TWO-WAY VIDEO COMMUNICATION BASED ON NETWORK CODING

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ABSTRACT

We consider a practical system design for a wireless video conference application, where we exploit the broadcast nature of wireless radio links using network coding. With network coding, the number of necessary downlink transmissions from the intermediate node to the two users is reduced, and thus the throughput is increased. We develop two systems, one based on amplify-and-forward and another on decode-and-forward technique, and compare them to traditional communication systems that do not use network coding. Our practical designs exploit the latest in 3-D wavelet-based scalable video coding and channel coding. Simulation results confirm the advantages of the proposed video communication schemes over conventional ones.

Index Terms— Network coding, video transmission, error control coding, wireless communications

1. INTRODUCTION

Wireless full-duplex transmission, where users exchange their information over a wireless radio channel with the help of intermediate nodes, has become a common communication scenario. Possible applications range from classic cell-phone voice conversations, interactive image/message exchange, and file sharing, to wireless videophony/video conference. Another emerging scenario is multimedia transmission over multihop wireless ad hoc and sensor networks, where intermediate network nodes serve as relays.

Motivated by increased bandwidth and power demands for these applications, in this paper we consider wireless interactive multimedia communications. We start with a simple scenario. Suppose that two users exchange their information packets via a base station. User 1 generates its message, encodes it, and sends the resulting packet X_1 over an uplink wireless channel to the base station. Similarly, User 2 encodes its message and sends encoded packet X_2. We assume that all communication has to go via the base station, that is, User 1 cannot overhear signals sent directly from User 2, and vice versa. Then the role of the base station is to relay the signals it has received. Traditionally, this is done by transmitting X_2 (the signal received from User 2) to User 1 and X_1 to User 2 via two orthogonal downlink channels.

However, instead of transmitting X_1 and X_2 separately, the base station can broadcast $X_1 + X_2$ over the wireless radio link. Then User 1 subtracts X_1 from the received (possibly corrupted) $X_1 + X_2$ and reconstructs the desired message. Similarly, User 2 subtracts X_2 and recovers X_1. Thus, instead of two separate transmissions from the base station, only one transmission is needed, which decreases power consumption. Indeed, if we suppose that X_1 and X_2 are two bitstreams of equal length, $+$ is the modulo-2 operation, and each transmission consumes the same amount of power, then the length of the packet $X_1 + X_2$ broadcast from the base station is the same as the length of X_1 (or X_2), and the required transmitter power is half that of separate transmissions of X_1 and X_2.

The idea outlined above exploits the broadcast nature of the wireless medium and is based on the concept of *network coding*. Network coding [1] is a novel technique originally proposed for multicasting information over wireline networks of noiseless channels. As opposed to traditional routing (where intermediate nodes act only as routers that forward received packets), network coding can achieve multicast capacity [2]. It is based on combining received information packets; that is, each intermediate network coding node computes a certain encoding function of the received packets and forwards the resulting packet towards its destination. It is enough to restrict the encoding function to time-invariant linear functions and still achieve the multicast capacity [3]. This has made network coding practical for many application scenarios, where each intermediate node needs only to compute a linear combination of the symbols from the received packets (i.e., a weighted sum of the symbols with weighted coefficients randomly taken from a large enough field), and forward the generated packet toward its destinations. We stress again that the resulting packet is of the same size as the incoming packets.

Strong potentials of network coding (in terms of reduced transmission power) in wireless packet networks were recently pointed out in [4, 5, 6, 7, 8] and references therein. It was shown in [5] that the full-duplex packet communication can be viewed as a single virtual multicast session, and thus network coding is the optimal solution. The provided scheme, dubbed physical piggybacking, achieves a throughput improvement compared to conventional routing based on two unicast sessions (each for sending one information packet).

In this paper we develop practical coding systems for the two-way video communication setup (e.g., 3G videophony). The video sequences are encoded using a wavelet-based scalable video coder [9] (though any other video coder can be used), protected with short-length low-density parity-check (LDPC) codes [10], modulated, and sent over wireless channels.

