Reliable Transmission of Video over Ad-hoc Networks Using Automatic Repeat Request and Multi-path Transport

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Abstract— **The increase in the bandwidth of the wireless channels and the computing power of the mobile devices makes it possible to offer video service for wireless networks in the near future. In an ad-hoc network, strong error protection is required because of the lack of fixed infrastructure. On the other hand, the mesh structure of an ad-hoc network implies that there may be multiple paths existing between a source and destination, which can be used to enhance video transmissions. In this paper we propose a simple but robust scheme for reliable transmission of video in bandwidth limited ad-hoc networks.**

In our scheme, a video stream is layer coded. The *Base Layer* (BL) packets and the *Enhancement Layer* (EL) packets are transmitted separately on two disjoint paths using *Multi-path Transport* (MPT). BL packets are protected by *Automatic Repeat Request* (ARQ), and a lost BL packet is retransmitted through the path where EL packets are transmitted. An EL packet has lower priority than a retransmitted BL packet and may be dropped at the sender when congestion occurs. Simulation results show that this scheme can guarantee a graceful video quality in adverse channel conditions. It is effective for video transmission over the high loss environment found in ad-hoc networks.

Keywords— **Layered Video Coding, Multi-path Transport, Automatic Repeat Request, Ad-hoc Network.**

I. Introduction

With the increase in the bandwidth of wireless channels and the computing power of the mobile devices, it is expected that video service will be offered for wireless networks in the near future. An ad-hoc network, which is a dynamically re-configurable wireless network with no fixed infrastructure, poses a great challenge to video transmissions [1]. In such networks, links go down frequently and packet loss on a multi-hop wireless path is more frequent than wired networks and cellular wireless networks. There needs to be strong error protection for video traffic in order to achieve an acceptable quality.

Forward Error Correction (FEC) and *Automatic Repeat Request* (ARQ) are two basic error control techniques widely used in various settings to combat channel errors [2][3][4][5][6]. FEC is efficient for random bit errors or burst errors of limited length. It is traditionally used for real-time multimedia traffic because it requires no feedback. ARQ, on the other hand, requires lower overhead than FEC since retransmission only happens when needed. But in some cases the propagation and other delays are so big that retransmissions may be unacceptable.

It is well-known that a wireless channel has burst errors, which generally cause either the loss of a whole packet or a large portion of the packet is corrupted. It is very difficult to use FEC only to recover such damaged packets with reasonable overhead and delay. Furthermore, most video coding standards use motion prediction and compensation to remove the temporary redundancy in the frames and achieve a high compression efficiency [7]. The loss of a packet not only affects the quality of the current picture, but also causes error propagation through the following video frames. In [2], different retransmission schemes for error control in multicast protocols geared toward real-time multimedia application are analyzed. The authors claim that retransmission schemes are not only appropriate for such applications, but in fact can be quite effective. Hence it is necessary and applicable to include ARQ in the transmission scheme to combat burst errors.

Layered video coding is very useful in coping with the heterogeneity of user access rates and network link capacities in video multicasting [8]. A scalable coder codes video into several layers. The base layer guarantees a basic display quality, and each enhancement layer correctly received by the receiver results in better video quality. Users can subscribe up to a certain layer according to their capacities. Enhancement layer packets are dropped at the congested node to let the base layer packets get through. Khansari, *et al*, proposed an architecture for low bit-rate layered video transmission over fading wireless channels using ARQ, in which the enhancement layer packets are dropped at the source node when its ARQ buffer occupancy exceeds a predefined threshold [5]. Saparilla and Ross investigated the two-layer coded video streaming over the Internet [9]. The dynamic allocations of the available bandwidth to the two layers are studied in order to minimize the impact of client starvation.

Ad-hoc networks are different from wired networks and cellular networks because of the lack of a fixed infrastructure and high user mobility. On the one hand, lack of infrastructure and user mobility make link conditions worse; on the other hand, the meshed structure and user mobility provide multi-user diversity and increase the capacity of ad hoc networks [10]. Compared with wired and cellular wireless networks, ad-hoc networks have several features that are not fully exploited yet. These features are:

- An ad-hoc network usually has smaller size, both in the number of the nodes and the geographical coverage. So the propagation delay, which is a major element of the end-to-end delay for the Internet, could be negligible for such ad-hoc networks, which results in a lower bandwidth-delay product. The smaller *Round Trip Time* (RTT) opens up the possibilities of a larger window of time for retransmissions.
- An ad-hoc network is still bandwidth limited, even as higher link bandwidths have been proposed. However, since more

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computing power is being integrated into mobile devices, it is possible to implement more complex control schemes in the nodes to better utilize the limited bandwidth.

