Experimental Evaluation of Analog Joint Source-Channel Coding in Indoor Environments

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Abstract—Recently, analog joint source-channel coding has been proposed as a means of achieving near-optimum performance for high data rates with a very low complexity. However, no experimental evaluation showing the practical feasibility of this scheme has been performed to date. In this paper, we describe a software-defined radio implementation of an analog joint source-channel coded wireless transmission system. Experimental evaluation carried out in an indoor environment making use of a wireless testbed shows that the performance perfectly matches that originally reported by simulations in Additive White Gaussian Noise (AWGN) channels for signal-to-noise ratio values below 20 dB.

I. INTRODUCTION

A digital communication system based on separation between source and channel coding is optimal for information transmission along many channels such as the Additive White Gaussian Noise (AWGN) channel [1]. The complexity of those systems, however, can be very high when they are designed to perform close to the Shannon capacity limit. In addition, they will introduce significant delays motivated by the long block lengths required to approach the theoretical limits. Moreover, full redesign of the digital system is required whenever we want to change the code rate or the distortion target.

Interestingly, analog communications are optimal under some circumstances. For instance, direct transmission of uncoded Gaussian samples over AWGN channels is optimal because Gaussian sources are perfectly matched to Gaussian channels. Thus, it is very interesting to consider analog transmission of discrete-time continuous amplitudes when possible. This fact has recently motivated several work [2–7] on finding analog transformations aimed at matching sources with channels. These schemes perform analog compression at symbol level and hence no delays are introduced. They also exhibit very low complexity, making them very attractive for practical applications.

Optimal Performance Theoretically Attainable (OPTA) occurs when the rate distortion function equals channel capacity. System performance is thus measured in terms of Signal-to-Distortion Ratio (SDR) versus Signal-to-Noise Ratio (SNR). Simulation results presented in [5] show that for a wide range of compression rates the mentioned analog compression schemes attain near-optimal performance, comparable to that of state-of-the-art digital systems.

However, the actual performance of these schemes over realistic scenarios has never been analyzed to date. In order to assess the actual performance of analog joint source-channel coding we employ a software-defined radio testbed implementing this transmission scheme. When carrying out measurements over the software-defined radio testbed, we have addressed the problems that analog compression schemes pose in practice. First, the Peak-to-Average Power Ratio (PAPR) of the transmitted signals can be very high, so careful normalization of the transmitted signals is needed to prevent performance degradation due to DAC/ADC limited resolution and the non-linear effects that the power amplifier may introduce. Second, the transmitted signals in analog compression schemes are parameterized depending on the SNR, so it was necessary to carry out closed-loop measurements. This is similar to digital systems that approach capacity in a wireless channel, which also use a closed-loop to adapt transmit power, modulation format and code rate to the channel response.

II. ANALOG JOINT SOURCE-CHANNEL CODING

We consider the analog transmission of discrete-time continuous-amplitude sources over an AWGN channel. At the transmitter, $N$ independent and identically-distributed (i.i.d.) source symbols are encoded into $K$ channel symbols. Such symbols are transmitted through an AWGN channel with noise variance $\sigma_n^2$. At the receiver, symbols can be decoded using either a Maximum Likelihood (ML) or a Minimum Mean Square Error (MMSE) approach. Due to its better performance, MMSE decoding will be considered along this work.

The distortion between a source symbol $x = \{x_i\}_{i=1}^N$ and a decoded symbol $\hat{x} = \{\hat{x}_i\}_{i=1}^N$ is calculated according to the MSE, defined as

$$\text{MSE} = \frac{1}{N} \mathbb{E}\{\|x - \hat{x}\|^2\}.$$

Consequently, the system performance can be measured in terms of the output SDR with respect to the SNR. Assuming the source symbols have unit variance, the SDR is defined as

$$\text{SDR} = 10\log_{10}\left(1/\text{MSE}\right).$$

Given $N$ and $K$, the OPTA is calculated by equating the
rate distortion function to the AWGN channel capacity [8]

\[ N \log (1 / \text{MSE}) = K \log_{10} \left( 1 + 1 / \sigma_n^2 \right) , \]

where the transmitted symbols are normalized to unit mean power.

Figure 1 shows the block diagram of an \( N:1 \) analog joint source-channel coding system where the source generates blocks of \( B \) i.i.d. symbols from a Gaussian distribution that are encoded into \( B/N \) channel symbols. Without loss of generality, we assume a Gaussian distribution with zero mean and unit variance for the source, and that the mean of transmit power is equal to one.

