# Single sideband modulation for v.h.f. private mobile radio

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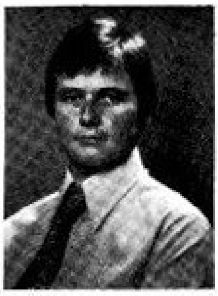
**Summary** In view of the increasing popularity of v.h.f. mobile radio communication, heavy congestion of presently available frequency allocations is becoming a major problem in some areas. The adoption of single sideband modulation with narrower radio channel spacing has been proposed as a possible solution to this problem.

This article briefly examines the implications of such a solution, and compares the main features of single sideband modulation with those of other popular modulation processes.

The implementation of a single sideband system is discussed with particular reference to recent studies on this subject.



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## Introduction

The present channel spacing employed by private mobile radio in Great Britain is 12·5 kHz. The ever increasing demand for channel allocation at private mobile radio frequencies will eventually create the need to reduce this spacing significantly without seriously increasing adjacent channel interference or adding to the complexity of the equipment used. Single sideband (s.s.b.) modulation appears to offer a solution to this problem, although the relative advantages and disadvantages of s.s.b. over other modulation systems need adequate consideration before any conclusions may be drawn.

## S.S.B. and other linear modulation systems

S.S.B. modulation is a 'linear' process where the amplitude of the modulated output varies with the amplitude of the modulating input. The simplest form of linear modulation is conventional amplitude modulation (a.m.). From a theoretical viewpoint, s.s.b. appears to have a substantial advantage over a.m. Removal of the carrier and one of the two identical sidebands which constitute an a.m. radio signal would allow an idealized transmitter to radiate eight times more 'usable power' using single sideband suppressed carrier (s.s.b.s.c.) than could be radiated by a.m. with the same peak power. Considering only thermal noise, s.s.b.s.c. can therfore give a 9dB signal-to-noise ratio improvement over a.m. operating at the same peak power. Morever, this improvement is achieved using half the required a.m. transmission bandwidth. For speech transmission, where the mean signal power is very much lower than the peak power, the use of suppressed or diminished carrier techniques can lead to an additional improvement of up to 4dB in the mean-to-peak power ratio of the output signal (see figure 1). The mean power of a speech signal is approximately 10dB lower than the peak power, and the speech modulated d.s.b. (or s.s.b.) signal directly preserves this relationship. In conventional a.m. with 100% modulation, the mean signal power is only 6dB lower than the peak, since the average amplitude is about half the peak amplitude. Thus, in terms of mean power, s.s.b. modulation has a 13dB (9dB + 4dB) advantage over conventional a.m. for speech transmission.

The improved performance of this system is paid for by the increased complexity of its components. S.S.B.S.C. demodulation requires the reinsertion of a carrier at the receiver, and any difference between the true carrier frequency and the reinserted (local) carrier frequency will appear as an equal frequency translation of each Fourier component of the receiver audio output. Speech remains reasonably intelligible with offsets up to ±150 Hz, although for offsets greater than ±50 Hz, the naturalness of speech is entirely lost. Phase coherence is not necessary for speech in s.s.b. The frequency stability of a crystal oscillator of the type used in most mobile radios is usually stated to be ±5 parts per million, or ±500 Hz at 100 MHz (v.h.f.), but, for speech transmission in a private mobile radio environment, it would be reasonable to