We design two schemes: the first one resembles amplify-and-forward coding, where the base station does not perform channel decoding; it only applies network coding on the received signals. In the second scheme, based on decode-and-forward, the base station performs channel decoding of received signals, before applying network coding on the reconstructions. Our first scheme is more suitable for real-time applications since decoding at the base station is avoided.

Several research groups have addressed the problem of practical

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network coding for wireless communications. For example, channel coding and network coding are combined for *one-way* communication with one intermediate relay node in [11]. In [12], transmission schemes that enable a set of terminals to communicate with a common destination are proposed; the schemes combine distributed source coding, channel coding, and network coding.

The paper is organized as follows. In Section 2 we describe our proposed systems and two conventional systems used as benchmarks. In the next section we describe our practical system design using the video coder of [9] and two LDPC codes. In Section 4 our experimental results are presented. The last section concludes the paper and suggests future work.

2. SYSTEM DESCRIPTION

In this section we describe the two proposed systems based on network coding and two corresponding benchmark systems that do not exploit network coding.

We consider a simple two-way communication scenario for exchanging messages between two users via the base station. User i, $i = 1, 2$, encodes its video sequence, protects it, and sends the resulting signal X_i over a wireless channel to the base station. The two signals received at the base station are: $Y_i = X_i + Z_i$, where Z_i is a Gaussian noise independent of the source signals. We consider two coding techniques at the base station: amplify-and-forward (AF) and decode-and-forward (DF).

In AF, the base station does not decode the signal it has received, but only amplifies it before forwarding to the users. In the benchmark system the signals received by Users 1 and 2, respectively, are: $Y_3 = A_{AF_b}(X_2 + Z_2) + Z_3$ and $Y_4 = A_{AF_b}(X_1 + Z_1) + Z_4$, where Z_3 and Z_4 are i.i.d. Gaussian noises, and $A_{AF_b} \geq 1$ is the amplification coefficient of the (b)enchmark scheme.

In the proposed system by applying network coding at the base station, we aim at reducing communication load. The base station simply sums the signals it has received from the two users, amplifies the resulting signal, and broadcasts it. Then, the signals received by User 1 and User 2, respectively, are: $Y_3 = A_{AF_p}(X_1 + Z_1 + X_2 + Z_2) + Z_3$ and $Y_4 = A_{AF_p}(X_1 + Z_1 + X_2 + Z_2) + Z_4$, where $A_{AF_p} \geq 1$ is the amplification constant of the (p)roposed scheme. User 1 subtracts $A_{AF_p}X_1$ from Y_3 yielding $Y_3' = A_{AF_p}X_2 + A_{AF_p}(Z_1 + Z_2) + Z_3$, and then decodes it to reconstruct the source. Similarly, User 2 subtracts $A_{AF_p}X_2$ from Y_4 yielding $Y_4' = A_{AF_p}X_1 + A_{AF_p}(Z_1 + Z_2) + Z_4$.

In DF, the base station decodes the received signal, reencodes and modulates it, and then forwards the result to the users. In the benchmark system the base station decodes separately Y_1 and Y_2, reconstructing source signals (encoded video). Reconstructed signals are reencoded and modulated to X_3 and X_4, respectively. Then the base station sends X_3 to User 1 and X_4 to User 2. Note that, the base station does not need to use the same codebook as the users. The signals received by Users 1 and 2, respectively, are: $Y_3 = A_{DF_b}X_3 + Z_3$ and $Y_4 = A_{DF_b}X_4 + Z_4$.

In the proposed system, as in conventional DF, the base station recovers the sent codewords from Y_1 and Y_2, and then modulates them to X_3 and X_4; X_3 and X_4 are summed and broadcast to the users. Thus, the received signals by User 1 and 2, respectively, are: $Y_3 = A_{DF_p}(X_3 + X_4) + Z_3$ and $Y_4 = A_{DF_p}(X_3 + X_4) + Z_4$. User 1 subtracts $A_{DF_p}X_3$ from Y_3 getting $Y_3' = A_{DF_p}X_4 + Z_3$, and then decodes it before reconstructing the source. Similarly, User 2 subtracts $A_{DF_p}X_4$ from Y_4 getting $Y_4' = A_{DF_p}X_3 + Z_4$ and reconstructs the encoded video signal.

