VIDEO TRANSMISSION OVER A RELAY CHANNEL WITH A COMPRESS-FORWARD CODE DESIGN

A Thesis

by

CHAITANYA POLAPRAGADA

Submitted to the Office of Graduate Studies of
Texas A&M University
in partial fulfillment of the requirements for the degree of

MASTER OF SCIENCE

May 2008

Major Subject: Electrical Engineering

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Approved by:

Chair of Committee,	Zixiang Xiong
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ABSTRACT

Video Transmission Over a Relay Channel with a Compress-Forward Code Design. (May 2008)

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Chair of Advisory Committee: Dr. Zixiang Xiong

There is an increasing demand to support high data rate multimedia applications over the current day wireless networks which are highly prone to errors. Relay channels, by virtue of their spatial diversity, play a vital role in meeting this demand without much change to the current day systems. A compress-forward relaying scheme is one of the exciting prospects in this regard owing to its ability to always outperform direct transmission. With regards to video transmission, there is a serious need to ensure higher protection for the source bits that are more important and sensitive. The objective of this thesis is to develop a practical scheme for transmitting video data over a relay channel using a compress-forward relaying scheme and compare it to direct and multi-hop transmissions. We also develop a novel scheme whereby the relay channel can be used as a means to provide the required unequal error protection among the MPEG-2 bit stream.

The area of compress-forward (CF) relaying has not been developed much to date, with most of the research directed towards the decode-forward scheme. The fact that compress-forward relaying always ensures better results than direct transmission is an added advantage. This has motivated us to employ CF relaying in our implementation.

Video transmission and streaming applications are being increasingly sought after in the current generation wireless systems. The fact that video applications are bandwidth demanding and error prone, and the wireless systems are band-limited and

unreliable, makes this a challenging task. CF relaying, by virtue of their path diversity, can be considered to be a new means for video transmission.

To exploit the above advantages, we propose an implementation for video transmission over relay channels using a CF relaying scheme. Practical gains in peak signal-to-noise ratio (PSNR) have been observed for our implementation compared to the simple binary-input additive white Gaussian noise (BIAWGN) and two-hop transmission scenarios.

To my Parents and my dear brother

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TABLE OF CONTENTS

LIST OF FIGURES

FIGURE Page

LIST OF TABLES

CHAPTER I

INTRODUCTION

A. Compress-forward code design

The unreliability of wireless channels has spurred an interest in developing novel algorithms and techniques to enable successful transmission of data. This becomes especially important for the next generation cellular systems which are required to support multimedia applications. To support these applications, next generation systems must be able to provide a high data rate and guarantee Quality of Service. One of the most important concerns while transmitting data over a wireless channel is that there would be a wide fluctuation in the signal attenuations known as fading. Diversity techniques, where independently fading copies of the source signal are transmitted, are the best way to counter the effects of fading. They would be an ideal choice for meeting the growing demands of next generation wireless systems. Spatial diversity is one of the important diversity schemes, where the source signal is transmitted from multiple geographically separated locations to the destination, thus providing multiple paths to the destination. This way signals along different paths might encounter independent fading, thus improving the chances of reliable transmission.

One of the methods of achieving spatial diversity in cellular systems is by the use of multiple antennas. However, this is practically unfeasible due to size constraints. One of the exciting developments in this regard is cooperative communication [1], where the source makes use of a partner node to transmit data to the destination. A simplest case of cooperative communication is that of a relay channel, where the partner does not send any data by itself but just relays data from the source to the destination node. The idea of using relay channels for improving communication was first proposed by Van Der Muelen [2]. An exact expression for the capacity of a general relay channel is yet to be

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determined. Cover and El Gamal [3] later derived much tighter bounds and proposed some schemes to achieve the lower bound. The relay signal processes the information received from the source before forwarding it to the destination. This along with the data sent by the source is used to recover the original source information. Relay channels can be broadly classified as Decode-Forward and Observe-Forward schemes based on the type of processing that takes place at the relay node. Over the past few years, extensive research has been directed towards developing practical Decode-forward relaying schemes [4] [5]. In these schemes, the relay decodes the source transmitted data and re-encodes it before forwarding it to the destination. Hence, this scheme relies heavily on the strength of the source-relay channel. It performs much better whenever source and the relay nodes are physically close to each other. On the other hand, when the channel between the source and relay nodes is worse than the source-destination link, Decode-Forward relaying would perform even worse than direct transmission.

Compress-Forward (CF) is an observe-forward scheme, where the relay node quantizes the source transmitted data before forwarding it to the destination. CF relaying is best suited whenever the relay and destination nodes are close to each other. More importantly, it always outperforms direct transmission. In spite of this, very few practical CF schemes have appeared in the literature. Arnab *et al.* [6] proposed a practical scheme for compress-forward relaying assuming a half-duplex Gaussian relay channel. Half-Duplex communication is assumed realizing the complexity of performing accurate interference cancellation and shielding at the relay. The inputs to the relay channel are assumed to be Gaussian and are made to operate at low SNR, where a significant gain is achieved with Compress-Forward relaying. The paper uses scalar quantization for forming the estimate of the relay received signal. Maximum Ratio Combining [7] of the relay transmitted and the Broadcast mode Destination received signal is used to exploit spatial diversity. However, the source node is made inactive in Multiple Access mode. While this reduces the complexity to a great extent, the source node now sends less data to the destination, thus causing a loss in degrees of freedom. By keeping implementation complexity to a minimum and using the powerful properties of LDPC codes, this scheme comes to within 1.67dB of the achievable rate of a compress-forward relaying scheme. Another practical compress-forward code design was proposed by Liu *et al* [8]. In this scheme, the source is kept active during the entire time of communication. The relay received signal is Wyner-Ziv Coded and channel coded jointly using a DJSC encoder before being sent to the destination. As a result of using powerful source and channel coding techniques at the relay, this practical CF code outperforms the existing practical DF schemes, even in rate regions where theoretically it is inferior. It comes close to 0.76dB from the theoretical CF limit.

Achievable rate for a general CF relay was proposed in [9]. However, in this work we assume that the inputs to the CF relay are binary modulated. Later, we derive the achievable rate assuming BPSK inputs making use of rate-distortion theory. The relay is assumed to be operating in a half-duplex fashion for practical purposes. Also, time-division duplexing instead of a frequency-division duplex is assumed for ease of implementation.

B. Video transmission over relay channels

There is an increasing demand to support multimedia applications like video streaming in the next generation wireless networks. Video transmission is bandwidth demanding and error prone. On the other hand, wireless networks are unreliable and bandwidth limited. Hence, there are strong interests to use diversity techniques [10] to ensure reliable transmission of video. It must be noted that not all source bits in a video bit stream are equally important. Hence, there is a need to ensure higher protection for the source bits that are more important. This can be easily achieved by the use of relay channels. The partition set of the video data which has a higher importance is transmitted by the source during the Broadcast mode. This would be processed by the relay node and along with the MAC mode source data will be transmitted to the destination. By sending the less important video information from the source during the MAC mode, the required unequal error protection can be achieved. This scheme would provide spatial diversity only for the important source information. However, there is a need to develop a scheme which would partition the video bit stream into two sets of unequal importance.

In this work, we employ MPEG-2 video coding [11]. Apart from providing higher data rates compared to MPEG-1, MPEG-2 also has algorithms which enable efficient encoding of interlaced video. MPEG-2 video coding also provides a number of subsets of the full standard called profiles, which define the set of algorithmic tools and levels, which set the constraints on the parameter values. It provides bit rate reduction by exploiting both spatial and temporal redundancies present in video data. To exploit unequal error protection, we divide the MPEG-2 bit stream into two data sets, one consisting of the I-frame information and the other consisting of P and B-frame information.

Wireless networks being highly unreliable cause significant signal loss during transmission. Hence, there is a serious need to employ error concealment techniques [12] [13] for video transmitted over wireless channels. This becomes even more essential when dealing with the transmission of MPEG-2 data, where even a single bit error might cause much damage to the video quality. This is because of error propagation owing to the use of variable length coding, slice techniques etc. For example, in the latter case, an error in the first bit in a slice might render the remaining information in the slice undecodable. The first and the most important step towards error concealment in video is error detection [14] [15]. Inaccuracies in detecting the erred regions might cause the video quality to degrade further by error concealment. We primarily use transform domain techniques and information from the MPEG-2 decoder to detect erred regions in the video frames. In this work, we deal only with the error concealment techniques which involve post-processing at the decoder side. By using these techniques, we need not add any redundancy to the source data and do not need to make any changes to the encoder. The intra-frame information in the video is concealed exploiting the spatial redundancy, while inter-frame information is recovered using a motion-compensated temporal prediction [12]. This is much better than the simple copy concealment [14], which might cause a high degradation in the video quality when there is a large motion between the frames.

The partitioning of MPEG-2 data, error detection and error concealment of transmitted video are the key principles we have used in our implementation of video transmission over relay channels using a CF code design.

C. Contribution of this work

The primary aim of this research work is to present the advantages of transmitting MPEG-2 data over a relay channel as opposed to direct and two-hop channels. In the process, we propose a practical scheme for video transmission over a relay channel with a Compress-Forward relaying scheme. We also present a scheme to partition the MPEG-2 data so as to provide more protection to the important source information. By doing so, we provide unequal error protection to video data, thus exploiting the gains of spatial diversity provided by the relay channel. For protecting video data from the transmission errors, we employ some of the error detection and concealment techniques which help in recovering from these errors. A rate-distortion approach for finding the achievable rate of a compress-forward relay with BPSK inputs has also been proposed. Finally, we compare the PSNR values of the decoded video obtained by transmission over a simple BIAWGN channel and a two-hop channel with our proposed compress-forward relaying scheme.

D. Organization of thesis

The rest of the thesis is organized as follows. Chapter II deals with the area of Compress-Forward relaying scheme. In this chapter, we provide the system model, code designs and achievable rates for a CF relaying. Video transmission over wireless channels is dealt with in Chapter III. Starting with the current research and related work in this area, we give an overview of MPEG-2 coding followed by various error detection and concealment schemes. Chapter IV presents our proposed implementation, simulation scenarios and results. Chapter V concludes this thesis and throws light on future scope for this work.

CHAPTER II

COMPRESS-FORWARD RELAYING

In this chapter, we discuss some of the theoretical aspects of Compress-Forward relaying scheme and present its achievable rate results. The organization of this chapter is as follows. In Section A, we introduce a simple three node relay channel and its operation. A system model for the relay channel with CF scheme is presented in Section B. Finally, we present the achievable rates for a general and binary-input relay channels with a CF relaying scheme in Section C.

A. Relay channel

Fig. 1 shows the basic set-up of a three node relay channel, consisting of a source, destination and an intermediate relay node. γ_{SR}, γ_{SD} and γ_{RD} denote the gains of the source-relay, source-destination and relay-destination channels respectively. By overhearing the source transmitted data before forwarding it to the destination, the relay node provides a path diversity between the source and the destination nodes. The destination thus receives two noisy versions of the source transmitted data. By jointly decoding them and thus exploiting spatial diversity [16], a more reliable estimate of the source information is available at the destination. This spatial diversity is highly important for the current day wireless networks which are prone to outage. A more recent development in this direction is that of cooperative diversity [1, 10], where along with sending its own data, every user also needs to relay his partner's data to the destination. A relay channel is thus a basic building block for cooperative communication.

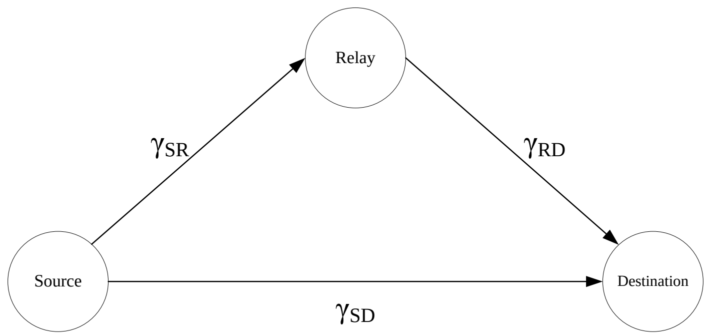

Fig. 1. A three-node relay channel.

