CHANNEL ERROR CORRECTION IN COMPRESSED IMAGE TRANSMISSION

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Abstract— **An a posteriori channel error correction method based on subband synthesis is presented in this paper. Using not downsampled subband signals, we estimate the positions and amplitudes of errors inserted during transmission, thus allowing a correction of the subband signals and a resynthesis of a (nearly) error free signal. The method can be extended to multiple dimensions. Examples of transmitted (compressed) images are shown with and without this method. The results are very promising.**

Keywords— **a posteriori Channel Error Correction, Subband/Wavelet Synthesis.**

Technical subject area— **Error Control Coding, Signal Processing for Communications.**

I. Introduction

In the area of image and video coding, subband decomposition and its special case, the discrete wavelet transform, are two popular techniques [1], [3]. Most applications are based on the two-channel scheme in which the original signal is split up into two subbands, each with half the size of the original. This process can be repeatedly applied to one or both subbands resulting in a tree-structured splitting of a certain number of levels. The subband coefficients are quantized, compressed with data compression techniques, transmitted, and at the receiver the original signal (actually, an estimate of it) is reconstructed by filtering and adding the subbands in reverse order. If the quantization is not too coarse and if there were no errors during transmission, the synthetized signal is very similar to the original. However, this is not always the case. In order to keep the number of errors introduced by real channels low, channel or error control coding is used, which consumes a certain amount of the available channel capacity. As a consequence, the quantization has to be made coarser in order to achieve a better compression of the subband coefficients. This, in turn, degrades the quality of transmitted images.

In this paper, we present a method for correcting residual errors after the channel decoding based on the cross-redundancy that is hiding in the subband signals and on information about the distribution of errors. The method works so well, that on graceful channels, channel coding can completely be dropped, freeing channel capacity for fine quantized subband data.

In Section II, the basics of the method are discussed on a simple example. The extension to two dimensions can be found in Section III, where the transmission chain used for compressed image transmission is described. Special attention is payed to error spreading due to decompression (run-length and Huffman decoding). Simulation results in Section IV illustrate the potential of the method, and some yet unanswered questions are listed in the Conclusion.

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II. The principle

In this section, the principle of our a posteriori channel error correction method based on subband synthesis for the one-dimensional case is described in detail. We use a simple example to show the effects we utilize for the correction. The filters used in the analysis/synthesis stages of the two-band decomposition are (e.g. [1]) $h_0 = [1, 3, 3, 1]$, $h_1 = [-1, -3, 3, 1]$, $g_0 = [-1, 3, 3, -1]$ and $g_1 = [-1, 3, -3, 1]$.

Consider the signal in Figure 1a. This signal may have an arbitrary form, but for simplicity, we have chosen a smooth signal with a mainly low pass characteristic, which is typical for lines of grayscale images. A usual subband decomposition of this signal (filtering with both the low pass and high pass analysis filters and downsampling by a factor of 2) yields subband signals depicted with dotted lines in Figures 1b and c. Assume that during transmission of the samples of these two subbands two errors occur: One in the low pass subband and one in the high pass subband. For simplicity, again, let us assume that both errors have positive amplitudes, and that they are not on close relative positions in the both subbands. Hence, the received subband coefficients differ from the original coefficients, and they are represented with solid lines in Figures 1b and c.

It is obvious, that a subband synthesis based on the received erroneous subbands cannot yield the original signal. Looking at the resulting signal depicted in Figure 1d (solid line), one can easily notice the differences to the original (dotted line) and locate the impulse responses of the synthesis filters g_0 and g_1, compare Figure 2.

A subsequent analysis (filtering, but *no* downsam-

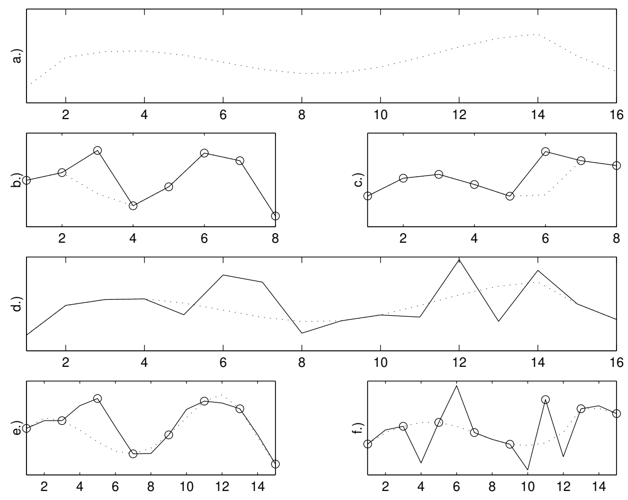

Fig. 1. The principle. a.) The original signal to be decomposed. b.) The (erroneous) low pass subband. c.) The (erroneous) high pass subband. d.) The synthetised signal. e.) and f.) The subbands after a new analysis *(without downsampling!)*.

