Radio Reception Developments

New techniques for communications systems described at I.E.R.E. conference

Technical development in radio receivers has been extremely slow, compared with that in, say, electronic computers and control systems. The superheterodyne was invented in 1919 and is still the basis of the majority of receivers; likewise, such techniques as synchronous detection, single-sideband operation multiplexing and diversity reception, which are periodically brought out of the cupboard, dusted and re-applied, have been known for many decades. The only technical changes which have broken new ground are not in basic systems at all but those which have been forced or made desirable by the demand for more channels, i.e. the gradual move into higher and higher frequency bands, and the availability of new active devices – the replacement of the valve by the transistor and the transistor by the integrated circuit. One reason for this slowness of development is that, if one includes sound and television broadcasting, for every one transmitter there are thousands of receivers – an enormous, widely spread investment which places an economic restraint on changes of transmission systems. If the system must remain in a particular form for many years there is obviously not much scope for the receiver designer.

In radio telephones and radio communication systems, however, considered as a separate class, there is much less of a disparity between the numbers of receivers and transmitters, and consequently any change of transmission system – for example, between types of modulation in mobile radio – is not such a drastic matter. It was perhaps for that reason that a recent conference on Radio Receivers and Associated Systems, organized by the I.E.R.E. in association with the I.E.E. and I.E.E.E. at the University College of Swansea (4th-6th July), was largely devoted to developments in radio communication – as distinct from, say, broadcasting or radio navigation. Thirty-three papers were presented, in five sessions, and the following notes are a selection from these.

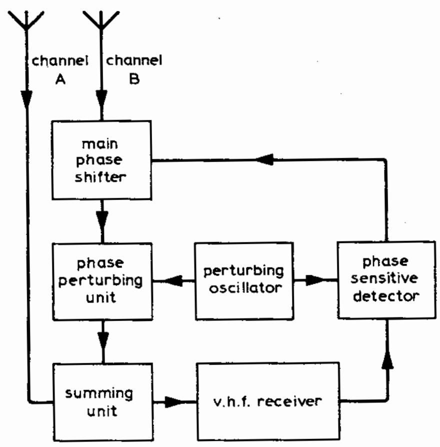

Fig. 1. Basic principle of self-phasing aerial system for space diversity reception in mobile radio. (All three channels are not shown.)

Diversity reception for mobile radio. Birmingham University's Department of Electronic and Electrical Engineering has been working on spatial diversity reception systems for aircraft and land vehicles for some years. At the conference five of its members presented a group of papers on automatic methods of compensating for the rapid fading (hundreds of fades per second, upwards) due to multipath reception which occurs in v.h.f. mobile radio links. Propagation at v.h.f. is principally by way of scattering, and the multiple reflection and diffraction from buildings and other obstacles cause standing-wave patterns to be set up, through which the mobile receiver moves. Amplitude variations of typically 20dB are experienced at the receiver (see Fig. 2). Since the distance between adjacent maxima and minima in the standing-wave pattern lies in the region of 0.5m to 1m it becomes feasible to mount an array of spaced aerials on the vehicle and combine additively the separate signals in such a way that the fading in the summed signal will be diminished.

The Birmingham schemes are based on automatic adjustment of the phases of the signals from the separate aerials; and one paper, for example, by J. D. Parsons and P. A. Ratcliff, described a three-channel "adaptive" system, using this principle, which was built around a conventional v.h.f./a.m. receiver. Fig.1 shows the basic components of the method, not including, for simplicity, all three channels. One aerial signal (channel A) is considered as a reference signal and is connected directly to a summing unit. The other (channel B) is passed through a phase-shifter capable of changing the signal phase through 2π

