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COMBINED AMPLITUDE-PHASE MODULATION FOR A VHF COMMUNICATION LINK

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ABSTRACT

In this work a combined amplitude - phase modulation communication system is considered. Amplitude modulation is used to transmit a voice signal, phase modulation is used for data transmission. Such scheme was studied as a possible realization of a data channel between aircraft and ground stations.
The performance of the combined modulation system is evaluated through computer simulation in the two cases AM-PSK and AM-MSK.

INTRODUCTION

This paper describes a communication system able to transmit simultaneously an "audio" signal and data on the same radio link. This problem is of particular interest for the realization of a "data" channel between aircraft and ground stations. This data channel could support the "voice channel", which present organization of Air Traffic Control (ATC) completely relies upon for ground-air-ground information exchanges.
The availability of a data channel could enable information exchanges in digital form between ground and on-board processors, thus allowing an automatic service management and a consequent decreasing of the voice channel load, which is today very near to saturation in critical ATC areas.

The two essential methods for implementing a ground-air-ground digital channel are:

1) the data channel is integrated in the secondary radar, and is used only for ATC functions (DABS) (1);

2) the data channel is realized using a VHF radio link and can be used not only for ATC functions, but for general data communication.

In the following a particular implementation for the second approach will be considered, which has the advantage of creating a data channel without relevant modifications to the service organization and to present equipment. On the contrary, the DABS, while allowing greater information capacity, requires new equipment, and is feasible only in a secondary surveillance radar (SSR) environment.

We set the following conditions for the implementation of the universal data link:
- data transmission is simultaneous and compatible with voice transmission and is radiated on the same radio channel;
- modification to present on-board equipment should be minimized, for data communication, while present on-board equipment must receive voice information without modifications and without noticeable interference due to data transmission.

The proposed method uses compound modulation of a single carrier, AM for voice transmission, PSK or MSK for data. In an ideal system the envelope and angle modulations do not interfere (2) and their compatibility is guaranteed. A block diagram of the communication system is shown in fig 1; essentially non-linear elements, such as converters and limiters, and bandpass elements, not present in an ideal system, are included. In particular, the "transmitter filter" represents all the operations which are carried out on the signal before it is radiated to limit and shape its spectral bandwidth. The receiver filter includes all the operations carried out by the receiver preselector, converter and IF stages in order to achieve the necessary selectivity.

The non-ideal nature of the actual system makes the envelope and angle modulations no longer independent of each other and sets an upper bound for the system performance, in terms of signal-to-noise ratio for the voice channel and error probability for the data channel. In the following sections some considerations are developed for the evaluation of the mutual interferences between the two information channels.

PSK AND MSK MODULATIONS

A phase-shift-keying (PSK) signal has the form:

$$s(t)=\sqrt{2E}\,\cos(\omega_c t+\psi_k) \qquad kT \leqslant t \leqslant (k+1)T,\ k \text{ integer} \qquad (1)$$

where ω_c is the carrier angular frequency, T the bit time interval and E the signal mean power. If a_k is the binary data sequence to be transmitted, $a_k = \pm 1$, we have

$$\psi_k=0 \text{ if } a_k=-1 \qquad \psi_k= \text{ if } a_k= 1 \qquad (2)$$

The spectral power density of a PSK baseband signal is given by (ref.3)

$$G(\omega) = T\, {\frac{\sin(\omega T/2)}{(\omega T/2)}}^{2} \qquad (3)$$

and this expression is plotted in fig 2, normalized to the maximum value, corresponding to the carrier frequency. Spectral occupancy is considerable in comparison with other digital modulations, due to the signal sharp phase changes. For this reason, modified PSK modulations with narrower bandwidths are often to be preferred (refs 4, 5). In this study PSK was considered for its theoretical importance and for comparison purposes, not for an actual practical interest.

The minimum-shift-keying modulation (MSK) has an actual interest because of its characteristics. MSK is a particular frequency-shift-keying and the signal has the following properties:

1) the envelope of the modulated signal is constant (as for PSK);
2) the istantaneous frequency deviation is ± 1/4 of the bit rate;
3) the istantaneous phase is continuous and varies linearly by $\pm\pi/2$ during every bit interval;
4) spectral occupancy is very little, if compared with other digital modulations with the same transmission rate. Therefore the signal tolerates severe bandpass filtering without great degradation in performance.

