WIDEBAND BURST-BY-BURST ADAPTIVE MODULATION WITH TURBO EQUALIZATION AND ITERATIVE CHANNEL ESTIMATION

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Abstract - In this contribution the performance of adaptive modulation - applied in conjunction with Turbo Equalization (TE) - is characterized in a noise limited environment over a slowly varying wideband multi-path Rayleigh fading channel. The iterative structure of the turbo equalizer was also exploited in order to invoke an iterative Least Mean Square (LMS) channel estimator. Finally, the throughput performance of the adaptive modulation scheme was compared to that of its constituent modulation modes, where a gain of 1.5dB to 1.7dB was recorded.

1. INTRODUCTION

Adaptive Quadrature Amplitude Modulation (AQAM) is based on a modulation mode switching regime, in which the modulation modes are selected according to the prevalent channel conditions. When the channel quality is low, lower order modulation modes are selected, in order to improve the mean Bit Error Rate (BER) performance and similarly, higher-order modulation modes are chosen, when the channel quality is favourable, in order to increase the average throughput. Recent AQAM-related advances over narrowband channels have been accomplished amongst others by Webb *et al* [1], Sampei *et al* [2], Goldsmith *et al* [3] and Torrance *et al* [4]. In a wideband environment, the presence of intersymbol interference (ISI) is combated not only by the employment of AQAM, but also by equalization. Hence we utilized the signal to noise plus residual ISI ratio at the output of the equalizer - termed as the pseudo-SNR - in order to switch the modulation modes. Then we compared the throughput of conventional fixed modulation and AQAM schemes at a certain target BER [5]. In this contribution, we will study the performance of AQAM in conjunction with turbo equalization. Turbo equalization [8] was first introduced by C.Douillard, A.Picart, M.Jézéquel, P.Didier, C.Berrou and A.Glavieux in 1995 for a serially concatenated rate $R = 0.5$ convolutional coded BPSK system. Specifically, Douillard *et al* demonstrated that the turbo equalizer was capable of mitigating the effects of Inter-Symbol Interference (ISI) by considering the channel's memory, when performing joint equalization and decoding iteratively.

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The iterative equalization and decoding nature of turbo equalization facilitated the introduction of iterative channel estimation, where the Channel Impulse Response (CIR) estimates are refined and improved after each iteration, in order to obtain more accurate channel estimates. In a practical and realistic AQAM system, the channel quality perceived by the receiver during the current receiver time-slot will differ from the actual channel quality experienced by the next received burst, when the previous channel quality estimate is actually taken into account due to the AQAM mode signalling latency incurred by the system. Hence, the modulation mode selected by the transmitter is not optimum with regards to the actual channel quality. Therefore, the impact of this latency will also be demonstrated in this contribution.

2. SYSTEM OVERVIEW

The schematic of the system is shown in Figure 1 where, the transmitted source bits are convolutionally encoded, interleaved and mapped to a specific AQAM mode selected by the adaptive mapper. The encoder utilized a half-rate recursive systematic convolutional (RSC) code having a constraint length of $K = 5$ and octal generator polynomials of $G0 = 35$ and $G1 = 23$. A random channel interleaver of 4032-bit memory was also implemented. The AQAM mode was selected based on the pseudo-SNR, γ_{dfe}, where the switching methodol-

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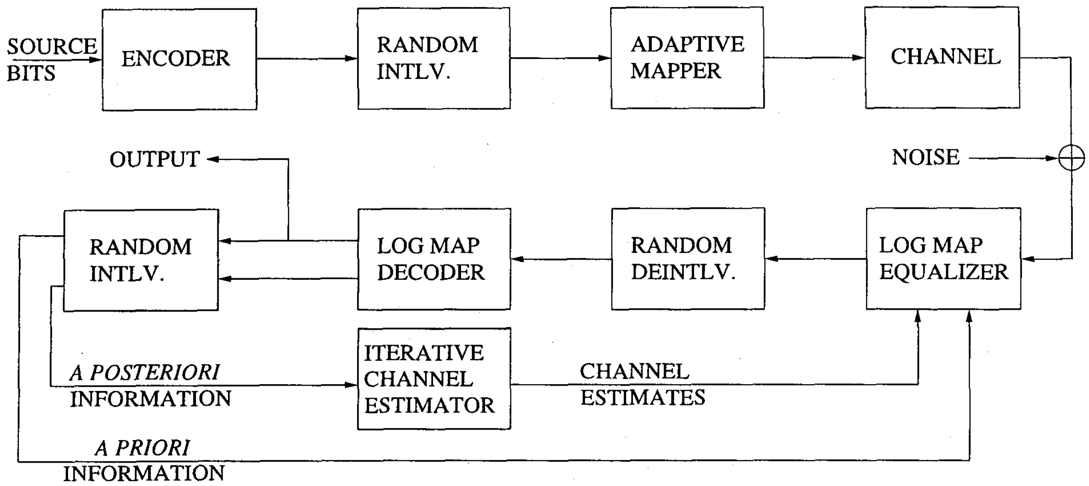