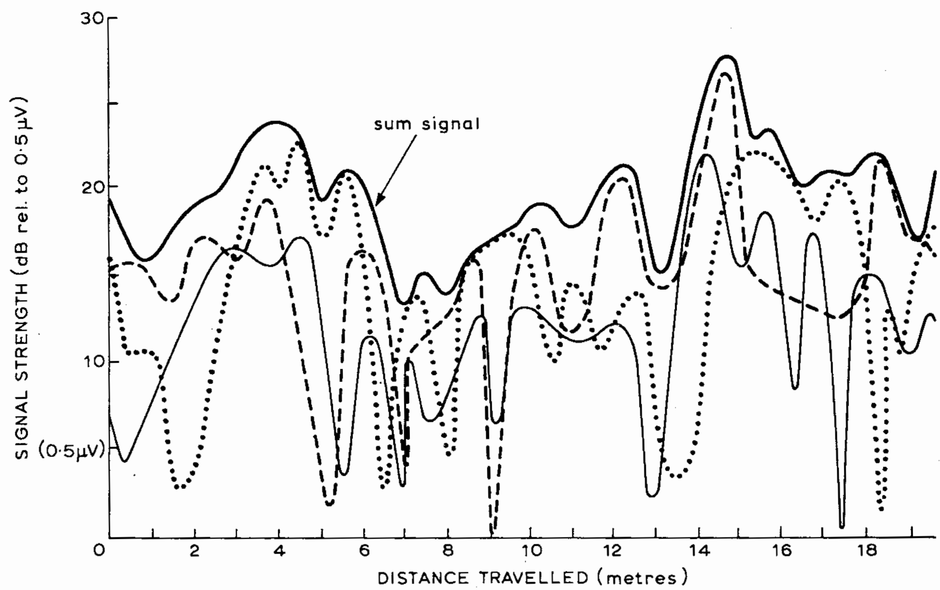

Fig. 2. Recordings of signal amplitudes in three aerials mounted at 0.4 wavelength spacing on a van, travelling through central Birmingham. The sum signal is given by the three-channel self-phasing aerial system shown schematically in Fig. 1.

radians. The phase-shifter output is passed to a phase-perturbing unit, a small-deviation phase modulator, and provides phase-difference information which is used to set the main phase shifter. The phase-perturbed signal then enters the summing unit, the output of which feeds the receiver.

Amplitude modulation at the perturbing frequency is produced in the sum signal. This modulation is in phase with the perturbing frequency when signal B lags signal A and the sum signal, and in anti-phase when signal B leads signal A and the sum signal. When signal B is in phase with signal A only a small amplitude second harmonic of the perturbing frequency is produced. An a.m. error signal is extracted from the detector of the receiver and phase detected with respect to the phase-perturbing oscillator. The polarity of the filtered d.c. output thus obtained indicates the direction in which the main-phase shifter must be changed to co-phase the signals A and B. This basic method is extended to three channels, a separate control loop being provided for each aerial and a separate perturbing frequency for each channel.

Results obtained in a severe "multi-path" area in the centre of Birmingham are shown in Fig.2. The individual aerial signal strengths are plotted in broken and thin lines, with variations as high as 27dB, but the maximum variation of the sum signal is only 15dB and its fading rate is greatly reduced.

Another diversity system includes a new technique for phase measurement. Described by J. D. Parsons and M. Henze, the system has not yet been tried out but a prototype is being built for use in the 450-470MHz mobile radio band. The phase measuring technique, based on single-sideband modulation with carrier, is used to detect the phase-difference between each of the received signals and their sum; and control signals are generated which adjust quantized phase-shifters in such a way that the signals are put approximately in phase. The system includes a single, standard radio receiver capable of detecting either amplitude-modulated or frequency-modulated signals. There is no need for modification to the receiver circuitry.

The third diversity system from Birmingham University, described by M. J. Withers, is based on a principle similar to that of a technique used already in radar. It utilizes a phase shifter which is swept continuously and therefore causes two independently received signals to come into phase periodically. By making the periodicity short compared with the modulation it is possible to satisfy the sampling theorem of information theory and extract the original modulation. A phase modulator, in the signal path from one aerial, repeatedly changes the signal phase in steps of $\pi/4$ from 0 to 2π. The combined signals are fed into a conventional v.h.f./a.m. receiver and the original modulation is retrieved by an envelope detector. Mobile tests have not yet been made but Mr. Withers said that an "on-the-air" trial at a fixed location showed no degradation of speech quality. If the system is successful in mobile tests its main advantage will be that a basic area coverage receiver can be used with a minimum of modification and the addition of an extra aerial, a combining "hybrid" and a phase modulator.