The MSK signal can be expressed as (ref 6)

$$s(t)=\sqrt{2E}\,\cos(\omega_c t+ \frac{\pi}{2T}a_k t+x_k), \qquad kT\leqslant t\leqslant (k+1)T,\ k \text{ integer} \qquad (4)$$

where a_k are the binary data to be transmitted, and the constants x_k are determined for phase continuity as:

$$x_k= x_{k-1}+(a_{k-1} - a_k)k\frac{\pi}{2} \qquad (5)$$

with $x_o=0$ and therefore $x_k=0$ or π (modulo 2π). It can be shown that in this case the MSK signal can be written as:

$$s(t)=\sqrt{2E}\sum\left[y_{2n-1}g(t-2nT)\cos(\omega_c t)- -y_{2n}g(t-2nT-T)\sin(\omega_c t)\right] \qquad (6)$$

where

$$g(t)=\begin{cases}\cos(\frac{\pi}{2T}t), & -T\leqslant t\leqslant T\\ 0, & \text{elsewhere}\end{cases} \qquad (7)$$

and the sequence of binary symbols $y_n=\pm 1$ is obtained recursively from the a_k as:

$$\begin{cases} y_o=1\\ y_1=1 \oplus (a_1 \oplus a_o)\\ y_n=y_{n-2} \oplus (a_n \oplus a_{n-1}), \quad n=2,3,.. \end{cases} \qquad (8)$$

where the $\oplus$ operation is defined by:

$$\begin{array}{c|cc} \oplus & -1 & 1\\ \hline -1 & -1 & 1\\ 1 & 1 & -1 \end{array} \qquad (9)$$

It follows that a MSK signal is OK-QPSK type, being formed by two sequences of pulses with duration 2T and shape given by g(t), modulating two quadrature channels respectively.
The spectral power density of a MSK baseband signal is given by:

$$G(\ )=\frac{4\pi^2 T\ 1+\cos(\omega T)}{(\omega^2 T^2 -\pi^2)} \qquad (10)$$

The performance of the optimum receiver, in terms of bit error probability, in the presence of a linear, infinite-bandwidth channel with additive, white gaussian noise is given by:

$$P(\text{error})= Q(\sqrt{E/N_o}) \qquad (11)$$

both for MSK and PSK, where the function Q(x) is defined by (12), and N_o is the total noise power.

$$Q(x) = \frac{1}{2} \int_{x}^{\infty} \exp(-x^2/2)dx \qquad (12)$$

This result refers to coherent detection, which increases the complexity and cost of the receiver. Therefore it is often advantageous to use differential technique which allows the demodulation process to be performed in a relatively simpler and cheaper manner.

FILTER EFFECTS

The envelope and phase modulated signal has the form:

$$s(t) = \left[1+mf(t)\right] \cos\left[\omega_c t + \varphi(t)\right] \qquad (13)$$

where m is the amplitude modulation index, $|m|<1$, f(t) is the voice signal, bounded to ±1, $\varphi(t)$ the data modulated phase.
The signal in (13) theoretically has an infinite bandwidth, while in a practical communication system both the transmission signal bandwidth and the receiver bandpass must be bounded. These band limitations produce mutual interference between the data and voice channels. For istance, they make the signal envelope no longer constant when the voice signal f(t)= 0.
The OK-QPSK form (6) is valid also for the PSK case, setting $y_{2n}=0$, $T=\frac{1}{2}T_s$, where T_s is the symbol time interval and g(t) is given in this case by:

$$g(t) = \begin{cases} 1 & \text{for } |t| \leqslant T \\ 0 & \text{elsewhere} \end{cases} \qquad (14)$$

The envelope of a OK-QPSK signal

$$i(t) = \left\{ \left[\sum_{n=-\infty}^{+\infty} y_{2n-1} g(t-2nT) \right]^2 + \left[\sum_{n=-\infty}^{+\infty} y_{2n} g(t-2nT-T) \right]^2 \right\}^{\frac{1}{2}} \qquad (15)$$

and its value is always 1 both for PSK and for MSK. If the modulated signal is passed through a bandpass filter, whose transfer function is supposed symmetrical around the carrier frequency, the signal at the output can be written again in the form (6), with g(t) substituted by q(t); q(t) is given by:

$$q(t) = g(t) * h(t) \qquad (16)$$

in which g(t) is the symbol shape and h(t) is the complex envelope of the filter impulse response. Therefore the signal is still OK-QPSK type, but the symbol shape is now determined by q(t), instead of g(t). The convolution operation makes the duration of q(t) longer than that of g(t), so intersymbol interference is born between even symbols, in-quadrature channel, and odd symbols, in-phase channel.

