Turbo-Detected Unequal Protection Audio and Speech Transceivers Using Serially Concantenated Convolutional Codes, Trellis Coded Modulation and Space-Time Trellis Coding

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Abstract – **The MPEG-4 TwinVQ audio codec and the AMR-WB speech codec are investigated in the context of a jointly optimised turbo transceiver capable of providing unequal error protection. The transceiver advocated consists of serially concatenated Space-Time Trellis Coding (STTC), Trellis Coded Modulation (TCM) and two different-rate Non-Systematic Convolutional codes (NSCs) used for unequal error protection. A benchmarker scheme combining STTC and a single-class protection NSC is used for comparison with the proposed scheme. The audio and speech performance of both schemes is evaluated, when communicating over uncorrelated Rayleigh fading channels. An E_b/N_0 value of about 2.5 (3.5) dB is required for near-unimpaired audio (speech) transmission, which is about 3.07 (4.2) dB from the capacity of the system.**

1. MOTIVATION AND BACKGROUND

In recent years, joint source-channel coding (JSCC) has been receiving significant research attention in the context of both delay- and complexity-constrained transmission scenarios. JSCC aims at designing the source codec and channel codec jointly for the sake of achieving the highest possible system performance. As it was argued in [1], this design philosophy does not contradict to the classic Shannonian source and channel coding separation theorem. This is because instead of considering perfectly lossless Shannonian entropy coders for source coding and transmitting their bitstreams over Gaussian channels, we consider low-bitrate lossy audio and speech codecs, as well as Rayleigh-fading channels. Since the bitstreams of the speech and audio encoders are subjected to errors during wireless transmission, it is desirable to provide stronger error protection for the audio bits, which have a substantial effect on the objective or subjective quality of the reconstructed speech or audio signals. Unequal error protection (UEP) is a particular manifestation of JSCC, which offers a mechanism to match the error protection capabilities of channel coding schemes having different error correction capabilities to the differing bit-error sensitivities of the speech or audio bits [2].

Speech services are likely to remain the most important ones in wireless systems. However, there is an increasing demand for high-quality speech transmissions in multimedia applications, such as video-conferencing [3]. Therefore, an expansion of the speech bandwidth from the 300-3400 Hz range to a wider bandwidth of 50-7000 Hz is a key factor in meeting this demand. This is because the low-frequency enhancement ranging from 50 to 200 Hz contributes to the increased naturalness, presence and comfort, whilst the higher-frequency extension spanning from 3400 to 7000 Hz provides a better fricative differentiation and therefore a higher intelligibility. A bandwidth of 50 to 7000 Hz not only improves the intelligibility and naturalness of speech, but also adds an impression of transparent communication and eases speaker recognition. The Adaptive Multi-Rate Wideband (AMR-WB) voice codec has become a 3GPP standard, which provides a superior speech quality [4].

For the sake of supporting high-quality multimedia services over wireless communication channels requires the development of techniques for transmitting not only speech, but also video, music, and data. Therefore, in the field of audio-coding, high-quality, high-compression and highly error-resilient audio-coding algorithms are required. The MPEG-4 Transform-domain Weighted Interleaved Vector Quantization (TwinVQ) scheme is a low-bit-rate audio-coding technique that achieves a high audio quality under error-free transmission conditions at bitrates below 40 kbps [5]. In order to render this codec applicable to wireless systems, which typically exhibit a high bit-error ratio (BER), powerful turbo transceivers are required.