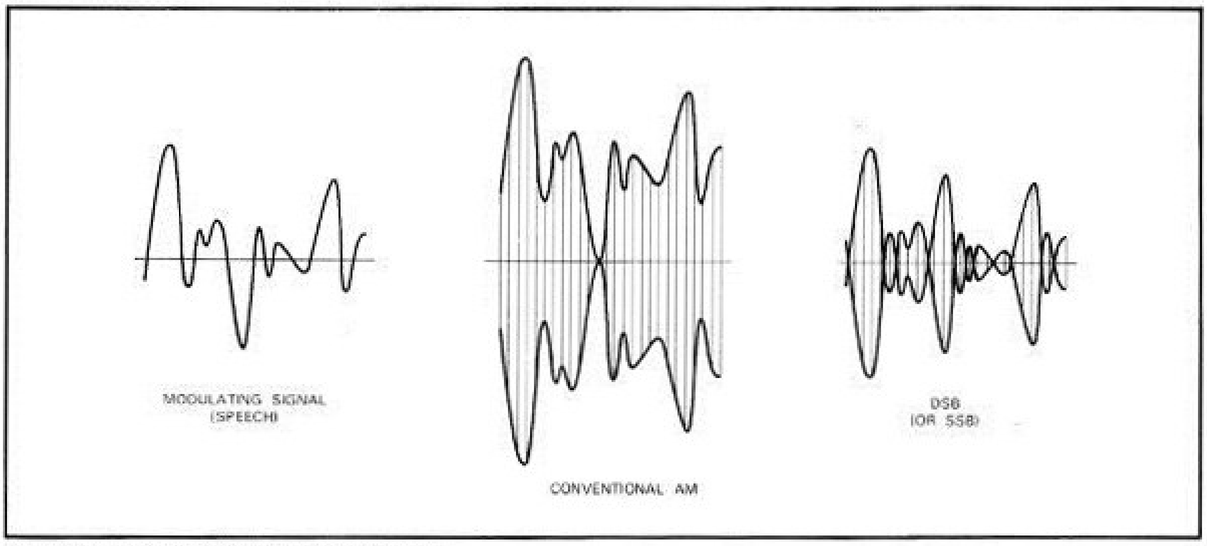


*Fig. 1. Comparison of speech modulation levels*

specify that the reinserted carrier frequency should not differ from the true carrier frequency by more than ±25 Hz. Since it is likely that future equipment will employ frequency synthesizers having only one reference crystal oscillator, the use of high stability temperature compensated crystals is possible. However, the cost of such oscillators with a specified frequency stability of ±0·25 parts per million (±25Hz at 100 MHz) could be unacceptable to the mobile radio market.

Since the information in the s.s.b. signal is conveyed by its amplitude, an automatic gain control (a.g.c.) is required to overcome fluctuations of the input signal strength. Conventional a.m. systems have an amplitude reference in the carrier component, upon which the a.g.c. can be based. S.S.B.S.C. systems only transmit radio frequencies with the actual speech syllables, and not in the gaps between words. Consequently, if the a.g.c. is based upon the mean signal amplitude (as in a.m.) the audio signal will be compressed, and the interword gaps will be filled with loud receiver noise since the a.g.c. gain is then at a maximum. Co-channel interference is also accentuated by the increased interword gain. In h.f. receivers these problems are overcome by using a fast attack, slow decay type of a.g.c. which holds the gain low for a fixed time between words, but this method is not suitable for v.h.f. where multipath propagation causes a very rapid fading which the a.g.c. cannot follow.

One answer to both frequency stability and a.g.c. problems is the transmission of a continuous pilot signal with the s.s.b. signal. The pilot signal takes the form of a diminished carrier, or a pilot tone inserted in the audio spectrum, and is used as a frequency and amplitude reference at the receiver. This slightly reduces the useful output power of the transmitter since some 'wasted' power is radiated at the pilot frequency.

A third method of linear modulation is double sideband (d.s.b.) modulation, a technique which generates the two sidebands of the full a.m. signal without the carrier component. A d.s.b. signal occupies the same transmission bandwidth as the equivalent a.m. signal, but can provide up to 10dB improvement in signal-to-noise ratio for a given mean power output over a.m. with speech modulation. D.S.B. systems suffer from similar frequency stability and a.g.c. problems to those of s.s.b., although the effects of frequency drifts at the receiver are more severe in d.s.b. than in s.s.b. since frequency translations occur in both sidebands. D.S.B. demodulation also requires strict phase coherence between transmitted and local carriers, even for speech transmission, since phase drifts in d.s.b. modulation appear as fading effects in the receiver audio output. In s.s.b. systems, phase drifts appear as delay distortion which is acceptable up to fairly high levels in speech transmission. The necessary local carrier reference can be extracted from a d.s.b. signal using the symmetrical property of the sidebands, although this can prove to be difficult with multi-transmitter co-channel operation, so both s.s.b. and d.s.b. transmissions might benefit from the inclusion of a pilot reference signal.