- Most ad-hoc routing protocols (e.g., ZRP, DSR, AODV, TORA) essentially provide multiple paths between the source node and the destination node [1][11], which makes it possible to use *Multiple-path Transport* (MPT) to achieve better protection of the video traffic [12][13][14].

It is desirable to take advantage of these features to improve the quality of the video transmitted. In this paper, we propose reliable transmission of layered video over ad-hoc networks using Selective Retransmission (SR) and NACK only ARQ scheme and MPT. The objective is to achieve an acceptable quality of the video under a highly error prone ad-hoc network environment. Although the scheme proposed in this paper is for uni-cast video transmission, it can be easily extended to video multicasting.

The remainder of the paper is organized as follows. Our scheme is discussed in detail in Section II. Section III presents the ad-hoc network model we used for the simulation. Our simulation results are reported in Section IV. Section V concludes our paper.

II. Video Transmission with ARQ and MPT

In this section, we discuss in detail the scheme we propose for video transmission over ad-hoc networks. The block diagram of Fig.1 shows the components of the architecture.

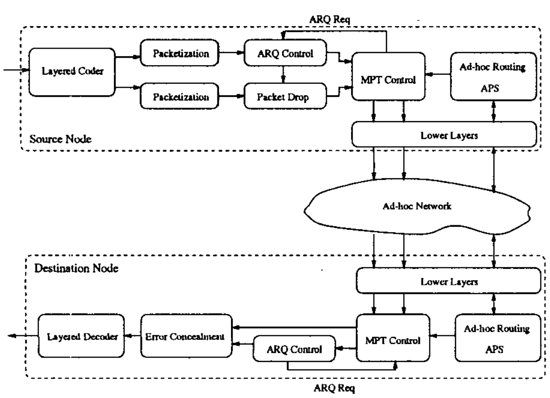

Fig. 1. An architecture for layered video transmission over ad-hoc networks using ARQ and MPT

In our scheme, a raw video stream is coded into two layers, i.e., the base layer (BL) and the enhancement layer (EL). Fig.2 illustrates the coding scheme. A BL frame is encoded using the standard predictive video coding technique, and then the prediction error between the original frame and the reconstructed BL picture is encoded into one or several enhancement layers [7]. Then BL frames and EL frames are packetized for transmission.

The receiver sends negative acknowledgments (ARQ requests) back to the sender to report packet losses. The size of such feedback is small enough so that the transmission time for these packets is negligible. Furthermore, the node works much faster than the channel and a packet received is quickly forwarded to the next hop, which results in a minimal delay within the node. However there will be some delay due to congestion build up on the wireless link. Another issue associated with MPT is the delay due to resequencing of packets that has to be performed at the destination. Gogate and Panwar [15] suggest that resequencing delay is moderate if used in conjunction with enhanced versions of current transport and network layer protocols. Thus the RTT mainly consists of the transmission delay of a video packet.

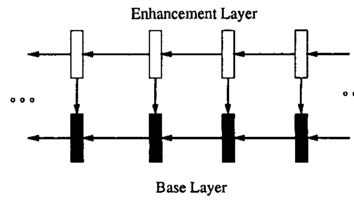

Fig. 2. The layered coding scheme

BL packets, which are more important, are protected with ARQ. The receiver sends ARQ requests to the sender if a BL packet is lost. These requests are sent to both paths, and are assumed to be error free as in [2]. Since the retransmission will introduce delay to the following packets for a bandwidth limited channel, we retransmit the BL packet on the EL path and discard the EL packet being transmitted at that time instance[1], as shown in Fig.3. Received packets are stored in a buffer waiting for the retransmitted BL packets to arrive. There is a deadline d for the packets. So the maximun number of possible transmission of a packet is less than or equals to $\lfloor d/RTT \rfloor$. Overdue packets are regarded as lost.