Trellis Coded Modulation (TCM) [6–8] constitutes a bandwidth-efficient joint channel coding and modulation scheme, which was originally designed for transmission over Additive White Gaussian Noise (AWGN) channels. Space-Time Trellis Coding (STTC) [7, 9] is a joint spatial diversity and channel coding technique. STTC may be efficiently employed in an effort to mitigate the effects of Rayleigh fading channels and render them Gaussian-like for the sake of supporting the operation of a TCM code. Recently, a sophisticated unequal-protection turbo transceiver using twin-class convolutional outer coding, as well as joint coding and modulation as inner coding combined with STTC-based spatial diversity scheme was designed for MPEG-4 video telephony in [1, 10]. Specifically, maximal minimum distance Non-Systematic Convolutional codes (NSCs) [11, p. 331] having two different code-rates were used as outer encoders for providing unequal MPEG-4 video protection. Good video quality was attained at a low SNR and medium complexity by the proposed transceiver. By contrast, in this paper, we study the achievable performance of the AMR-WB and the MPEG-4 TwinVQ speech and audio codecs in conjunction with the sophisticated unequal-protection turbo transceiver of [1, 10].

2. THE AMR-WB CODEC'S ERROR SENSITIVITY

The synthesis filter's excitation signal in the AMR-WB codec is based on the Algebraic Code Excited Linear Predictor (ACELP) algorithm, supporting nine different speech codec modes having bitrates of 23.85, 23.05, 19.85, 18.25, 15.85, 14.25, 12.65, 8.85 and 6.6 kbps [4]. Like most ACELP-based algorithms, the AMR-WB codec interprets 20 ms

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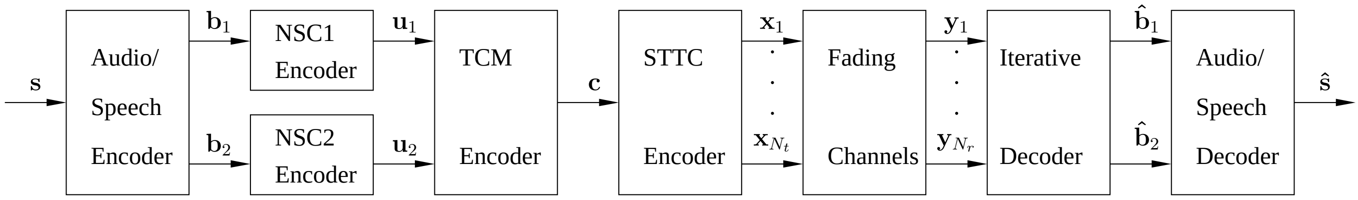

Figure 1: Block diagram of the serially concatenated STTC-TCM-2NSC assisted audio/speech scheme. The notations $\mathbf{s}$, $\hat{\mathbf{s}}$, $\mathbf{b}_i$, $\hat{\mathbf{b}}_\mathbf{i}$, $\mathbf{u}_i$, $\mathbf{c}$, $\mathbf{x}_j$ and $\mathbf{y}_k$ denote the vector of the audio/speech source symbol, the estimate of the audio/speech source symbol, the class-i audio/speech bits, the estimates of the class-i audio/speech bits, the encoded bits of class-i NSC encoders, the TCM coded symbols, the STTC coded symbols for transmitter j and the received symbols at receiver k, respectively. Furthermore, N_t and N_r denote the number of transmitters and receivers, respectively. The symbol-based channel interleaver between the STTC and TCM schemes as well as the two bit-based interleavers at the output of NSC encoders are not shown for simplicity. The iterative decoder seen at the right is detailed in Figure 4.

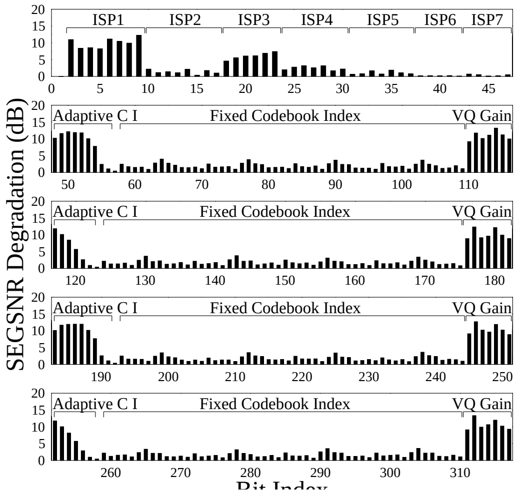

Figure 2: SegSNR degradations versus bit index due to inflicting 100% Bit Error Rate (BER) in the 317-bit, 20 ms AMR-WB frame

