TURBO-EQUALISED H.263-BASED VIDEO TELEPHONY FOR GSM/GPRS

P. Cherriman, B. L. Yeap, L. Hanzo

Dept. of ECS, Univ. of Southampton, SO17 1BJ, UK.
Tel: +44-703-593 125, Fax: +44-703-594 508
Email: `lh@ecs.soton.ac.uk`
`http://www-mobile.ecs.soton.ac.uk`

ABSTRACT

The performance of turbo-equalised GSM/GPRS-like videophone transceivers is studied over dispersive fading channels as a function of the number of turbo-equalisation iterations. Iteration gains in excess of 4 dB were attained. The proposed system is capable of providing low-resolution videophone services over GSM/GPRS.

1. SYSTEM PARAMETERS

In this contribution a GSM/GPRS-based videophone scheme is proposed and characterised. The associated system parameters are summarised in Table 1, while the system's schematic is portrayed in Figure 1. An advanced feature of the system is that it employs joint channel decoding and channel equalisation, which is referred to as turbo equalisation. The system uses the GSM frame structure [1], and the COST-207 Hilly Terrain (HT) channel model characterised in Figure 2. Each transmitted packet is interleaved over two GSM TDMA frames, in order to disperse burty errors.

The GPRS system allows the employment of multiple timeslots per user. We studied both a GSM-like system using 1 slot per TDMA frame, and a GPRS-like arrangement with 4 slots per TDMA frame, a scenario where the user is assigned half the maximum capacity of an 8-slot GPRS/GSM carrier. The bitrates associated with 1 and 4 slots per TDMA frame are shown in Table 2.

The effective video bitrates that can be obtained in conjunction with half-rate convolutional coding are 10 and 47.5Kbit/s for the 1 and 4 slots per TDMA frame scenarios, respectively. Again, the system's schematic is shown in Figure 1.

2. OVERVIEW OF TURBO-EQUALISATION

Turbo equalisation [2] was proposed by Douillard, Picart, Jézéquel, Didier, Berrou and Glavieux in 1995 for a serially concatenated rate $R = \frac{1}{2}$ convolutional-coded Binary Phase Shift Keying (BPSK) system. Specifically, Douillard *et al.* demonstrated that the turbo equaliser was capable of mitigating the effects of Inter-Symbol Interference (ISI), provided that the channel impulse response (CIR) is known. Instead of performing the equalisation and error correction decoding independently, better performance can be achieved by considering the channel's memory, when performing joint equalisation and decoding iteratively. Gertsman and Lodge [3] then showed that the iterative process of turbo equalisation can compensate for the degradations due to imperfect channel estimation.

Simulation Parameters	
Channel Model	COST-207 Hilly Terrain
Carrier Frequency	900MHz
Vehicular Speed	30mph
Doppler Frequency	40.3Hz
Modulation	GMSK, $B_n = 0.3$
Channel Coding	Convol.(n,k,K) = (2,1,5)
Octal Generator Polynomials	23, 33
Channel Interleavers	Random (232, 928)
Turbo-Coding Interleavers	Random (116, 464)
Max Turbo-Equaliser Iterations	10
No. of TDMA Frame per Packet	2
No. of Slots per TDMA Frame	1, 4
Convolutional Decoder Algorithm	LOG-MAP
Equaliser Algorithm	LOG-MAP

Table 1: System Parameters

Different iteration termination criteria [4] – such as the so-called cross-entropy [5] – were also investigated in order to minimise the number of iteration steps for the turbo equaliser. Turbo equalisation was also proposed by Bauch and Franz [6] for the Global System of Mobile Communications – known as GSM – where different approaches were investigated for overcoming the dispersion of the so-called *a priori* information due to the interburst interleaving scheme used in GSM. Further research into combined turbo coding using convolutional constituent codes and turbo equalisation has been conducted by Raphaeli and Zarai [7].

The basic philosophy of the original turbo equalisation technique stems from the iterative turbo decoding algorithm consisting of two Soft-In/Soft-Out (SISO) decoders, a structure, which was proposed by Berrou *et al.* [8]. Before proceeding with our in-depth discussions, let us briefly define below the terms *a priori*, *a posteriori* and extrinsic information, which we employ throughout this treatise.

