Analogue and Digital Linear Modulation Techniques for Mobile Satellite

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Abstract

The choice of modulation format for a mobile satellite service is complex. This paper summarises the subjective performance of a variety of candidate schemes and voice coder technologies. It is shown that good performance can be achieved with both analogue and digital voice systems, although the analogue system gives superior performance in fading. The results highlight the need for flexibility in the choice of signaling format. Linear transceiver technology capable of utilising many forms of narrowband modulation are described.

Introduction

The mobile satellite environment places severe constraints on the selection of modulation systems. It is important to balance the power efficiency of any scheme that is chosen with its bandwidth utilisation. The selection should seek to minimise the bandwidth required for a single voice channel (or corresponding data channel) while maximising the quality of performance at any given carrier to noise level. This paper discusses the subjective performance of various modulation schemes and speech processing techniques with application to the mobile satellite service. The discussion is based on the mean opinion score (MOS) assessment of voice quality with measurements drawn from a number of recent publications and laboratory measurements. A comparison is made between the schemes on the basis of both bandwidth and speech quality at given carrier to noise levels.

It is interesting to note how the performance of an analogue form of linear modulation (ACSSB) compares with that of a digitised voice source used in conjunction with an appropriate data modulation scheme. For a given power/bandwidth budget, the linear analogue system performs at least as well as digital equivalents using the present generation of voice coder technology. This is perhaps surprising given the current trend towards digital implementations. It should be noted that the analogue scheme is most favourable in a fading environment (Ricean) due to the absence of any sharp performance threshold as received signal strength is lowered.

Future advances in vocoder technology will undoubtedly make digital voice more attractive and it is prudent to select a modulation system, and consequently a transceiver design, that can efficiently convey both digital and analogue traffic with the same hardware. A pilot aided form of linear modulation based on the Transparent Tone in Band (TTIB) technique [1] has been shown to be directly compatible with both analogue and digital traffic and would allow a smooth and cost effective transition between analogue and digital voice operation. The design of mobile terminal equipment intended for pilot aided linear modulation is briefly reviewed here before discussing the comparison of modulation systems.

The Linear Modulation Mobile Terminal

Recent developments in digital signal processing technology has made the exciting prospect of a reconfigurable transceiver, capable of handling a variety of analogue and digital signaling formats a realistic goal. Prototype equipment based on direct conversion techniques has been developed at the Centre for Communications Research at the University of Bristol for use in a variety of frequency bands [2]. Such a system is normally described as *linear* so long as its operation is
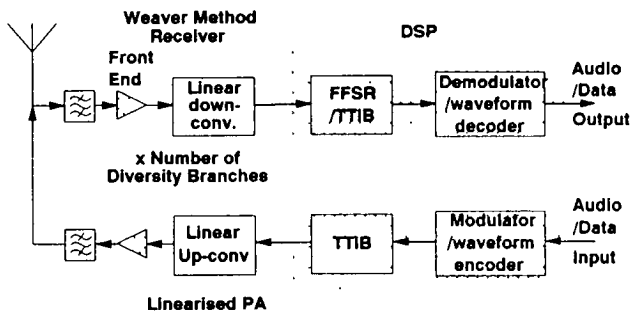
Figure 1: A Pilot Aided Linear Transceiver

Figure 2: Feedback Linearisation of an RF Amplifier

independent of the amplitude and phase characteristics of the signaling format.

Both the transmitter and receiver systems are based on the Weaver frequency translation technique. This means that each frequency translation requires a quadrature local oscillator to drive the mixers. An advantage of using the Weaver method is that the image products fall within the users own channel, thus minimising the accuracy needed in the gain and phase match on the two quadrature signal paths.

Figure 1 gives a block diagram of a proposed transceiver architecture for a pilot aided linear transceiver. In such a system, distortions due to fading are removed in the receiver baseband processing by ascertaining the fading characteristics from the nature of the received pilot tone. The transceiver is divided into the fixed hardware sections of the linearised transmitter and the zero IF/single IF receiver, and the software reconfigurable baseband processing sections. In a pilot aided system, the baseband processing includes the generation of TTIB and fading correction using feedforward signal regeneration [3]. Processing is also required for whatever signaling format is to be transmitted through the system. It is intended that the particular signaling format chosen is not limited by the transceiver design.