segments of speech as the output of a linear synthesis filter synthesized from an appropriate excitation signal. The task of the encoder is to optimise the filter as well as the excitation signal and then represent both as efficiently as possible with the aid of a frame of binary bits. At the decoder, the encoded bit-based speech description is used to synthesize the speech signal by inputting the excitation signal to the synthesis filter, thereby generating the speech segment. Again, each AMR-WB frame represents 20 ms of speech, producing 317 bits at a bitrate of 15.85 kbps. The codec parameters that are transmitted over the noisy channel include the so-called imittance spectrum pairs (ISPs), the adaptive codebook delay (pitch delay), the algebraic codebook excitation index, and the jointly vector quantized, pitch gains as well as algebraic codebook gains.

Most source coded bitstreams contain certain bits that are more sensitive to transmission errors than others. A common approach used for quantifying the sensitivity of a given bit is to consistently invert this bit in every speech or audio frame and evaluate the associated Segmental SNR (SegSNR) degradation [12]. The SegSNR degradation is computed by subtracting from the SegSNR recorded under error-free conditions the corresponding value when there are channel-induced bit-errors.

The error sensitivity of the various encoded bits in the AMR-WB codec determined in this way is shown in Figure 2. The results are based on samples taken from the EBU SQAM (Sound Quality Assessment Material) CD, sampled at 16 kHz and encoded at 15.85 kbps. It can be observed that the bits representing the ISPs, the adaptive codebook delay, the algebraic codebook index and the vector quantized gain are fairly error sensitive. The least sensitive bits are related to the fixed codebook's excitation pulse positions, as shown in Figure 2. This is because, when one of the fixed codebook index bits is corrupted, the codebook entry selected at the decoder will differ from that used in the encoder only in the position of one of the non-zero excitation pulses. Therefore the corrupted excitation codebook entry will be similar to the original one. Hence, the algebraic codebook structure used in the AMR-WB codec is quite robust to channel errors.

3. THE MPEG-4 TWINVQ CODEC'S ERROR SENSITIVITY

The MPEG-4 TwinVQ scheme is a transform coder that uses the modified discrete cosine transformation (MDCT) [5] for transforming the input signal into the frequency-domain transform coefficients. The input signal is classified into one of three modes, each associated with a different transform window size, namely a long, medium or short window, catering for different input signal characteristics. The MDCT coefficients are normalized by the spectral envelope information obtained through the Linear Predictive Coding (LPC) analysis of the signal. Then the normalized coefficients are interleaved and divided into sub-vectors by using the so-called interleave and division technique of [5], and all sub-vectors are encoded separately by the VQ modules.

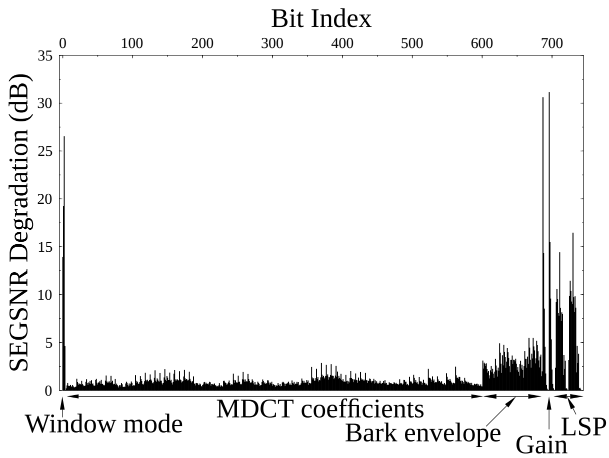

Figure 3: SegSNR degradations due to inflicting a 100% BER in the 743-bit, 23.22 ms MPEG-4 TwinVQ frame