A priori The *a priori* information associated with a bit v_m is the information known before equalisation or decoding commences, from a source other than the received sequence or the code constraints. The apriori information is also often referred to as intrinsic information, in order to contrast it with the extrinsic information, which is described next.

Extrinsic The extrinsic information associated with a bit v_m is the information provided by the equaliser or decoder based on the received sequence and on the *a priori* information of all bits with the exception of the received and *a priori* information explicitly related to that particular bit v_m.

A posteriori The *a posteriori* information associated with a bit

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Bitrates etc		
Slots/TDMA frame	1	4
Coded Bits/TDMA slot	116	116
Data Bits/TDMA slot	58	58
Data Bits/TDMA frame	58	232
TDMA frame/packet	2	2
Data Bits/packet	116	464
Packet Header (bits)	8	10
CRC (bits)	16	16
Video Bits/packet	92	438
TDMA frame length	4.615ms	4.615ms
TDMA frames/s	216.68	216.68
Video Packets per sec	108.34	108.34
Video bitrate (kbps)	10.0	47.5
Video framerate (fps)	10	10

Table 2: Summary of System-specific Bitrates

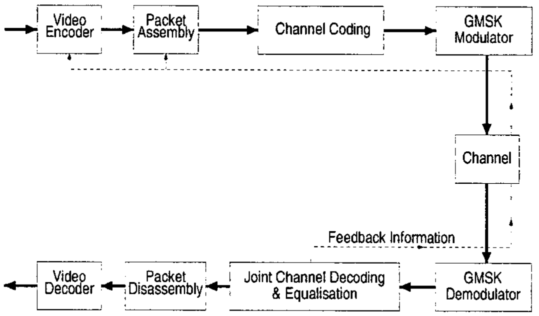

Figure 1: System schematic for turbo-equalised video system.

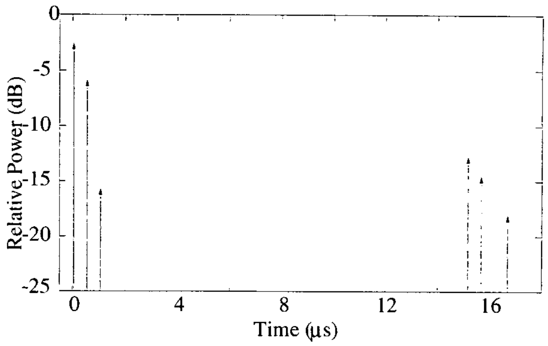

Figure 2: COST207-Hilly Terrain Channel Model

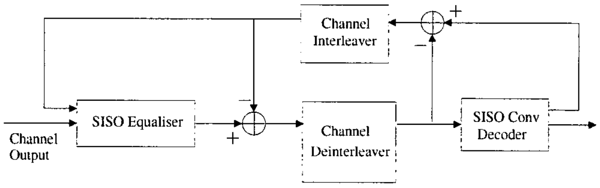

Figure 3: Structure of original turbo-equaliser introduced by Douillard *et al.* [2].

is the information that the SISO algorithm provides taking into account *all* available sources of information about the bit u_k.

As mentioned previously, the turbo equaliser of Figure 3 consists of a SISO equaliser and a SISO decoder. The SISO equaliser in Figure 3 generates the *a posteriori* probability upon receiving the corrupted transmitted signal sequence and the *a priori* probability provided by the SISO decoder. However, at the initial iteration stages – *i.e.* at the first turbo equalisation iteration – no *a priori* information is supplied by the channel decoder. Therefore, the *a priori* probability is set to $\frac{1}{2}$, since the transmitted bits are assumed to be equiprobable. Before passing the *a posteriori* information generated by the SISO equaliser to the SISO decoder of Figure 3, the contribution of the decoder – in the form of the *a priori* information – accruing from the previous iteration must be removed, in order to yield the combined channel and extrinsic information. This also minimises the correlation between the *a priori* information supplied by the decoder and the *a posteriori* information generated by the equaliser. The term 'combined channel and extrinsic information' indicates that they are inherently linked – in fact they are typically induced by mechanisms, which exhibit memory – and hence they cannot be separated. The removal of the *a priori* information is necessary, in order to prevent the decoder from "reprocessing" its own information, which would result in the so-called 'positive feedback' phenomenon, overwhelming the decoder's current reliability-estimation of the coded bits, *i.e.* the extrinsic information.