The envelope of the filtered signal is given by (15) by substituting g(t) with q(t) and in general it is not constant, unless q(t) satisfies particular conditions. Figs 3 and 4 show typical behaviours of the envelope of filtered PSK and MSK signals respectively, with F T=2.5, being F the single-side -3 dB bandwidth of the bandpass filter (Butterworth, fourth order)

The considerations in this section show that in the amplitude-and-phase communication system the envelope of the received signal is detrmined both by the voice-AM modulating signal and by the data phase modulation because of the bandwidth limitations. A theoretical analysis is much more complicated in this case. The overall performance of the considered communication system was then evaluated by a suitable simulation program.

THE SIMULATED COMMUNICATION SYSTEM

The simulation of the communication system was performed using discrete techniques and sampling of the signals. The equivalent baseband system was considered, taking into account the finite spectral occupancy of the signals.

The bandpass filters in the communication chain were modeled as Butterworth, with the following characteristics:
- transmitter filter: fourth order, with -3 dB single-side bandwidth 7.5 kHz;
- receiver filter: eighth order, with -3dB single-side bandwidth 5 kHz.

Filtering was performed in frequency domain, while non-linear operations, as modulation and demodulation were simulated in time domain. The transformations from one domain to the other were done using the Fast Fourier Transform (FFT) algorithm.

The signal was processed in blocks of 2048 samples; the sampling frequency was chosen 19200 Hz. A general block diagram of the simulation program is depicted in fig 5.

In order to estimate the effects of data modulation on the detected voice signal, the error signal, defined by

$$e(t) = s_o(t) - mf(t) \qquad (17)$$

was computed; f(t) is the voice modulating signal, m is the AM modulation index, s_o is the signal at the AM detector output.

RESULTS AND COMPARISONS

The performance of the voice-data communication system in the two cases AM-PSK and AM-MSK was evaluated utilizing the simulation program described in the previous section. Results are presented for four different signals, modulating in the voice channel:

1) a tone with frequency 937.5 Hz;
2) a tone with frequency 1875 Hz;
3) a sum of five tone, given by:

$$f(t)=\sum_{i=1}^{5} a_i \cos(2\pi f_i t) \qquad (18)$$

in which the coefficients a_i and the frequencies f_i have the values indicated in table 1;

i	a_i	f_i
1	2/9	468.75 Hz
2	1/3	937.5 Hz
3	2/9	1406.25 Hz
4	1/9	1875.0 Hz
5	1/9	2343.75 Hz

Tab.1 Values for a_i and f_i in (18)

4) a real voice signal, having time duration of four seconds.

Several values for the AM modulation index were considered, i.e. 0.2, 0.4, 0.6, 0.8, 0.9. Binary pseudo-random data sequences were utilized in the simulation, with equiprobable symbols. The following data rates were considered: 300, 600, 1200, 2400 bits/s.

As stated in the previous section, amplitude modulation of the carrier influences the data demodulation operation and data modulation produces distortion in the envelope of the modulated signal. In order to characterize these mutual effects the following parameters were computed:

- root mean square error in the voice signal at the AM detector output;
- peak error in the same signal as above;
- bit error rate after data demodulation.

The first and the second parameters permit to estimate the distortion on the voice signal due to the presence of data modulation and to the channel noise. The two following signal-to-noise ratios were computed:

$$\begin{aligned}(S/N)_o &= 10\log(0.5/N)\\ (S/N)_1 &= 10\log(S/N)\end{aligned} \qquad (19)$$

where S and N are respectively the signal and noise power at the AM detector output. The first definition is useful to consider the overall noise after the envelope detector independently of the modulation index and of a particular voice signal. In this case the noise power is related to the power of the signal which would be obtained at the AM detector output if a signal modulated by a sinusoid, with a modulation index equal to 1 is fed to the input. $(S/N)_1$ is a normal signal-to-noise ratio.

The two considered signal-to-noise ratios are shown in figs 6 and 7 versus data transmission rate, for the considered voice modulating signals, for the AM-PSK system; the results for the same cases for the AM-MSK system are reported in figs 8 and 9. In this figures the AM modulation index was set 0.9, while a noiseless channel was assumed, in order to estimate the effects of data modulation on the detected voice signal.