Selective calling for mobile radio. Moving into higher and higher frequency bands is not the only way of meeting the demand for extra communications channels: there is the reduction of channel spacing (e.g. in mobile radio from 50kHz to 25kHz and now to 12.5kHz) and the sharing of particular channels. The last expedient is now commonplace in mobile radio, and though simple to achieve, it has the serious practical disadvantage of causing any one "mobile" to listen to an almost continuous stream of speech, only a small proportion of which is directed at him. This problem has been investigated over the past few years by the introduction of various systems of selective calling. In these, by the use of signalling codes, a particular "mobile" can be called while the remainder are kept muted. There are single-tone and multi-tone systems, and the multi-tone systems can be subdivided into simultaneous and sequential systems. J. E. Philips and E. J. Slevin, of G.E.C. Mobile Radio (part of Marconi Communications Systems Ltd), described a five-tone sequential system which they claimed "would appear to provide the optimum solution for present requirements" — optimum, that is, in its balance of price and calling capacity.

The tones available — chosen from a C.C.I.R. marine specification so that the system can be used for ships – are ten in number, between 1124 and 2100Hz. To call a "mobile" from the base station any five of these tones are transmitted in sequence, so that in all 10^5 tone combinations, or codes, are available. The duration of each tone, determined by the hardware consideration of the Q of filter coils in a decoder at the receiving end, is 40 milliseconds.

In the mobile station, the audio signal from the receiver is fed via the decoder to the loudspeaker. The h.t. for the decoder is taken from the transmitter/receiver supplies and so no internal connections to the set need be made. Since the set is disconnected from the speaker except when the "mobile" is called no speech and/or noise is heard. Thus the required setting of the volume control is not known. To overcome this an h.t. switch has been included which connects the decoder to 9V and 5V supply rails when the incoming noise and/or signal level is above a predetermined threshold, which may be reached by adjustment of the volume control. The 5V rail is then used to light a lamp on the front panel, indicating to the operator that the volume control of the receiver is adjusted correctly.

The first tone received merely operates the h.t. switch. The second tone, forming the first of the actual calling code, is fed to a limiter which enables the decoder to work over a large dynamic range (this may be as much as 25dB since at low s/n levels the receiver's a.g.c. system is inoperative). From the limiter the tone signal is fed to a tuned circuit (filter), the frequency of which is predetermined by the particular code of the "mobile". The code to which each decoder responds is determined by a wire matrix which selects various tappings on a coil to give the various tuning frequencies. These tappings are selected in the required order by sequencing logic. The first code tone, on entering the filter, will cause the tuned circuit of the filter to resonate because the circuit is programmed to tune this frequency. Once the filter response has built up above a certain level an integrator begins to operate and its output, upon reaching the trigger level of a Schmitt trigger, will actuate it. This Schmitt trigger then operates a sequencing network which selects the next tap of the coil – determined by the wire matrix code – which causes the tuned circuit to tune to the second code tone, and so on, until all five tones have been decoded. After the last tone has been decoded a relay is actuated; and this causes a lamp to be lit on the front of the decoder unit and connects the audio signal from the receiver to the loudspeaker.

The authors demonstrated a selective calling equipment working on this principle and stated that the decoder to be carried by the "mobile" would eventually be manufactured as a 2in × 1in thick-film circuit.

Improving s/n in f.m. demodulation. In a communications f.m. receiver, when the noise accompanying the carrier at the receiver input is large enough for the so-called threshold region* to be entered, the envelope of the carrier-plus-noise combination shows a significant number of dips. These are experienced as sharp noise peaks, heard as clicks, in the receiver output. J. H. Roberts, of Plessey Avionics and Communications, described several

"Threshold" is defined as the point, on the graph of input carrier/noise ratio vs. output signal/noise ratio, at which the output signal/noise ratio departs by 1dB from its linear relationship with the input carrier/noise ratio.