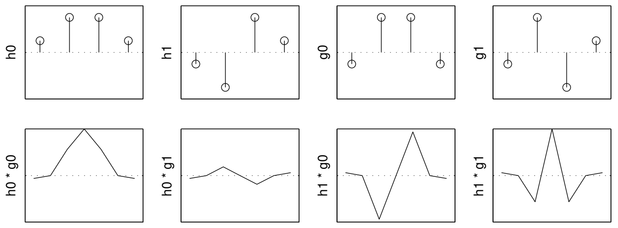

Fig. 2. The analysis and synthesis filters used in the example ($h0$, $h1$, $g0$ and $g1$) and their pair-wise convolutions.

pling) of this signal yields subband signals like in Figures 1e and f (solid lines). Please note that these signals are depicted in their original clock-rate versions (i.e. not downsampled) - this plays a crucial role in the following considerations.

Taking a closer look at the subband signals in Figures 1e and f, one can find the superimposed impulse responses of $h_0 * g_0$, $h_0 * g_1$, $h_1 * g_0$, and $h_1 * g_1$, compare Figure 2. On the position of the low pass band error, there is a superimposed impulse response $h_0 * g_0$ in the low pass subband, and a superimposed impulse response $h_1 * g_0$ in the high pass subband. On the position of the high pass band error, there is a superimposed impulse response $h_1 * g_1$ in the high pass subband, and a superimposed impulse response $h_0 * g_1$ in the low pass subband. Please note, that this information gets lost by downsampling, and then the (downsampled) subband signals exactly match those in Figures 1b and c (circles).

Based on this information, a fairly accurate estimation of the positions of errors and their amplitudes is possible, which allows a simple and powerful, even though not perfect, error correction. To accomplish it, following problems have to be solved:

- how to find the position of errors; if the estimate of the real, not downsampled versions of the subband signals is needed for that search, how to obtain them, and
- how to estimate the amplitude of the errors.

A. How to find the positions?

The method of finding the positions of errors itself is not crucial as long as it yields reliable results. Reliable means here, that no additional errors are (wrongly) detected. For some awkward signal forms it is almost impossible to tell whether the error-candidate is perhaps only the signal itself or the signal with superimposed error. Therefore, the decision algorithm should be rather defensive and in case of doubt leave the error position unmarked, which is a good choice, as we will see later.

Upon reception of the subband samples x_0 and x_1 (see Figure 3, a classical interpolation (e.g. spline interp.) is performed in order to get a coarse approximation of the subband signals in the higher clock-rate (box "interp$\uparrow$ 2" in Figure 3). At the same time, a subband synthesis (resulting in $\hat{x'}$) followed by a new analysis (x'_0 and x'_1) is performed.

The decision (boxes "D0" and "D1") is basically performed as a thresholdig in two signals simultaneously. For a certain position in the low pass band to be marked as an error, both correlation signals on the input of "D0" have to be above certain thresholds. The outputs of both boxes "D0" and "D1" are logical values in the high clock-rate.

In Figure 3, the gray background marks blocks which operate on the higher clock-rate (i.e. the signals are not downsampled).

B. How to estimate the amplitudes?

The amplitude of high pass band errors can directly be computed from x'_1, and they simply have to be subtracted from the received value in order to get correct samples.

One easy way to estimate the original amplitude of an erroneous low pass subband sample is to interpolate it from neighbouring samples. In Figure 3, this is done by the block "gap interp". On erroneous positions this value is inserted in place of the received sample.

Finally, the second subband synthesis results in signal $\hat{x}$, which, ideally, contains no more errors.