A particular type of parameterized space-filling continuous curves, called spiral-like curves, are used to encode the \( x = (x_1, x_2) \) Gaussian samples. These curves were proposed for transmission of Gaussian sources over AWGN channels by Chung and Ramstad [2–4]. For the case of \( 2:1 \) compression (i.e. \( N=2 \)), they are formally defined as

\[ x_t(\theta) = \left( \text{sign}(\theta) \frac{\delta}{\pi} \theta \sin \theta, \frac{\delta}{\pi} \theta \cos \theta \right) , \quad (1) \]

where \( \delta \) is the distance between two neighboring spiral arms, and \( \theta \) is the angle from the origin to the point \( x_\theta \) on the curve. Therefore, each pair of Gaussian samples, \( x_1 \) and \( x_2 \), represent a specific point in \( \mathbb{R}^2 \) that is matched to the closest point \( x_\theta = (x_{\theta,1}, x_{\theta,2}) \) on the spiral. The angle from the origin to that point on the spiral, \( \theta \), will be the channel symbol for \( x_1 \) and \( x_2 \), i.e.

\[ \hat{\theta} = M_\delta(x) = \arg \min_\theta \{ ||x - x_\delta||^2 \} . \quad (2) \]

Since our goal is the minimization of the MSE, the bidimensional space has to be filled by the spiral in the best possible way for every SNR value. On one hand, by changing the \( \delta \) value, we manage to optimize this matching and to improve the system performance. On the other hand, it is possible to achieve higher compression rates (i.e. \( N:1 \)) by extending (1) to generate more complex curves [9, 10].

The next step consists in defining an invertible function of \( \theta \) —with the corresponding normalization factor to ensure the transmit power constraint. In [2, 4, 11], the invertible function \( T_n(\theta) = \theta^\alpha \), with \( \alpha = 2 \) was proposed. However, as shown in [5], the system performance can be improved if \( \alpha \) and \( \delta \) are numerically optimized for each different SNR value. Therefore, the channel symbol is \( T_n(\hat{\theta}) \sqrt{\gamma} \), where \( \sqrt{\gamma} \) is the normalization factor. In summary, the received symbol \( y \) at the decoder can be expressed as

\[ y = T_\alpha(M_\delta(x)) + n \sqrt{\gamma} , \quad \text{with} \quad T_\alpha(\hat{\theta}) = \text{sign}(\hat{\theta}) |\hat{\theta}|^\alpha . \]

Given a received symbol \( y \), MMSE decoding is performed at the receiver to calculate an estimation of the corresponding source symbol. Optimal MMSE decoding can be expressed as

\[
x_{\text{MMSE}} = \mathbb{E} \{ x | y \} = \int X p(x | y) dx = \frac{1}{(p(y))} \int X p(y | x) p(x) dx , \quad (3)
\]

where the mapping function \( M_\delta(\cdot) \) is used to obtain the conditional probability \( p(y | x) \). Note that the integral in (3) can only be calculated numerically because \( M_\delta(\cdot) \) is discontinuous and highly non-linear. To do so, \( x \) is firstly discretized using a uniform step and a mapped value is calculated for each discretized point according to (2). As a result, we obtain a discretized version of \( p(y | x) \). Next, \( p(x) \) is also computed for each point, and hence the calculation of the integral is reduced to multiplicative and additive operations. Since this discretization does not depend on the received symbol, it is calculated once off-line and stored in the decoder.

The proposed analog system can be readily modified to adapt the compression rate from \( N:1 \) to \( N:K \) as shown in Figure 2. Now, blocks of \( c_1 + c_2 \) source symbols are generated at the transmitter: \( c_1 \) symbols are sent through a 1:1 uncoded system and the rest are encoded into \( c_2 / 2 \) channel symbols using a 2:1 system. Since the total number of symbols transmitted through the channel is \( c_1 + c_2 / 2 \), the overall compression rate is \( N/K = (c_1 + c_2) / (c_1 + c_2 / 2) \). By properly choosing \( c_1 \) and \( c_2 \), any compression rate between 1:1 and \( N:K \) can be achieved as long as the condition \( K < N < 2K \) is fulfilled. The total average transmit power per channel symbol should be constrained to one, but the power invested to transmit the \( c_1 \) uncoded symbols and the \( c_2 / 2 \) coded symbols can be different. For this reason, we added a Power Allocation (PA) module in the encoder (see Figure 2) that decides how much power is allocated to each sub-system in order to achieve the best performance.