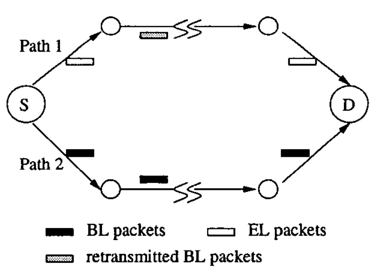

Fig. 3. The two-path layered video transmission model with end-to-end ARQ for BL packets

We assume an ad-hoc network with bursty losses and frequent path breakdowns. There is a lower routing layer providing and maintaining an active set of available paths (APS) from the source to the destination. Some paths in the APS are used in the video transmission, while others are kept as backups. The routing daemon probes the backup paths periodically and starts a rerouting process to find new paths if all the backup paths are down. Thus the impact of rerouting delay on the video traffic is minimized. The technique in [1] is used

[1] A variation of the EL packet dropping scheme is to dynamically reduce the rate of the EL when there is a request for the retransmission of a BL packet. This requires a more intelligent layered coder.

to attempt to keep the paths as disjoint as possible, so they are unlikely to be down at the same time.

A signaling protocol reserves bandwidth along the path for the video session. Although the wireless channel may have higher bandwidth, we only reserve what is needed. Instead of reserving a large bandwidth along a single path for both layers, we choose two disjoint paths to introduce diversity and transmit EL and BL packets separately on these two paths. The remaining bandwidth of the paths can be used to transmit other traffic.

Error concealment schemes [3] are used at the receiver and intra macroblock refreshment techniques are used at the sender to cope with the error propagation caused by lost BL packets (either due to failure of the ARQ scheme when both paths are unavailable, or a retransmitted packet arrives at the receiver too later).

III. Modeling of the Ad-hoc Network

We propose a new model for ad-hoc networks, which is described below and is used in our simulations.

Link Model Although wireless channels are time varying, experiments for various types of channels show that the basic channel parameters can be stable for short time intervals. Therefore a non-stationary wireless channel can be adequately represented by a set of stationary channel models [6][16][17][18]. The classical two-state Gilbert-Elliott model for burst noise channels has been widely used [16][17]. In [6][18], K-state Markov models are used in which the received instantaneous signal-to-noise ratio (SNR) is partitioned into K ranges where each range of SNR corresponds to one state. Various techniques are proposed to derive the transition probabilities of the Markov process given the partitioning and channel coding schemes.

In this work, we use a finite state discrete time Markov process to model a wireless link in the Ad-hoc network. Let $S = \{s_0, s_1, \ldots, s_N\}$ denotes the state space of the Markov process. As shown in Fig 4, the transition probabilities are $\{a_i, i = 1, 2, \ldots, N\}$ and $\{b_i, i = 1, 2, \ldots, N\}$, respectively. Each state i is associated with a Bit Error Rate (BER) P_i. We assume P_i's are in decreasing order, with $P_0 = 1$ (the link is down) and $P_N = 0$ (no loss at all). The parameters can be derived from available measurement data.

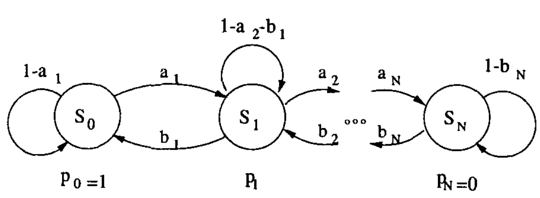

Fig. 4. The wireless link model

We assume that there is no abrupt change of the link quality, i.e., transitions only happen between adjacent states. This is reasonable because the movement of the mobiles, fading, the power drain on batteries, and interference of the nearby channels happen in a relatively slow time scale. Abrupt changes are not likely if we don't consider events such as the sudden failure of a mobile node.

Path Model Most previous work on wireless channel modeling focuses on modeling a fading channel with a single hop [6][16][18]. In Ad-hoc networks, a path consists of a number of wireless links, which can be dependent or independent of each other. Furthermore, in the multi-path case, the availability of a path, and the correlation of the paths are also important factors in addition to the loss rates and the available bandwidth [1]. Therefore the single hop model may be inadequate if we wish to take into account such issues.