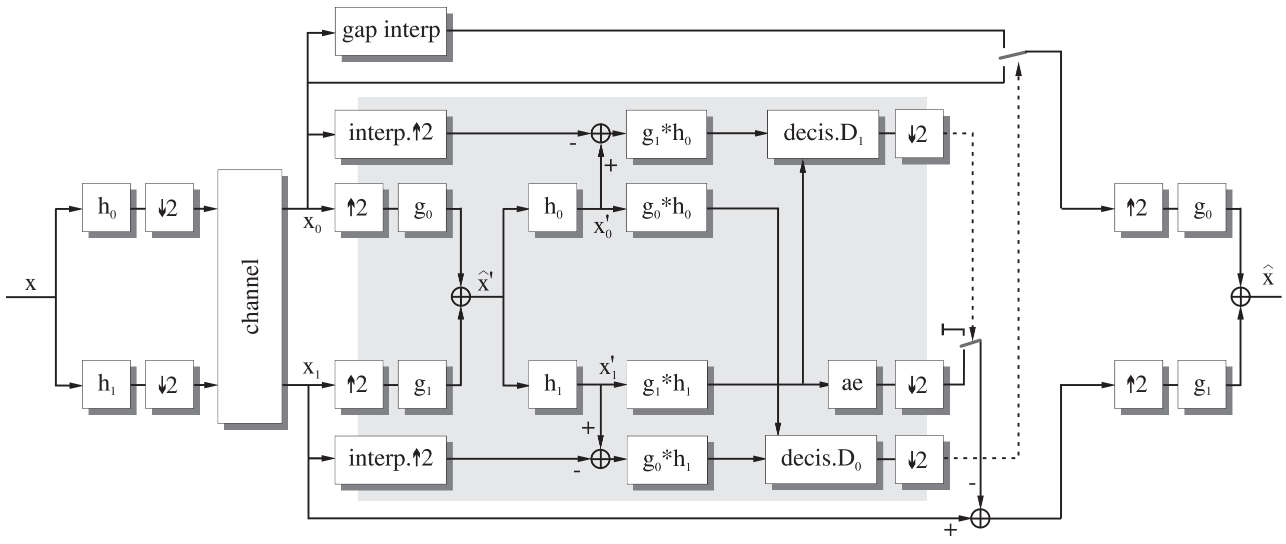

Fig. 3. The 1D subband analysis and error correcting synthesis.

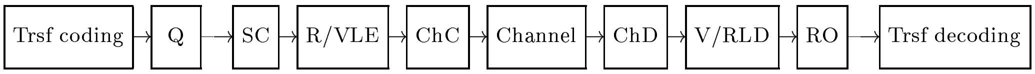

Fig. 4. Compressed transmission of images.

III. Compressed Image Transmission

The technique presented in the previous section can be extended to multiple dimensions. It is even more advantageous to do so, because errors, that have been left undetected or uncorrected in one direction can be handled successfully in the next one. Subband/wavelet image coding with separable filters is the ideal application in this sense.

Usually, the transmission of compressed images is performed in the following steps. First, an appropriate transform coding is performed (in Figure 4, this is represented by the block "Trsf coding"). In order to be able to achieve significant compression ratios, quantization follows (block "Q"). The quantized coefficients of the transform coding are usually scanned ("SC") in a way which allows the run-length and variable-length encoding ("R/VLE") to perform best. In order to mitigate the influences of the channel, channel coding is used ("ChC"). After the channel decoding ("ChD"), the run-length and variable-length decoding ("R/VLD") is followed by reordering ("RO") the obtained samples into a layout which is needed for transform decoding.

In case of subband/wavelet decomposition as the transform coding (which we use in this paper), the original image is decomposed in the well-known manner [1] into four subbands. First, the rows of the image are decomposed with the analysis filter bank into two subbands, after which a further decomposition of the columns of these two subbands follows. Please note, that the last decomposition takes place in vertical direction. An example of such a decomposition is shown in Figure 5.

The scanning of the samples (Figure 4, block "SC") is usually done subband-wise. Despite the spatial orientation property of the wavelet decomposition, we scan the samples in horizontal direction (i.e. row-wise) in all subbands. This decision was motivated by considerations described later on in this section. In the simulations in this paper, no channel coding was used.

Typically, high pass subbands contain a lot of samples whose values are below the quantization level. Therefore, the use of run-length coding [2] is justified for precompressing these regions of the subbands. Run-length coding maps byte-sequences of various lengths onto byte-blocks of fixed lengths. To achieve further compression, entropy coding (variable-length coding techniques such as arithmetic coding or Huffman coding) is used. Huffman coding maps a byte onto a (short) bit-stream of variable length. It is self-evident that such a compressed bit-stream is very vulnerable against channel noise. Every bit-error desynchronizes the Huffman decoder, so a self-synchronizing decoder has to be implemented. Such a decoder would deliver byte-streams that sporadically contain bursts of erroneous bytes. The run-length decoder should cope with that byte-stream, i.e. be robust against error bursts and be able to deliver a byte-stream of the same length as it had on the input of the run-length

Fig. 5. Original image and its subbands after decomposition.

coder. The easiest way to implement such run-length and Huffman coder/decoder pairs is to insert resynchronisation and position marks in the byte- and bitstreams. An example of the distribution and patterns (expressed via the shade of grey) of such an error propagation in the reordered (subband-wise, row by row) byte-plane is depicted in Figure 6. This error distribution is highly signal-dependent.

