Abstract: In this paper we have considered using turbo channel coding to encode the 8 kbits/s output from the G.729 speech codec. The source and channel coded bits are then transmitted using a wideband Orthogonal Frequency Division Multiplexing (OFDM) system in the framework of the Mode-I FRAMES proposals [1]. We illustrate the benefits of using OFDM with channel coding to alleviate some of the problems associated with wideband fading channels. Furthermore, we discuss how OFDM can be used in conjunction with the G.729 speech codec and half rate channel coding in order to utilise one speech/data FRAMES sub-burst. Finally some of the issues and problems associated with using turbo coded OFDM in speech transmission systems are considered using the system characterised in Table 1. In Figure 6 a channel SNR of 6dB appears sufficiently high under the stipulated system conditions for near-unimpaired speech transmission.

1 Introduction

The European Research in Communications Equipment (RACE) project endeavoured to comparatively study Time Division Multiple Access (TDMA) and Code Division Multiple Access (CDMA) for the third generation Universal Mobile Telecommunications System (UMTS). Many sophisticated techniques have been proposed and the Advanced TDMA (ATDMA) framework emerged as an attractive candidate for supporting a variety of services requiring different bit-rates. Although the ATDMA proposal gives cognizance to the different expected delay-spread constraints imposed by the various propagation environments, it is inevitable that in some cases the prevailing dispersion will exceed the expected values, which results in serious Bit Error Rate (BER) degradation in conventional serial modems if no equaliser is used. The European FRAMES consortium [1] is advocating a hybrid air-interface for their UMTS proposal, which may incorporate the so-called Orthogonal Frequency Division Multiplex (OFDM) system in its modulation tool-box. Hence, as an attractive design alternative, it is beneficial to study the applicability of OFDM in conjunction with sophisticated channel coding and G.729 speech coding for the emerging UMTS.

In this paper we have studied the use of OFDM in a FRAMES like system. The recently standardised G.729 speech codec is used to encode narrow-band speech at 8 kbits/s, and the encoded speech bits are channel coded using turbo coding. The various components of our proposed system are detailed in Section 2, and the results of our simulations are presented in Section 3.

2 System Description

2.1 System Overview

The system model employed in this paper is depicted in Figure 1. At the transmitter, a G.729 speech coder generates data packets of 80 bits per 10 ms from a speech file, and this speech data is encoded by a half-rate channel encoder. The encoded bits are modulated by a Quadrature Phase Shift Keying
(QPSK) modulator and the resulting signals are transmitted using an OFDM modem to the receiver. During transmission, the signal is corrupted in the frequency-selective time-varying channel, and white Gaussian noise is added at the receiver's input stage. At the receiver, the OFDM signal is demultiplexed and demodulated, and the resulting bits are passed to the channel decoder. The received bits are decoded by the G.729 decoder, and the segmental signal-to-noise ratio (SEGSNR) degradation of the recovered speech is evaluated.

2.2 The G.729 Speech Codec

The G.729 speech codec [2, 3] was standardised by the International Telecommunications Union (ITU) in 1995. It provides speech quality equivalent to that of 32 kbits/s G.726 ADPCM, but at a bit rate of only 8 kbits/s. It provides good toll quality speech even after three tandems, and its speech quality is only slightly degraded by a random bit error rate (BER) of 0.1%. This good performance is achieved using Conjugate-Structure Algebraic CELP (CS-ACELP) with a 10 ms frame-length. The codec is described in detail in [2]. More recently a reduced complexity version, called G.729 Annex A [3], has been standardised. This provides nearly as good speech quality as G.729, but with a complexity of only about 12 MIPS, which is about half that of G.729. Because of their good speech quality, robustness to random bit and frame errors and due to their relatively low delay and complexity requirements it seems likely that both G.729 and G.729A will become widely used. For this reason we have chosen to use G.729 as the speech codec in our system. Let us now consider the proposed channel codec.

2.3 Turbo Channel Encoding

Turbo coding is a novel form of channel coding, which is reported to produce excellent results [4]. The information sequence is encoded twice, with an interleaver between the two encoders serving to make the two encoded data sequences approximately statistically independent of each other. In our simulations we have used half rate Recursive Systematic Convolutional (RSC) encoders, but turbo coding is also possible with other constituent codes [5]. Each constituent RSC encoder produces a systematic output, which is equivalent to the original information sequence, as well as a stream of parity information. The two parity sequences are then punctured before being transmitted along with the original information sequence to the decoder. This puncturing of the parity information allows a wide range of coding rates to be realised. We have chosen to use the commonly adopted scheme of sending alternative parity bits from each encoder. Along with the original data sequence this results in an overall coding rate of 1/2.