We model an ad-hoc path as follows:

Let $\{L_i, i = 1, 2, \ldots, N\}$ represents a pool of links that can be used to form a set of paths from a source node to a destination node, e.g., the link pool in [1]. Each link in the pool is modeled as shown in Fig 4. A path is constructed by randomly choosing a subset of links from the pool. A packet goes through the links one at a time along the path and experiences different losses. The number of common links shared by two paths determines the degree of their correlation. Common links are perfectly correlated while different links are assumed independent. More common links therefore implies higher correlation. A path is down when one of its links is in state s_0. We can either choose a new set of links after a rerouting delay, i.e., to model the routing process; or wait until the link transits from state s_0 to s_1, i.e., the link is up again. In our simulation reported in Section V, when the link is down, all packets through this link are dropped and we wait for the link to go up again without path switching.

Packet Error Rate (PER) In our simulation, we simulate the bit error on the link, and then transfer it to path packet loss probability according the length of the packet being transmitted and the number of hops of the path.

Suppose a path consists of h links, the n_{th} packet with length l_n is being transmitted along the path and its first bit is transmitted on the first link at time t. Assume the BER of the link models, $p(t)$, has taken FEC into account, i.e., a packet is lost if one or more bits of the packet are corrupt. Then the loss probability for the n_{th} packet over the first link is:

$$PER_1^n(t) = 1 - \prod_{i=1}^{l_n} [1 - p(t + i - 1)] \tag{1}$$

For store-and-forward networks, the loss probability over the entire path for the n_{th} packet starting at time t is:

$$PER^n(t) = 1 - \prod_{k=1}^{h} [1 - PER_k^n(t + (k-1) \times l_n)] \tag{2}$$

IV. Simulation Results

To verify the performance of the new scheme, we simulate video transmission over two multi-hop wireless paths. A traditional scalable coding scheme without ARQ has also been simulated for the comparison purpose.

For all the experiments reported in this section, two disjoint paths are used for each video session. A path consists of 4 links and each link is modeled by a *three-state* Markov process [12]. BERs for the states are $p_0 = 1$, $p_1 \in [10^{-6}, 10^{-4}]$, and $p_2 = 0$, respectively. The average packet size is about 3000 bits, which gives path PER from 1% to 20%. RTT is assumed to be 300ms, which equals to 3 frame times. End-to-end SR and NACK only ARQ scheme is used to protect the BL packets by which lost BL packets are retransmitted on the EL path. We also assume the deadline $d = 2 \times RTT$, hence only one retransmission for each lost BL packet is allowed in our simulations. For the comparison scheme, BL and EL packets are sent to different paths without ARQ protection.

The "Foreman" video sequence is used in the simulations. Raw video frames are coded with a coder extended from the UBC H.263+ codec [19]. The SNR scalable coding technique in H.263+ standard (annex O) is used to generate BL and EL frames. The format of the video is QCIF, and our target frame rate is 10 frame per second (fps) with coded frame interval of 100ms. A GOB is mapped into one packet so that no GOB is split and loss of a packet only corrupts one GOB. This packetization strategy enhances the video's error resilient capability. Intra macroblock refreshment technique is used to mitigate the error propagation effect when packets are lost. The intra refresh rate is 20% for our scheme and 30% for the comparison scheme. The data rates of BL and EL are about 255kpbs each. In the decoder of our scheme, if one BL packet is lost after the retransmission, the decoder copies the corresponding GOB of the last decoded frame. If one EL packet is lost, the corresponding GOB from the base layer is copied. The decoder of the SNR scalable codec uses the same error concealment technique.

Table I shows the average PSNR obtained from a set of experiments we performed. Tests 1 and 2 are for symmetric paths conditions, while Tests 3, 4, 5, and 6 are for asymmetric paths conditions and different path allocations. It can be seen that for the same path conditions and same path allocation, our scheme outperforms the comparison scheme in all the cases. For PER on one path as high as 20% our scheme still gives average PSNR higher than 30dB. The difference in average PSNR, $PSNR_{ARQ} - PSNR_{NARQ}$, in Test 3 is smaller than that in Test 5. This is because the BL path is better in Test 3 and less BL packets are lost and hence retransmitted on the EL path. Tests 3, 4 and Tests 5, 6 also show how path allocations affect the received average PSNR. In Test 4, where the worse path is used for BL instead of the better path as in Test 3, the average PSNR of our scheme decreases 1.3dB, while the average PSNR for the comparison scheme decreases 5.7dB. In Test 6, when the worse path is used for BL instead of the better path as in Test 5, the average PSNR of our scheme decreases 2.6dB, while the average PSNR for the comparison scheme decreases 5.4dB. These show that our scheme is much more resilient to errors in path estimations than the comparison scheme. This is important for layered video transmission where the source is generally required to estimate the path conditions and choose the better path for the BL stream.