S.S.B. systems are occasionally subject to a 'sideband capture' effect where the local oscillator, locked to a low level pilot carrier, is 'pulled' out of synchronization by the introduction of a large sideband level, although it is possible to minimize the effect by careful choice of phase locked loop parameters. D.S.B. systems, having sidebands symmetrically arranged about the carrier should not experience this effect.

Other linear modulation methods, such as vestigial sideband (v.s.b.), independent sideband (i.s.b.), and quadrature amplitude modulation (q.a.m.), are not usually considered suitable for use in the private mobile radio environment because of the cost or complexity of their implementation.

**Exponential modulation systems**

Exponential modulation processes vary the frequency or phase of a constant amplitude carrier in relation to the amplitude of the modulating signal. The two basic methods of exponential modulation are therefore known simply as frequency modulation (f.m.) and phase modulation (p.m.). It is generally accepted that narrow band frequency modulation (n.b.f.m.) at 12·5 kHz channel spacing gives little or no improvement over a.m. Signal-to-noise ratios are of a similar order in the two cases, although for low level signals, a.m. exhibits a more graceful deterioration of audio output into noise as the signal is reduced.

N.B.F.M. signals contain large carrier components which can cause severe carrier interaction problems in multi-transmitter co-channel operation. At 25 kHz channel spacing, f.m. is undoubtedly superior to a.m., but such generous channel allocations are no longer practicable at private mobile radio frequencies. However, f.m. systems have one important advantage over linear systems at 25 kHz or 12·5 kHz in that they do not require linear r.f. power amplifiers. The linear amplifiers presently employed in s.s.b. at h.f. often have d.c. to r.f. conversion efficiencies as low as 15% and any non-linearities cause spurious emissions in the output. Class C power amplifiers of the type used in most f.m. transmitters have conversion efficiencies of 60% to 70% so that, for a given power output, the battery drain of an s.s.b. transmitter would be about four times that of an f.m. transmitter. This increase in power consumption could be an important factor in the design of hand-held transceivers powered by dry-cell batteries, although, since the transmit duty cycle of mobile radios is normally expected to be 5% to 10% of their total operating time, the effect might not be significant.

P.M., whilst having certain advantages in the field of data transmission (phase shift keying), is subject to the same limitations as f.m. It should be noted that n.b.f.m. signals in mobile radio are normally produced by the indirect f.m. method, where the input signal is integrated and then passed through a narrow band phase modulator. The resulting signal can be shown mathematically to be equivalent to the frequency modulated version of the input signal. Although p.m. is used successfully in mobile radio at 12·5 kHz spacing, adaptation of the technique for use in narrower channels is unlikely to yield acceptable results.

It has been suggested that more efficient management of the private mobile radio frequency allocations, particularly incorporating 'spread spectrum' and 'dynamic channelling' techniques, might permit the use of f.m. having 25 kHz transmission bandwidths without overcrowding. Techniques such as these would require fairly complex additions to existing designs together with a complete reorganization of frequency assignations. However, the constantly decreasing cost and size of micro-computer components will make digitally controlled 'frequency hopping' and 'dynamically channelled' mobile radios much more feasible as time progresses.

**General considerations**

Impulsive noise interference is particularly relevant to the study of mobile radio where it is derived mainly from the ignition systems of petrol engines. It has been claimed[1] that, for s.s.b., since the reduced bandwidth causes longer ringing times in the receiver, the effect of impulsive noise is more severe than for a.m. or n.b.f.m. However, recent papers[2,3] suggest that the reduced bandwidth allows less energy to pass from the impulse, making it less unpleasant after demodulation. It has also been shown that careful matching between receiver stages, particularly at the i.f. filter, can significantly reduce the effects of impulsive interference. In receivers using fast attack; slow decay a.g.c. methods, each impulse will trigger the receiver into a low gain state for a relatively long interval, which is most undesirable. Clipping and noise blanking can provide some improvement, although effective noise blanking is difficult since the reduced bandwidth makes pulse detection less precise.

Adjacent channel interference tests have been performed[4] giving a comparison between f.m., a.m. and s.s.b. at various channel spacings and with various modulation signals. The results of the tests show, that for each modulating signal, s.s.b. performance at 5 kHz channel spacing was always better than that of either one or both of the 12·5 kHz spaced a.m. and f.m. systems used.