The data sent from the source to the relay is processed before sending it to the destination. As was first proposed in [3], based on the type of processing, the relay channel may be broadly classified as either Decode-Forward or Observe-Forward. In the Decode Forward scheme, the relay node fully decodes the source transmitted codeword and re-encodes it before sending it to the destination. The destination node then tries to recover the source message from the source and relay received data using methods such as backward decoding, successive interference cancellation etc. Since the relay node must be able to decode the source transmitted codeword, it is highly necessary that the source-relay channel is strong. However, if the source-relay channel is poor compared to the source-destination channel, decode-forward relaying performance would be even worse than direct transmission. On the other hand, no decoding is involved in the Observe-forward scheme and the relay just processes the source codeword before sending it to the destination. In Amplify-Forward, one of the Observe-Forward schemes, the relay node just amplifies the source message. However, this is technologically non-trivial. Also, if the source-relay channel is bad, amplification would further increase noise. Compress-forward is an important Observe-Forward scheme, where the relay compresses the source data before sending an estimate to the destination for joint decoding.

The relay node can either act in a Half-Duplex or a Full-Duplex manner. In Full-Duplex mode, the relay node can transmit and receive in the same time/frequency band. While Full-duplex relaying is desirable, it is impractical as the difference in transmit and receive signal levels necessitates an isolation between the two [17]. On the other hand, in half-duplex mode, the relay node transmission and reception occur at different times (Time-Duplexing) or at different frequencies (Frequency-Duplexing). Theoretically both time-duplexing and frequency-duplexing should yield identical results. However, time-duplexing has significant implementation gains over frequency-duplexing. This is based on the fact that in frequency-duplexing, isolating transmit and receive signals at the relay requires higher-order filters or a large guard band, which has a high implementation complexity. On the other hand, time-duplexing just requires packet-level synchronization, a common feature in the current day wireless systems. Hence, we will use time-division duplexing for our work.

The achievable rates for a Decode-Forward and Compress-Forward Relays were proposed in [9] assuming a Time-Division Half-Duplex relaying. The same results are also applicable to the Frequency-Division case when the inter-user channel gains γ_{SR}, γ_{SD} and γ_{RD} are fixed. Practical implementations of Decode-Forward Relaying were discussed in [4] [5]. Compress-Forward Relaying implementations were given in [6] [8] [18]. While Decode-Forward Relaying is helpful when the source-relay link is good, it can perform worse than direct transmission if the link is in outage. On the other hand, Compress-Forward (CF) would give best results when the Relay-Destination link is strong but always outperforms direct transmission.

B. System model

Fig. 2 illustrates how communication takes place in a half-duplex relay channel. The inter-user channel is assumed to be Gaussian and that the channel coefficients γ_{SR}, γ_{SD} and γ_{RD} are fixed during the time of communication. We consider a time-division duplex relaying, that is the relay node cannot transmit and receive data at the same time. The total time of communication is considered to be 1 unit and divided into two time fractions of durations α and $(1- \alpha)$ respectively. During the first time fraction called the Broadcast (BC) mode, source transmits information to the relay and destination nodes.

During the second time fraction called the Multiple Access (MAC) mode, the relay and source nodes transmit information which is received by the destination mode. The naming of the modes is consistent with the type of channel in each case.

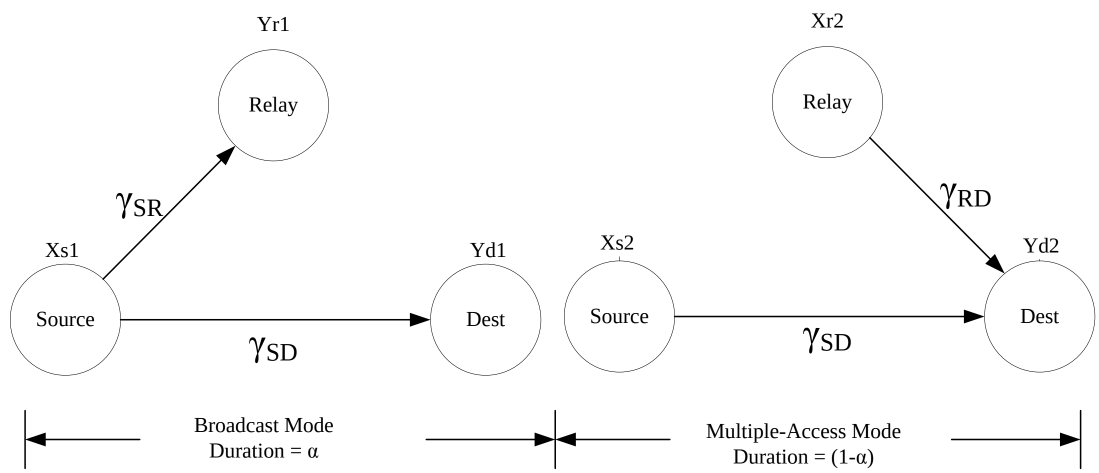

Fig. 2. Illustration of the broadcast and MAC modes of a half-duplex relay channel.

The notation used for various signals in Fig. 2 is as follows. Signal names X, Y and V are used to denote the transmitted and received signals respectively. The s, r and d in the first part of subscript denote the signals at source, relay and destination nodes respectively. In the second part of the subscript, 1 denotes a BC mode signal, while 2 denotes a MAC mode signal. With this notation, the system model can be represented by the following equations.

$$Y_{d1} = \gamma_{SD} * X_{s1} + w_1 \tag{1}$$

$$Y_{r1} = \gamma_{SR} * X_{s1} + w_2 \tag{2}$$

$$\overline{Y}_{r1} = Y_{r1} + w_q \tag{3}$$

$$Y_{d2} = \gamma_{SD} * X_{s2} + \gamma_{RD} * X_{r2} + w_3, \tag{4}$$

with the power constraints

$$E[X_{s1}^2] \leq P_{s1} \tag{5}$$

$$E[X_{s2}^2] \leq P_{s2} \tag{6}$$

$$E[X_{r2}^2] \leq P_{r2}, \tag{7}$$

where P_{s1}, P_{s2} denote the average powers of the source signal in BC and MAC modes respectively and P_{r2} denotes the average power of relay signal in MAC mode. From equations (5)-(7), the average total power constraint for the relay channel is given by

$$P = \alpha * P_{s1} + (1\text{-}\alpha) * (P_{s2}+P_{r2}) \tag{8}$$

Here α denotes the time fraction of BC mode of transmission. For fairness in comparison, we assume that the source transmission power in the direct transmission case is equal to the total relay power P given by (8). Note that X_{r2} is formed after channel coding and modulation of $\overline{Y}_{r1}$. Also, w_1, w_2 and w_3 represent the inter-channel noise which is assumed to be zero mean, unit-variance Gaussian sources ~ $N(0, 1)$. The quantization noise w_q is assumed to be zero-mean Gaussian with variance σ_q^2.

For simulation purposes, we assume that the source, relay and destination nodes are collinear. This would give a simple characterization of the relay position without affecting the results. As shown in Fig. 3, the source-destination distance is assumed to be unity and the source-relay distance is assumed to be d.

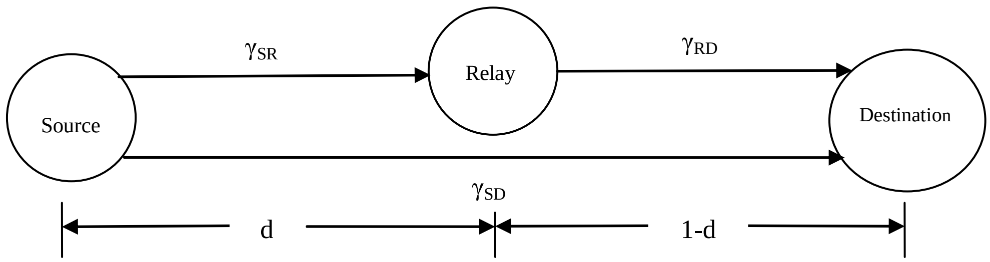

Fig. 3. Simulation set-up of the relay channel.

With the set-up in Fig. 3, the channel gains will be as follows. Normalizing γ_{SD} to 1, the source-relay channel gain, $\gamma_{SR} = 1/d^{\beta}$ and the relay-destination channel gain, $\gamma_{RD} = 1/(1\text{-}d)^{\beta}$, where β is the channel attenuation exponent. We make an important assumption that all the transmitters and receivers will have complete knowledge of these channel gains known as channel state information at transmitters and receivers (CSITR). Also, we only consider the non-fading case i.e. the channel gains are assumed to be constant during the time of communication. Based on this system model, we derive the capacity of a general Relay channel.

C. Achievable rate of a CF relaying scheme

In this section, we give the achievable rate for a general-input relay channel and its associated CF coding scheme [9]. Based on this, we derive the achievable rate of the relay channel with BPSK inputs.

1. General relay channel

The CF coding scheme [6] employed for the relay channel is as follows. The total information is divided into two parts namely (M_1, M_2). We intend to transmit the M_1 data during BC Mode and M_2 data during MAC mode. First, we encode and modulate the M_1 data to obtain the codeword X_{s1}. During the BC mode, X_{s1} is transmitted to both the relay and the destination. This codeword is received as Y_{r1} by the relay and as Y_{d1} by the destination. Neither the relay nor the destination tries to decode the received data at this point of time. The relay will quantize Y_{r1} to obtain $\bar{Y}_{r1}$, which is then encoded to generate the codeword X_{r2}. The rate of compression R_2 is chosen such that it is less than or equal to the capacity of the relay-destination link. M_2 data is encoded to generate the codeword X_{s2}. During the MAC mode, the source sends X_{s2} to the destination, while relay sends X_{r2}. We send X_{s2} at a rate R_1, which could be decodable at the destination.

Fig. 4 shows the set-up of the relay channel during the Broadcast mode which resembles a 1x2 Single-input Multiple-Output (SIMO) channel. If lossless communication over the relay-destination link were possible, the capacity of the SIMO channel would have been $I(X_{s1}; Yr1, Yd1)$. However, the capacity of the channel is only $I(X_{s1}; Yd1, \bar{Y}_{r1})$, since we can only send $\bar{Y}_{r1}$ reliably.

Fig. 5 shows the set-up of the relay channel during the Multiple Access mode which resembles a 2x1 Multiple-input Single-Output (MISO) channel. Note that X_{s2} is formed by encoding $\bar{Y}_{r1}$. We can communicate X_{s2} over the Source-Destination channel at a rate R_1 and X_{r2} over the Relay-Destination channel at a rate R_2 reliably if R_1 and R_2 are rate points in the capacity region of a Multiple-Access channel.

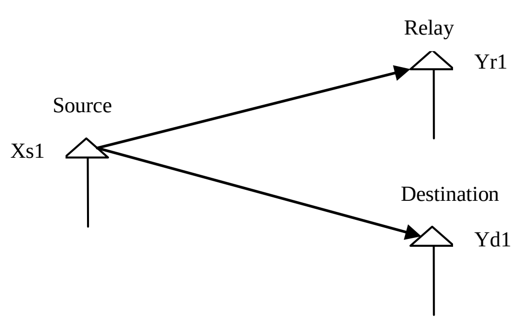

Fig. 4. A 1x2 SIMO system during the broadcast mode.

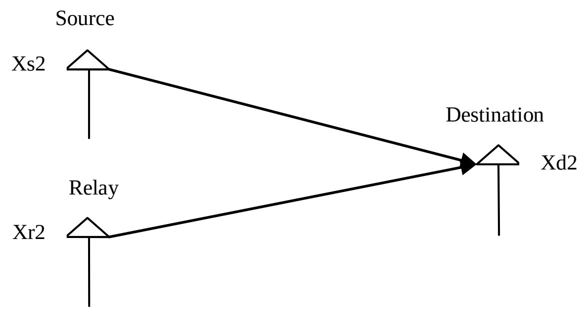

Fig. 5. A 2x1 MISO system during the multiple-access mode.