The combined channel and extrinsic information is channel-deinterleaved and directed to the SISO decoder, as depicted in Figure 3. Subsequently, the SISO decoder computes the *a posteriori* probability of the coded bits. Note that the latter steps are different from those in turbo decoding, which only produces the *a posteriori* probability of the source bits, rather than those of all channel coded bits. The combined deinterleaved channel and extrinsic information is then removed from the *a posteriori* information provided by the decoder in Figure 3 before channel interleaving, in order to yield the extrinsic information. This is to prevent the channel equaliser from receiving information based on its own decisions, which was generated in the previous turbo equalisation iteration. The extrinsic information computed is then employed as the *a priori* input information of the equaliser in the next channel equalisation process. This constitutes the first turbo equalisation iteration. The iterative process is repeated, until the required termination criteria are met [4]. At this stage, the *a posteriori* information of the source bits, which has been generated by the decoder is utilised to estimate the transmitted bits.

In our investigations we have employed convolutional coding for the proposed turbo-equalised GSM-like video system, since it has been shown in reference [9] that convolutional-coded GMSK systems are capable of providing large iteration gains – *i.e.* gains in SNR performance with respect to the first iteration – using successive turbo equalisation iterations. Furthermore, our turbo equalisation experiments have shown that the rate $R = 0.5$ convolutional-coded GMSK system outperformed the rate $R = 0.5$ convolutional-coding based turbo-coded GMSK scheme. Motivated by these trends, we propose a convolutional-coded GSM-like videphone system, which employs turbo equalisation, in order to enhance the video performance of the system. For the purpose of benchmarking, we have also simulated the turbo-equalised GSM-like system employing convolutional-coding based turbo codes.

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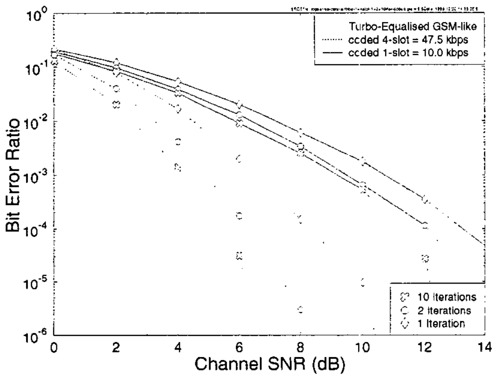

Figure 4: BER versus channel SNR for 1 and 4 slots per TDMA frame, and for 1, 2 and 10 turbo-equaliser iterations, over the channel of Figure 2.

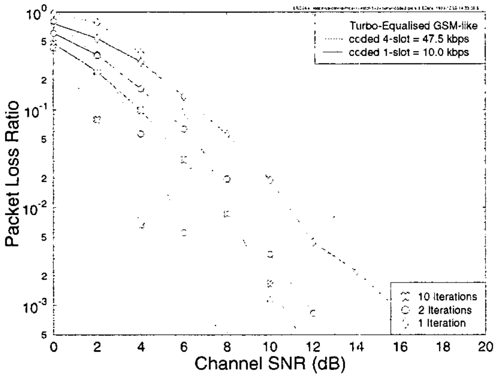

Figure 5: Video PLR versus channel SNR for 1 and 4 slots per TDMA frame, and for 1, 2 and 10 turbo-equaliser iterations, over the channel of Figure 2.

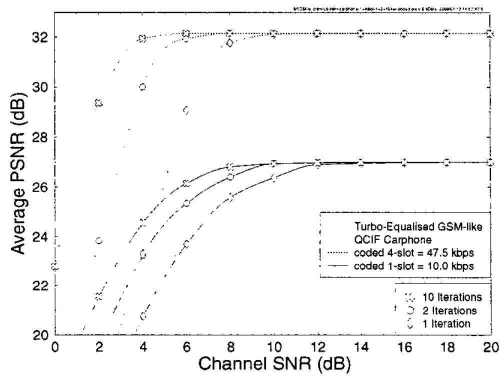

Figure 6: Video quality in PSNR (dB) versus channel SNR for 1 and 4 slots per TDMA frame, and for 1, 2 and 10 iterations of the turbo-equaliser upon using the highly motion active "Carphone" video sequence over the channel of Figure 2.