The four subbands in Figure 5 bit-wise `xor`-ed with the corresponding error values in Figure 6 build the input data set of the transform decoding.

At synthesis (transform decoding), we begin with columns of subbands lying one upon the other. Horizontal error bursts represent single errors for vertical processing, so our method described in Section II can effectively correct them. This interleaving-effect is why we scan all the subbands solely horizontally. The reason for using a defensive error searching strategy in Section II is, that errors not having been marked in the vertical synthesis still can be detected and corrected in the subsequent horizontal processing.

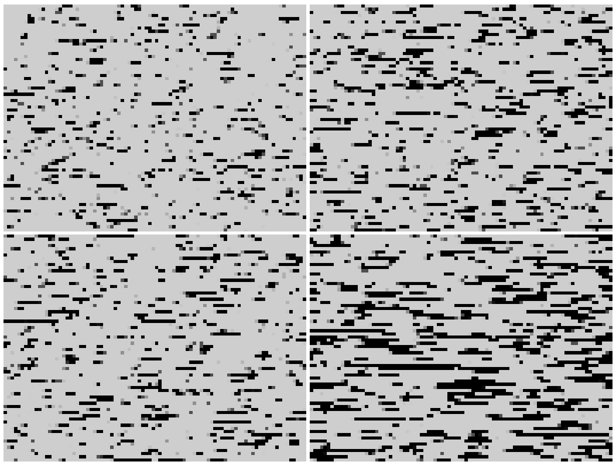

Fig. 6. Error distribution and patterns in the reordered subband byte-plane (example).

A. Advantages of the method

This error correcting method allows us to drop channel coding (or to keep only a necessary minimum) and use the bandwidth for transmitting more source coded data. In general, this increases the transmitted image quality. This method *is not* an image restoration or post-processing method, and bluring - inherent to the most of these methods - is not present. The method is a receiver-site-only method, which relocates the computation onto the receiver solely. This can be advantageous in applications, that have to be enhanced without altering the sender.

B. Drawbacks of the method

The main disadvantage of the method is, that for some specific signal forms, errors will be detected even though there are none. The estimation of the not downsampled versions of the received subband signals is far from perfect (especially on edges in the image) and the amplitude of errors can not be estimated accurately enough.

IV. Simulation results

In Figures 7 and 8, simulation results for two channel bit-error probabilities are given: $P_e = 0.001$ and $P_e = 0.01$. These probabilities represent mean bit-error probabilities on the used binary symmetric channel (BSC). Due to run-length and Hamming-decoding, the number of errors just before synthesis is much higher than right after the channel; it depends on the signal, and has no force of expression therefore.

It is obvious, that the method presented here yields very nice results in terms of surpressing burst errors stemming from run-length and Hamming-decoding of

Fig. 7. Image transmission example. Mean bit-error probability on the channel: $P_e = 0.001$. First image: without our correction method. Second image: using our method.

Fig. 8. Image transmission example. Mean bit-error probability on the channel: $P_e = 0.01$. First image: without our correction method. Second image: using our method.

erroneous bit-streams. Figure 8 is a good demonstration of its potentials. However, in situations like error bursts in both vertical and horizontal direction, it is inefficient. Such artefacts can be found in both Figures 7 and 8.

Conclusion

The technique presented here is a new technique which allows the correction of channel errors a posteriorily, i.e. after reception. There are still many questions to be answered, among others:

- Does the method perform better with longer (better) filters (i.e. less aliasing between the subbands)? If so, which filters are the most appropriate?
- Is there an optimum in the tradeoff between channel coding and quantization (that influences the compressability of the image)? Can channel coding in real transmission situations be completely left out?
- Does it generally make sense to perform the error correction presented here in more stages consecutively (i.e. iteratively)? If so, how the thresholds have to be adjusted?

The method for correcting channel errors based solely on the source decoder, which we presented in this paper, is a very attractive one, because it allows usage of channel codes of higher rates ("weaker" codes), and therefore the transmission of more image data in the same bandwidth.

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