With the above coding scheme, we can describe the achievable rate of a general CF relay channel. The achievable rate of a Half-Duplex Relay Channel with CF relaying [9] is given as follows:

$$R_{CF} = \sup_{0\leq\alpha\leq 1}[\alpha * I(X_{s1};Y_{d1},\overline{Y}_{r1}) + (1-\alpha) * R_1], \tag{9}$$

subject to the constraint

$$\alpha * I(Y_{r1};\overline{Y}_{r1}|Y_{d1}) \leq (1\text{-}\alpha) * R_2 \tag{10}$$

The rates R_1 and R_2 must belong to the achievable rate region of a multiple-access channel. Hence, we have the following constraints on R_1 and R_2

$$\begin{aligned} R_1 &\leq I(X_{s2};Y_{d2}|X_{r2}), \\ R_2 &\leq I(X_{r2};Y_{d2}|X_{s2}), \\ R_1+R_2 &\leq I(X_{s2},X_{r2};Y_{d2}). \end{aligned} \tag{11}$$

2. Relay channel with BPSK input

The achievable rate in [9] has been derived assuming that the data transmitted over the channels have a Gaussian distribution. With this assumption, it is relatively easy to derive the achievable rate for a CF relay. However, in our work, we assume that the channel inputs are BPSK signals. We will now derive the achievable rate of the relay channel assuming that the channel inputs X_{s1}, X_{s2} and X_{r2} are BPSK signals [8].

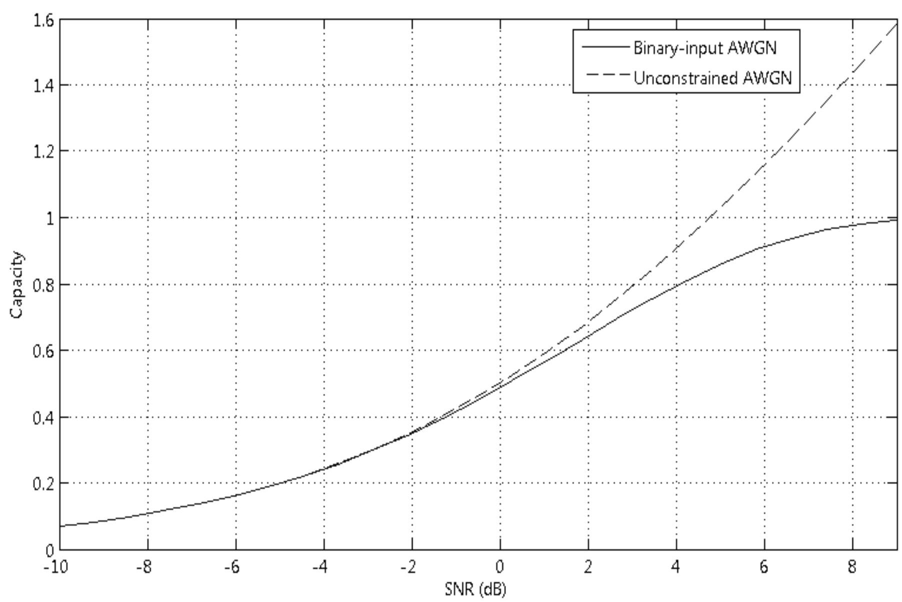

Fig. 6. Comparison of capacity curves for a binary AWGN and unconstrained AWGN channels.

The capacity of a binary-input AWGN (BIAWGN) channel [16] with a signal-to-noise ratio SNR is denoted by C(SNR) and defined as given in (12). Fig. 6 shows the comparison between the capacities of a binary-input AWGN channel and an unconstrained AWGN channel. We will use the capacity result in (12) for deriving the achievable rate of a binary-input relay channel.

$$C(SNR) = 1 - \int_{-\infty}^{\infty} \frac{e^{\frac{\tau^2}{2}}}{\sqrt{2\pi}} \log\left(1 + e^{-2\sqrt{SNR}\tau - 2SNR}\right) d\tau \quad (12)$$

Consider the set-up of a relay channel as shown in Fig. 3. Let the distortion in reconstructing Y_{r1} at the decoder side be $D_{CF}(R)$, with the quantization rate of compression being R. By considering the quantization noise to be Gaussian, the quantized signal can be written as $\bar{Y}_{r1} = Y_{r1} + w$, where w is a Gaussian noise with a variance $D_{CF}(R)$. We require the quantization rate to be less than or equal to the capacity of the relay-destination link i.e.

$$R \leq \frac{1-\alpha}{\alpha} * C\left(\frac{\gamma_{RD}^2 P_r}{1 + \gamma_{SD}^2 P_{s2}}\right), \quad (13)$$

where $P_{s1} = \alpha^* Ps/k$, $P_{s2} = (1-\alpha)^* Ps/(1-k)$ are the powers allocated to the source signal in the BC and MAC modes respectively with k being the varying factor and P_s being the total source power. P_r denotes the total power allocated to the relay transmitted signal in MAC mode.

The achievable rate of CF relay with BPSK inputs is given by

$$R_{CF} \leq \max_{0 \leq \alpha \leq 1} [\alpha C(\gamma_{SD}^2 P_{s1} + \frac{\gamma_{SR}^2 P_{s1}}{1 + D_{CF}(R)}) + (1-\alpha) C(\gamma_{SD}^2 P_{s2})] \quad (14)$$

The achievable rate in (14) depends on the Rate-Distortion function $D_{CF}(R)$ for the Gaussian mixture signal obtained at the relay during the BC mode. In practice, it is not straightforward to obtain $D_{CF}(R)$ directly. On the other hand, it is easier to find rate and distortion as a function of the quantization noise, σ^2_n. We can obtain the Rate-Distortion curve indirectly by obtaining the plots for both rate as well as distortion as functions of σ^2_n. Using this, we can find the achievable rate of a CF relay from (14). We now proceed with the procedure for finding $D_{CF}(R)$.

Since X_{s1} is a BPSK signal capable of taking one of the two values $+\sqrt{P_{s1}}$ or $-\sqrt{P_{s1}}$, the signal received at the Y_{r1} would be a Gaussian mixture signal. The distribution function of Y_{r1} would be as follows.

$$f(y_{r1})=\frac{1}{2}*\frac{1}{\sqrt{2\pi}}\exp\left(\frac{-(y_{r1}-\gamma_{SR}\sqrt{P_{s1}})^2}{2}\right)+\frac{1}{2}*\frac{1}{\sqrt{2\pi}}\exp\left(\frac{-(y_{r1}+\gamma_{SR}\sqrt{P_{s1}})^2}{2}\right)$$

Quantization distortion at a zero rate: With the quantization rate being 0, the best value of $\bar{Y}_{r1}$to choose would be the mean value of the Gaussian mixture signal which is 0. Hence, the zero-rate distortion with a mean-square error measure would be $D_{CF}(0) = E(y_r^2) = \int_{-\propto}^{\propto} y_r^2 * f(y_{r1})\, dy_r$

Rate computation: Let $\bar{Y}_{r1}=Y_{r1}+N$, where $N \sim \mathcal{N}(0,\sigma^2_n)$ is the quantization reconstruction. We have the quantization rate R given by,

$$R= I(Y_{r1};Z)= H(Y_{r1})- H(Y_{r1}|Z), \tag{15}$$

where $H(Y_{r1}) = \int -f(y_{r1})* \log\big(f(y_{r1})\big)\, dy_{r1}$. Similarly, $H(Y_{r1}|Z)$ is given as follows

$$H(Y_{r1}|Z)= -\int f(z) \int f(y_{r1}|z)\log(f(y_{r1}|z)dy_{r1}\, dz \tag{16}$$

We have $f(z)= \int f(y_{r1},z)dy_{r1}$ and $f(y_{r1}|z) = f(y_{r1},z)/f(z)$, where $f(y_{r1},z) = f(y_{r1})*f(z|y_{r1})$ is given as follows

$$f(y_{r1},z) = \left[\frac{1}{2}*\frac{1}{\sqrt{2\pi}}\exp\left(\frac{-(y_{r1}-\gamma_{SR}\sqrt{P_{s1}})^2}{2}\right)+\frac{1}{2}*\frac{1}{\sqrt{2\pi}}\exp\left(\frac{-(y_{r1}+\gamma_{SR}\sqrt{P_{s1}})^2}{2}\right)\right] * \left[\frac{1}{\sqrt{2\pi\sigma_n^2}}*\exp\left(\frac{-(z-y_{r1})^2}{2\sigma_n^2}\right)\right]$$

Using the above result and substituting (16) in (15), we can compute the quantization rate R in terms of the quantization noise variance σ_n^2.

Distortion computation: The reconstruction of Y_{r1} from $\overline{Y}_{r1}$, $\hat{Y}_{r1}$ is given by

$$\hat{Y}_{r1} = E[Y_{r1}|z] = \frac{\int y_{r1} f(y_{r1},z)\,dy_{r1}}{\int f(y_{r1},z)\,dy_{r1}} \tag{17}$$

Using (17), the mean-square distortion can be calculated as

$$D = E\left[\left(Y_{r1}-\hat{Y}_{r1}\right)^2\right] = \int f(z) \int \left(y_{r1}-\hat{Y}_{r1}\right)^2 * f(y_{r1}|z)\,dy_{r1}\,dz \tag{18}$$

Using (18), we can find the quantization distortion in terms of the quantization noise variance σ_n^2.

We can derive $D_{CF}(R)$ using the Rate- Distortion results from (15) and (18). This value is then substituted in the r.h.s of (14) and then optimized over α to obtain the achievable rate of a CF relay with BPSK inputs.

CHAPTER III

VIDEO TRANSMISSION OVER WIRELESS CHANNELS

In this chapter, we discuss the various issues involved with the transmission of video over wireless channels. The organization of this chapter is as follows. In Section A, we give a brief overview of current research and related work. MPEG-2 video coding principles are discussed in Section B. In Section C we discuss the various error detection and concealment techniques for recovering transmitted video from channel errors. Finally, in Section D we present some experimental results.

A. Current research and related work

Use of multiple antennas at the source node would provide spatial diversity necessary for video transmission. But this is impractical because of size and power limitations. User cooperation diversity [1] provides an exciting alternative to this. To the best of our knowledge, no work dealing directly with video transmission over compress forward relay networks has been done. We discuss some of the works which use multi-path transport and cooperation diversity as a means of transmitting multimedia data.

In [19], the incoming video bit-stream is divided into several sub-streams and transmitted over multiple paths (MPT) to exploit spatial diversity. Based on whether the sub-streams have equal or unequal importance, the source coding is classified as Multiple Description coding (MDC) or layered coding (LC) respectively. The paper discusses also briefly about the various techniques which could be used to achieve multi-path diversity. The means of using cooperation diversity for video transmission are discussed in [20]. A joint optimization of video coding, channel coding and cooperation is done. It tries to establish how layered cooperation, a combination of unequal error protection (layered coding) and cooperation can be used to obtain best results for video transmission. However RCPC codes are used for channel coding, which are not capacity approaching.

There is an increasing need to incorporate error control and concealment techniques for video communication because of the unreliable nature of wireless channels. Several techniques [12] have been proposed in this regard. In the Forward Error Concealment techniques, the source stream comes with an added redundancy to improve the error resilience. Other class of techniques would perform post-processing of the video at the decoder to exploit the spatial and temporal characteristics for Error Concealment. A Motion-Compensated Temporal Prediction has been widely used because of its simplicity and effectiveness. In this method, the intra-coded frame errors are recovered using spatial error concealment, while motion-compensated temporal prediction is used for the non-intra coded frames. However, several works dealing with error concealment [13] do not consider error detection in a detailed manner. In reality, Block error detection is a major challenge.