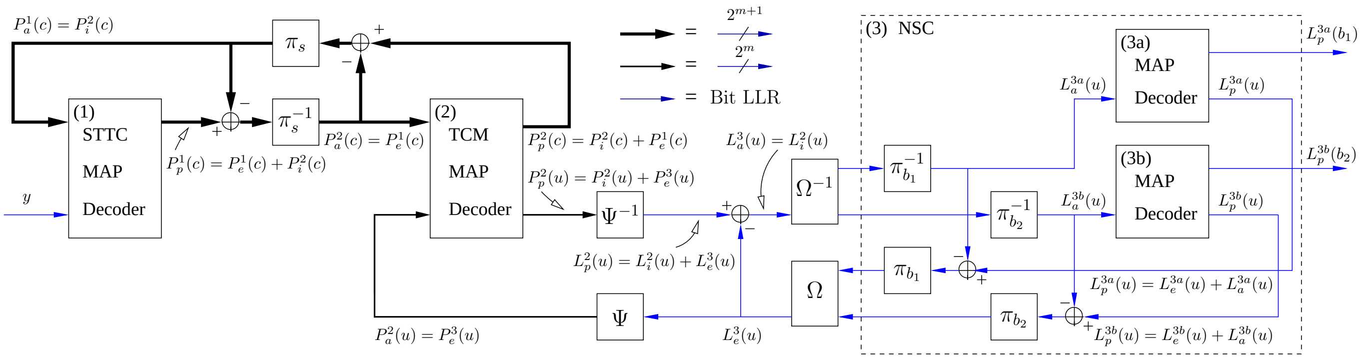

Figure 4: Block diagram of the STTC-TCM-2NSC turbo detection scheme seen at the right of Figure 1. The notations $\pi_{(s,b_i)}$ and $\pi^{-1}_{(s,b_i)}$ denote the interleaver and deinterleaver, while the subscript s denotes the symbol-based interleaver of TCM and the subscript b_i denotes the bit-based interleaver for class-i NSC. Furthermore, Ψ and Ψ^{-1} denote LLR-to-symbol probability and symbol probability-to-LLR conversion, while Ω and Ω^{-1} denote the parallel-to-serial and serial-to-parallel converter, respectively. The notation m denotes the number of information bits per TCM coded symbol [10] ©IEE, 2004, Ng, Chung and Hanzo.

Similarly, bit error sensitivity investigations were performed in the same way, as described in the previous section. Figure 3 shows the error sensitivity of the various bits of the MPEG-4 TwinVQ codec for a bitrate of 32 kbps. The results provided are based on a 60 seconds long excerpt of Mozart's "Clarinet Concerto (2nd movement - Adagio)". This stereo audio file was sampled at 44.1 kHz and again, encoded at 32 kbps. Since the analysis frame length is 23.22 ms, which corresponds to 1024 audio input samples, there are 743 encoded bits in each frame. This figure shows that the bits representing the gain factors, the Line Spectral Frequency (LSF) parameters, and the Bark-envelope are more sensitive to channel errors, compared to the bits representing the MDCT coefficients. The bits signalling the window mode used are also very sensitive to transmission errors and hence have to be well protected. The proportion of sensitive bits was only about 10%. This robustness is deemed to be a benefit of the weighted vector-quantization procedure which uses a fixed-length coding structure as opposed to using an error-sensitive variable-length structure, where transmission errors would result in a loss of synchronisation.

