required sinusoidal 19kHz pilot tone, which is then added to the signal to allow the suppressed 38kHz carrier waveform to be regenerated within the decoder circuit.

Additional carrier signals

In the USA, supplementary sideband components have been added to the FM stereo signal for some years. They are called SCA (Subsidiary Communication Authorisation) or Storecast, and consist of signals transmitted as a 10% level double-sideband modulation based on a 67kHz sub-carrier. These signals are usually relatively low-quality continuous broadcasts intended to provide background music for restaurants and supermarkets; stereo FM receivers designed for use in the USA need care in decoder design to prevent inadvertently demodulated SCA signals from interfering with the wanted programme.

In Europe, an additional low-level, (3% modulation) HF signal is now added as a carrier for Radio Data System (RDS) broadcasts, which provide time, station identification and programme and road traffic information. This is transmitted as a phase-shift keyed, 7.5kHz bandwidth modulation of a 57kHz sub-carrier, initially locked in quadrature to the pilot-tone third harmonic to avoid interference with other 57kHz sub-carrier modulation.

Data is transmitted in 16-bit words at 1187.5bit/s and allows a variety of supplementary data to be broadcast. A full explanation of this system and its potential has been given by Shute[10], and commercial RDS decoders are now available as DIY add-on components for existing FM receivers. They usually require effective screening to avoid interference with the audio output signals.

Frequency synthesiser systems

Domestic users of FM receivers demand some means of accurate, preset, push-button station selection. This required accurate and stable tuning mechanisms — better, probably, than could be obtained from the existing Varicap-diode voltage-controlled tuning systems — and has encouraged the development of relatively low-cost, IC-based frequency-synthesiser techniques, using variations of the method outlined in **Fig.12**.

In its simplest form, the outputs of both the local oscillator X in the receiver frequency changer (which is voltage controlled because of the varicap tuning sys-

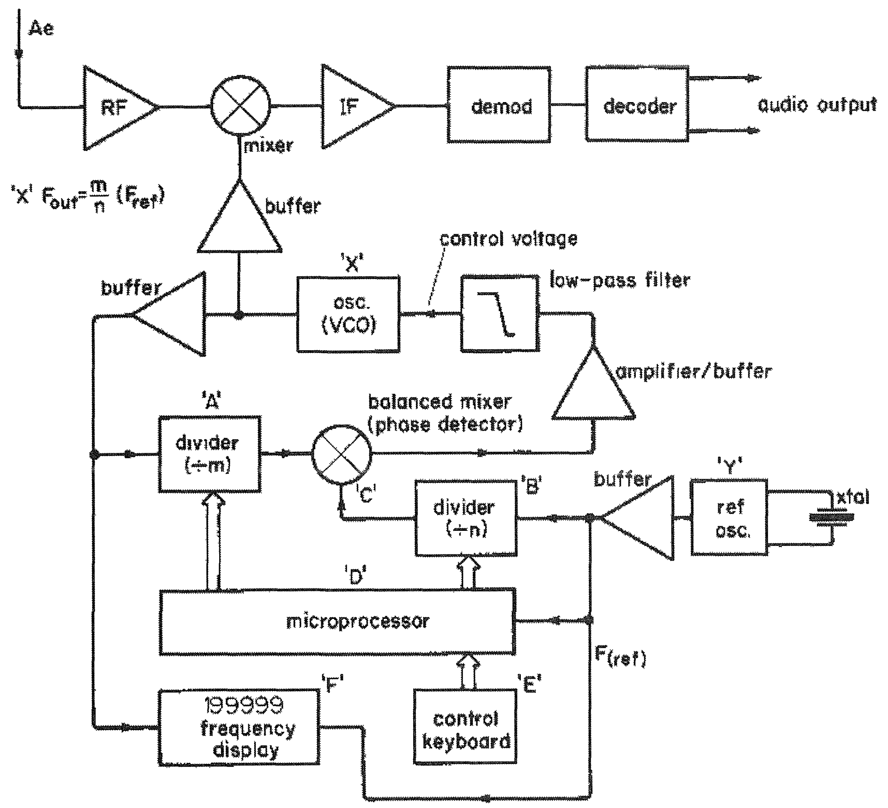


Fig. 12. Phase-locked-loop frequency synthesiser provides simple and accurate tuning for domestic users.

tem) and a highly stable, quartz crystal controlled oscillator Y are fed through a pair of frequency dividers A and B to a phase detector C. The output from this, after amplification and filtering, is then used to control the frequency of the local oscillator in a phase-locked loop, so that the output frequency is precisely held to mF_{ref}/n.

Precise values of the division ratios m and n are controlled by a microprocessor D, in response to the input commands from a push button station selector E; the tuned frequency is also displayed by a frequency counter system F. It is common practice for all the function blocks A-F to be combined in a single IC.

Requirements of the frequency dividers are simplified somewhat by the fact that, by international agreement, the operating frequencies of all VHF FM transmitters within the 88-108MHz band are held to exact multiples of 100kHz.

The elaboration of the receiver system will depend on its price bracket, but it is fairly common for current models to offer switched options for tuner selectivity, and also for stereo/mono blend to optimise both selectivity and signal-to-noise ratio. Receivers also incorporate a battery back-up system for preserving the microprocessor memory, so that the tuning data for a given channel can be stored during switch-off.

With the growing availability of RDS information, it is practicable for details of both the station selected and the programme being received, to be displayed on an information panel.

It is also feasible for the tuner to select automatically either the type of programme material required — speech, drama, pop or classical music — or the transmitter giving the best signal strength for that programme. This will be of particular value to car radio users travelling through the reception areas of local transmitters.

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