### III. Testbed Description

A testbed developed at the University of A Coruña (see Figure 3) was used to experimentally evaluate the performance of the analog joint source-channel coding system described above. The testbed has been constructed using Commercial-Off-The-Shelf (COTS) modules from Sundance Multiprocessor [12] for the implementation of the baseband functionalities, and Radio Frequency (RF) front-ends from Lyrtech [13].

The hardware of the testbed is completed with a distributed, multilayer software architecture specifically designed to ease the interaction with the testbed hardware [14–16].

Figure 4 shows a block diagram containing the software
In this section we describe the measurement of the mean performance—in terms of SDR with respect to SNR—of the analog joint source-channel coding scheme described in Section II in an indoor environment. The measured results are compared with both the OPTA and the results obtained from simulations.

A. Closed-Loop Set-up

Narrowband single-antenna measurements were carried out for simplicity reasons and because we already had past experience on similar experiments [14–18]. A feedback channel was implemented to send to the transmitter the SNR estimated at the receiver and select the optimum coding parameter $\delta$ accordingly.

An important feature of our experimental setup is that all signal processing operations were implemented in MATLAB (see Figures 4 and 5). This way our experiments take into account all hardware effects introduced by the hardware (e.g. ADC, DAC and power amplifiers, among others) caused by the specific features of the resultant signals after the encoder such as the high PAPR. Note that the alternative approach of using a channel sounder that measures the channel coefficients to be later utilized in a simulation does not allow us to measure the hardware impairments.

B. A Typical Indoor Environment – the Channel

We evaluated experimentally the performance of the analog joint source-channel coding in an office. The transmit monopole antenna was placed on a table right adjacent to a WiFi access point of the public wireless network at the University of A Coruña. The receive antenna was placed on a table used by one of the employees in the office, hence emulating the actual position that a desktop or a laptop computer would occupy. The transmit and receive antennas were at a distance of approximately 9 m.

C. Quantization of the Transmitted Signals

One of the major hindrances that has to be taken into account when implementing the analog joint source-channel coding is the elevated PAPR of the transmitted signals. For the correctness of the measurement, it is important to ensure that every generated sequence has the same mean power value. In order to guarantee such a constraint we have implemented the following procedures:

- After the encoding stage (see Figure 2), the resulting symbols are normalized to ensure that the whole sequence—including the uncompressed symbols—always has a mean power equal to one.
- Once the symbols are converted to discrete IF signals, they have to be first scaled and then quantized according to the 16 bits DAC resolution (see Figure 4). Given the

IV. EXPERIMENTAL MEASUREMENTS

and hardware elements utilized at the transmit side. Once the Gaussian source is generated, the resulting samples are encoded using the scheme shown in Figure 2 (including the power allocation when it is required). Next, the following steps are carried out:

- The encoded samples are split into two branches.
- For each branch, an up-sampler followed by a squared root-raised cosine pulse-shape filter are responsible of generating the discrete-time baseband signals.
- The resulting signals are I/Q modulated.
- The passband signal is scaled and quantized.
- The resulting signal is stored in the buffer available at the hardware testbed.
- When the transmitter is triggered, such buffer is read cyclically and in real-time by the DAC, which generates a signal at the IF of 5 MHz.
- The resulting analog signal is sent to the RF front-end to be transmitted at the RF center frequency of 5.605 GHz.

At the receiver side, once the transmitter has been triggered, the following steps are carried out (see Figure 5):

- The RF front-end down-converts the signal received by the antenna to the 5 MHz IF frequency.
- The signal is digitized by the ADC and stored in real-time in the buffer.
- Time and frequency synchronization are carried out.
- I/Q demodulation followed by filtering and decimation.
- Next, channel estimation and equalization are performed.
- The resulting observations are normalized and sent to the decoder (see Figure 2).
high PAPR of the resulting signals, we have determined by computer simulations the maximum absolute value that a symbol can reach—including all possible rates and power allocation schemes—and it turns out to be approximately equal to 8.8.

D. Measurement Procedure

The transmission of a symbol sequence involves the following operations:

- The transmitter uses a training sequence to estimate the SNR at the receiver.
- The transmitter, based on the estimated SNR value, selects the optimum value of \( \delta \), generates a new Gaussian sequence, and encodes it using the selected \( \delta \) value.
- The sequence is normalized to ensure a transmit power value equal to one. The resulting normalization factor has to be sent to the receiver to perform the inverse operation before decoding (see Figures 1 and 2).
- Next, the sequence is pulse-shape filtered, scaled, and I/Q modulated (see Figure 4).
- The transmit frame is assembled. In addition to the encoded signal, a preamble, a silence and a training sequence are added for time and frequency synchronization, for noise power spectral density estimation and to equalize the channel at the receiver, respectively.
- The assembled frame is then transmitted.
- At the receiver, the channel is least-squares estimated and equalized; then, the SNR is also estimated; and, finally, the observations are evaluated (see Figure 5).