It was also shown that the 12·5 kHz a.m. and f.m. equipments modified to operate at 6·25 kHz spacing were significantly inferior in adjacent channel performance to the original 12·5 kHz versions. It should also be noted that the relative performance of different modulation systems depends considerably on the amount of speech processing used. A recent study[5] in the U.S.A. concluded that the use of frequency and amplitude compression s.s.b. systems could reduce the required channel spacing to 2·5 kHz. The additional cost and complexity of such techniques is apparently very low.

Fading effects caused by interference between different paths of the same signal are particularly noticeable at v.h.f. Multipath fading in a linear modulation system is quite different from that in an exponential system because of the sharp f.m. noise threshold. Multi-transmitter quasi-synchronous[6] operation is another area of interest to the v.h.f. mobile radio user. It is generally agreed that linear modulation systems are better suited to this mode of operation than exponential systems.

**Implementation of s.s.b. systems**

An s.s.b. waveform can be generated by one of two basic methods. The first, and most commonly used method, is known as the filter method (see figure 2). The audio input is fed to a balanced modulator where it is mixed with an intermediate frequency (i.f.) signal from the i.f. oscillator to produce a d.s.b. waveform. The i.f. d.s.b. signal passes through a filter which removes one of its two

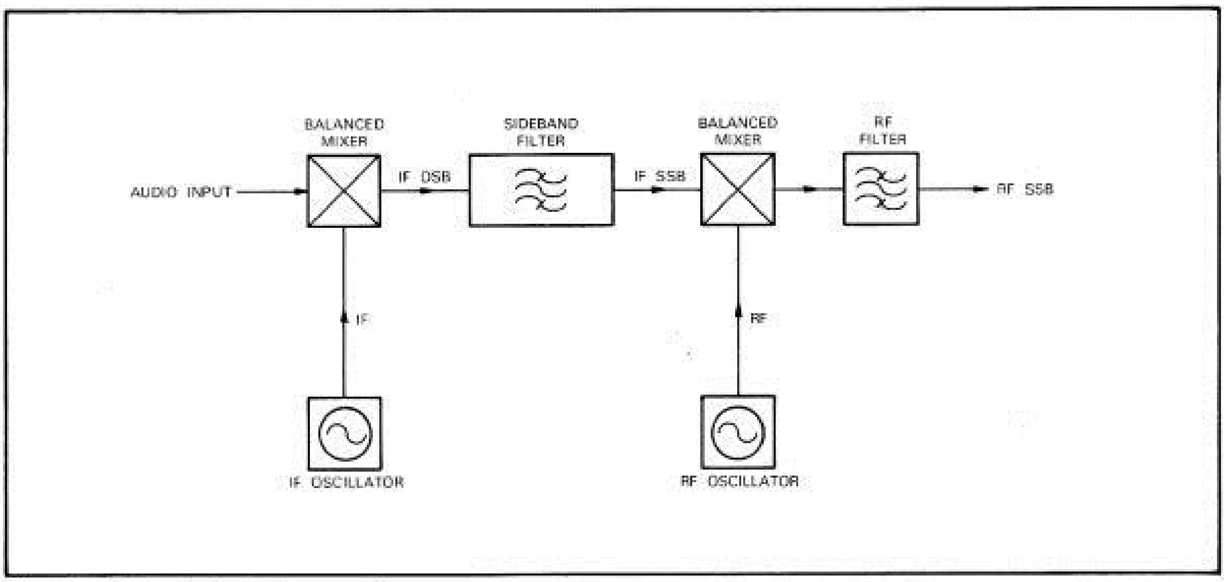


*Fig. 2. The filter method of s.s.b. generation*

sidebands to leave the required s.s.b. signal. The s.s.b. signal is then mixed with a radio frequency (r.f.) signal which translates it to the final output frequency of the system. In some systems, two or more i.f. stages with sideband filters are used to ensure the unwanted sideband is completely rejected. The r.f. s.s.b. signal, generated at a low power level, must be amplified to the required output power level by linear amplifiers.

The second method is known as the phasing method (see figure 3). The audio and r.f. signals are passed through 90° phase splitters which produce two quadrature components of each signal. One audio and one r.f. signal are fed to each of two balanced mixers to generate two d.s.b. waveforms. The two d.s.b. signals are then added so that one sideband cancels and the other is reinforced. Either sideband can be produced by reversing the audio or the r.f. connections to the modulators.