The original, near-Shannonian, performance results for turbo codes were achieved using a very long block length \( L \) of 65,536 bits. It is well known that the performance of the codes decreases as the frame length \( L \) decreases, but that good performance is still achievable with relatively short frame lengths. It is also well known that the design of the interleaver used within the turbo coder has a vital influence on its performance. For long frame lengths random interleavers are used, but for shorter frame lengths of 100 or 200 bits, such as in a speech transmission system, Jung and Naflhan [6] reported that block interleavers should be used. However Jung and Naflhan used a 12x16 block interleaver in their work, while we have found that block interleavers with an odd number of rows and columns significantly out perform those with an even number of rows or columns. This is because, as noted by Barbulescu and Pietrobon [7], with an odd number of rows and columns the odd and even data bits are kept separated. When alternate puncturing from each constituent encoder is used, as it most often is, this ensures that
for each information bit one and only one parity bit is transmitted. This "odd-even" separation improves
the performance of the turbo code [7], especially for short frame length systems in our experience.

As mentioned above, the G.729 speech codec provides 80 coded speech bits per 10 ms frame. All our
simulations have used two constraint-length three RSC constituent encoders, with generator polynomials
expressed in octal form as 7 and 5. The Maximum A Posteriori (MAP) [8] algorithm has been used with
8 decoding iterations. For each 10 ms G.729 frame to be turbo encoded separately we need to convey
80 information bits, plus two bits for trellis termination, where the number of trellis terminating bits
required to flush the encoder's shift-register corresponds to the number of shift-register stages in the
encoder. This gives a required interleaver length of 82, which is very close to the interleaver length of 81
given by a 9x9 square interleaver.

For BER comparisons we have simulated systems using both a square L = 81 interleaver, which can
transport 79 data bits per turbo coded frame, and a system with an L = 82 interleaver. Because of
the known benefit of using block interleavers for short frame transmission systems [6] we generated this
length-82 interleaver by merely copying the elements of a square 81 interleaver, and leaving the final
additional element in the L = 82 interleaver un-interleaved. We have also simulated a system with
L = 169, using a 13 x 13 square interleaver. As described later this turbo encoder is used to code the
160 bits from two 10 ms G.729 frames. Finally, in order to characterise the near-optimum performance
that can be achieved with turbo coding, we have simulated a system using a random interleaver with
L = 10,000. Naturally, such an encoder could not be used for speech systems because of the delay it
would introduce, but it may be useful for video or data transmission. Let us now consider the frame-structure
of the proposed system.

2.4 OFDM in the FRAMES Speech/Data Sub-Burst

The emerging UMTS standard will have to accommodate a wide range of user profiles and data rates.
The Advanced Communications Technologies and Services (ACTS) programme's FRAMES project [1]
aims to propose such a system, incorporating a wide variety of possible system parameters. For these
experiments, the FRAMES Mode 1 Speech/Data sub-burst was chosen, offering sufficient data bandwidth
for half-rate coded speech transmission. Figure 2 shows the timing of the frame and the chosen time
slot, where the frame and the Speech/Data sub-burst durations are 4.615 µs and 72.1 µs, respectively,
and the channel symbol rate is 1.3 MHz. Originally, the FRAMES proposal specified Offset-QPSK as
the modulation scheme in these slots, leading to a channel bit rate of 2.6 Mbits/s.

The FRAMES Speech/Data sub-burst offers a convenient environment for 64 sub-carrier OFDM
transmission, as is demonstrated in Figure 2. The 64 data samples of the OFDM symbol are preceded
by a 24-sample cyclic extension, which allows operation in wideband channels with an impulse response
length of up to 24 samples or 18.5 µs without inter-burst interference. Let us now consider our wide-band
channel model in the next Section.

2.5 Channel model

All experiments were conducted utilising the COST207 bad urban (BU) compliant impulse response [9].
The continuous COST207 BU impulse response was discretised to a seven-path model exhibiting a delay
spread of 2.45 µs and a maximum delay of 7.7 µs, as seen in Figure 3. Each of the paths constituting the impulse response was faded independently, employing a Rayleigh fading channel. The carrier frequency and the vehicular velocity were set to 2 GHz and 50 km/h, respectively, which leads to a Doppler frequency of 92.6 Hz for the Rayleigh channel. The normalised Doppler frequency is therefore $6.7664 \cdot 10^{-5}$.