4. THE TURBO TRANSCEIVER

Once the bit error sensitivity of the audio/speech codecs was determined, the bits of the AMR-WB and the MPEG-4 TwinVQ codec are protected according to their relative importance. Figure 1 shows the schematic of the serially concatenated STTC-TCM-2NSC turbo scheme using a STTC and a TCM scheme as well as two different-rate NSCs as its constituent codes. Let us denote the turbo scheme using the AMR-WB codec as STTC-TCM-2NSC-AMR-WB, whilst STTC-TCM-2NSC-TVQ refers to the turbo scheme using the MPEG-4 TwinVQ as the source codec. For comparison, both schemes protect 25% of the most sensitive bits in class-1 using an NSC code rate of $R_1 = k_1/n_1 = 1/2$. By contrast, the remaining 75% of the bits in class-2 are protected by an NSC scheme having a rate of $R_2 = k_2/n_2 = 3/4$. The code memory of the class-1 and class-2 encoders is $L_1 = 3$ and $L_2 = 3$, respectively. The class-1 and class-2 NSC coded bit sequences are interleaved by two separate bit interleavers, before they are fed to the rate-$R_3 = 3/4$ TCM scheme [6–8] having a code memory of $L_3 = 3$. Code termination was employed for the NSCs, as well as for the TCM [6–8] and STTC codecs [7, 9]. The TCM symbol sequence is then symbol-interleaved and fed to the STTC encoder as seen in Figure 4. We invoke a 16-state STTC scheme having a code memory of $L_4 = 4$ and $N_t = 2$ transmit antennas, employing $M = 16$-level Quadrature Amplitude Modulation (16QAM) [8]. The STTC scheme employing $N_t = 2$ requires a single 16QAM-based termination symbol. In the STTC-TCM-2NSC-AMR-WB scheme the 25% of the bits that are classified into class-1 includes 23 header bits, which gives a total of 340 NSC1-encoded bits. In the ITU stream format [13], the header bits of each frame include the frame types and the window-mode used.

Hence, the overall coding rate of the STTC-TCM-2NSC-AMR-WB scheme becomes $R_{AMRWB} = 340/720 \approx 0.4722$. By contrast, the overall coding rate of the STTC-TCM-2NSC-TVQ scheme is $R_{TVQ} = 744/1528 \approx 0.4869$. The effective throughput of the STTC-TCM-2NSC-AMR-WB and STTC-TCM-2NSC-TVQ schemes is $log_2(M) \cdot R_{AMRWB} \approx 1.89$ Bits Per Symbol (BPS) and $log_2(M) \cdot R_{TVQ} \approx 1.95$ BPS, respectively.

At the receiver, we employ $N_r = 2$ receive antennas and the received signals are fed to the iterative decoders for the sake of estimating the audio bit sequences in both class-1 and class-2, as seen in Figure 1. The STTC-TCM-2NSC scheme's turbo decoder structure is illustrated in Figure 4, where there are four constituent decoders, each labelled with a round-bracketed index. The Maximum A-Posteriori (MAP) algorithm [7] operating in the logarithmic-domain is employed by the STTC and TCM schemes as well as by the two NSC decoders, respectively. The iterative turbo-detection scheme shown in Figure 4 enables an efficient information exchange between the STTC, TCM and NSCs constituent codes for the sake of achieving spatial diversity gain, coding gain, unequal error protection and a near-channel-capacity performance. The information exchange mechanism between each constituent decoders is detailed in [10].