Figure 1: Schematic of the joint turbo equalization and adaptive modulation system, as described in Section 2.

ogy can be summarized as follows[5]:

$$\text{Modulation Mode} = \begin{cases} NOTX & \text{if } \gamma_{dfe} < f_1 \\ BPSK & \text{if } f_1 \le \gamma_{dfe} < f_2 \\ 4QAM & \text{if } f_2 \le \gamma_{dfe} < f_3 \\ 16QAM & \text{if } f_3 \le \gamma_{dfe}, \end{cases} \tag{1}$$

where $f_n, n = 1...3$ are the pseudo-SNR switching thresholds and the NOTX mode is used to temporarily disable transmissions under severely degraded instantaneous channel conditions.

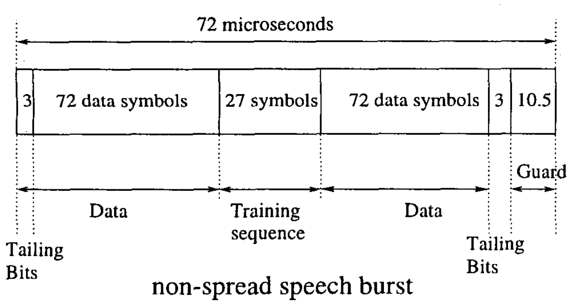

Figure 2: Transmission burst structure of the FMA1 non-spread speech burst of the FRAMES proposal [6]

The transmission burst structure used in this system is the FMA1 non-spread speech burst as specified in the Pan-European FRAMES proposal [6], which is seen in Figure 2. A three-path, symbol-spaced fading channel of equal weights was utilized, where the Rayleigh fading statistics obeyed a normalised Doppler frequency of 3.3615×10^{-5}. At the receiver, the CIR was estimated using the Least Mean Square (LMS) algorithm [7] and the training symbols of the transmission burst shown in Figure 2. The initial step-size of the LMS algorithm was set to 0.05. This initial channel estimate was then utilized by the Soft-In/Soft-Out (SISO) equalizer. We have used the Log-*Maximum A Posteriori* (Log-MAP) algorithm [9] for the SISO equalizer, since the Log-MAP algorithm achieves optimal performance, despite having reduced computational complexity compared to the original *Maximum A Posteriori* (MAP) algorithm [10]. At the next iteration the CIR was re-estimated and refined with a smaller step-size of 0.01 using the entire transmission frame's symbols derived from the *a posteriori* information of the SISO decoder, which also employed the Log-MAP algorithm. This was repeated for each turbo equalization iteration. The *a posteriori* information was transformed from the log domain to modulated symbols using the approach employed in Reference [11].

Since the Log-MAP equalizer did not provide an SNR estimate in order to ascertain the channel quality on a burst-by-burst basis, the output pseudo-SNR of the DFE was utilized as the AQAM mode switching criterion [5]. In order to justify its utilization, the channel quality, which was quantified in terms of the output pseudo-SNR of the DFE must also indicate the performance of the Log-MAP equalizer in terms of the BER. Consequently, we evaluated the correlation between the number of erroneous decisions produced by the Log-MAP equalizer and the output pseudo-SNR of the DFE on a burst-by-burst basis. This is shown in Figure 3, where a 4QAM mode was utilized at a channel SNR of 8dB over a symbol-spaced, equal-weight three-path fading channel. Referring to this figure, the error

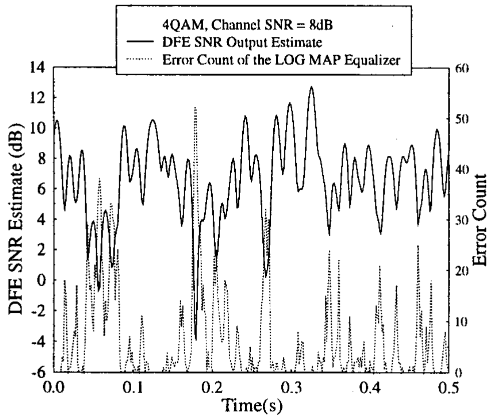

Figure 3: Variation of the number of errors per AQAM burst using the burst format of Figure 2 produced by the Log-MAP equalizer and the corresponding output pseudo-SNR estimate of the DFE using a 4QAM mode at a channel SNR of 8dB over a symbol-spaced, equal-weight three-path fading channel.

count of the Log-MAP equalizer displayed a good correlation with the output pseudo-SNR of the DFE, where the error count increased whenever the output SNR of the DFE decreased and vice versa. Consequently, we can justify the utilization of the output pseudo-SNR of the DFE as an AQAM mode switching criterion in the Log-MAP equalizer.