The results of simulation of the AM-PSK system show that $(S/N)_o$ is independent of the AM modulation index, and decreases as data rate is increased. It is therefore possible to obtain the variation of the parameter $(S/N)_1$ versus the AM modulation index using the results in fig 7. In particular, $(S/N)_1$ mostly depends on the detected signal power, which is essentially determined by the power of the modulating signal, then by the particular voice signal and the modulation index.

The obtained results for the AM-MSK system show some differences with respect to AM-PSK: $(S/N)_1$ is nearly independent of the modulation index, while $(S/N)_o$ depends greatly on it. Both of them vary with the type of voice modulating signal and decrease as data rate is increased. The RMS and peak values of the error signal are much lower for the AM-MSK system than for AM-PSK.

These differences are principally due to the spectral spread of PSK with respect to MSK. In the case of AM-PSK the unwanted amplitude modulation caused by data modulation is essentially due to the bandpass filtering of the signal; so $(S/N)_o$ does not depend significantly neither on the voice modulating signal, nor on the modulation index, because the spectral spreading due to amplitude modulation is little, if compared with the spectral occupancy of the PSK signal.

If MSK modulation is used for data, the available channel bandwidth is large enough to transmit the phase modulated signal without considerable envelope fluctuations, for the examined data rates. Amplitude modulation of the MSK signal causes a certain spectrum spreading, particulary in the presence of high frequencies modulating in the AM channel. This fact accounts for the observed dependence of the parameter $(S/N)_0$ versus the type of AM modulating signal and the AM modulation index, in the AM-MSK case. The parameter $(S/N)_1$ depends very little on the modulation index, in the AM-MSK system, because an increase in this index causes both the power of the information signal and of the noise at the envelope detector output to increase.

CONCLUSIONS

The results of computer simulation of the considered communication system show that the modulation MSK is much less sensitive to bandwidth limitations than PSK, because of its spectral compactness.
In the most interesing case, real voice modulating signal, signal-to-noise ratios about 15 dB for data rate 300 b/s, 12 dB for 600 b/s, 9 dB for 1200 b/s were obtained for the AM-PSK system, while signal-to-noise ratios greater than 20 dB were found for the AM-MSK system, for data rates 300 and 600 b/s. This value is usually considered satisfactory for voice communication via a radio channel.

REFERENCES

1. P.R.Drouilhet: "DABS: A System Description", Lincoln Laboratory, ATC-42, FAA-RD-74-189, November 1974.

2. J.C.Hancock, R.W.Lucky: "Performance of Combined Amplitude and Phase Modulated Communication Systems", IRE Trans. on Comm., December 1960.

3. S.A.Gronemeyer, A.L.Mc Bride: "MSK and Offset QPSK Modulation", IEEE Trans. on Comm., August 1976.

4. J.W.Bayless, R.D.Pedersen: "Efficient Pulse Shaping Using MSK or PSK Modulation" IEEE Trans. on Comm., June 1974.

5. L.J.Greenstein, P.T.Fitzgerald: "Envelope Fluctuations Statistics of Filtered PSK and Other Digital Modulations" IEEE Trans. on Comm., April 1979.

6. T.Masmura, S.Sahaejima, Y.Morihiro, H.Fuketa: "Differential Detection of MSK with Nonredundant Error Correction", IEEE Trans. on Comm., June 1979.

7. H.R.Mathwich, J.F.Balcewicz, M.Hecht: "The Effect of Tandem Band and Amplitude Limiting on the E_b/N_0 Performance of Minimum (Frequency) Shift Keying (MSK)", IEEE Trans. on Comm., October 1974.

8. C.R.Cahn: "Combined Digital Phase and Amplitude Modulation Communication Systems", IRE Trans. on Comm. Systems, September 1960.

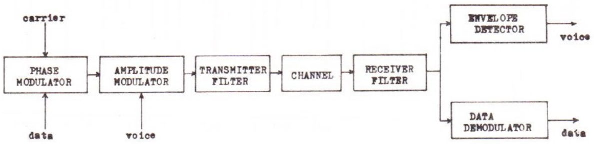

Fig. 1 Block diagram of the communication system

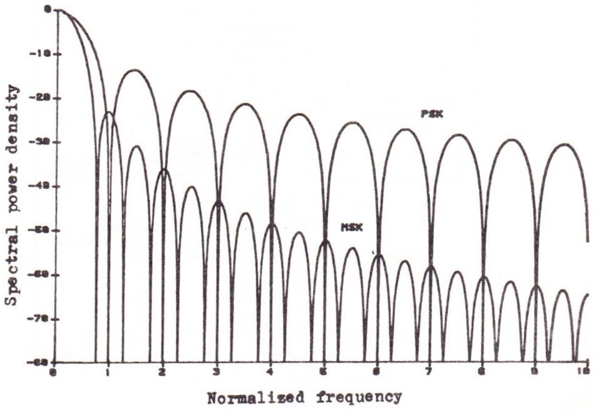

Fig. 2 Spectral power density of PSK and MSK (normalized with respect to the carrier value)