Linear Transmitters

The transmit section of the transceiver must meet stringent requirements for linearity and power efficiency. Considerable success has been achieved at Bristol using feedback amplifier linearisation techniques, although feedforward techniques also have potential. A block diagram of feedback amplifier linearisation is shown in figure 2. The baseband control function may take a number of forms, the simplest and most well understood being the cartesian feedback system [4] in
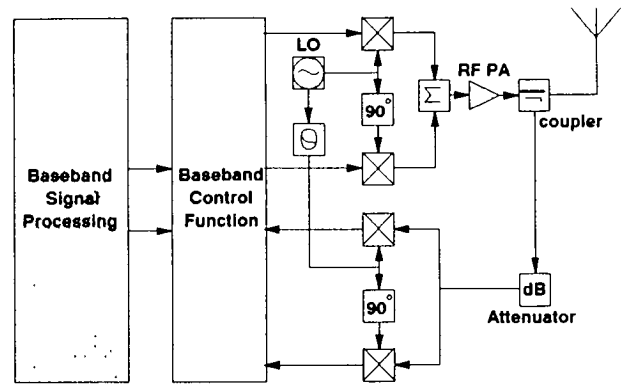
which the control function consists of a pair of low-pass differential amplifiers. A well known alternative is adaptive predistortion [5] in which the feedback path is sampled (not necessarily continuously) and the gain and phase response of the amplifier assessed across its operating range. The transmitted signal is predistorted to correct for any non linearities in the amplifiers characteristic.

With careful design, feedback linearisation can give excellent performance. Linearised class C amplifiers yield third order intermods better that 60dB down on PEP for a two tone test. Performance is limited by the loop gain that can be achieved within the limits of stability.

The gain and phase match of the I/Q downconverter limits the image rejection at the transmitter output. 40 dB image rejection requires a phase match of 1 degree and an amplitude match of 0.1 dB. This can be achieved over a reasonable bandwidth using RF techniques alone. To extend the bandwidth further, an automatic calibration procedure can be carried out in production to calculate correction factors for any gain and phase mismatch. In service, the baseband processing can use these to predistort the drive to the amplifier. In a transmitter fed with I and Q baseband signals, DC offsets will coincide with the pilot tone and can easily be made small enough not to dominate (offsets 45 dB down on PEP can be achieved, while the pilot is often around 10 dB down).

Linear Receivers

Weaver method zero IF receiver systems (direct conversion) have been suggested for linear transceivers [6]. They offer the advantage of minimising the component count, but require two linear mixers operating at the incoming frequency and an accurate 90 degree phase
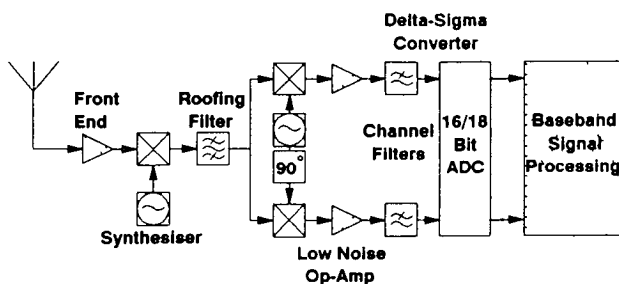
Figure 3: A Single IF Weaver Method Receiver.

Figure 4: Placing DC in the demodulated I/Q signals.

shift network which operates across the entire range of the systems frequency synthesiser. Carrier leakage can also be problematical at high frequencies, causing large DC offsets in the demodulated I/Q signals. These problems mean that direct conversion will probably be unsuitable for use at mobile satellite frequencies for the immediate future.

Figure 3 shows a block diagram for a single IF linear receiver. This alleviates many of the problems by performing the I/Q generation at a low, fixed IF. If a suitable frequency is chosen, then mixers with good linearity, low carrier leakage and low cost can be obtained. The quadrature phase shift network need only operate at a single frequency, making accuracy and stability easy to achieve. Unfortunately, a roofing filter will now be required to remove the image frequencies which, after the first mixing stage, do not lie within the wanted channel. Such filters can be obtained cheaply if a prudent IF is chosen.

Carrier leakage around the final mixers results in a DC offset in the resulting I/Q baseband signals. Such offsets can be removed by filtering, but they can alter slightly with incoming signal strength. When operating in weak signal areas, this variation means that DC rejection must be achieved up to 5 to 10 Hz. Such filtering removes some of the information content of the received pilot. It may be prudent to frequency lock the receiver so that DC falls in the gap between the signaling modulation and the pilot. DC rejection can then be achieved with no information loss and removal of the frequency offset is straightforward (figure 4).