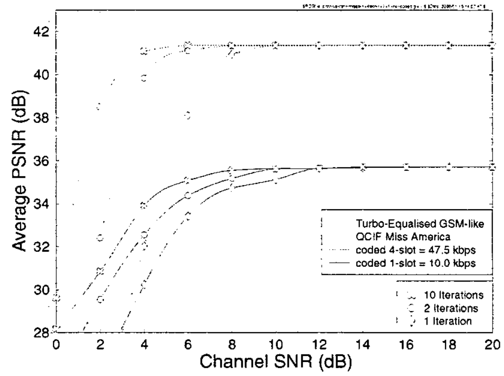

Figure 7: Video quality in PSNR (dB) versus channel SNR for 1 and 4 slots per TDMA frame, and for 1, 2 and 10 turbo-equaliser iterations, using the low-activity "Miss America" video sequence over the channel of Figure 2.

3. TURBO-EQUALISATION RESULTS

Let us now characterise the performance of our system. Figure 4 shows the bit error ratio (BER) versus channel SNR for the 1- and 4-slot scenarios, after 1, 2 and 10 iterations of the turbo equaliser. The figure shows the BER performance improvement upon each iteration of the turbo-equaliser, although there is only a limited extra performance improvement after 5 iterations. The figure also shows that the 4-slot scenario has a lower bit error ratio, than the one slot scenerio. This is because the 4-slot scenario has a longer interleaver, which renders the turbo-equalisation process more effective due to its increased time-diversity.

Let us now consider the packet loss ratio (PLR) versus channel SNR performance in Figure 5. The PLR is a more pertinent measure of the expected video performance, since our video scheme discards all video packets, which are not error-free. Hence our goal is to maintain as low a PLR, as possible. Observe in Figure 5 that the associated iteration gains are more pronounced in terms of the PLR, than in bit error ratio.

It should further be noted that for low SNRs the packet loss performance of the 4-slot system is inferior to that of the 1-slot system, while the bit-error ratio is similar or better at the same SNRs. This is because the probability of having a single bit error in the 4-slot video packet is higher due to its quadruple length. This phenomenon will be further augmented in Section 3.2.

3.1. Video Performance

The video quality of our system is directly related to its PLR performance. Figure 6 shows the associated average PSNR versus channel SNR performance, demonstrating that an improved video quality can be maintained at lower SNRs, as the number of iterations increases. Additionally, the higher bitrate of the 4-slot system corresponds to a higher overall video quality. Explicitly, up to 6dB SNR-gain can be achieved after 10 turbo-equalisation iterations, as seen in Figure 6.

Figure 6 characterised the performance for the highly motion-active "Carphone" sequence. However, the achievable performance improvements are similar for the low-activity "Miss America" video sequence, as seen in Figure 7. Observe, furthermore that the lower-

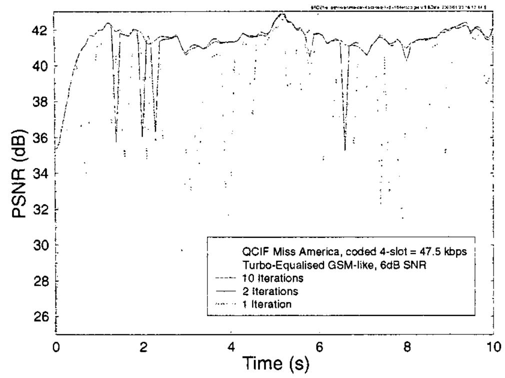

Figure 8: Video quality in PSNR (dB) versus time using 4 slots per TDMA frame, and for 1, 2 and 10 iterations of the turbo-equaliser for the low-activity "Miss America" video sequence.

Channel SNR for 1dB loss of PSNR		
Slots/TDMA frame	1	4
1 Iteration	9.0dB	7.5dB
2 Iterations	7.2dB	5.2dB
3 Iterations	6.44dB	3.9dB
4 Iterations	6.37dB	3.7dB
10 Iterations	5.8dB	3.4dB

Table 3: Minimum required operating channel SNR for the QCIF Carphone sequence over the channel of Figure 2.