The magnitude of the resulting time- and frequency-variant channel transfer function for a duration of 200 frames or 0.923 seconds is shown in Figure 4. Although the transfer function exhibits considerable variations in the frequency domain, the average received sub-carrier energy per OFDM symbol, indicated by the bold line, shows little fluctuation. This relative stability of the OFDM symbol energy, over a period of time substantially longer than the inverse of the Doppler frequency, is an effect of the inherent multipath diversity. This leads to a more even distribution of errors, which enables the channel codec to work more efficiently [10].

### 2.6 System parameters

Since the end-to-end delay of speech transmission should be less than 100 ms, the speech frame length should ideally not exceed 20–30 ms. The performance of a turbo decoder, on the other hand, improves with an increasing number of coded bits per block. As a compromise, a 20 ms speech block size was chosen. The G.729 speech codec produces 80 data bits per 10 ms input speech, resulting in a total of 160 data bits per speech block. We will demonstrate that the performance of turbo codes is very dependent on the internal interleaver’s algorithm and latency. For short block lengths, as stated earlier, square interleavers with an odd number of rows and columns exhibit the best performance. The smallest square interleaver holding 160 input bits is $13 \times 13 = 169$ bits long, allowing for the transmission of 160 data, two termination and seven unused padding bits.

The 169 uncoded data bits produce 338 coded output bits, which are transmitted using OFDM in one time slot over four consecutive frames. Employing QPSK as the modulation scheme for the OFDM sub-carriers, only 43 of the 64 sub-carriers in each OFDM symbol are employed for data transmission. The 21 remaining sub-carriers are used for Pilot-Symbol Assisted Modulation (PSAM), allowing coherent detection of the symbols at the receiver. This PSAM was not simulated, but instead perfect channel estimation was used at the receiver. This means that both the demodulator and the turbo decoder...
Table 1: Turbo-coded OFDM system for speech transmission — system parameters

<table>
<thead>
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<th>System Parameters</th>
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<tr>
<td>Carrier Frequency:</td>
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<td>Normalised Doppler Frequency:</td>
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<tr>
<td>Cyclic Extension:</td>
<td>24 samples</td>
</tr>
<tr>
<td>Data Sub–Carriers:</td>
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</tr>
<tr>
<td>Pilot Sub–Carriers:</td>
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<td>Modulation Scheme:</td>
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<td>Number of Iterations:</td>
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operated with perfect estimates of both the fading amplitude and the noise variance. Having described the system, let us now focus our attention on the results.

3 Results

Figure 5 shows the BER performance of our system for the various turbo encoder / interleaver combinations described earlier, as well as for a constraint-length three convolutional code for comparison. It can be seen that the $L = 10,000$ turbo code gives an extremely impressive performance even in the Rayleigh fading channel. The $L = 81$ and $L = 169$ turbo decoded systems both give performances significantly better than the convolutional coded system, showing that turbo codes can be useful in speech transmission schemes. However, disappointingly, the $L = 82$ system performs much worse than the $L = 81$ system, illustrating the importance of choosing a good interleaver for use with turbo encoders.

The $L = 169$ turbo coded system described above was used to transmit G.729 coded speech. The segmental SNR degradation relative to the performance of G.729 over a perfect channel, against channel SNR for both this and the convolutional coded system is shown in Figure 6. It can be seen that the turbo coded system gives a gain of about 3 dB in channel SNR over the convolutional coded system in segmental SNR degradation region of less than 1 dB, which corresponds to near-unimpaired speech quality. We note, however that this is achieved at the cost of an increased decoding complexity due to the eight decoding iterations employed.

4 Conclusions

In conclusion, the attractive G.729 speech codec can be advantageously combined with turbo coding and OFDM transmission using the system of Table 1. Due to the multipath diversity of wideband channels the OFDM modem performance is quite impressive. Furthermore, the error distribution is less bursty than over narrow-band channels and hence the channel codec is less frequently overloaded by channel errors. In Figure 6 a channel SNR of 6 dB appears sufficiently high under the stipulated system conditions for near-unimpaired speech transmission.

5 Acknowledgements

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Figure 5: The effect of Frame Length on BER

Figure 6: The Segmental SNR Degradation with Convolutional and Turbo Encoding

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References