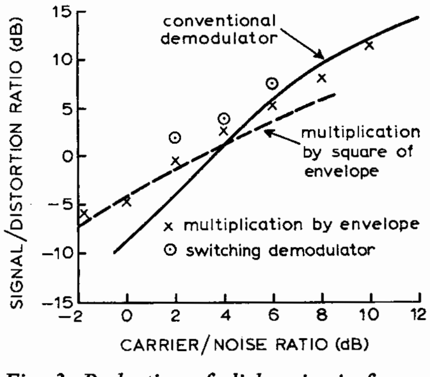

Fig. 3. Reduction of click noise in f.m. reception by multiplying the base-band output by a function of the i.f. output. The graphs of signal/distortion ratio vs. carrier/noise ratio are for different multiplying functions.

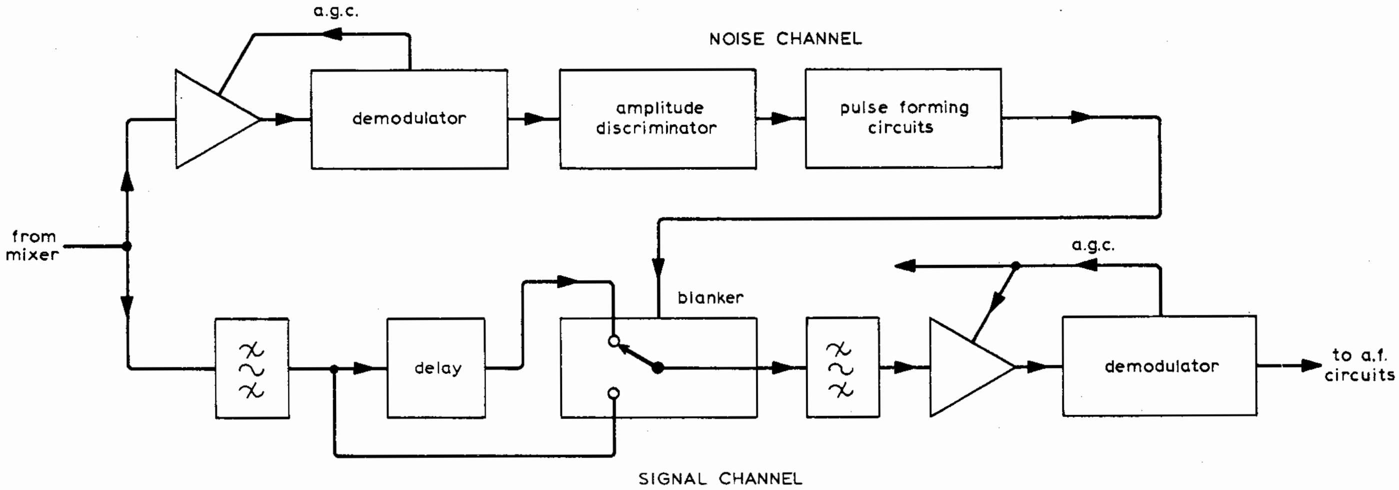

Fig. 4. Impulse noise blanker fed from mixer of a.m. receiver. In one channel each noise impulse initiates a blanking pulse which is applied to the "signal" channel to reduce its gain at the appropriate time.

methods by which this effect could be mitigated, all based on the idea of multiplying the base-band output of the f.m. receiver by a function of the envelope of the i.f. input. This has the effect of reducing the intensity of the click noise. In fact the click noise is exchanged for other types of base-band interference, but an overall improvement in signal/distortion ratio is said to be obtained at carrier/noise ratios well below the threshold point in the receiver. The practical situation in which Mr. Roberts was interested was with low carrier/noise ratios of -2dB to $+6$dB.

The envelope multiplication was tried out by means of computer simulation and some of the results are presented in Fig 3, which is for the top end (1MHz) of a 240-channel f.m. radio system with a base-band range of 60-1052kHz. Generally, there is improvement in signal/distortion ratio at low values of carrier/noise ratio but degradation at high values. Multiplication by the envelope function seems to be superior to multiplication by the square of the envelope. The best performance (circled dots) is obtained by what is described as a switching demodulator. In this there is a continual switching between conventional demodulation (solid line) and multiplication by the square of the envelope (broken line) – each switch to multiplication being made when the instantaneous level of the envelope falls below a specified threshold and a noise click is likely to occur.

