DESIGN AND PERFORMANCE OF AN ADAPTIVE MFSK HF TERMINAL

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INTRODUCTION

The paper will describe the implementation of a real-time integrated modem/codec system for long-range radio communications, e.g. HF. The system has been developed by the Hull-Lancaster Communication Research Group. The system has been implemented using a proprietary DSP-based architecture which uses the AT&T DSP32C digital signal processor and the transmission format comprises MFSK modulation combined with RS coding. At the conference the main features of the system will be described and results from on-air HF trials over both static and aeromobile channels will be presented.

An adaptive MFSK modem based on the PICCOLO concept [1] has been developed. The DSP based approach has resulted in a highly flexible implementation in which the key features are outlined below:-

1. Variable number of tones from 2 to 32.

2. Flexible tone placement; each tone is independently agile within a 2 kHz bandwidth with a 1 Hz resolution.

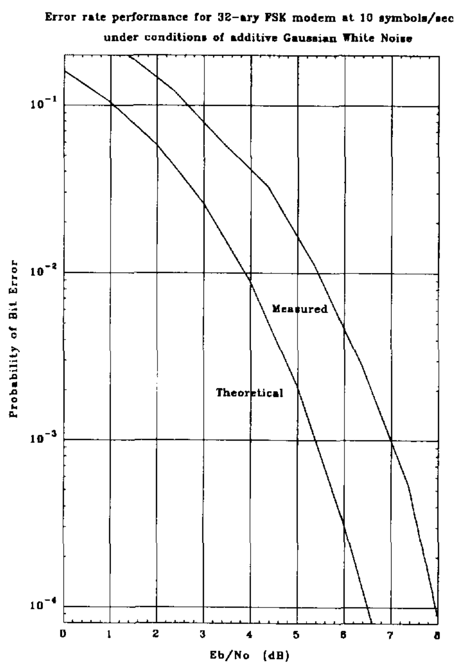
3. Flexible symbol transmission rates from 10 symbols/s to 100 symbols/s.

4. A frequency offset compensation system provides accurate modulator to demodulator frequency alignment over channels which exhibit relatively high time-varying frequency offsets due to equipment drift and Doppler shift induced by terminal motion.

5. The demodulator contains highly agile demodulation filters which allow frequency tracking and re-tuning to be accomplished on-line in real-time.

6. An embedded synchronisation system obtains symbol synchronisation by utilising demodulator filter outputs. This synchronisation technique adds no transmission overheads to the system.

The figure below shows a plot of the measured uncoded error rate performance of the modem in additive Gaussian white noise versus the theoretical performance.



This modem system has been integrated with an adaptive RS codec. The codec can be configured to use several RS coding schemes in order to give the system a flexible response to the coding problem in terms of both the size of the coding symbol set and the code rate and hence error correcting power of the code. The codec is currently capable of supporting the following coding schemes: RS(7,3), RS(15,9), RS(31,15), RS(31,25), and RS(31,29). Additional codes can be added when required.

The rest of the paper will describe the minimum weight decoding (MWD) algorithm that is used in the decoder in conjunction with channel measurement information to provide an efficient decoding algorithm that makes use of soft-decision information.

RS CODING EMPLOYING CHANNEL MEASUREMENT INFORMATION.

This section of the paper describes a decoding algorithm that utilises channel measurement information, in addition to the conventional use of the algebraic properties of the code to improve the performance of the overall system. A performance increase is achieved by using channel measurement (soft-decision) information to provide a measure of the reliability of each received symbol. This scheme has been applied to a form of RS decoding known as Minimum-Weight decoding [2], which is a low complexity decoding algorithm.

The minimum-weight RS decoding algorithm.

Minimum Weight Decoding (MWD) [2] is a combination of error-trapping, systematic search, trial-and-error and step-by-step decoding. Before the decoding algorithm was implemented it was studied carefully to determine which stages had the greatest effect on the overall decoding time. One such area was the generation of the new syndrome which is equivalent to a long division over a Galois Field, this was replaced by a logical operation by changing the representation of the data. The real advantage of using MWD is its ease of implementation and speed of execution when compared to other RS decoding techniques. As mentioned previously, MWD is a trial and error technique as it reaches a stage where if the error has not been trapped, symbols in the received word are changed arbitrarily. This is the section of the algorithm which can take advantage of the soft information. Instead of using a predetermined sequence to change each symbol in turn, the symbol with the lowest signal-to-noise ratio in the codeword can be replaced with the symbol which was received during the same symbol period but has the second highest signal-to-noise ratio. Thus the trial and error stage has now been replaced with a statistical method for determining which symbols in the received word to replace and when.

PRINCIPLES OF THE CHANNEL MEASUREMENT TECHNIQUE.

The channel measurement information is obtained from the outputs of the correlators being used in the modem. The output of the matched correlator will form the estimate of the received signal level, while the output of the non-matched correlators forms the noise level estimate. The noise and signal level estimates are found by averaging the output of the correlators over the previous symbol interval. The noise estimate is found more accurately by taking the average of the non-matched correlators, while the signal level is found more precisely by subtracting the average noise level from the received signal + noise level. From the signal and noise level estimates it is possible to find an estimate of the channel SNR for an AWGN channel.

The variance of the received signal σ^2, for the matched filter is given by [3]:

$$\sigma^2 = \frac{N_o B^2 T_s}{4}$$

(1)

where B is the receiver bandwidth and N_0 is the one sided power spectral density of the noise component $n(t)$ and is given by:

$$N_0 = \frac{\sigma_s^2}{W} \tag{2}$$

Here $\sigma_s{}^2$ is the variance of the noise, which is equivalent to the averaged noise power, and W is the noise bandwidth and is given by $1/2T$ where T is the sampling rate. Substituting into equation 2 gives the following where T_s is the symbol duration.

$$\sigma^2 = \frac{\sigma_s^2 B^2 T_s T}{2} \tag{3}$$

The channel SNR is equal to a constant over σ^2 and can readily be obtained if required. Once the estimate for the channel SNR has been obtained this information can readily be used by the modified MWD algorithm as stated in the section above.

CONCLUDING REMARKS

The paper describes several important design features of an adaptive MFSK modem and RS codec system, implemented using two DSP PC expansion cards. The host IBM PC-compatible which contains the DSP cards is connected to a commercially available amateur grade HF transceiver. The modem/codec combination exhibits a far greater degree of responsiveness to both user requirements and channel state than previous implementations.

The modem/codec system to be described forms an essential component of an automatic and adaptive HF MFSK transmission system, currently under development by the Hull-Lancaster Communications Research Group in the UK. This system is based upon the use of a flexible digital architecture with both PC and DSP elements.

Results showing the performance of the modem and codec over both static and aeromobile HF channels will be presented at the conference.

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ACKNOWLEDGEMENT

The authors wish to express their thanks to the Defence Research Agency (Aerospace Division) at Malvern for their support of the work detailed in this paper.