The availability of modern delta-sigma analogue to digital converters with 18 bit resolution means that conversion into the digital domain can be achieved with dynamic range well in excess of that required for mobile satellite receivers. The signals can then be filtered, amplified and mixed as required with no fear of instability. (High resolution ADCs are becoming cost effective due to the economies of scale in the digital audio market). The limitation in receiver performance; adjacent channel performance, sensitivity, selectivity and
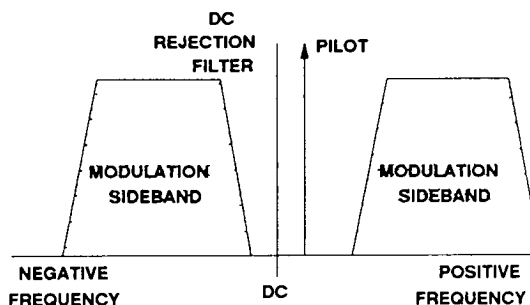
overload point. are now largely dependant on the linearity of the RF pre-amplifier and subsequent mixer stages. Field testing of a 900MHz single IF linear receiver is currently in progress at Bristol and prototype hardware for 1.7 GHz operation is under development.

Subjective Comparison of Modulation Systems

Before comparing some modulation formats which may be used in a mobile satellite system (whether linear of not) it must be stated that voice assessment based on MOS is only truly valid if all modulation candidates are tested by the same panel of listeners under the same test conditions. (Clearly, the source tape must be common for all systems under trial and the playback to assessment panels, customers etc, must be randomised to prevent learning of the information content) As the results for this assessment are drawn from a number of sources [7,8,9,10], this condition is not fully satisfied, however, the subjective performance of analogue linear modulation with respect to a high quality 9.6kbps codec has been validated at the laboratories at Bristol University and thus provide a good reference point for scaling the other results. It is on this basis that the results are presented and discussed, giving, it is felt, and accurate picture of modulation and coder performance with current technology. Further work is in progress at Bristol to complete the analysis for a full range of coder types and modulation techniques.

Modulation Systems

All modulation systems will be compared on the basis of average signal power with respect to noise power density (dB/Hz) required to maintain a given voice quality (MOS). Three digital modulation schemes have been studied, namely:

- 25 % raised cosine filtered 16-QAM

  (coherent detection).

  Bandwidth efficiency = 3.2 bps/Hz

  BER of 10^{-3} at an $\frac{E_b}{N_0}$ of 13.5 dB

  BER of 10^{-2} at an $\frac{E_b}{N_0}$ of 11 5 dB

- 25 % raised cosine filtered QPSK

  (coherent detection).

  Bandwidth efficiency = 1.6 bps/Hz

  BER of 10^{-3} at an $\frac{E_b}{N_0}$ of 9 dB

  BER of 10^{-2} at an $\frac{E_b}{N_0}$ of 7 dB

- 100 % raised cosine filtered trellis coded 8-PSK

  Bandwidth efficiency = 1 bps/Hz

  BER of 10^{-3} at an $\frac{E_b}{N_0}$ of 5.8 dB

  BER of 10^{-2} at an $\frac{E_b}{N_0}$ of 3.8 dB

All three systems assume a 2 dB implementation loss.

Three vocoder rates are investigated of 9.6kbps, 4.8kbps, and 2.4kbps. Channel bandwidth for the digital systems is calculated as the vocoder rate divided by bandwidth efficiency plus 300 Hz guard band to allow for Doppler spread.

The carrier to noise ratio, $\frac{C}{N_0}$ is calculated as

$\frac{E_b}{N_0} \times 10 \log (\text{vocoder rate})$

The fourth system is an analogue form of linear modulation based on TTIB. The $\frac{E_b}{N_0}$ values for this system are measured as the average received signal power when the speaker is active (including pilot tone) divided by the received noise power density. It should be noted that this figure is somewhat pessimistic as it does not take account of periods of voice inactivity. Clearly, the average received power would be lower if voice inactivity were accounted for resulting in lower power consumption in both the mobile and the satellite. For the equivalent power saving in a digital voice system, a robust voice activity detection is required.

The channel bandwidth for the analogue system is 3.7 kHz, consisting of a 300 to 3000 Hz voice bandwidth, a 600 Hz TTIB notch and a 400 Hz guard band.