Fig. 5, 6, and 7 are the received PSNR versus frame numbers. In all the figures the solid-line curves are the PSNR of our scheme, while the dashed-line curves are the PSNR of two-path transport without ARQ. Although only one retransmission is allowed for each lost BL packet, the gain in PSNR is significant. It can be seen that our scheme outperforms the comparison scheme in all cases for most of the frames.

TABLE I
COMPARISON OF THE SCHEMES: AVERAGE PSNR

-	PER_1	PER_2	$PSNR_{ARQ}$	$PSNR_{NARQ}$
Test 1	8%	9%	36.3dB	30.5 dB
Test 2	16%	18%	31.5dB	25.3dB
Test 3	2%	20%	36.3dB	36.1dB
Test 4	20%	2%	35.0dB	30.4dB
Test 5	8%	19%	35.1dB	29.8dB
Test 6	19%	8%	32.5dB	24.4dB

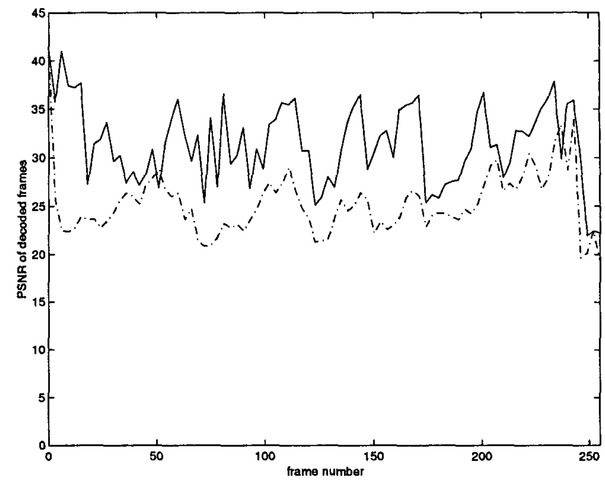

Fig. 5. PSNR vs. frame numbers for Test 2

V. CONCLUSIONS

In this paper we propose a simple but robust scheme for the reliable transmission of video over bandwidth limited ad-hoc networks. A raw video stream is layer coded and BL and EL packets are transmitted separately on two disjoint paths. The BL packets are protected by ARQ retransmission, by which the retransmissions of the BL packets take place on the path that EL packets are transmitted.

Simulation results show that this scheme is robust to path estimation errors and can guarantee a graceful video quality under adverse channel conditions. It is effective for video transmission over ad-hoc networks with burst errors.

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REFERENCES

[1] M. Pearlman, Z. J. Hass, P. Sholander, and S. S. Tabrizi, "On the impact of alternate path routing for load balancing in mobile Ad Hoc networks," *Proceedings of MobiHOC 2000*, pp:3-10, August 2000.