For the sake of benchmarking both audio schemes advocated, we created a powerful benchmark scheme for each of them by replacing the TCM and NSC encoders of Figure 1 by a single-class NSC codec having a coding rate of $R_0 = k_0/n_0 = 1/2$ and a code memory of $L_0 = 6$. Note that if we reduce the code memory of the NSC constituent code of the STTC-NSC benchmarker arrangement from L_0=6 to 3, the achievable performance becomes poorer, as expected. If we increased L_0 from 6 to 7 (or higher), the decoding complexity would double, while the attainable performance is only marginally increased. Hence, the STTC-NSC scheme having L_0=6 constitutes a good benchmarker scheme in terms of its performance versus complexity tradeoffs. We will refer to this benchmarker scheme as the STTC-NSC-TVQ and the STTC-NSC-AMR-WB arrangement designed for the audio and the speech transceiver, respectively. Again, all audio and speech bits are equally protected in the benchmarker scheme by a single NSC encoder and a STTC encoder. A bit-based channel interleaver is inserted between the NSC encoder and STTC encoder. Taking into account the bits required for code termination, the number of output bits of the NSC encoder of the STTC-NSC-TVQ benchmarker scheme is $(744+k_0L_0)/R_0 = 1500$, which corresponds to 375 16QAM symbols. By contrast, in the STTC-NSC-AMR-WB scheme the number of output bits after taking into account the bits required for code termination becomes $(340 + k_0L_0)/R_0 = 692$, which corresponds to 173 16QAM symbols. Again, a 16-state STTC scheme having $N_t = 2$ transmit antennas is employed. After code termination, we have $375 + 1 = 376$ 16QAM symbols or $4(376) = 1504$ bits in a transmission frame at each transmit antenna for the STTC-NSC-TVQ. The overall coding rate is given by $R_{TVQ-b} = 744/1504 \approx 0.4947$ and the effective throughput is $\log_2(16)R_{TVQ-b} \approx 1.98$ BPS, both of which are very close to the corresponding values of the STTC-TCM-2NSC-TVQ scheme. Similary, for the STTC-NSC-AMR-WB scheme, after code termination, we have $173 + 1 = 174$ 16QAM symbols or $4(174) = 696$ bits in a transmission frame at each transmit antenna. This gives the overall coding rate as $R_{AMRWB-b} = 340/696 \approx 0.4885$ and the effective throughput becomes $\log_2(16)R_{AMRWB-b} \approx 1.95$ BPS. Again, both of the values are close to the corresponding values of the STTC-TCM-2NSC-AMR-WB scheme. A decoding iteration of each of the STTC-NSC benchmarker schemes is comprised of a STTC decoding and a NSC decoding step.

We will quantify the decoding complexity of the proposed STTC-TCM-2NSC schemes and that of its corresponding benchmarker schemes using the number of decoding trellis states. The total number of decoding trellis states per iteration of the proposed scheme employing 2 NSC decoders having a code memory of $L_1 = L_2 = 3$, using the TCM scheme having $L_3 = 3$ and the STTC arrangement having $L_4 = 4$, becomes $S = 2^{L_1} + 2^{L_2} + 2^{L_3} + 2^{L_4} = 40$. By contrast, the total number of decoding trellis states per iteration for the benchmarker scheme having a code memory of $L_0 = 6$ and for the STTC having $L_4 = 4$ is given by $S = 2^{L_0} + 2^{L_4} = 80$. Therefore, the complexity of the proposed STTC-TCM-2NSC scheme having two iterations is equivalent to that of the benchmarker scheme having a single iteration, which corresponds to 80 decoding states.

5. SIMULATION RESULTS

In this section we comparatively study the performance of the audio and speech transceiver using the Segmental Signal to Noise Ratio (SegSNR) metric.

Figures 5 and 6 depict the audio SegSNR performance of the STTC-TCM-2NSC-TVQ and that of its corresponding STTC-NSC-TVQ benchmarker schemes, respectively, when communicating over uncorrelated Rayleigh fading channels. It can be seen from Figures 5 and 6 that the non-iterative single-detection based performance of the STTC-NSC-TVQ benchmarker scheme is better than that of the STTC-TCM-2NSC assisted MPEG-4 TwinVQ audio scheme. However, at the same decoding complexity quantified in terms of the number of trellis decoding states the STTC-TCM-2NSC-TVQ arrangement performs approximately 0.5 dB better in terms of the required channel E_b/N_0 value than the STTC-NSC-TVQ benchmarker scheme, both exhibiting a SegSNR of 13.8 dB. For example, at the decoding complexity of 160 trellis decoding states, this corresponds to the STTC-TCM-2NSC-TVQ scheme's 4th iteration, whilst in the STTC-NSC-TVQ scheme this corresponds to the 2nd iteration. Therefore, we observe in Figures 5 and 6 that the STTC-TCM-2NSC-TVQ arrangement performs by 0.5 dB better in terms of the required channel