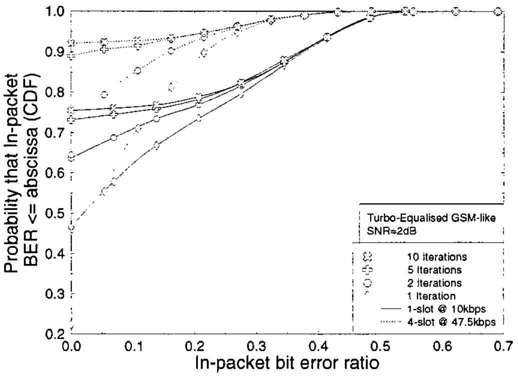

Figure 9: CDF of the "in-packet" BER at a channel SNR of 2dB over the channel of Figure 2, and for various number of iterations for the turbo-equalised 1- and 4-slot systems.

activity video sequence is represented at a higher video quality, given the same video bitrate. A deeper insight into the achievable video quality improvement in conjuction with turbo-equalisation can be provided by plotting the video quality measured in PSNR (dB) versus time, as seen in Figure 8 for the "Miss America" video sequence using the 4-slot system at a channel SNR of 6dB for 1, 2 and 10 iterations of the turbo equaliser.

Specifically, the bottom-trace of the figure shows how the video quality varies in the one-iteration scenario, which is equivalent to conventional equalisation. The sudden reductions in video quality are caused by packet-loss events, which result in parts of the picture being "frozen" for one or possibly several consecutive video frames. The sudden increases in video quality are achieved, when the system updates the "frozen" part of the video picture in subsequent video frames. The PLR for this scenario was 10%.

The video quality improved significantly with the aid of two turbo-equaliser iterations, while the PLR was reduced from 10% to 0.7%. During the time interval shown in Figure 8 there are eight lost video packets, six of which can be seen as sudden reductions in video quality. However, in each case the video quality recovered with the update of the "frozen" picture areas in the next video frame.

We have found that the maximum acceptable PSNR video-quality degradation was about 1dB, which was associated with near-unimpaired video quality. In Table 3 hence we tabulated the corresponding minimum required channel SNRs that the system can operate at for a variety of scenarios, extracted from Figure 6. As it can be seen from the table, the minimum operating channel SNR for the 1- and 4-slot system using one iteration is 9 and 7.5dB, respectively. This corresponds to a system using conventional equalisation. A system using two turbo-equalisation iterations can reduce these operating SNRs to 7.2dB and 5.2dB, respectively. The minimum operating SNRs can be reduced to as low as 5.8dB and 3.4dB for the 1- and 4-slot systems, respectively, when invoking 10 iterations.

3.2. Bit error statistics

In order to more explicity demonstrate the benefits of turbo-equalisation, we investigated the mechanism of how turbo-equalisation reduces the bit-error and packet loss ratios. We found that the distribution of the bit-errors in video packets after each iteration provided us with interesting insights. Hence the Cummulative Density Function (CDF) of the number of bit-errors per video packet was evaluated. In order to allow a fair comparison between the 1- and 4-slot system, we normalised the number of bit-errors per packet to the video packet size, hence producing the CDF of "in-packet" BER.

Figure 9 shows the CDF of the "in-packet" BER for a channel SNR of 2dB and for 1, 2, 5 and 10 iterations, for both the 1- and 4-slot systems. It should be noted that the value of the CDF for an "in-packet" BER of zero is the probability that a packet is error-free, and hence can be interpreted as the packet success ratio (PSR). The packet loss ratio is equal to one minus the PSR. For example, the 4-slot system characterised in Figure 9 has a packet success ratio of 0.22 after one iteration, which corresponds to a PLR of 78%.

Both the 1- and 4-slot system increase the PSR as the number of iterations increases. For example, the 4-slot system increases the PSR from 22% to 92%, as the number of iterations is increased from 1 to 10. This corresponds to a PLR reduction from 78% to 8%. However, the CDF of "in-packet" BER can provide further insight into the system's operation. It can be seen in Figure 9 that the turbo iterations reduce the number of packets having "in-packet" BERs of less than 30%. However, the probability of a packet having an "in-packet" BER higher than 35% is hardly affected by the number of iterations, since the number of bit errors is excessive, hence overwhelming the turbo equaliser, despite its powerful error correction capability.