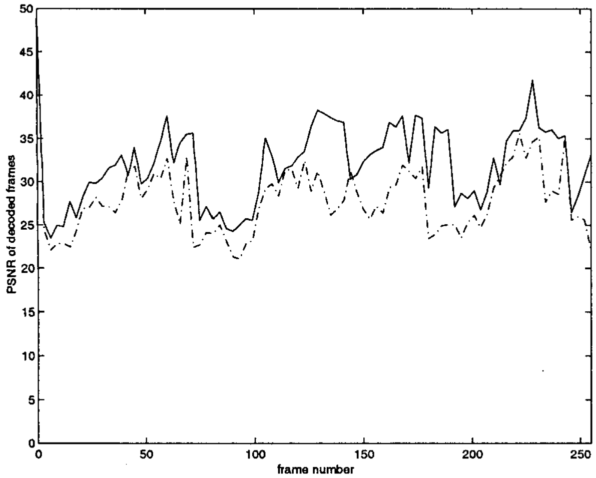

Fig. 6. PSNR vs. frame numbers for Test 5

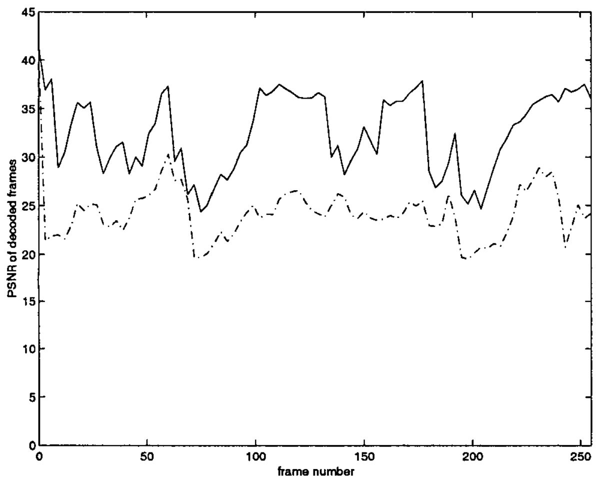

Fig. 7. PSNR vs. frame numbers for Test 6

[2] S. Pejhan, M. Schwartz, and D. Anastassiou, "Error control using retransmission schemes in multicast transport protocols for real-time media," *IEEE/ACM Transactions on Networking*, Vol.4, No.3, pp:413-427, June 1996.

[3] Y. Wang and Q. Zhou, "Error control and concealment for video communication: a review," *Proceedings of The IEEE*, Vol.86, No.5, pp:974-996, May 1998.

[4] Q. Zhang and S. A. Kassam, "Hybrid ARQ with selective combining for fading channels," *IEEE Journal on Selected Areas in Communications*, Vol.17, Issue 5, pp:867-880, May 1999.

[5] M. Khansari, A. Jalali, E. Dubois, and P. Mermelstein, "Low bit-rate video transmission over fading channels for wireless microcellular systems," *IEEE Transaction on Circuits and Systems for Video Technology*, Vol.6, No.1, pp:1-11, February 1996.

[6] J. Lu, K. B. Letaief, and M. L. Liou, "Robust video transmission over correlated mobile fading channels," *IEEE Trans. on Circuits and Systems for Video Technology*, Vol.9, No.5, pp:737-751, August 1999.

[7] R. Schafer and T. Sikora, "Digital video coding standards and their role in video communications," *Proceedings of The IEEE*, Vol. 83, No. 6, pp: 907-924, June 1995.

[8] S. Sarkar and L. Tassiulas, "Fair allocation of discrete bandwidth layers in multicast networks," *Proceedings of IEEE Infocom 2000*, pp.1491-1500, 2000.

[9] D. Saparilla and K. W. Ross, "Optimal streaming of layered video," *Proceedings of IEEE Infocom 2000*, pp:737-746, April 2000.

[10] M. Grossglauser and D. Tse, "Mobility increase the capacity of ad-hoc wireless networks," *Proceedings of IEEE Infocom 2001*, pp:1360-1369, April 2001.

[11] E. M. Royer and C.Toh, "A review of current routing protocols for ad hoc mobile wireless networks," *IEEE Personal Communications*, Vol.6, Issue 2, pp:46-55, April 1999.

[12] S. Lin, S. Mao, Y. Wang, and S. S. Panwar, "A reference picture selection scheme for video transmission over Ad-Hoc networks using multiple paths," *Proceedings of IEEE ICME 2001*, August 2001.

[13] Gogate, N. and S. S. Panwar, "Supporting applications in a mobile multihop radio enviroment using route diversity: I. Non-realtime data," *Proceedings of IEEE ICC'98*, pp:802-806, June 1998.

[14] N. Gogate and S. S. Panwar, "Supporting video/image applications in a mobile multihop radio environment using route diversity," *Proceedings of IEEE ICC'99*, pp:1701-1706, June 1999.

[15] N. Gogate and S. S. Panwar, "On a resequencing model for high speed networks," *Proceedings of IEEE Infocom'94*, pp:40-47, June 1994.

[16] E. N. Gilbert, "Capacity of a bursty-noise channel," *Bell Syst. Tech. J.*, Vol.39, No.9, pp:1253-1265, September 1960.

[17] E. O. Elliott, "Estimates of error rates for codes on burst-noise channels," *Bell Syst. Tech. J.*, Vol.42, pp:1977-1997, September 1963.

[18] Q. Zhang and S. A. Kassam, "Finite-state Markov model for Rayleigh fading channels," *IEEE Transaction on Communication*, Vol.47, No.1, pp:163-171, November 1999.

[19] University of British Columbia, "H.263+ codec," avaliable at ftp://dspftp.ece.ubc.ca.