Figure 5 shows the results for a static AWGN. As would be expected, the analogue system out-performs the digital systems at high signal strengths, the latter exhibiting an upper threshold due to vocoder algorithm limitations which degrade as the coder rate is reduced. As the signal strength falls, certain digital systems out-perform the analogue scheme. For example, TC-8PSK with a 9.6 kbps coder is much better than ACSSB at 45 dB/Hz. The improvement in performance, however, is achieved at the price of bandwidth expansion. The TC-8PSK system with a 9.6
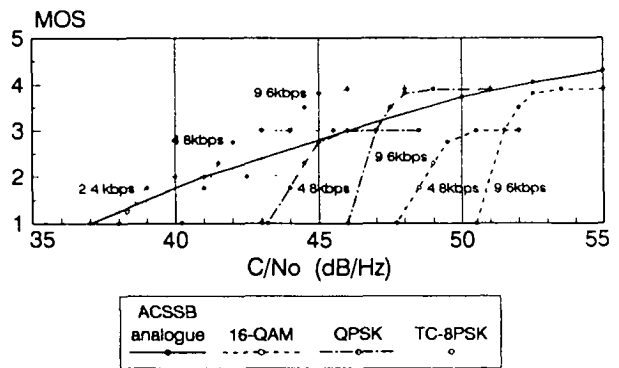

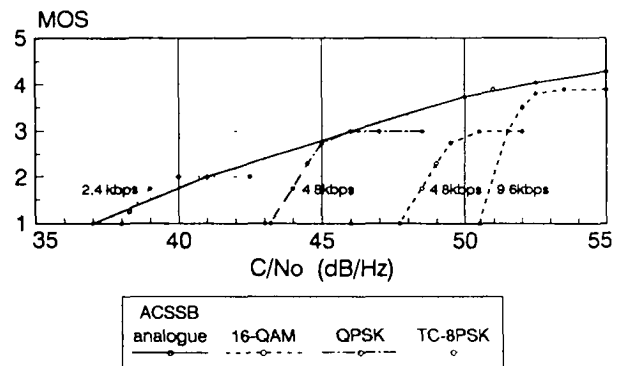Figure 5: Subjective Performance of all Systems.

Figure 6: Subjective Performance of Systems with Channel Bandwidth below 4 kHz.

kbps coder requires a channel bandwidth of approximately 9.9 kHz, compared with the 3.7 kHz channel bandwidth for ACSSB.

Taking into account the narrow band requirement for mobile satellite operation, figure 6 highlights the performance of those systems which require a channel bandwidth of less than 4 kHz. It is now clear that AC-SSB performs as well as any of the digital competitors, except for a 2.4 kbps coder with TC-8PSK at very low signal strengths. It is well recognised that 2.4 kbps coders do not, at present, provide sufficient quality of service for a public telephony mobile satellite service. If services with a MOS below 2.5 are neglected, then the choice of modulation system becomes that of figure 7. It is now clear that ACSSB performs at least as well as present generation modulation and voice coding technologies. At low signal strengths the amplitude companding maintains a usable voice channel without recourse to bandwidth expansion.
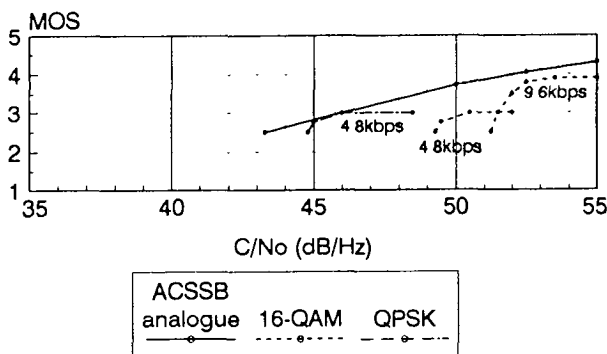
Figure 7: Subjective Performance of Systems with Channel Bandwidth below 4 kHz and MOS above 2.5.

Performance in Fading

Taking an operating point of 51 dB/Hz as a typical link budget for the M-SAT system, the 9.6 kbps 16 - QAM system will not operate satisfactorily, 4.8 kbps 16 - QAM will allow a fade margin of 2 dB, 4.8 kbps QPSK will give a fade margin of 6 dB, and ACSSB a fade margin of 7 - 8 dB.

Conclusions

With the application of feedback amplifier linearisation techniques, along with recent advances in device technology (particularly ADC, DAC and DSP technology) a universal linear transceiver can be realised. This could convey voice traffic via an analogue or a digital modulation format. Thus a smooth and cost effective transition between signaling formats could be achieved as and when superior speech vocoders become available for the mobile satellite service.

Based on some recent subjective performance measurements of candidate modulation schemes and speech coders for mobile satellite, it is concluded that digital vocoder technology, with a suitable data modulation format can now approach the performance af an analogue linear modulation system in a given power/bandwidth budget. Speech quality, channel occupancy, fade margin and power consumption, however, are all superior in the analogue implementation.

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