We note that all benchmark systems effectively have double the rate in the downlink regime: in the proposed network coding schemes, the base station broadcasts one packet, while in the conventional schemes the base station sends one packet to User 1 and another one to User 2.

3. PRACTICAL SYSTEM DESIGN

In this section, we describe our two practical system designs one based on AF and the other on DF, illustrated in Fig. 1.

In both systems, the encoder at the user side is the same. User i, $i = 1, 2$, compresses its video signal and protects the resulting block using an LDPC code; then BPSK modulation is performed to obtain signal X_i which is sent to the base station.

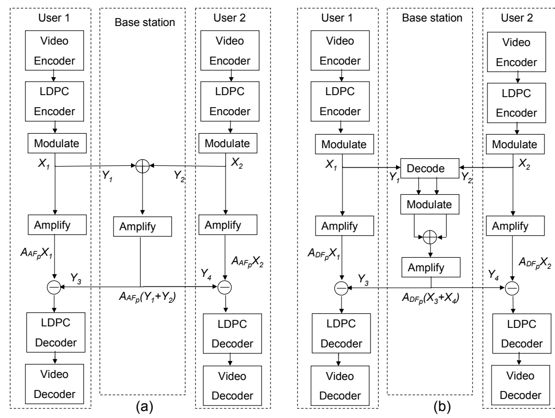

Fig. 1. The block scheme of the proposed systems: (a) AF, (b) DF.

In the proposed AF system, the base station sums the two signals received from the users, Y_1 and Y_2, and broadcasts the result using power $A^2_{AF_p}$. Neither error protection/video decoding/encoding nor demodulation are required at the base station. All network coding operations are conducted on the physical layer in the signal domain. Upon receiving a signal from the base station, User i subtracts $A_{AF_p}X_i$ (which is perfectly known), and attempts to reconstruct the source block, using LDPC decoding and then performs video decoding. Thus each user performs one conventional LDPC and video encoding and decoding.

In the proposed DF system, the base station first independently decodes the two received signals Y_1 and Y_2 using an LDPC decoder. Next, it modulates the recovered codewords to X_3 and X_4, and forms the sum $X_3 + X_4$ which is broadcast with power $A^2_{DF_p}$. Decoding at the users' side is the same as with the AF system. With DF, each user needs one LDPC encoder/decoder pair, one video encoder/decoder pair, and the base station needs one LDPC decoder.

To evaluate the performance of the proposed systems we now compare them to conventional AF and DF systems, which exploit the same video and error protection coder, but do not rely on network coding. In the benchmark AF scheme the base station simply amplifies the signal received from User 1 and forwards the result to User 2; similarly, the signal received from User 2 is amplified before being broadcasted. In benchmark DF scheme, the base station decodes signals received from the two users, reencodes them and modulates to X_3 and X_4, and sends separately the two signals to User 1 and User 2.

Note that the proposed systems do not increase complexity compared to traditional AF and DF systems, as they require only one simple signal summation. The AF scheme is of lower complexity and might be preferable in real-time applications as it does not require decoding at the base station.

For video coding, we resort to the latest technology in scalable video compression (3-D wavelet video coding) of [9]. Recent research results (see [13] and references therein) on 3-D scalable wavelet video coders based on the framework of motion-compensated temporal filtering (MCTF) [14] have shown excellent performance, competing with the best MC-DCT based standard video coder. The coder can be viewed as the extension of wavelet-based coding in JPEG2000 from 2-D images to 3-D video. It nicely combines scalability features of wavelet-based coding with motion compensation, which has been proven to be very efficient and necessary in MC-DCT based standard video coders. In the next section, we show our results obtained using 3-D wavelet coder of [9]. However, we note that our proposed systems do not depend on the choice of video code; hence for time constrained applications, standard low-complexity MC-DCT coders can be applied.