No sideband filters are required, and there are no intermediate frequencies since the audio signal is mixed directly with the final r.f. This method is not generally used in mobile radios because it is diffiult to maintain an equal 90° phase shift across the entire audio spectrum. Correct operation of this system relies on very accurate adjustment of all the system components.

A third method of generation and detection of s.s.b. signals was described by D. K. Weaver in 1956[7]. This was another type of phasing method which did not require a 90° phase splitter. Unwanted components in the output caused by unbalanced modulators or inaccurate phasing were located within the user's own transmission bandwidth, thus reducing adjacent channel interference. Although the inventor predicted that

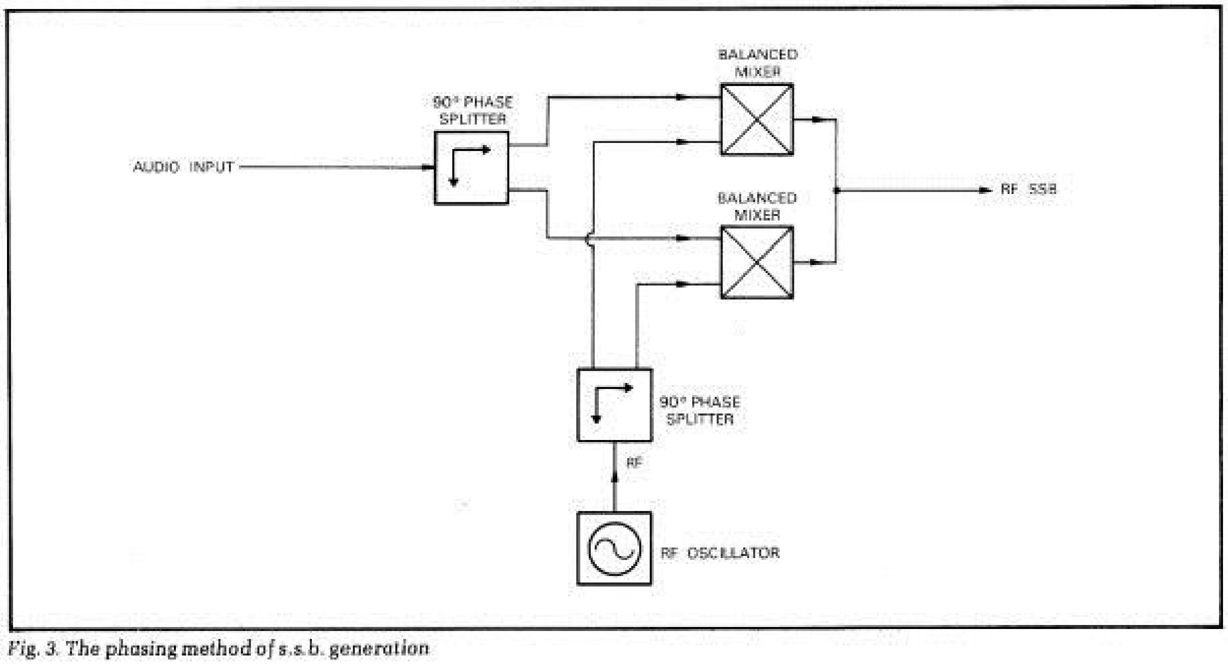


*Fig. 3. The phasing method of s.s.b. generation*

his design would play an important role in the future of s.s.b. systems, little use has been made of the method, possibly because its correct operation again relies on critical adjustment of the components. Both of these phasing type methods of s.s.b. generation should not, however, be completely disregarded. The appearance of single-package balanced modulators, which are less susceptible to changes in ambient conditions, makes these configurations more reliable. It is also possible to build excellent wide-band phase shift networks using analogue c.c.d. delay lines, and although these components are at present fairly expensive, their cost will follow the usual downward trend as their popularity increases.