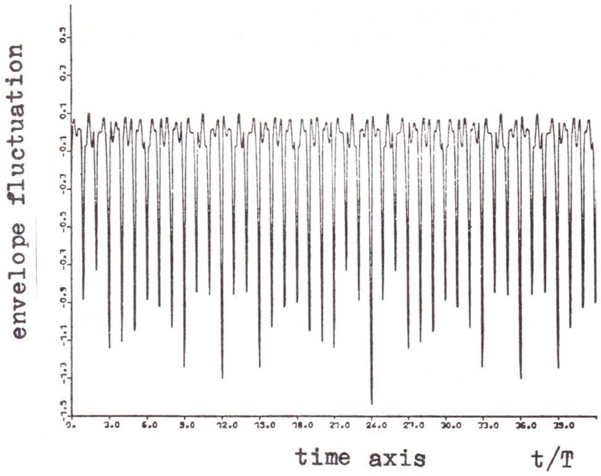

Fig. 3 Typical envelope behaviour of a filtered PSK signal

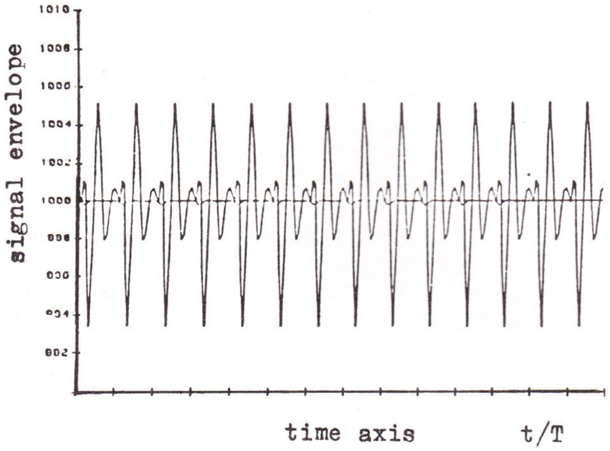

Fig. 4 Typical envelope behaviour of a filtered MSK signal

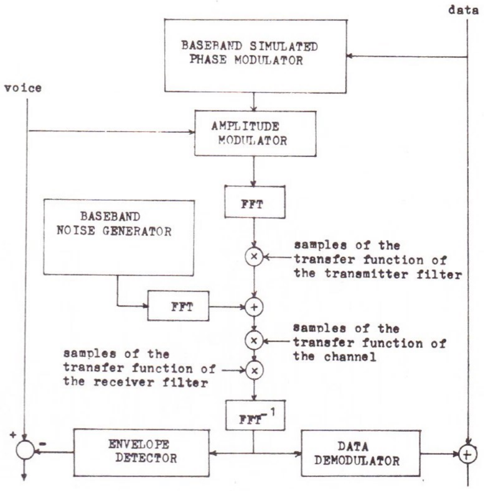

Fig. 5 General block diagram of the simulation program

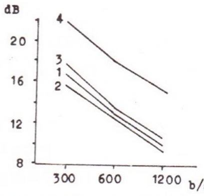

Fig. 6 $(S/N)_o$ for AM-PSK

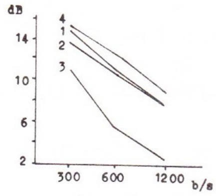

Fig. 7 $(S/N)_1$ for AM-PSK

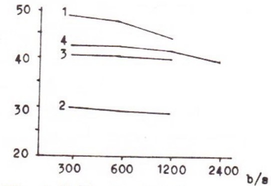

Fig. 8 $(S/N)_o$ for AM-MSK

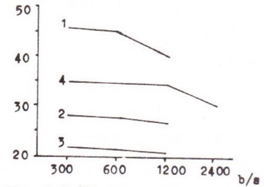

Fig. 9 $(S/N)_1$ for AM-MSK

Modulating voice signal:
-1- tone, 937.5 Hz; -2- tone, 1875 Hz; -3- sum of five tones;
-4- real voice signal.

Fig. 5 General block diagram of the simulation program