LDPC codes [10] are capacity-approaching linear codes, suitable for wireless communications due to their excellent performance and low decoding complexity. We apply two different high-performance LDPC codes, which are designed such that the constraints imposed on the construction of the parity check matrix ensure a bipartite graph with good structure and connectivity as well as low number of short cycles. The first code is MacKay's pseudo-random construction [15]. However, this code suffers high encoding complexity as it requires large amount of information to specify nonzero elements in the party-check matrix. For a more practical system, we employ structured LDPC codes, Quasi-Cyclic (QC) LDPC codes, with lower complexity and comparable or better performance than code of [15]. The QC LDPC codes can be encoded using simple shift-registers with complexity linearly proportional to the code length [16]. They are constructed following the method of [17], based on permutation matrices. We ensure that these regular LDPC codes have no short cycles of length four in the Tanner graph; hence, good performance is achieved with low encoding complexity.

Since scalable codes are very sensitive to channel errors (due to error propagation), we stop decoding when the first uncorrected error is detected [18, 19]. (We assume perfect error detection with the employed LDPC codes.) The receiver in that case reconstructs the video using only packets decoded before the first error.

4. SIMULATION RESULTS

We compare the two proposed systems with the benchmark AF and DF without network coding. The same video coder and LDPC code are always used by both users. We encoded 300 frames of the 352×288 CIF "Foreman" video sequence at 30 f/s. The communication channels are modeled as independent additive white Gaussian noise (AWGN) channels. BPSK modulation is always assumed.

We present results for two different LDPC codes. One is a pseudo-random LDPC code generated using MacKay's code-six construction method [20]. The second code is QC LDPC code designed using the method of [17]. Both codes are decoded with the sum-product algorithm, with a maximum of 50 decoding iterations. Since our goal is to design a low-complexity coding system suitable for real-time communication scenarios, we use small block lengths of up to 5000 bits. In that regard, our video coder may be replaced by one with lower complexity.

To show the impact of the block length N on the performance, in the first experiment we compare two proposed AF schemes: one with $N = 1519$ bits and the other with $N = 4000$ bits. Both schemes use the QC LDPC code of [17] with $A_{AF_p} = 2$. The transmission rate R for both schemes is 40 kilobits per second [kb/sec]. The results in the form of expected PSNR of the Y-component averaged over all 300 frames as a function of SNR in the uplink are shown in Fig. 2. The figure shows a loss of about 1 dB due to a smaller codeword size N. The code rates are 0.84 for the code with $N = 4000$ and 0.87 for the second code with $N = 1519$.

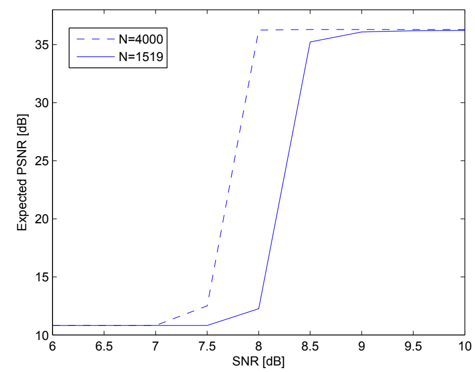

Fig. 2. The expected PSNR [dB] as a function of SNR at the uplink. Results are given for the proposed AF scheme for two different codeword lengths N.

Fig. 3 compares the proposed AF scheme to the AF benchmark at different transmission rates. Both schemes use the QC LDPC code of [17] with $A_{AF_p} = 2$. The SNR in the uplink is fixed at 9 dB. A gain of roughly 3 dB due to network coding can be observed.