More recent developments in s.s.b. transmitter design have occurred at the University of Bath, where Professor W. Gosling and V. Petrovic have produced a 'polar-loop' transmitter[8]. The polar-loop transmitter resolves a conventionally generated, low level s.s.b. signal into a baseband amplitude-related component, and a constant amplitude, angle-modulated component (polar co-ordinate form). The angle-modulated signal is amplified to the final output power level by a low-linearity class C or D amplifier, and the resulting signal is amplitude modulated by the scaled baseband component. In this way, a reconstructed s.s.b. signal is created at a high power level without the need for inefficient linear amplifiers. The overall linearity of this method is increased by configuring the transmitter as one large phase-locked loop. This arrangement, claim the inventors, provides an efficient means of generating an s.s.b. signal with a very high spectral purity. The technique can also be applied to other types of modulation such as d.s.b., v.s.b., i.s.b., or even a.m.

Demodulation of an s.s.b. signal is simply the reverse of the modulation process. Filter or phasing methods may be used, although, once again, the former is more popular in mobile radio. Most mobile equipments are transceivers in which the transmit and receive circuits share a number of common components. A typical s.s.b. transceiver (see figure 4) utilizes the antenna, r.f. and i.f. oscillators and sideband filter for both transmit and receive functions.

The r.f. s.s.b. signal enters the antenna, is amplified by the r.f. amplifier and then mixed down to i.f. by the receiver mixer. The i.f. waveform is passed through the sideband filter to remove any unwanted frequencies, and is then fed to the detector via an i.f. amplifier. The detector is typically a diode ring demodulator which uses the signal from the i.f. oscillator to generate a resultant audio output. A.G.C. may act on both r.f. and i.f. amplifiers, and these must, of course, be linear.

S.S.B. systems using a pilot carrier as a frequency and a.g.c. reference require additional circuitry to insert and extract the diminished carrier signal. Carrier insertion at the transmitter can be achieved by unbalancing the audio to i.f. mixer, and shifting the i.f. so that the resulting carrier component lies within the passband of the sideband filter. Alternatively, and perhaps preferably, a portion of signal from the i.f oscillator can be fed forward around the sideband filter to the input of the second transmitter mixer. Extrac-