3. FIXED MODULATION PERFORMANCE

The BER performance of the BPSK, 4QAM and 16-QAM modes is shown in Figures 4, 5 and 6, respectively, where a total of four turbo equalization iterations were implemented. In each of these figure, the performance of the system employing perfect CIR estimation and four turbo equalization iterations was also depicted for comparisons. As it can be seen from the respective figures, the performance of the iterative CIR estimator approached that of the perfect CIR estimator upon employing four turbo equalization iterations. This was as a result of the iterative nature of the turbo equalization, where the reliability of the decoded information increased for every subsequent iteration. Consequently, the iterative channel estimator exploited the increased reliability of the decoded symbols in order to yield an improved channel estimate for the Log-MAP equalizer. This will then yield a more reliable output from the Log-MAP equalizer. In Figure 4, the BPSK iteration gain - *i.e* the gain in SNR performance with respect to the first iteration - achieved after the fourth iteration was approximately 1dB at BER= 10^{-4}, while the iteration gain observed in Figure 5, for the 4QAM modulation mode after four iterations was approximately 2dB at BER= 10^{-4}.

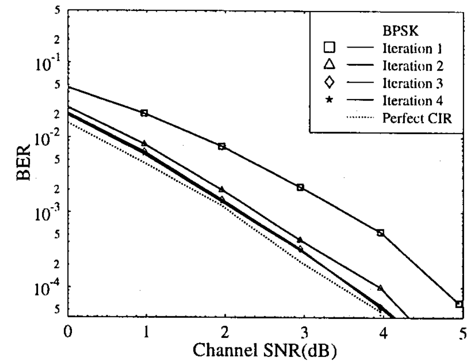

Figure 4: BER performance of the **BPSK** fixed modulation mode over the three-path, equal-weight, and symbol-spaced fading channel using imperfect channel estimation, turbo equalization after four iterations and the transmission burst structure of Figure 2.

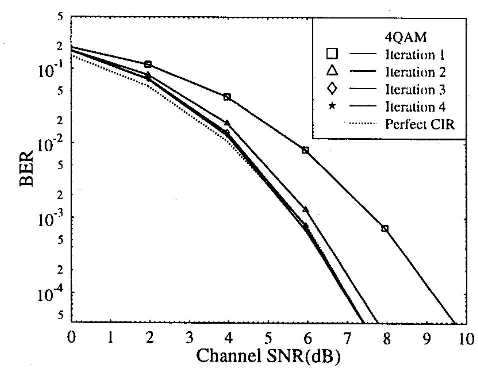

Figure 5: BER performance of the **4QAM** fixed modulation mode over the three-path, equal-weight, and symbol-spaced fading channel using imperfect channel estimation, turbo equalization after four iterations and the transmission burst structure of Figure 2.

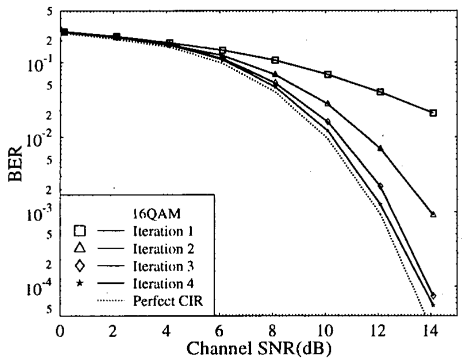

Figure 6: BER performance of the **16QAM** fixed modulation mode over the three-path, equal-weight, and symbol-spaced fading channel using imperfect channel estimation, turbo equalization after four iterations and the transmission burst structure of Figure 2.

4. AQAM PERFORMANCE

In this section, we will analyse the BER and Bits Per Symbol (BPS) performance of the joint turbo equalization and half-rate, K=5 RSC-coded AQAM system. In this joint scheme, the switching thresholds, f_n of Equation 1 were chosen in order to achieve a target BER of approximately 0.01%. The switching thresholds were set experimentally as follows: $f_1 = -1.5$dB, $f_2 = 2.5$dB and $f_3 = 6.5$dB. In our forthcoming experiments the modulation mode of the next transmitted AQAM burst was selected based on the channel quality of the previous received AQAM burst, which corresponded to a delay of one TDMA frame or 16 time-slots. This scenario represents a practical burst-by-burst adaptive QAM scheme. Finally, the iterative LMS CIR estimator was also implemented, in order to provide a realistic joint turbo equalization and AQAM system.