Reducing impulse noise. Much of the noise which affects a.m. radio communication is impulsive in character, notably that generated by the ignition systems of motor vehicles. One method of countering this is by the use of an amplitude limiter, or "clipper", but an alternative approach at present being investigated is the use of a "blanker" – a circuit which reduces the receiver gain to a very low value for the duration of the noise pulse. As an example, Professor W. Gosling described the work that is being done at the University College of Swansea on a.m. and s.s.b. receivers, particularly that of analysing the process of blanking and deciding on the best blanking function (pulse shape) to use. The basic technique is to take the signal from the receiver's mixer and split it into two channels (Fig. 4), a "noise" channel, in which the noise impulses are detected and used to initiate blanking action, and a "signal" channel. The signal channel is conventional except that it includes a blanking circuit and also that a delay line is inserted to ensure that the noise pulse is given sufficient time to bring the blanking circuit into operation *before* the noise pulse arrives at it.

A trapezoidal blanking function has been chosen, in preference to a rectangular function, for a variety of reasons; for example, the Fourier components, which may fall within the passband of the receiver, decline more rapidly, and there is less drastic "ringing" of the i.f. filter which follows the blanking circuit. Trapezoidal blanking almost eliminates high-frequency ringing (about 3kHz), but a lower frequency ring remains, and will typically worsen the s/n ratio by 3-4dB.

Fundamentally this difficulty arises because the narrow band i.f. filter through which the received signal must pass cannot be designed both to give the required adjacent channel selectivity and also to have a good response to transients, whether noise pulses or blanked breaks in the received signal. The blanker reduces the amplitude of the transient to equal the instantaneous amplitude of the received signal, and this is its advantage over a receiver not equipped with noise reduction circuits, but a serious transient still remains. A possible approach to this problem is to reduce the magnitude of the blanking transient, for example by holding the signal amplitude constant during the blanking period rather than reducing it to zero. Since blanking operations must be performed at i.f., and at a signal level of only tens of microvolts, circuits which will perform this function adequately present serious difficulties.

One solution under investigation at Swansea is shown in Fig. 4. Here the i.f. is 5.2MHz, the pre-blanker i.f. filter has a bandwidth of 100kHz while the post-blanker filter is a conventional s.s.b. crystal i.f. filter with a passband of 2.5kHz. The necessary delay is provided by a 64μs ultrasonic glass delay line. The received signal normally passes through the broad-band delay line, but when a noise pulse is detected the delay line is switched out of circuit and the signal passes directly to the subsequent receiver circuits. Provided the delay is short compared with the shortest modulation period, there will be little change in the signal amplitude; hence the transient superimposed on the incoming signal will be much reduced.

Automated frequency changes for h.f. Readers who follow the regular feature H.F. Predictions in this journal will be well aware that the operating frequencies of h.f. radio communications systems have to be changed throughout the day in order to take advantage of the changing ionospheric propagation conditions along different routes. This normally entails a good deal of repetitive and time consuming manual operation, but the Post Office radio telegraph station at Somerton, Somerset, has introduced an automated system for changing the frequencies of receivers which relieves staff of this work, reduces human errors and allows the operator to concentrate on fault localization and clearance. The engineer in charge of the station, Mr. L. Wilks, described the system.

The basic method adopted is to use the distant-end transmitter to control the receiver frequency changes within a basic schedule provided by a 60-position drum-type timer making one revolution in 24 hours. This timing schedule is set by cams inserted in the drum and these operate switches which are arranged to give a "time slot" of 48 minutes during which a frequency change to a receiver can be made. To allow for transmitter and/or receiver frequency variations, the receiver performs a frequency searching operation ($\pm$500Hz at a speed of 40Hz per second) after making a frequency change. The receiver reverts to its normal a.f.c. locked condition when a signal of correct keying speed and acceptable telegraph distortion is recognized.