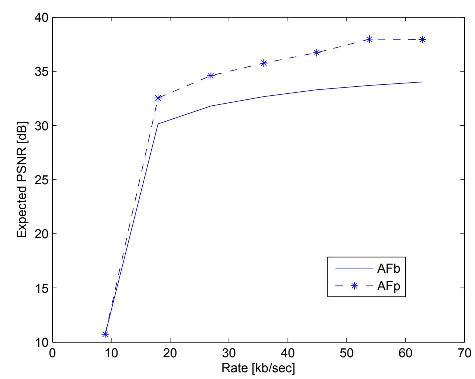

Fig. 3. The expected PSNR [dB] as a function of the transmission rate R. Subscript 'b' is used for the benchmark systems and 'p' for the proposed. SNR at the uplink is 9 dB.

In next simulations we compare the performance of the proposed AF and DF schemes to the corresponding benchmarks. The codeword length is fixed at 4000 and 4489 bits for the QC LDPC and pseudo-random LDPC code, respectively. The code rate is always 0.87 for the QC LDPC and 0.85 for the pseudo-random LDPC code. The amplified gains $A_{AF} = A_{DF}$ are fixed at 2. The obtained results are presented in Figs. 4 and 5. The main conclusions from the figures are the following. First, DF significantly outperforms AF as expected. Second, the proposed AF scheme shows better performance than the corresponding AF benchmark scheme. Third, the two DF schemes have similar performance. Finally, the schemes with QC LDPC code perform slightly better.

We note that in all simulations, in downlink, our network coding schemes use double the rate since one N-length packet is broadcast to both users, as opposed to the benchmark systems where one N-length packet is sent to User 1 and another one to User 2 over two

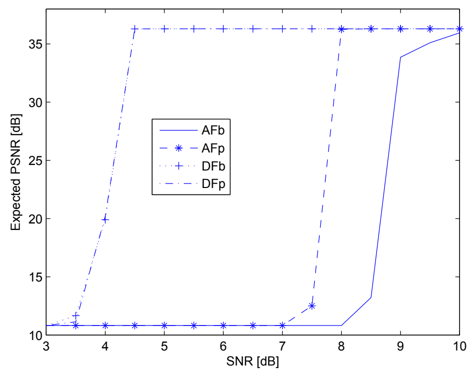

Fig. 4. The expected PSNR as a function of SNR at the uplink. Subscript 'b' is used for the benchmark systems and 'p' for the proposed. The QC LDPC code is used with $N = 4000$ bits.

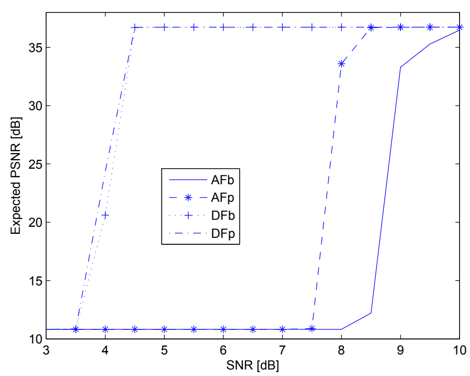

Fig. 5. The expected PSNR as a function of SNR at the uplink. Subscript 'b' is used for the benchmark systems and 'p' for the proposed. The pseudo-random LDPC code is used with $N = 4489$ bits.

orthogonal channels. This is not an issue in cellular systems where such communication is natural. (Since this is basically due to broadcasting gain over peer-to-peer connection, we did not include it in our comparison.) Our results show that even under such comparison conditions, our schemes based on network coding are competitive.

5. CONCLUSION AND FUTURE WORK

We develop two practical systems for two-way real-time video communication among two users via a base station. The systems use scalable video coding and are based on the concept of network coding that exploits the broadcast nature of the wireless radio link. This way, the number of necessary transmissions from the base station is reduced, and thus a higher throughput achieved. The proposed systems based on network coding also demonstrate similar or improved performance over conventional systems (without network coding) based on DF and AF, respectively.

Future work will include more practical setups with fading channels and more than one intermediate node, as in ad hoc networks. We also plan to investigate possible improvement with joint source-channel coding and unequal error protection [19]. Video conference with more than two users is another exciting research topic.

6. REFERENCES

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