A transform domain detection algorithm has been proposed in [15]. While this method proves to be effective, it can wrongly indicate sharp edges within a frame as a block error thus, degrading its quality further. Another interesting method has been proposed by Bhattacharya *et al.* [14], whereby false detection due to sharp edges is avoided by an edge detection algorithm. Next we give an overview of the MPEG-2 video compression followed by error detection and concealment techniques for video communications.

B. MPEG-2 video coding

The MPEG-2 ISO standard [11] [21] was developed as an extension to MPEG-1 for the digital compression of audio and video signals. Apart from providing higher data rates compared to MPEG-1, MPEG-2 also has algorithms which enable efficient encoding of interlaced video. It is capable of coding Standard definition Television (SDTV) signals at bitrates from 3 to 15Mbits/s and High Definition Television (HDTV) signals at 15-30Mbits/s. It consists of a wide array of algorithmic tools to support diverse applications. Making the decoders support the entire standard is a complete waste of bandwidth. Hence, a number of subsets of the full standard called profiles, which define the set of algorithmic tools and levels, which set the constraints on the parameter values, have been proposed. In this section, we give a brief overview of the compression techniques, followed by the structures of encoder and decoder and the different profiles and levels defined in MPEG-2.

1. Compression techniques

A picture with its Red, Blue and Green (RGB) components can also be represented in terms of luminance (Y) and chrominance (UV) components. It turns out that YUV representation is more effective than the RGB representation, since under-sampling the chrominance components would not have any drastic effect on the picture quality. A digital sampling of 4:2:2 indicates that the horizontal component of chrominance has been sub-sampled by 2, while 4:2:0 indicates that both the horizontal and vertical chrominance components have been sub-sampled by a factor of 2. In the case of interlaced video, each picture frame consists of two interlaced fields, each containing odd and even numbered lines respectively, sampled at different instants of time separated by 20ms. On the other hand, all the lines of a non-interlaced video are sampled at once and hence called a progressively scanned video. An uncompressed 720x576 line video with 25 frames/s and 8 bits for each of the Y, U and V pixels with a digital sub-sampling requires a bit rate of 124Mbits/s. With MPEG-2 compression, the same can be compressed down to 3-15Mbits/s with a reasonable good quality. This would give a general idea of the amount of compression achievable with MPEG-2 without sacrificing much in the quality. The techniques employed by the MPEG-2 standard to get the required bit-rate reduction are discussed next.

Two principal types of redundancies are exploited for reducing the bit rate of the uncompressed video. An intra-frame Discrete Cosine Transform (DCT) [22] is used to exploit the spatial redundancy, while an inter-frame Motion Compensated prediction is used to exploit the temporal redundancy.

Intra-frame compression: DCT transforms a frame from a pixel domain to frequency domain. Each DCT coefficient based on its position indicates its contribution to the horizontal and vertical frequency. For most natural images the transformation would make most of the high frequency coefficients zero, thus packing the energy within a few low frequency coefficients. By quantizing and coding the DCT coefficients, a bit-rate reduction can be achieved. Quantization would reduce the number of possible values the coefficients can take, thus reducing the bit rate. Since human visual system is more sensitive to the low frequency components than the high frequency ones, it is a common practice to quantize the high frequency components more coarsely. This way the original pixel values may not be retained after Inverse DCT, thus making the process a lossy one. The DCT coefficients are scanned in a zigzag fashion. A variable length code is used to perform entropy coding on these coefficients. For this, we group runs of zero coefficients with a non-zero coefficient, with the final group replaced by an EOB marker. Each group is then replaced by a variable length codeword.

Inter-frame prediction: For most natural video sequences, it is common to have very less change between adjacent frames in a frame sequence. Hence, instead of intra-coding each frame just exploiting the spatial redundancy, we could also exploit the temporal redundancy between frames. Here, we try to predict a frame from a previously coded reference frame. One of the simplest forms of inter-frame prediction is to try and predict a block of pixels from the co-sited block in the previous frame. However, this would amount to a very poor prediction especially if there a large motion between two adjacent frames. A much better method would be the motion-compensated prediction where we do a block-matching, by finding the block in the reference frame which would best describe the current block and use it for prediction. The error measure for block-matching depends on the application, while mean-square-error is generally used. In the motion-compensated prediction, there is a need to encode the motion-vector information, thus causing a slight bit-rate overhead.

2. Encoder and decoder

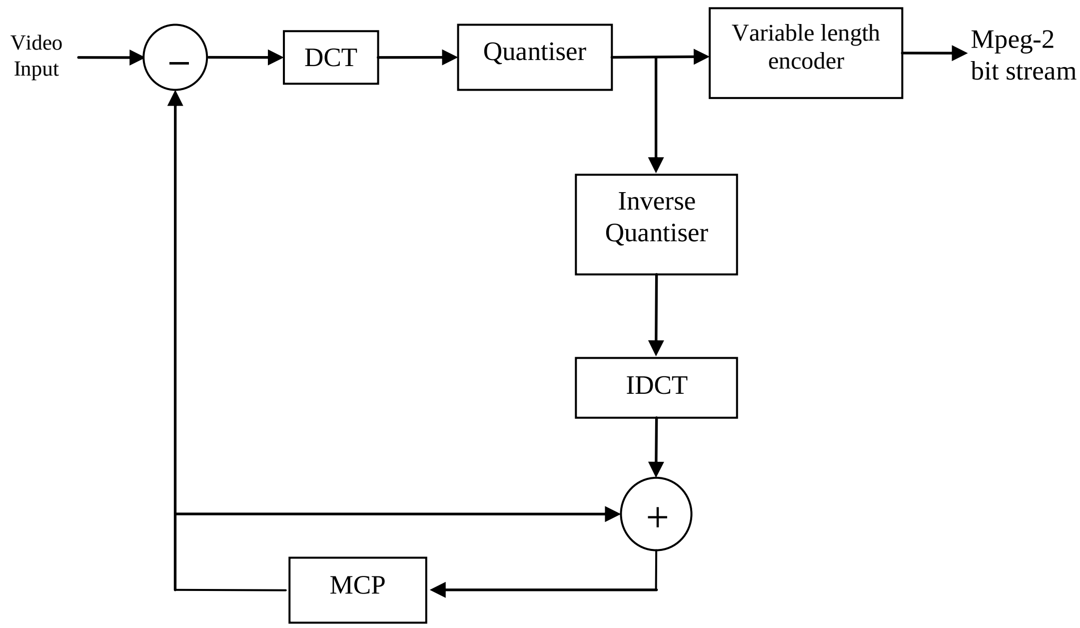

Fig. 7. An MPEG-2 encoder.

Fig. 7 shows the structure of an MPEG-2 encoder [21]. The combination of DCT and motion-compensation is used in an MPEG-2 encoding system. Prediction error is computed as the difference between the source frame and its motion-compensated prediction. After DCT transformation of the prediction error, it is quantized and encoded using a VLC coder. This along with the intra-frame information, motion vector information and other side information such as synchronization data from the MPEG-2 bit stream.

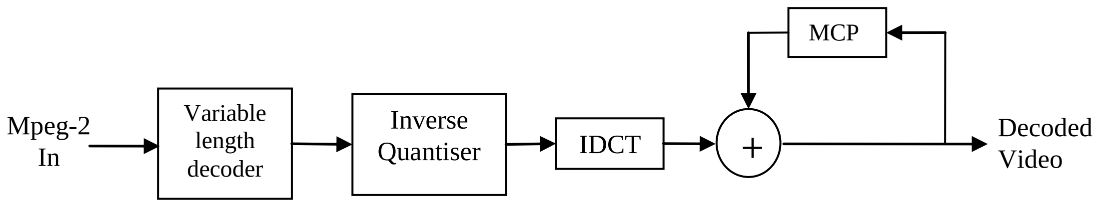

Fig. 8. An MPEG-2 decoder.

Fig. 8 shows the structure of an MPEG-2 decoder [21]. At the decoder side, the DCT coefficients of the prediction error are inverse transformed and then combined with the motion-predicted compensation to predict the frame from previously decoded frames. Several options are provided for predicting a block within a frame. It can be forward predicted, backward predicted or bi-directionally predicted. It can also be intra-coded in case the block-matching error is less than a predefined threshold. Based on the type of prediction modes used for blocks in a picture, they can be classified as I-type: all blocks are intra-coded, thus exploiting only spatial redundancy, P-type: blocks can be either forward-predicted or intra-predicted and B-type: blocks can be forward-predicted, backward-predicted, bi-directionally predicted or intra-predicted. Because of this, the size of the I-picture tends to be bigger than the P-picture which is in turn bigger than the B-picture. To facilitate backward prediction, the future reference frames need to be decoded first. Hence, the coding order needs to be different from the display order. Buffer control in an MPEG-2 system can be achieved by placing a fixed sized storage buffer between the coder and the channel. The optimal buffer size must be chosen based on the code rate of operation of the MPEG-2 system. A larger buffer size would not affect picture quality at the expense of delay while a smaller buffer size reduces the delay at the expense of picture quality.

3. Profiles and levels

The profiles supported by MPEG-2 [21] [11]are given in Tables 1.

Table 1

MPEG-2 profiles

Profile Name	Frames	Sampling	Layers	Features
Simple	I, P	4:2:0	1	No picture re-ordering, low-delay
Main	I, P & B	4:2:0	1	Picture re-ordering
4:2:2	I,P & B	4:2:2	1	Supports 4:2:2 sampling
SNR	I,P & B	4:2:0	1-2	SNR scalability
Spatial	I,P & B	4:2:0	1-3	Spatial (Resolution) scalability
High	I,P & B	4:2:2	1-3	Low, normal & high-quality decoding

For applications where delay constraints are tight, it is preferable not to use backward prediction. Hence, a simple profile has been included in the MPEG-2 system which does not support B-type pictures and hence has less delay. On the other hand, Main profile supports B-type pictures, thus improving picture quality at the expense of delay. It is one of the most commonly used profiles. An important feature of MPEG-2 is the scalable coding, where the total bit stream can be divided into a number of layers consisting of a base layer which can be decoded by it-self and several enhancement layers which either reduce quantization distortion (SNR Profile) or improve resolution (Spatial Profile).

Table 2

MPEG-2 levels

Level	Max. Picture size	Max. Frame rate (Hz)	Max. Bit rate (Mbps)
Low level	352x288	30	4
Main level	720x576	30	15
High 1440	1440x1152	30	60
High level	1920x1152	30	80

Table 2 gives the levels [11, 21] supported by the MPEG-2 standard. It supports four levels with each setting an upper bound on the picture size, frame rate and hence bit-rate of operation. Main profile at main level is the most commonly used setting.

C. Error detection and concealment

In this section, we give an overview and importance of error concealment, followed by error detection and concealment techniques.

1. Overview

There is a growing need for error control and concealment of video transmitted over wireless networks, which are highly unreliable, thus causing significant signal loss during transmission. This becomes even more significant when dealing with the transport of MPEG-2 data, which has a high degradation even with the loss of a single bit owing to the use of Variable length coding.

The error concealment techniques can be divided into three broad categories based on the roles played by the encoder and the decoder. In the Forward error concealment scheme, we would add redundancy to the source transmitted message, which would aid error recovery at the destination. Secondly, to avoid adding redundancy at the source, we could do post-processing at the decoder which attempts to recover lost information in the frames exploiting spatial information from neighboring macro-blocks or temporal information from previously decoded frames. In the third method, the encoder and decoder can interact with each other to accomplish error concealment. Several Error concealment techniques have been proposed in the literature falling into one of the above three methods. A survey of these techniques has been proposed by Wang and Zhu [12].

The first and the most important step towards error concealment in video is the detection of blocks in error [23]. If we can't detect the erred regions accurately, there is a possibility that the video quality gets degraded by error concealment. To give an illustration of the quality degradation caused by transmission errors, Fig. 9 shows the reconstruction of the I-frame of a Foreman sequence received over an AWGN channel with a bit-error rate (BER) of 10^{-5}. An error in a particular slice won't affect the decoding of any other slice. On the other hand, an error in the middle of a slice will make all the remaining blocks in the slice undecodable, as we can see from Fig. 9. Also, the damage of any block in an I-frame gets propagated to the subsequent P and B-frames which use it as a reference.

