Two-Hop Two-Path Voice Communications over a Mobile Ad-hoc Network

Jagadeesh Balam and Jerry D. Gibson
Department of Electrical and Computer Engineering
University Of California, Santa Barbara, California 93106-9560
Email: balam@ece.ucsb.edu and gibson@ece.ucsb.edu

***Abstract*— We consider two-hop communication of a delay-sensitive, memoryless Gaussian source over two independent paths in an ad-hoc network. To capture the behavior an ad-hoc network we combine a path availability model and a physical layer packet loss model. The path availability model includes the effect of path failures due to node mobility and route switching delays while the physical layer model accounts for the losses in the wireless channel. An analysis using the path availability model reveals potentially long connection down times due to path failures, suggesting that path diversity may be essential to support voice communications over a mobile ad-hoc network. We compare the performance of a few path diversity based communication methods involving multiple description coding and single description coding in an ad-hoc network with packet losses due to path failures and the physical channel.**

I. Introduction

We consider the communication of delay-sensitive sources like voice over a mobile ad-hoc network using path diversity. Interactive multimedia communication over an ad-hoc network may be hindered due to various factors such as bit errors, node failures, changing routes, and congestion. We analyze the effect of route failures due to node mobility and route switching delays, and losses in the wireless channel on the distortion of a delay-sensitive source communicated over two hops in an ad-hoc network.

We model the source as a delay-sensitive *i.i.d.* Gaussian source communicated over two hops in a two path ad-hoc network. For source coding, we consider multiple description (MD) coding with two descriptions of rate $R/2$ sent over two independent paths, a single description (SD) code of rate R sent over a single path, an SD code of rate $R/2$ duplicated over the two paths, and an SD code of rate R duplicated over the two paths. We consider only symmetric paths in this paper due to simplicity and the lack of space. In previous work [1], we compared different path diversity methods using parallel Gaussian channels. In the present work, we provide a theoretical analysis of the average distortion incurred by a source transmitted over an ad-hoc network that includes:

- A path availability model to model burst losses due to route failures caused by node mobility and route switching delays
- Allowable delay tolerance Δ before a packet is dropped
- Physical layer losses modeled by a two state Markov model
- Retransmissions to reduce packet losses in the wireless channel

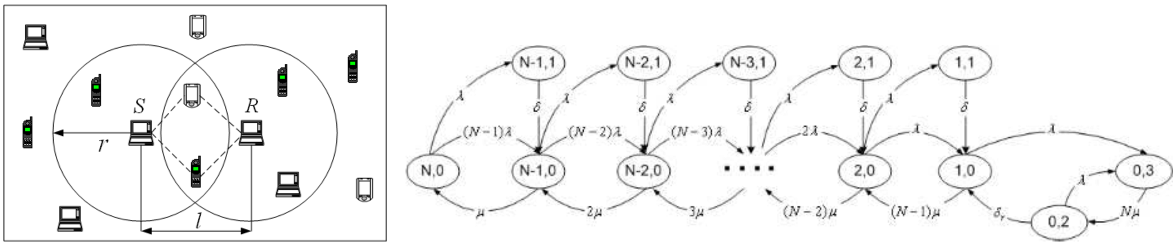
This research has been supported by the California Micro Program, Applied Signal Technology, Dolby Labs, Inc., Mindspeed, and Qualcomm, Inc., by NSF Grant Nos. CCF-0429884 and CNS-0435527, and by the UC Discovery Grant Program and Nokia, Inc.

In [2], the authors consider communication of delay-sensitive, memoryless Gaussian sources over wireline networks. They show that an MD coding system performs better than the SD system for high network loading. However, the authors do not consider packet headers, such as in IEEE 802.11, which could significantly affect the capacity and loading in a network when the payloads are small. We showed in our previous work [1], that a fair comparison between MD and SD should include packet overheads, because in networks such as IEEE 802.11 WLANs, the overheads can be significantly larger than the actual payloads, which mitigates any advantage due to MD coding. In this work, we analyze the performance of the path diversity methods considered in [1] when used for two-hop communication in an ad-hoc network.