The BER and BPS performance of the amalgamated scheme is shown in Figures 7 and 8, respectively. In Figure 7 the AQAM upper-bound performance is also shown for comparison, where perfect CIR estimation, a zero delay in channel quality estimation and four turbo equalization iterations were utilized. Referring to Figure 7, the approximate target BER of 0.01% was achieved and maintained after four iterations. Furthermore, the performance of the practical one TDMA-frame delay AQAM system approached that of the upper-bound scenario. The BPS throughput performance shown in Figure 8 highlighted the switching mechanism of the AQAM system, where as the channel SNR increased, the higher order AQAM modes were chosen

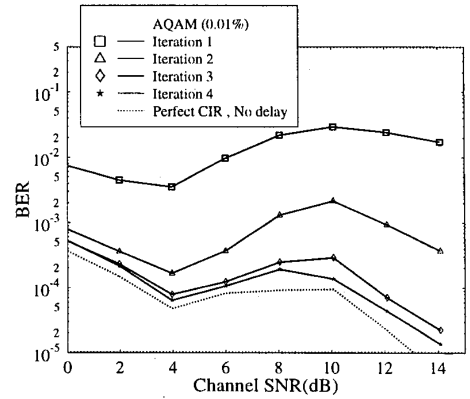

Figure 7: **BER** performance of the joint **AQAM** and turbo equalization scheme over the three-path, equal-weight, and symbol-spaced fading channel using imperfect CIR estimation, turbo equalization after four iterations and the transmission burst structure of Figure 2. The AQAM system was subjected to a realistic delay of one TDMA frame between channel quality estimation and mode switching. The zero delay scenario with perfect channel estimation after four turbo equalization iterations is also shown for comparison.

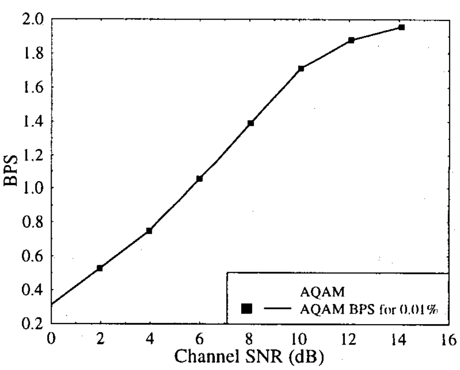

Figure 8: **BPS** performance of the joint **AQAM** and turbo equalization scheme over the three-path, equal-weight, and symbol-spaced fading channel using imperfect channel estimation, turbo equalization after four iterations and the transmission burst structure of Figure 2.

more frequently due to the improving channel quality. Consequently, the BPS throughput was higher as the channel SNR was increased. The performance of the fixed modulation schemes was also compared to that of AQAM in terms of the channel SNR required, in order to achieve a certain BPS throughput for a target BER of approximately 0.01%. The results were compiled from Figures 4, 5, 7 and 8 and tabulated in Table 1. An SNR gain of approximately 1.5dB to 1.7dB was recorded for the AQAM scheme at a BPS of 0.5 and 1.0, corresponding to the half-rate, K=5 RSC-coded throughput of BPSK and 4QAM, respectively.

	Required Channel SNR (dB)	
	BPS = 0.5	BPS = 1.0
Fixed Schemes	3.5	7.0
AQAM	1.8	5.5

Table 1: The approximate channel SNRs required, in order to achieve a Bits Per Symbol (BPS) throughput of **0.5** and **1.0** for the fixed modulation mode system and the amalgamated AQAM and turbo equalization system. The required channel SNRs were extracted for an approximate BER of 0.01% from Figures 4, 5, 7 and 8.

5. CONCLUSION

In this paper, we have introduced a practical and realistic AQAM system, which utilized turbo equalization at the receiver. In this respect, even with a one TDMA frame delay between the estimation of the channel quality and AQAM mode switching, the system exhibited only a slight performance degradation, when compared to the upper-bound performance - where perfect CIR estimation, a zero delay in channel quality estimation and four turbo equalization iterations were utilized - as evidenced by Figure 7. We have also exploited the iterative nature of the turbo equalizer by invoking an iterative LMS CIR estimator. Its performance was characterized in Figures 4, 5, 6 and 7, where the performance approached that of the perfect CIR estimation scenarios. Finally the BPS throughput was recorded in Table 1, where the AQAM scheme exhibited a gain of approximately 1.5dB to 1.7dB, when compared to that of the coded BPSK, 4QAM and 16QAM fixed modulation benchmarkers. Currently, we are studying a range of techniques, reducing the complexity of the turbo equalizer, in order to accommodate longer channel delays and higher order AQAM modes.

Acknowledgement

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