In this work, we only consider schemes that conceal errors by post-processing at the decoder. They do not require any change at the encoder side and do not add any redundancy. For the I-type frames a spatial error concealment scheme is used, while motion-compensated temporal error concealment is used for the P and B-type frames. The most obvious advantage of error concealment schemes is the improvement in video quality in terms of both visual as well as PSNR scales. One of the disadvantages of the error concealment schemes is that they tend to increase the delay, which may be critical in some multipoint applications. Also, there is an unavoidable processing complexity at the decoder side. In the forward error concealment schemes, there is an added redundancy thus increasing the bit rate of transmission.

Fig. 9. Foreman sequence at 910kbps received over an AWGN channel with BER=10^{-5}.

2. Error detection

Before any error-concealment techniques can be applied to a video corrupted by transmission errors, it is essential to find when and where a particular error has occurred. Error detection can be performed either at the transport coder/decoder. In this work, we discuss only the techniques which perform error detection at the video decoder.

A pixel domain error detection technique has been proposed by [24]. In this, the difference in the pixel values of the blocks in successive lines is used as a measure for error detection. If this value if greater than a pre-defined threshold, the block is declared to be in error. Lam and Reibman [15] developed a transform domain technique for error detection and concealment of images which performs much better than the pixel domain algorithm. It attempts to conceal the effect of errors from the received data which has been corrupted. The error detection technique employed in this paper can also be used for video applications. It does not require any changes to the encoder and does not change the bit-rate. Their error detection technique is described as follows. If t_i^{curr}(j), i=0, 1, …, 63 and t_i^{prev}(j), i=0, 1, …, 63 are the DCT coefficients of the j^{th} block in two consecutive lines, the weighted squared error is computed as follows

$$e(j) = \sum_{i=0}^{63} [w_i(|t_i^{curr}(j)| - |t_i^{prev}(j)|)] \quad (19)$$

The weights w_i's are chosen such that a higher weight age is given to the low frequency DCT coefficients than the higher frequency ones, consistent with the human visual system. The weighted squared error $e(j)$ is then used as a measure for finding the blocks which are in error.

An unsupervised error detector has been proposed by Chen [25], where the MPEG-2 decoder can detect the errors on itself. It proposes a number of techniques that can be used for error detection such as the deciphering of corrupted fixed and variable length code-words, the number of 8x8 DCT coefficients exceed 64, deciphering motion vector violations and looking for other semantic violations. A more interesting method for error detection has been proposed by Bhattacharyya *et al.* [14]. It would detect some of the synchronization errors which may not be detected by any of the previous methods, while being computationally efficient. It is based on the fact that average energy of a block does not change abruptly when compared to its neighboring blocks, except in the presence of a sharp edge. In case a strong edge is present, the edge direction is found and this would be used to find if the abrupt change in average energy is because of the presence of a genuine edge or due to the error. If F_{8x8} denotes the DCT coefficient matrix of the block for which we intend to find the edge direction, the average energy and the coarse edge direction are given by

$$E = \sum_{i=0}^{7}\sum_{j=0}^{7}|F_{i,j}| \qquad (20)$$

$$\theta = \tan^{-1}\frac{F[0][1]}{F[1][0]} \qquad (21)$$

To detect edge-activities, the average energy of a block given by (20) is compared with the eight blocks in its neighborhood. In case a sharp change in the average energy is observed, we look for any edges in the block and determine its orientation using (21). If the average energy along this edge direction θ is again greater than a pre-defined threshold, the block is declared to in error.

3. Error concealment

In this section, we will discuss some error concealment techniques under the assumption that the location of errors is already known. Also, we only consider the schemes which perform error concealment by post-processing at the decoder.

It is a known fact that the images of most natural scenes do not contain many dominant high-frequency components. This implies that the change in pixel values across a frame is very smooth. Also, the human visual system is more sensitive to a change in low frequency components than the high frequency ones. These facts can be used for error concealment at the decoder without adding any kind of redundancy at the encoder side. Since a synchronization codeword is inserted at the beginning of each scan row of macro blocks, an error in one of the rows does not affect any other scan rows. On the other hand, an error in the middle of a scan row will render the remaining macro blocks in that row undecodable. An intra-coded macro block can be concealed by replacing it with the average pixel values of its neighboring blocks which have been correctly received. This is known as spatial error concealment. Also, a damaged inter-macroblock in a frame can be replaced by the corresponding macro block in its reference frame based on the motion vector information. All the post-processing techniques use either spatially adjacent blocks or blocks in the previous frame for error concealment. Next, we will review some of the error concealment techniques we use in this work.

Motion-compensated temporal prediction: One of the simplest error concealment techniques is to replace a damaged macro block with the co-sited macro block in the previous frame. While this method involves less complexity, it performs very poorly if there a large motion between adjacent frames. A more efficient technique would be to replace a damaged macro block with its motion-compensated macro block (MC-MB) based on its motion vectors. A similar approach has been followed by [26], where a PSNR gain of 1dB is obtained even at high cell-loss rates. However, this approach can suffer major setbacks in case the coding mode and motion vectors get damaged. Damage to the coding mode of a macro block may lead to an intra-mode block compensated using temporal concealment producing catastrophic results. Loss in the motion vectors of a damaged macro block may also cause degradation in performance. Fig. 10 illustrates the phenomenon of Motion-Compensated Temporal Prediction.

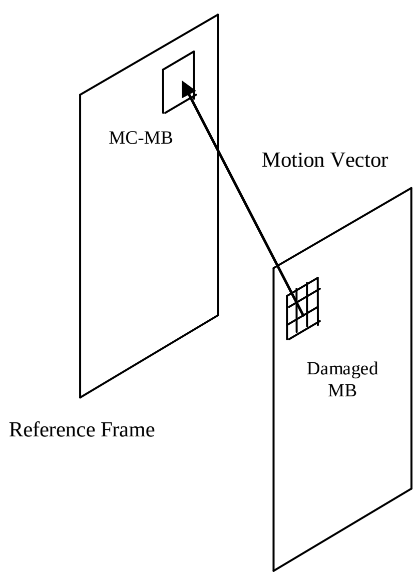

Fig. 10. Motion-compensated temporal prediction.

Pixel and frequency-domain interpolation: Aign and Fazel [13] proposed some techniques to implement spatial error concealment by interpolation technique. In the pixel domain technique, two types of interpolation methods can be used. In the first method, pixels of the damaged block are interpolated from the pixels in the two closest boundaries. In the second method, pixels of the damaged block are interpolated from the boundary pixels in all four directions. Fig. 11 shows how a damage pixel is interpolated from its four closest boundary pixels.

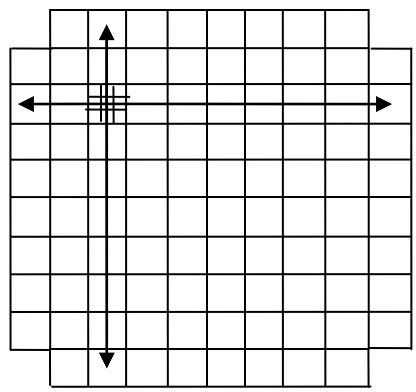

Fig. 11. Pixel-domain spatial interpolation.

Recovery of corrupted coding modes and motion vectors: As explained previously, damage to the coding modes and motion vectors can have a catastrophic effect on error concealment. Hence, there is a serious need to recover these using either spatial or temporal interpolation. Whenever damage to the coding mode of a block is detected, the block is simply set to intra-mode to avoid any drastic effects. Whenever damage to a motion vector of a block is detected, we can recover them using one of the following methods [27]:

- Using the same motion-vectors as that of the co-sited macro block in the previous frame.

- Using either the median or a weighted average of the motion vectors from spatially adjacent blocks.

D. Experimental results

Fig. 12 gives a comparison of the PSNR values for MPEG-2 streams encoded at rates 100kbps, 500kbps, 1Mbps and 3Mbps. As can be seen from the figure, there is an average PSNR difference of close to 10dB between the sequences encoded at 3Mbps and that at 100kbps. Fig. 13 shows a comparison of PSNR values of reconstructed Frame 0 for a Foreman sequence encoded at 100kbps and 300kbps.

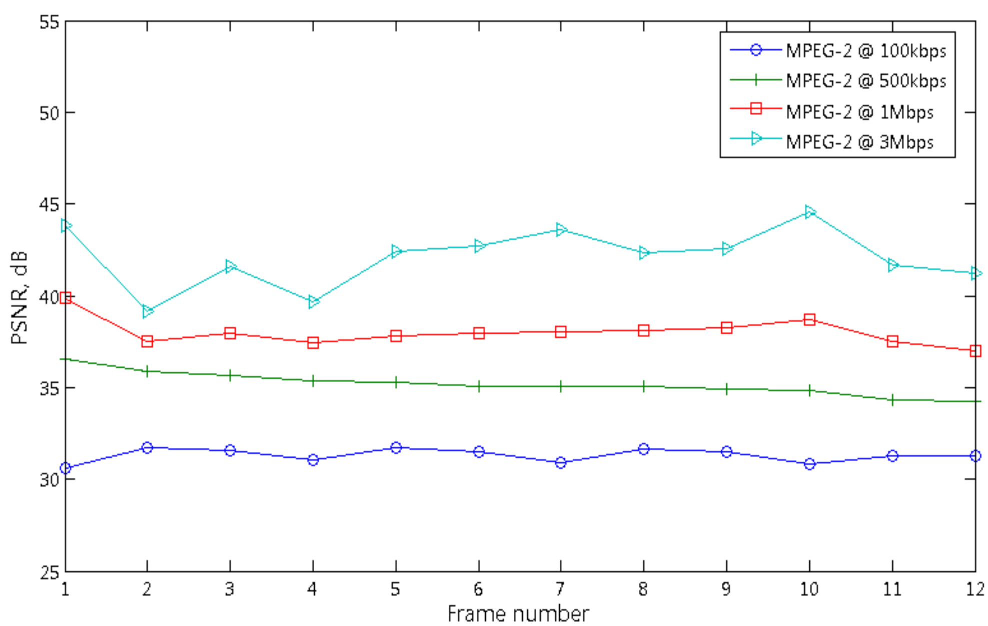

Fig. 12. PSNR comparison for different coding rates.

Bit-rate 100kbps: PSNR=30.6dB

Bit-rate 300kbps: PSNR=34.3dB

Fig. 13. PSNR comparison between reconstructed foreman sequences encoded at different rates.

CHAPTER IV

IMPLEMENTATION

In this chapter, we present a practical scheme for video transmission over relay channels with a CF code design followed by some simulation results. In Section A, we present the block diagram of our implementation and its operation. We present the simulation results in Section B.