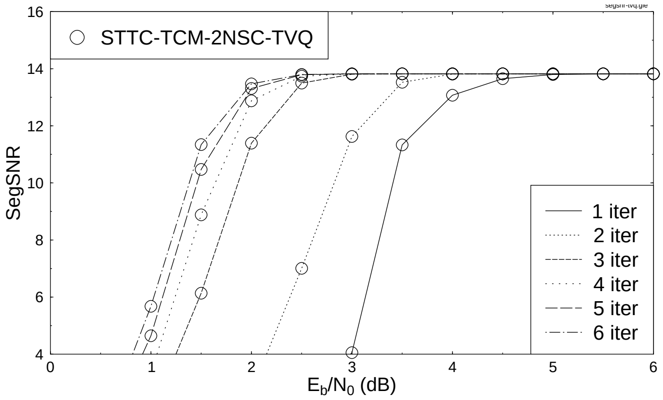

Figure 5: Average SegSNR versus E_b/N_0 performance of the 16QAM-based STTC-TCM-2NSC assisted MPEG-4 TwinVQ audio scheme, when communicating over uncorrelated Rayleigh fading channels. The effective throughput was **1.95 BPS**.

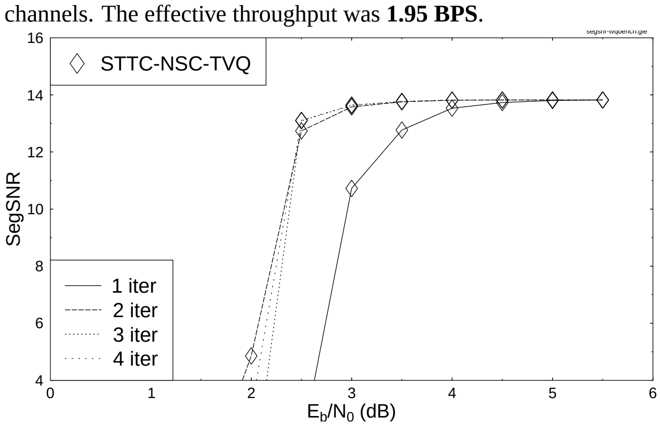

Figure 6: Average SegSNR versus E_b/N_0 performance of the 16QAM-based STTC-NSC assisted MPEG-4 TwinVQ audio benchmarker scheme, when communicating over uncorrelated Rayleigh fading channels. The effective throughput was **1.98 BPS**.

E_b/N_0 value than its corresponding benchmarker scheme.

Similarly, it can be observed from Figures 7 and 8 that at the decoding complexity of 160 trellis decoding states the STTC-TCM-2NSC-AMR-WB arrangement performs 0.5 dB better in terms of the required channel E_b/N_0 value than the STTC-NSC-AMR-WB scheme, when targetting a SegSNR of 10.6 dB. By comparing Figures 5 and 7, we observe that the SegSNR performance of the STTC-TCM-2NSC-AMR-WB scheme is inferior in comparison to that of STTC-TCM-2NSC-TVQ.

More explicitly, the STTC-TCM-2NSC-TVQ system requires an E_b/N_0 value of 2.5 dB, while the STTC-TCM-2NSC-AMR-WB arrangement necessitates $E_b/N_0 = 3.0$ dB, when having their respective maximum attainable average SegSNRs. The maximum attainable average SegSNRs for STTC-TCM-2NSC-TVQ and STTC-TCM-2NSC-AMR-WB are 13.8 dB and 10.6 dB, respectively.

This discrepancy is due to the reason that both schemes map the most sensitive 25% of the encoded bits to class-1. By contrast, based on the bit error sensitivity study of the MPEG-4 TwinVQ codec outlined in Section 3, only 10% of the MPEG-4 TwinVQ encoded bits were found to be gravely error sensitive. Therefore, the 25% class-1 bits of the MPEG-4 TwinVQ also includes some bits, which were found to be only moderately sensitive to channel errors. However, in the case of the AMR-WB codec all the bits of the 25%-