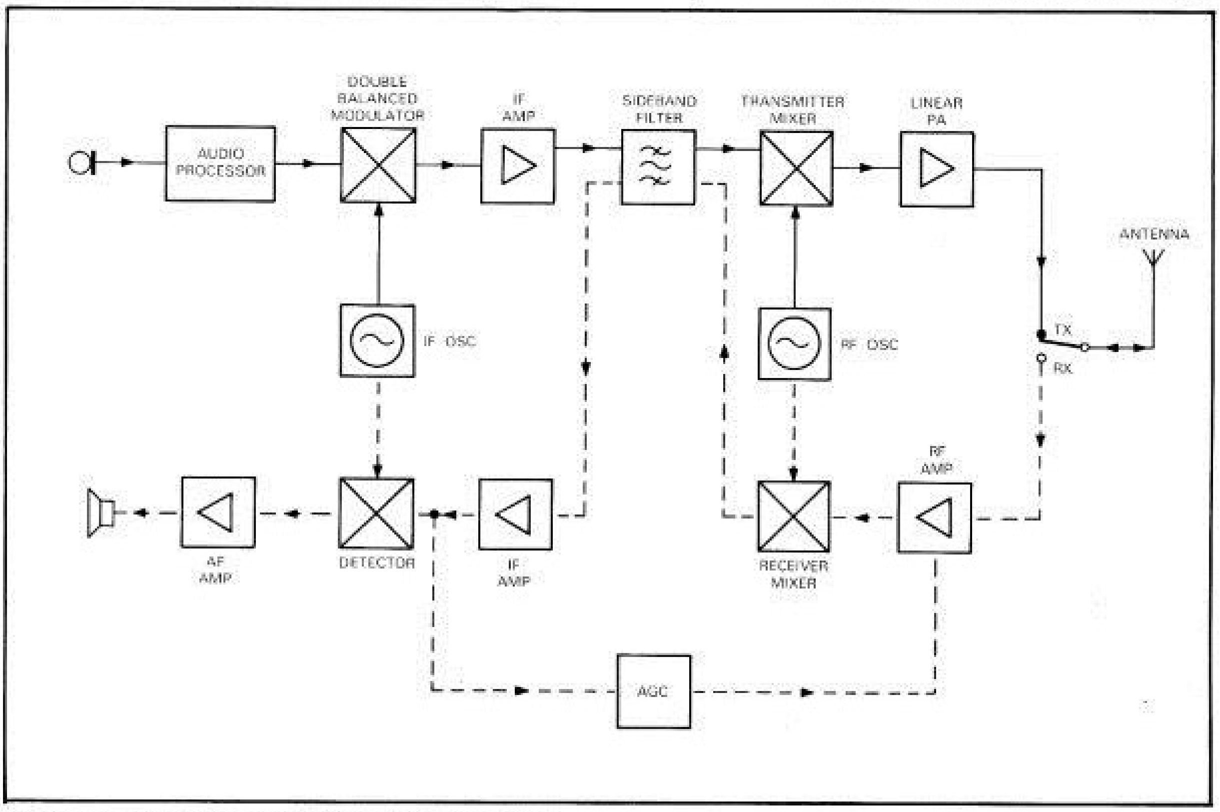


*Fig. 4. Simplified block diagram of a typical s.s.b. transceiver*

tion of the carrier at the receiver is a more difficult process which is usually performed at i.f., using a narrow-band pilot carrier filter or a variable bandwidth phase locked loop controlling the i.f. oscillator.

## Conclusions

S.S.B. exhibits a considerable power advantage over conventional a.m., and can provide a performance at least equal to that of n.b.f.m., whilst offering a significantly reduced transmission bandwidth. It is felt that the saving of bandwidth is of fundamental importance in v.h.f. mobile radio, where it could lead to improved operation within the present channel spacings, or even a reduction of the spacings themselves. Of the methods suggested for conservancy of the private mobile radio v.h.f. spectrum, s.s.b. appears to be the easiest to implement and the most promising at the present time.

Outstanding problems which require particular attention during the development of a working v.h.f. s.s.b. system, are those of frequency stability, application of a.g.c. and the effects of impulsive interference. Pilot carrier techniques could provide a solution to the first two problems, although some work is yet needed to perfect these techniques at a cost which will be acceptable to the mobile radio market. Recent developments in high efficiency transmitter configurations also add to the feasibility of v.h.f. s.s.b. mobile radios.

The overall conclusion must be that s.s.b. is worthy of serious consideration for use in mobile radio at v.h.f.

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**RESUME**

Par suite de l'expansion constante des télécommunications par radio mobile v.h.f., le fort encombrement des allocations de fréquences actuellement disponibles pose un grand problème dans certains secteurs. L'adoption de la modulation à bande latérale unique à espacement plus étroit entre les canaux radio a été proposée en vue de résoudre ce probléme.

L'article examine brièvement les implications d'une telle solution et établit la comparaison entre les principales caractéristiques de la modulation à bande latérale unique et celles d'autres procédés demodulation en faveur.

La mise en oeuvre du systéme est discutée ici, avec mentionparticuliére des études récentes dans ce domaine.



**ZUSAMMENFASSUNG**

Im Hinblick auf die zunehmende Beliebtheit fahrbarer UKW-Funkanlagen wird die starke Überlastung der gegenwärtig verfüglichen Frequenzen in verschiedenen Gebieten zu einem wesentlichen Problem. Als eine mögliche Lösung wurde die Einführung der Einseitenbandmodulation mit dichterer Funkkanalanordnung vorgeschlagen.

In diesem Artikel werden die Implikationen einer derartigen Lösung kurz untersucht und die Hauptmerkmale der Einseitenbandmodulation mit denen der anderen beliebten Modulationsverfahren verglichen.

Die praktische Einführung eines Einseitenbandsystems wird unter besonderer Bezugnahme auf kürzliche Studien auf diesem Gebiet diskutiert.



**SUMARIO**

En vista de la creciente popularidad de las comunicaciones de radio móvil de VHF, la enorme congestión hace que la actual disponibilidad de frecuencias disponibles se haya convertido en un importante problema en algunas zonas. Como solución a este problema se ha propuesto la adopción de modulación de banda lateral única con espaciación más estrecha de canales de radio.

El presente artículo examina las implicaciones de tal solución, y compara las principales características de la modulación de banda lateral única con las de otros populares procesos de modulación.

Se expone la realización de un sistema de banda lateral única con referencia particular a estudios recientes sobre este asunto.