A. Block diagram and operation

The block diagram of our proposed Compress-Forward scheme for Video Transmission is shown in Fig. 14. The various steps involved in our implementation are as follows:

- **Partitioning of MPEG-2 data**: In an MPEG-2 bit stream, I-frame data has to be given more importance than the P and B-frame data in terms of error protection. This is because an error in an I-frame will propagate to all the other frames in the GOP. This unequal error protection would help us in using the channel resources in the best possible manner. Unequal error protection could be easily achieved by the transmission over a relay channel. For this, we need to partitioning the MPEG-2 bit stream into two separate data sets one consisting only the I-frame data and the other consisting of the P and B-frame data. By ensuring that the spatial diversity of the relay channel is provided only for the I-frame data, we can ensure the required unequal error protection is achieved for the MPEG-2 bit stream.
- **Broadcast mode operation**: During the broadcast mode, we transmit I-frame data over the relay channel. The inter-channel noise is assumed to be a zero-mean unit-variance Gaussian signal. For error protection, we encode the I-frame

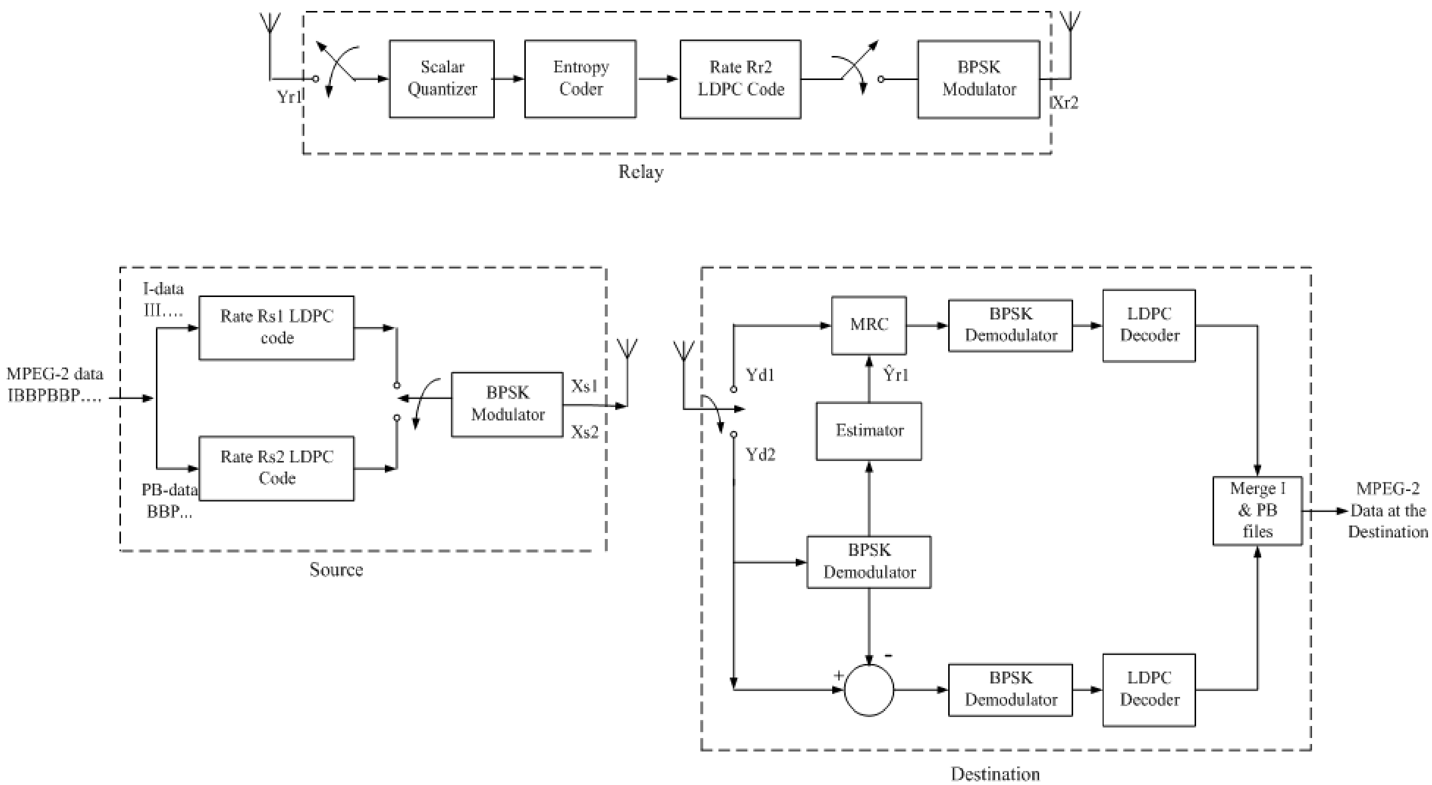

Fig. 14. Block diagram of our proposed CF relaying scheme for video transmission.

data with a rate R_{s1} LDPC code. It is a great advantage that there exist capacity approaching binary LDPC codes which have been optimized for the binary AWGN channel [28]. The obtained codeword is then BPSK modulated to obtain X_{s1}. X_{s1} is then transmitted to both the relay and the destination nodes. This is received as Y_{r1} by the relay and as Y_{d1} by the destination as in (1) and (2) respectively.

- **Relay processing**: The relay received signal Y_{r1}, which is a Gaussian mixture signal needs to be compressed before it is forwarded to the destination. For this we employ an entropy constrained scalar quantization. By this, we ensure that the mean square distortion is minimized by optimizing over the quantization thresholds, while constraining the entropy. Entropy coding is performed on the quantization indices followed by channel coding at a rate R_{r2} and BPSK modulation to obtain X_{r2}.
- **MAC mode operation**: During the MAC mode, the source transmits PB-frame data over the relay channel. We encode the PB-frame data with a rate Rs2 LDPC code and then perform BPSK modulation to obtain X_{s2}. The relay transmits X_{r2}, while the source transmits X_{s2} to the destination node. This is received as Y_{d2} by the destination as in (4).
- **Destination node processing**
  - Successive interference cancellation: For decoding the signals X_{s2} and X_{r2} from X_{d2} at the destination node, we need to perform joint decoding. However, this is a highly complex procedure. On the other hand, by choosing the rates R_{s2} and R_{r2} as the boundary points on the capacity region of a multiple-access channel shown in Fig. 6, we can decode them by a simple successive interference cancellation. After getting an estimate of X_{r2}, by quantization reconstruction, we obtain $\bar{Y}_{r1}$.
  - Maximum ratio combining: It should be noted that both Y_{d1} and $\bar{Y}_{r1}$are noisy version of X_{s1}. By performing maximum-ratio combining of Y_{d1} and $\hat{Y}_{r1}$, we can get an optimal estimate of X_{s1}.

- Error detection and concealment: I and PB frame data can be recovered after LDPC decoding. From the above procedure it is clear that spatial diversity is exploited only for the I-frame data. In other words, the more important I-frame data has higher protection than the PB-frame data (unequal error protection). After receiving I-frame and PB-frame data, they are merged at the destination to recover the original MPEG-2 bit-stream. The MPEG-2 bit stream is likely to have a very low average PSNR values because of being corrupted by the channel noise. We need to conceal these errors by post-processing at the decoder. For error detection, we use a combination of the methods proposed by Bhattacharyya *et al* [14], Lam *et al* along with the information obtained from the MPEG-2 decoder. We use a motion compensated temporal prediction for error concealment.

B. Simulation results

1. Simulation set-up

a. Direct channel set-up

Fig. 15 shows the simulation set-up of our BIAWGN channel for video transmission. Capacity-approaching binary LDPC codes are used for channel coding the MPEG-2 data after which it is BPSK modulated. We assume the inter-user noise is $N(0, 1)$.

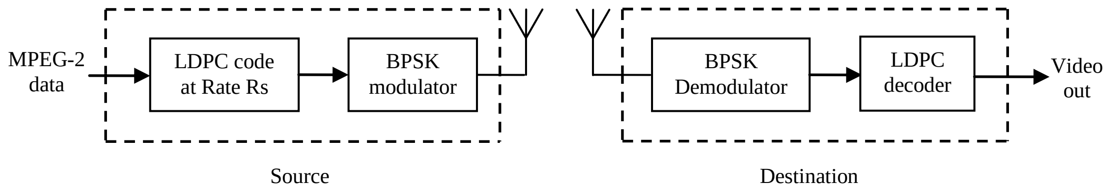

Fig. 15. Video transmission over a BIAWGN channel.

b. Two-hop channel set-up

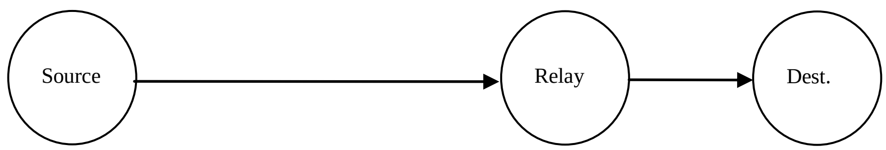

Fig. 16. Two-hop channel set-up.

Fig. 16 shows the simulation set-up of the two-hop channel. It is assumed that the source, relay and destination nodes are collinear. The source-relay, relay-destination and source-destination distances are assumed to be 0.95, 0.05 and 1.0 respectively. It should be noted that if the capacity of the source-relay link is C_1 and that of the relay-destination link is C_2, the capacity of the two-hop channel is given by $C_{two\text{-}hop}=\min(C_1,C_2)$.

c. Relay channel set-up

Fig. 14 shows the simulation set-up of the relay channel. It is assumed that the source, relay and destination nodes are collinear. The source-relay, relay-destination and source-destination distances are assumed to be 0.95, 0.05 and 1.0 respectively. The channel attenuation exponent is taken as 3, thus making the channel gains γ_{SD} = 1.0, γ_{SR}

$= 1/(1-0.95)^3 = 1.166$ and $\gamma_{RD} = 1/(1-0.05)^3 = 8000$. The inter-user channels are assumed to be $N(0, 1)$.

2. Results

a. CF relay with Gaussian inputs

Fig. 17 gives the comparison of the achievable rates of a Gaussian CF Relay with the source silent in MAC mode as proposed in [6] and our implementation with the source active in MAC mode. From Fig. 17, we can observe that the achievable rate reduces slightly with an active source in MAC at the expense of more data sent over the relay channel compared to the source silent case.

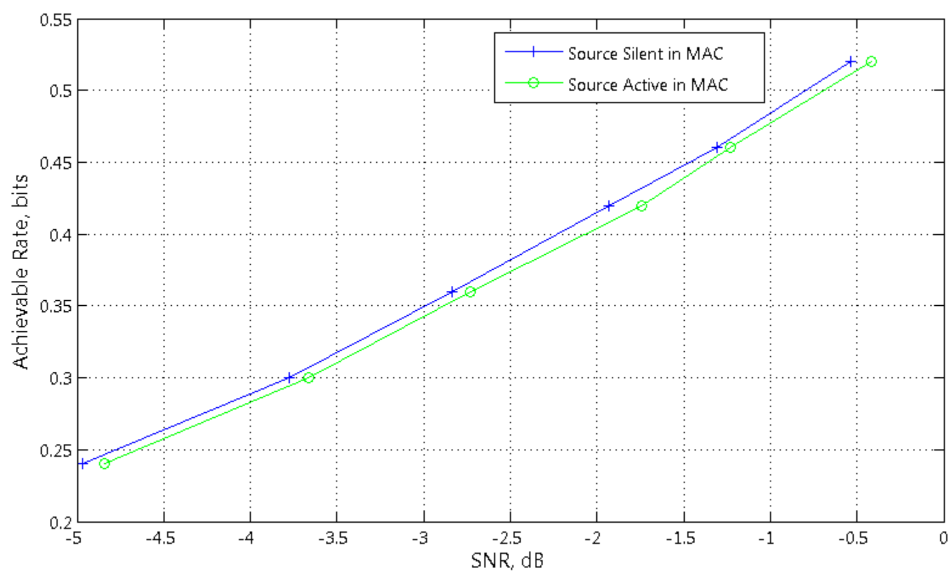

Fig. 17. Achievable rates of a Gaussian CF relay with source silent and active in MAC.

b. CF relay with BPSK inputs

The achievable rate for CF relay with BPSK inputs is given by (14). For computing the $D_{CF}(R)$ term in (14), we will require the Rate-Distortion curve. For this, we obtain the plots for R vs. σ_n^2 using (15) and D vs. σ_n^2 using (18), which would in turn give us the R-D curve. The variation of Quantization Rate and mean-square Distortion with the Quantization Noise Variance for a Gaussian mixture signal with the relay node at a position d=0.95 is given by Figs. 18 and 19 respectively.