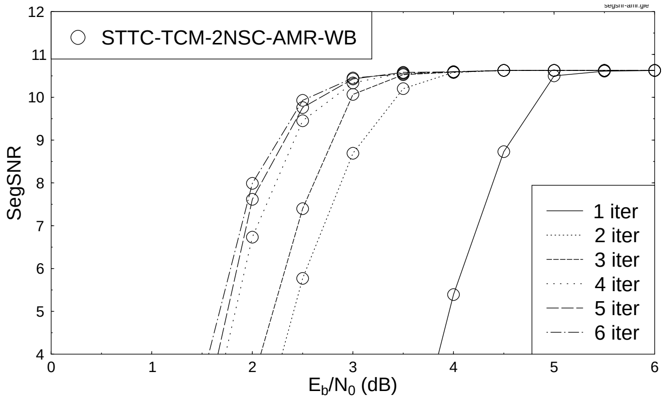

Figure 7: Average SegSNR versus E_b/N_0 performance of the 16QAM-based STTC-TCM-2NSC assisted AMR-WB speech scheme, when communicating over uncorrelated Rayleigh fading channels. The effective throughput was **1.89 BPS**.

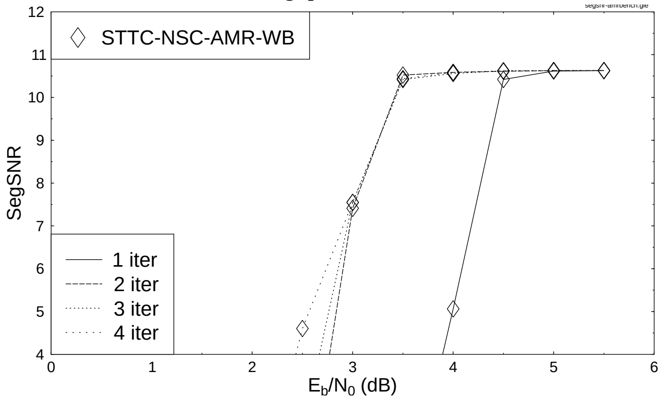

Figure 8: Average SegSNR versus E_b/N_0 performance of the 16QAM-based STTC-NSC assisted AMR-WB speech benchmarker scheme, when communicating over uncorrelated Rayleigh fading channels. The effective throughput was **1.95 BPS**.

partition were found to be quite sensitive to channel errors. Furthermore, the frame length of the STTC-TCM-2NSC-TVQ scheme is longer than that of the STTC-TCM-2NSC-AMR-WB arrangement and hence benefits from a higher coding gain.

It is worth mentioning that the channel capacity for the system employing the full-diversity STTC scheme with the aid of $N_t = 2$ transmit antennas and $N_r = 2$ receive antennas is -0.57 dB and -0.70 dB for the throughputs of 1.95 BPS and 1.89 BPS, respectively, when communicating over uncorrelated Rayleigh fading channels [14].

6. CONCLUSIONS

In this contribution, we comparatively studied the performance of the MPEG-4 TwinVQ and AMR-WB audio/speech codecs combined with a jointly optimised source-coding, outer unequal protection NSC channel-coding, inner TCM and spatial diversity aided STTC turbo transceiver. The audio bits were protected differently according to their error sensitivity with the aid of two different-rate NSCs . The employment of TCM improved the bandwidth efficiency of the system and by utilising STTC spatial diversity was attained. The performance of the STTC-TCM-2NSC scheme was enhanced with the advent of an efficient iterative joint decoding structure. Both proposed twin-class STTC-TCM-2NSC schemes perform approximately 0.5 dB better in terms of the required E_b/N_0 than the corresponding single-class STTC-NSC audio benchmarker schemes. This relatively modest advantage of the twin-class protected transceiver was a consequence of having a rather limited turbo-interleaver length. In the longer interleaver of the videohphone system of [1, 10] an approximately 2 dB E_b/N_0 gain was achieved. For a longer-delay non-realtime audio streaming scheme a simliar performance would be achieved to that of [10]. Our future work will further improve the achievable audio performance using the soft speech-bit decoding technique of [15].

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