II. Path Availability for two-hop communication

We consider a simplified ad-hoc network model for two-hop communication between two stationary sender and receiver nodes. There are N mobile nodes that are randomly scattered over a bounded area and each communication path between the sender and the receiver node requires one node out of the N nodes to serve as a router node. As shown in Fig. 1(a), the router node is located in the area of intersection of coverage of nodes S and D. In [3], this two-hop communication scenario is modeled as a continuous time Markov chain (CTMC) of N components with N repair facilities, with a failure rate λ and repair rate μ as shown in Fig. 1(b). Each state denoted (x, y) is identified by the number of nodes x in the intersection region, and the status of the connection y that can take values ranging from 0 to 3. '0' indicates that the connection is up between S and R, '1' indicates that the connection is down and the route needs to be switched to a new router node, '2' indicates that the connection is down and a new route needs to be established with the single node available in the intersection region and '3' indicates that there are no nodes that can act as a router for a connection to be set up. The average delay for route switching is $1/\delta$ and the average connection reestablishment delay is $1/\delta_r$.

Solving the balance equations for the CTMC, the steady



(a) Two-hop communication

(b) Markov model of an ad-hoc network (from [3])

Fig. 1. Markov model for two-hop communication in an ad-hoc network

state probabilities for each of the states ($\pi_{x,y}$) are given by the following set of equations [3]

$$\pi_{k,0} = \frac{N!}{k!(N-k)!}\left(\frac{\mu}{\lambda}\right)^k\left(\frac{\delta_r}{\lambda+\delta_r}\right)\pi_{0,3} \quad (1)$$

$$\pi_{j,1} = \frac{\lambda}{\delta}\frac{N!}{j!(N-j)!}\left(\frac{\mu}{\lambda}\right)^j\left(\frac{\delta_r}{\lambda+\delta_r}\right)\pi_{0,3} \quad (2)$$

$$\pi_{1,2} = \frac{N\mu}{\lambda+\delta_r}\pi_{0,3} \quad (3)$$

where $N \geq k \geq 1$, $N-1 \geq j \geq 1$ and $\pi_{0,3}$ after normalization can be determined as

$$\pi_{0,3}^{-1} = \sum_{k=1}^{N}\frac{N!}{k!(N-k)!}\left(\frac{\mu}{\lambda}\right)^k\left(\frac{\delta_r}{\lambda+\delta_r}\right)+\frac{N\mu}{\lambda+\delta_r}+ \sum_{k=2}^{N}\frac{\lambda}{\delta}\frac{N!}{k!(N-k)!}\left(\frac{\mu}{\lambda}\right)^k\left(\frac{\delta_r}{\lambda+\delta_r}\right)+1$$

The steady state connection availability (SSCA) is given by the summation of states with the connection in the 'up' state, i.e. (k,0) [3]

$$A_s = \sum_{k=1}^{N}\pi_{k,0} = \sum_{k=1}^{N}\frac{N!}{k!(N-k)!}\left(\frac{\mu}{\lambda}\right)^k\left(\frac{\delta_r}{\lambda+\delta_r}\right)\pi_{0,3} \quad (4)$$

We consider two independent paths set up between the sender and receiver nodes. The paths can be independent if the nodes have multiple radios and the communication is over a different channel for each router or if the routers time-share the channel. In such a scenario the path availability for each path in the steady state is similar to (4), the difference being that the number of nodes that can act as a router is reduced by one because one node is already acting as a router for the other path. The SSCA for each path is

$$A_{si} = \sum_{k=1}^{N-1}\frac{N-1!}{k!(N-1-k)!}\left(\frac{\mu}{\lambda}\right)^k\left(\frac{\delta_r}{\lambda+\delta_r}\right)\pi_{0,3} \quad (5)$$

$where\ i \in 1,2$. Equation (5) differs from (4) only in the number of nodes that affect the availability.

III. Source Coding Methods

We model the source to be transmitted as *i.i.d.* Gaussian with unit variance and the source is also assumed to be delay-sensitive, i.e. a transmitted packet is useless if received after a certain delay. For example, for voice communications, a packet delayed by more than about 150 - 250 ms may be discarded. In our previous work in [1], we compared various source coding methods along with path diversity in the presence of rate-dependent packet losses. Here, we compare the path diversity methods when used for two-hop communication over an ad-hoc network with packet losses due to route failures that are independent of rate and packet losses due to the physical channel that depend on the rate. The path diversity methods we consider are listed below.