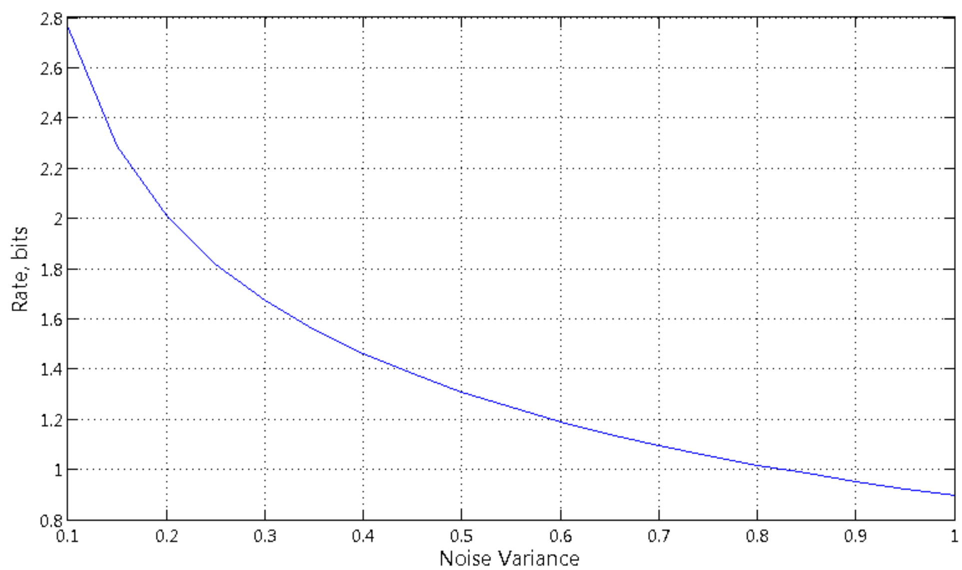

Fig. 18. Rate vs. noise variance for a Gaussian mixture signal with the relay at d=0.95.

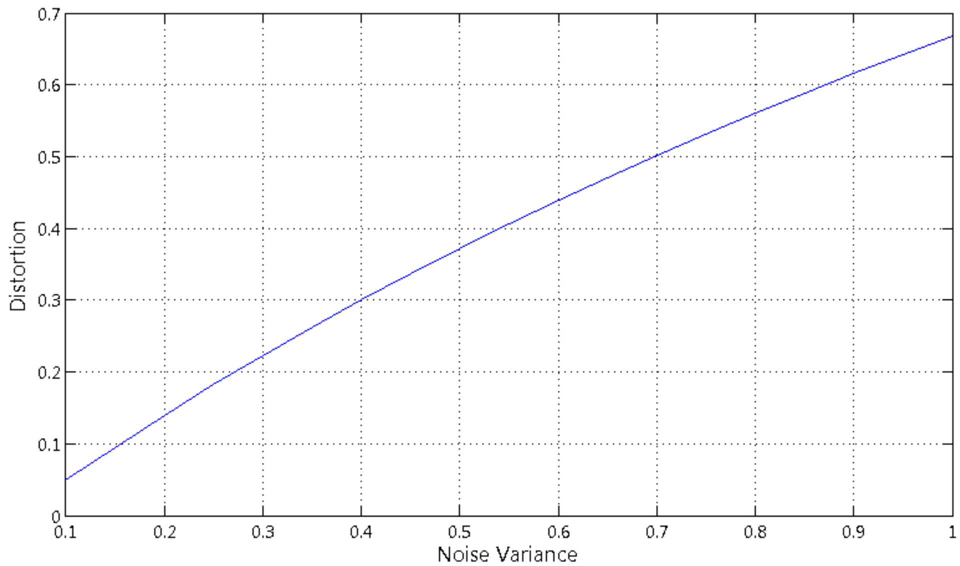

Fig. 19. Distortion vs. noise variance for a Gaussian mixture signal with the relay at d=0.95.

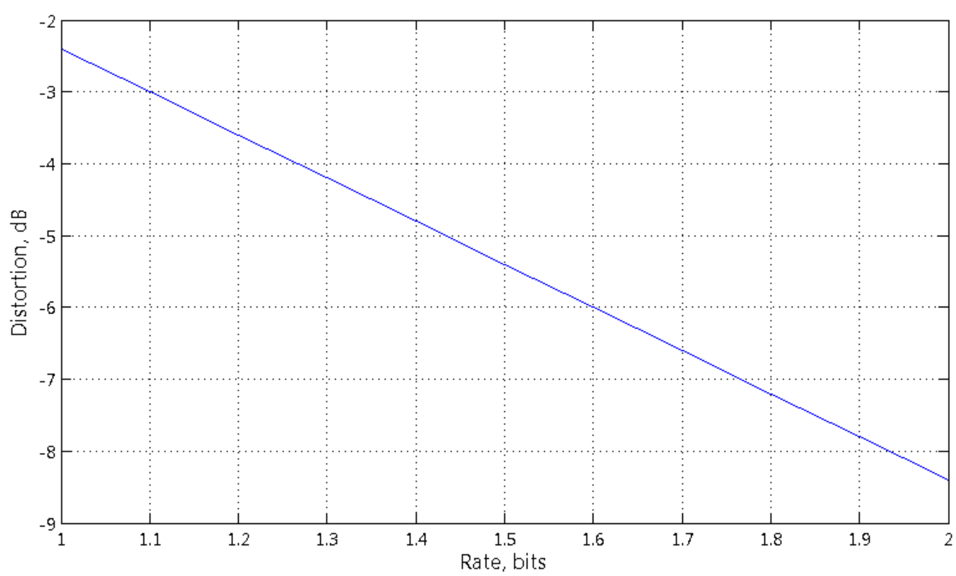

Fig. 20. Distortion vs. rate for a Gaussian mixture signal with the relay at d=0.95.

The R vs. σ_n^2 curves in Fig. 18 and D vs. σ_n^2 curve in Fig. 19 can be used to find the Rate vs. Distortion curve as shown in Fig. 20. Using this plot, we can find the achievable rate of CF relaying with BPSK inputs. We compare this with the achievable rate for CF relaying with Gaussian inputs in Fig. 21.

c. Comparison of direct, two-hop and relay

Our main aim is to transmit an MPEG-2 video stream over a BIAWGN channel (direct transmission), two-hop channel and a relay channel (relay transmission) and compare the PSNR's of the decoded video in these cases. For a fair comparison, we assume that the power per bit is fixed for all the three scenarios.

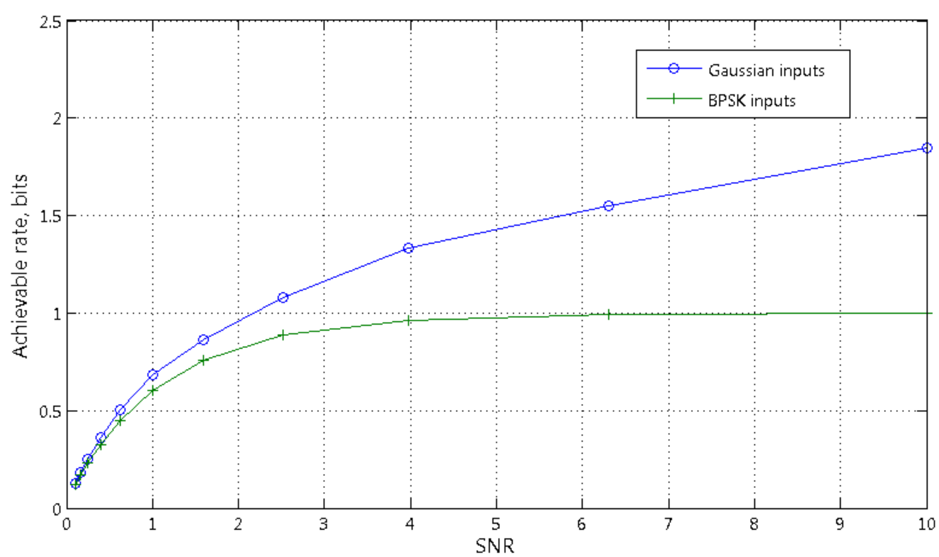

Fig. 21. Achievable rate of CF relaying with BPSK and Gaussian inputs.

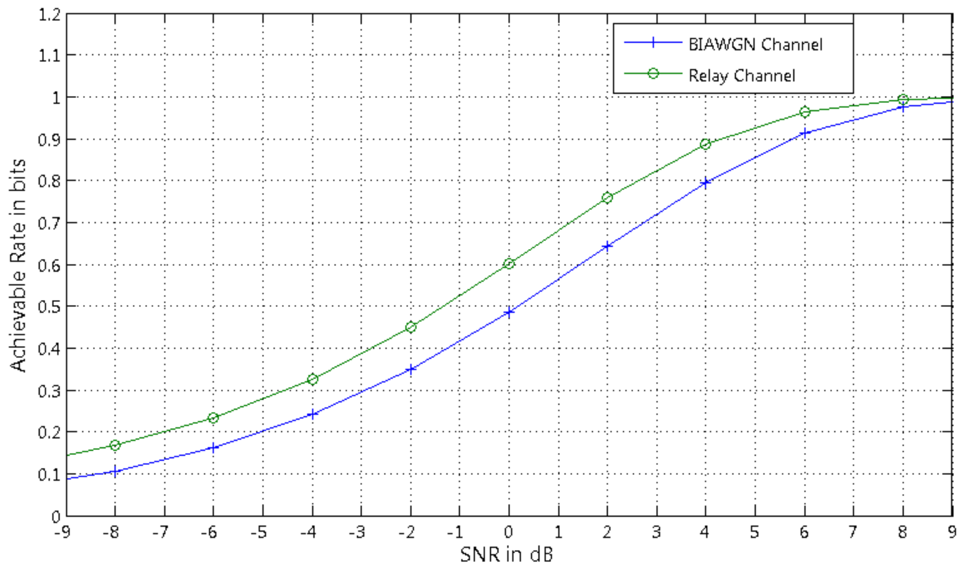

Fig. 22. Achievable rates of BIAWGN channel and relay channel with BPSK inputs.

Fig. 22 shows the theoretical achievable rates of a BIAWGN channel and a CF relay channel with BPSK input obtained using the results from section II-B.

- **Simulation parameters:** We fix the SNR per bit to be -1dB. As shown in Fig. 22, the theoretical achievable rates for an SNR of -1dB would be 0.41 for a BIAWGN channel and 0.525 for a CF relaying. In other words, we expect the achievable channel coding rate of the relay channel C_2 to be nearly 1.28 times that of the BIAWGN channel C_1. We will use this as a basis for comparing video transmission over the two channels. Next, we give the practical rates which have been achieved using a BIAWGN channel, two-hop channel and CF relaying. We aim to achieve a BER of 10^{-5} for the direct, two-hop and relay transmission schemes. The video coding rate for the direct transmission is fixed at 910kbps, with an SNR = -1dB and the achievable video coding rate for the relay transmission is found, while keeping the power constant.
- **Theoretical results:** Let S_1, C_1 and S_2, C_2 be the source coding and channel coding rates for two different transmission schemes respectively. Let k_1, n_1 and

k_2, n_2 denote the corresponding message and codeword lengths. For a fair comparison of SNR between the two schemes, the following relations in (22) and (23) must be true.

$$\frac{k_1}{n_1} = \frac{k_2}{n_2} \tag{22}$$

$$n_2 = n_1 * \frac{C_2}{C_1} \tag{23}$$

- **Practical Results:** We now discuss the practical results obtained by comparing direct, two-hop and relay transmission schemes.
  - ➢ BIAWGN channel: Fig. 15 shows the procedure for transmitting an MPEG-2 sequence over a BIAWGN channel. We consider a 12-frame Foreman sequence encoded at a rate S_1 = 910 kbps for direct transmission. A channel coding rate of C_1 = 0.36 was achieved with an SNR = -1dB and for a BER = 10^{-5}.