All frames have the same structure (see Figure 6): a Pseudo Noise (PN) sequence made up of 100 symbols (preamble), a silence of 50 symbols, 16 pilot symbols, and 7200 source symbols that are encoded according to one of the three possible compression rates considered, i.e. 2:1, 10:6, or 10:9. We use a symbol period of \( T_s = 20 \) samples per symbol, with sampling frequency for both the DAC and the ADC fixed to 40 Msample/s. The symbols are pulse-shape filtered using a squared root raised cosine filter with 20 \% of roll-off, resulting in a 2.4 MHz occupied bandwidth at the center frequency of 5.605 GHz. Table I shows the resulting transfer rates for the three different compression rate values, and the number of coded and uncoded symbols obtained when 7200 source symbols are used. Additionally, the number of I/Q symbols that compose the transmit frame, and the frame duration, are indicated. Note that Table I does not include the overheads due to preamble, pilot symbols, and normalization values.

E. Experimental Evaluation

We have experimentally evaluated the SDR of the transmitted sequences with rates 2:1, 10:6, and 10:9, using the MMSE receiver. For the 10:6 and the 10:9 cases, two schemes are evaluated. The first one transmits the whole sequence with the same mean power value. The second one uses the power allocation scheme defined in Section II. In total, five different schemes are measured. In order to obtain the curves shown in Section V, each scheme is measured using 10 different transmit power levels to get different SNR values. With the purpose of increasing the precision of the obtained SDR values, we repeated the experiments 100 times, which resulted in 1000 different realizations per scheme (5000 in total).

V. RESULTS

The obtained results are presented in Figures 7, 8, and 9 for the rates 2:1, 10:6, and 10:9, respectively. The points in the measurement curves are not equidistant because the effective SNR at reception depends on the particular channel realization resulting from each experiment. For the 10:6 and the 10:9 cases, the SDR is measured with and without Power Allocation (PA). For comparison purposes, in all graphs we have included the OPTA curve as well as the corresponding SDR curves when the five different schemes are simulated using an AWGN channel model. In all cases, the \( x \) and \( y \) axes of Figures 7, 8, and 9 respectively show the SNR and the SDR values, both expressed in decibels.

All graphs in Figures 7, 8, and 9 clearly show that the measurement curves perfectly match the simulated ones for SNR values below 20 dB. Note that, since we are considering the SNR at reception (i.e., taking into account the effect of the channel), the static wireless fading channel is actually seen as a deterministic AWGN channel. When the SNR is greater than 20 dB, the SDR curves exhibit saturation effects. Such effects are mainly caused by the limited number of resolution bits of the DAC (16 bits in this case), and also by the high PAPR of the transmitted signals. Notice that at high transmit power levels, the high PAPR makes the transmit power amplifier to
introduce non-linear effects (saturation). Looking at Table I, and at the graphs mentioned above, it is interesting to see how the SDR increases as the transfer rate decreases, as expected.

VI. CONCLUSION

Recently, analog joint source-channel coding has been introduced as a means of achieving near-optimum performance for high data rates with very low complexity. However, no experimental evaluation proving its feasibility in practical applications was previously carried out. In this work, we have described a software-defined radio implementation of a practical wireless communications system that uses analog joint source-channel coding. We have used this software-defined radio implementation to carry out closed-loop narrowband single-antenna measurements in a typical and very realistic indoor scenario, which has forced us to take into account the problems that analog joint source-channel coding schemes pose in practice. The obtained results showed that the analog joint source-channel coding measurements perfectly match those originally reported by simulations in AWGN channels for SNR values below 20 dB, thus clearly demonstrating the feasibility of these analog compression schemes in real environments. For SNR values above 20 dB there is a performance degradation caused by the limited number of resolution bits of the DAC (16 bits in this case), and also by the high PAPR of the transmitted signals, which caused non-linear effects introduced by the power amplifier at the transmit side.

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Fig. 8. 10:6 scheme: OPTA, simulated SDR with and without power allocation (PA), and measured SDR with and without power allocation (PA).

Fig. 9. 10:9 scheme: OPTA, simulated SDR with and without power allocation (PA), and measured SDR with and without power allocation (PA).