By referring to Figures 4 and 5, it can be seen that the four-slot system always has a lower BER, than the 1-slot system, although at low SNRs the PLR is higher for the 4-slot system. The CDF of Figure 9 assists in interpreting this phenomenon further. Ex-

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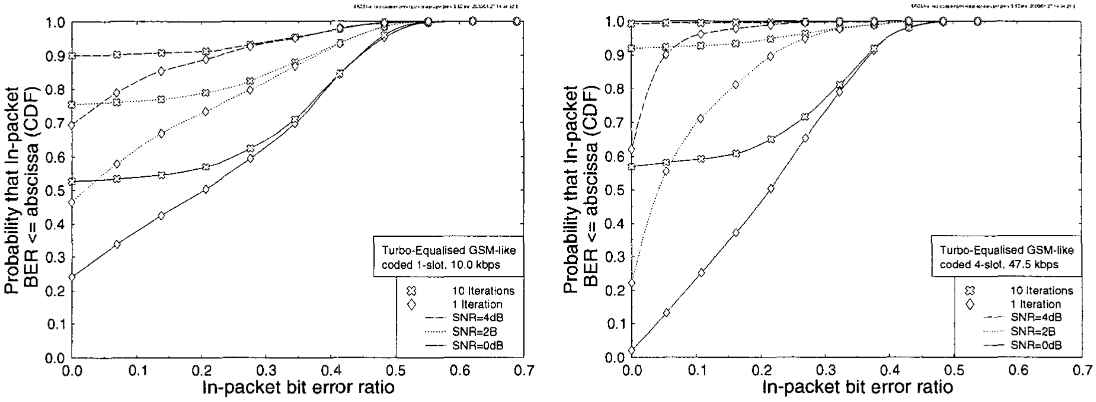

(a) 1-slot

(b) 4-slot

Figure 10: CDF of "in-packet" BER performance over the channel of Figure 2 for the turbo-equalised 1- and 4-slot system for channel SNRs of 0dB, 2dB and 4dB and 1 and 10 iterations.

plicitly, the CDF shows that the PSR improves more significantly for the 4-slot system, than for the 1-slot system, as the number of iterations increases. This is because the 4-slot system allows the employment of a longer interleaver, thereby improving the efficiency of the turbo-equaliser. However, the CDF also underlines a reason for the lower BER of the 4-slot system across the whole range of SNRs, demonstrating that the probability of packets having a high "in-packet" bit error rate is lower for the 4-slot system. Since packets having a high "in-packet" BER have a more grave effect on the overall BER, than those packets having a low "in-packet" BER, this explains the inferior overall BER performance of the 1-slot system.

Figure 10 shows the CDF of "in-packet" BER for conventional equalisation as well as with the aid of ten turbo-equaliser iterations for 0dB, 2dB and 4dB channel SNRs. Figure 10(a) represents a 1-slot system, while Figure 10(b) a 4-slot system. The figures also show the PLR performance improvements achieved with the aid of turbo-equalisation. Having studied the performance of our convolutionally coded turbo equalised system we then attempted to improve the system's performance upon invoking a more powerful turbo channel codec. Our expectation was that the system's performance may be further increased, since it employed a more complex turbo equaliser scheme. Surprisingly, no substantial further performance increase was achieved upon invoking turbo-coded turbo-equalisation.

4. SUMMARY AND CONCLUSIONS

The performance of turbo-equalised GSM/GPRS-like videophone transceivers was studied over dispersive fading channels as a function of the number of turbo-equalisation iterations. Iteration gains in excess of 4 dB were attained, although the highest per-iteration gain was achieved for iteration indices below five. As expected, the longer the associated interleaver, the better the BER and PLR performance. In contrast to our expectations, the turbo-coded, turbo-equalised system was outperformed by the less complex convolutional-coded turbo-equalised system. In conclusion, GPRS/GSM are amenable to video telephony and turbo equalisation is a powerful means of improving the system's performance. Our future work will improve the system's performance invoking the MPEG4 video codec and space-time coding.

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