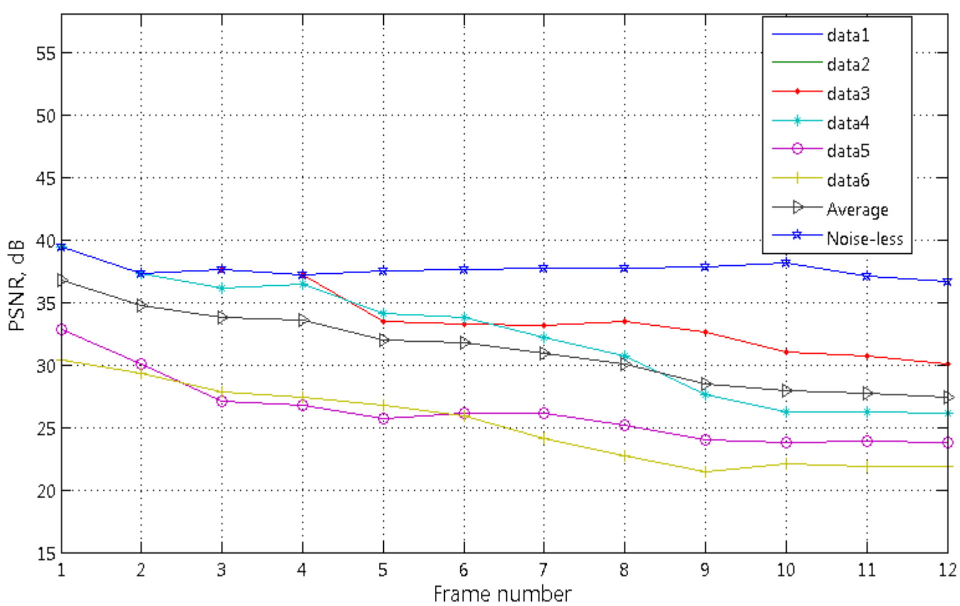

Fig. 23. PSNR results for video transmission over a BIAWGN channel.

Fig. 23 shows the PSNR results of the decoded video for six data sets after its transmission over a BIAWGN channel. It also shows the corresponding average PSNR.

- Two-hop channel: The practical achievable rate for the two-hop channel with an SNR = -1dB for a BER = 10^{-5} has been observed to be C_2 = 0.40. From (22) and (23), for a fair comparison with direct transmission, we can use a source coding rate up to (C_2/C_1)*910kbps ~ 1000kbps for the two-hop channel. Fig. 24 shows the PSNR results of the decoded 12-frame Foreman sequence at 1000kbps for five data sets, after their transmission over the two-hop channel. It also shows their average PSNR.

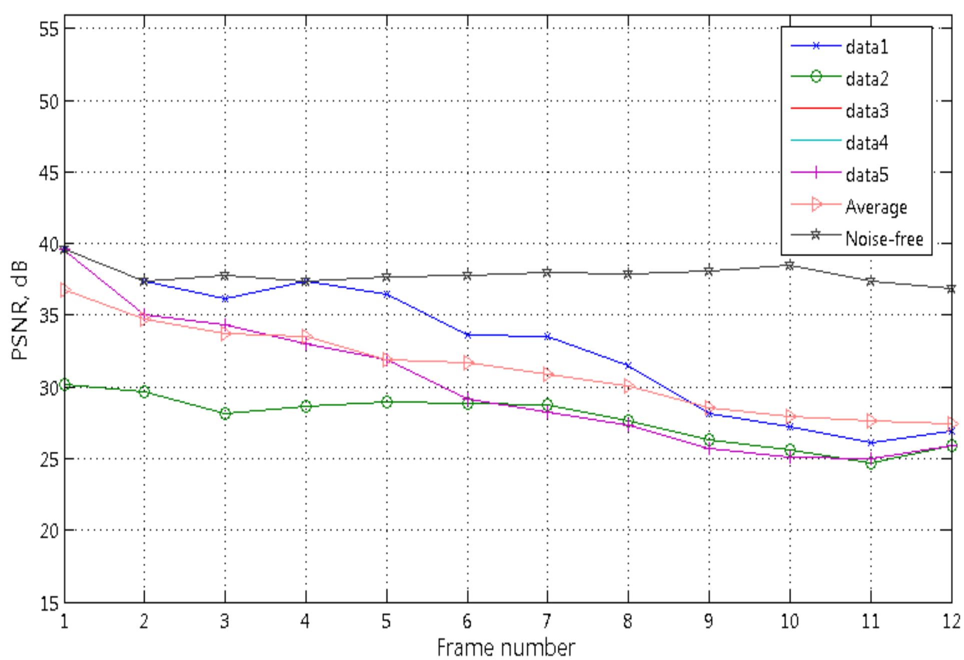

Fig. 24. PSNR results for video transmission over a two-hop channel.

- CF relaying: Since the relay-destination link is very strong, for simulation purposes we assume that the link is perfect. That is, we do not explicitly perform entropy coding and channel coding of the quantization index. Instead, it is assumed to be known at the destination node. The practical achievable rate for CF relaying with an SNR = -1dB with a BER = 10^{-5} is obtained as C_3 = 0.45. The relay channel parameters for achieving this optimal rate are given by Table 3. From (22) and (23), for a fair comparison with direct transmission, we can use a source coding rate up to (C_3/C_1)*910kbps ~ 1160kbps for the relay channel. Fig. 25 shows the PSNR results of the decoded 12-frame Foreman sequence at 1160kbps for five data sets, after its transmission over the relay channel. It also gives the corresponding average PSNR.

Table 3

Optimal relay channel parameters for an SNR = -1dB

Parameter	Value
A	0.38
K	0.36
R_{s1}	0.56
R_{s2}	0.38
R_{CF}	0.45

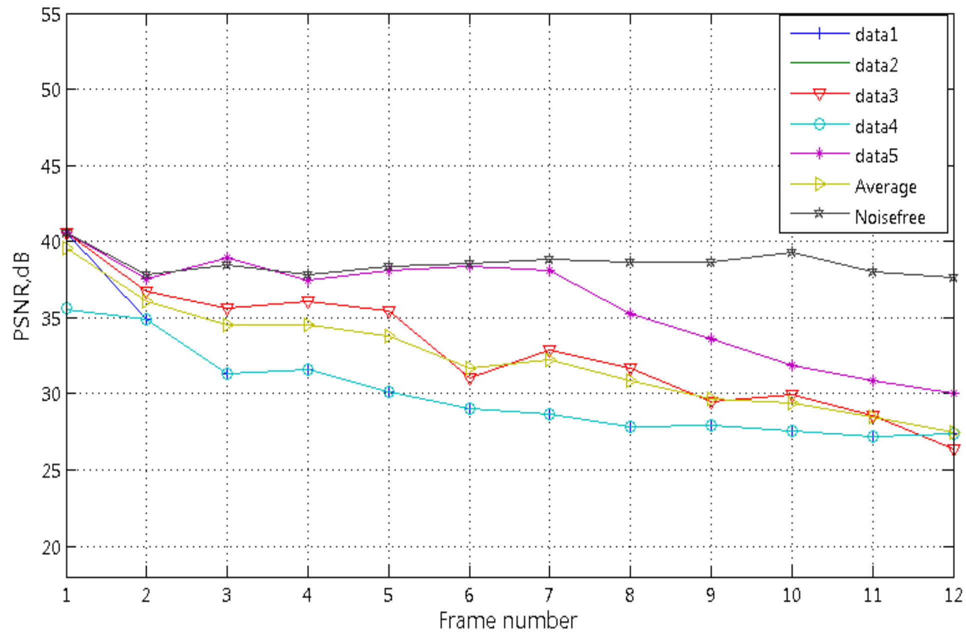

Fig. 25. PSNR results for video transmission over the relay channel.

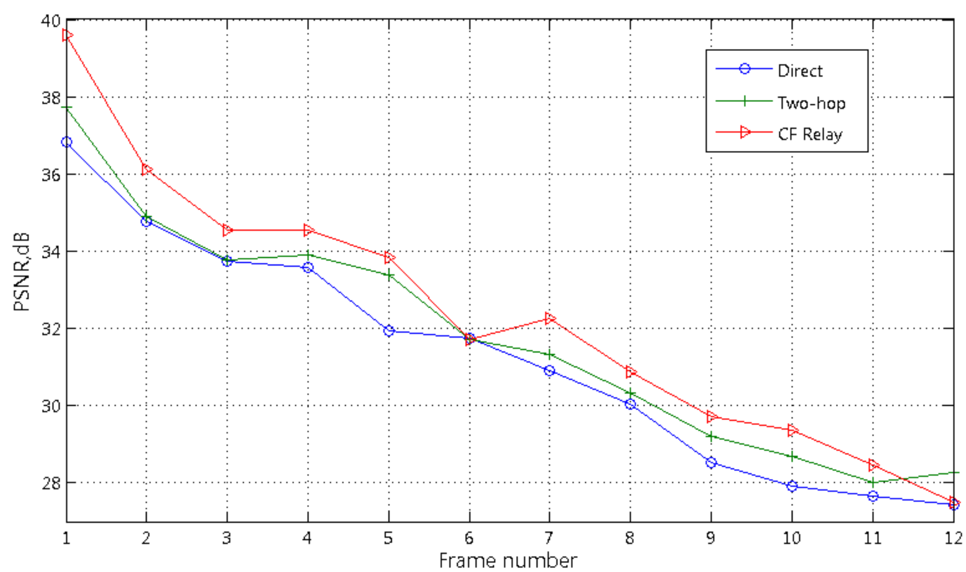

Fig. 26. Comparison of PSNR results for direct, two-hop and relay transmission schemes.

- Comparison between direct, two-hop and relay transmission schemes: Fig. 26 shows the comparison between the PSNR's of the decoded video obtained by transmission over a direct and a relay channel. It can be observed that relay transmission has a significant gain in PSNR values compared to direct transmission.

Table 4

Average PSNR values for direct, two-hop and relay transmissions

Frame Number	Direct (dB)	Two-Hop (dB)	Relay (dB)
1	36.82	37.73	39.58
2	34.77	34.92	36.13
3	33.75	33.77	34.54
4	31.95	33.89	34.55
5	31.74	33.39	33.83
6	30.92	31.72	31.72
7	30.04	31.32	32.26
8	28.52	30.34	30.88
9	27.91	29.20	29.71
10	27.68	28.69	29.37
11	27.68	28.02	28.47
12	27.42	28.27	27.48
Average PSNR	**31.26**	**31.77**	**32.38**

Table 4 gives the numerical values of the average PSNR's of the decoded video received over the direct, two-hop and relay channels respectively. It can be observed that relay transmission achieves an average PSNR gain of 1.12dB over direct transmission and 0.61dB over two-hop transmission. These values come close to the expected gains in theory.

CHAPTER V

CONCLUSIONS AND FUTURE WORK

This research work proposes a practical scheme for transmitting MPEG-2 video stream over a relay channel with a CF coding scheme. It also uses spatial diversity as a means to provide more protection to the intra-frame information in the video stream. This unequal error protection utilizes the spatial diversity provided by the relay channel in an optimal manner.

We presented some of the error detection and concealment techniques for recovering transmission errors that occur during video transmission. This is highly important for MPEG-2 data, where even a single bit error might have a drastic affect on the video quality. We also presented a rate-distortion approach for finding the achievable rate of a compress-forward relay scheme with BPSK inputs. Finally, we made a fair comparison between MPEG-2 transmission over a simple BIAWGN channel and a two-hop channel with our proposed compress-forward relaying. Practical gains in the PSNR values have been observed for the compress-forward relaying scheme compared to the binary AWGN channel and two-hop channel, thus proving the effectiveness of our proposed implementation.

There is a wide scope for future improvement of our work. The performance of our implementation can be further improved by using a much better partitioning scheme for the MPEG-2 data. By providing spatial diversity for the header and motion vector information instead of intra-frame information, received video quality can be further enhanced. Further, considering the inter-user channels to be Rayleigh fading instead of Gaussian would exploit the spatial diversity gains in a much better fashion.

Significant improvements can be made to the processing at the relay node. This includes using Wyner-ziv coding and much better channel coding schemes. There is also a need to employ more recent and much better video coding schemes such as H.264

instead of MPEG-2 video coding. Finally, the relaying scheme can be extended to the more recent development of cooperative communication [1, 10].

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