1) Multiple description (MD) coding with path diversity
2) Path diversity with a half-rate ($R/2$ $(bits/symbol)$) SD code
3) Path diversity with a full-rate (R $(bits/symbol)$ SD) code

The achievable distortion region for a Gaussian source with unit variance and a fixed rate R ($R/2$ for each description), using MD coding is given by [4]

$$D_1 \geq 2^{-R} \quad (6)$$

$$D_0 \geq 2^{-2R} \quad (7)$$

$$(D_0, D_1) = (a, \frac{1+a}{2} - \frac{1-a}{2}\sqrt{1-\frac{2^{-2R}}{a}}) \quad (8)$$

$for\ a \in [2^{-2R}, 2^{-R}/(2-2^{-R}]$ where D_0 is the distortion at the central decoder and D_1 is the distortion at the side decoders. For a packet loss rate p, the average distortion achieved at the receiver using a two-description coder is

$$D_{MD} = (1-p)^2 a + 2p(1-p)(\frac{1+a}{2} - \frac{1-a}{2}\sqrt{1-\frac{2^{-2R}}{a}}) + p^2 \quad (9)$$

The above equation can be used to find the optimal distortion possible when the packet loss rate p is known to encoder, which is not the case in practice on wireless channels. We call this the MD optimal case (MD-OPT). We also consider the no excess joint rate case (MD-NJR) and the no excess marginal rate case (MD-NMR) of MD coding [5].

The average distortion for each of the communication methods that involve an SD coder, with probability of packet loss p is given as follows:
Single description of rate R without path diversity (SD)

$$D_{SD} = (1-p)2^{-2R} + p \quad (10)$$

Half-rate coder with path diversity (DHR-PD)

$$D_{DHR-PD} = (1-p)^2 2^{-R} + 2p(1-p)2^{-R} + p^2 \quad (11)$$

Full-rate coder with path diversity (DFR-PD)

$$D_{DFR-PD} = (1-p)^2 2^{-2R} + 2p(1-p)2^{-2R} + p^2 \quad (12)$$

For a two-hop network, we show the effect of packet losses on the performance of each of the communication methods mentioned above through the SNR obtained at the receiver, where SNR for the unit variance Gaussian source is calculated as $10 * log10(\frac{1}{D_{av}})$ where D_{av} is the average distortion at the receiver. For our specific analyses we pick a rate $R = 4\ bits\ per\ symbol$ and assume that each packet contains 40 symbols, resulting in 160 bits per packet. Each packet is generated at 20 ms intervals. Such a packet length and packet rate are common in packet based voice communications using low bit-rate codecs such as G.729 [6].

IV. Burst Losses due to Path Unavailability

For a delay-sensitive source, path failures due to the dynamic topology of the ad-hoc network can result in long bursts of packet losses. If the source is not delay-sensitive, then the source packets can be added to a queue and transmitted when the path is reestablished, but for voice communications, a packet received after a certain delay is as good as lost. It is a common practice in speech error concealment to completely silence the speech when there are more than five or six consecutive packet losses and then the call is dropped. It is important for a network supporting voice communications to guarantee that such long burst errors occur infrequently.

We reduce the model to two states [7] ('up' and 'down') and deduce the equivalent failure rate (rate of transitions from the 'up' state to the 'down' state) and repair rate (transitions from 'down' state to the 'up' state). The equivalent failure rate λ_{eq} is given by [7]

$$\begin{aligned} \lambda_{eq} &= \frac{\sum_{k=1}^{N} \pi_{k,0}\lambda}{\sum_{k=1}^{N} \pi_{k,0}} && (13) \\ &= \lambda && (14) \end{aligned}$$

The equivalent connection set up rate or the rate at which connection goes from the down state to the up state is given by

$$\delta_{eq} = \frac{\sum_{k=1}^{N-1} \pi_{k,1}\delta + \pi_{0,3}\delta_{0,3}}{\sum_{k=1}^{N-1} \pi_{k,1} + \pi_{0,3}} \quad (15)$$

where $\delta_{0,3} = \frac{N\mu\delta_r}{\lambda+\delta_r}$ is the repair rate when the network reaches state (0,3).

The network connection can now be modeled as a two state model with an up and a down state. A_s from (4) gives the probability that the network connection is up and $1 - A_s$ is the probability that the connection is down. From this model, we find the average time that the network is 'down' each time it enters the 'down' state as $1/\delta_{eq}$ for a single path.

In Fig. 2, we plot the SSCA for a varying distance between the sender and the receiver for different number of nodes N scattered randomly in a $1000\ m \times 1000\ m$ bounded area, with the average velocity of the nodes varying between $5\ m/s$ and $0.5\ m/s$. The transmission radius for all the nodes is fixed at $250\ m$ and the other network related delays are calculated using the parameters provided in [3]. In Fig. 3 we plot the average length of down time for the network due to route failure $1/\delta_{eq}$ for varying distance between the nodes and different number of nodes. We plot average down time lengths only until 50 seconds to show more clearly the smaller burst lengths. We see that, except when the path availability is close to one, $1/\delta_{eq}$ is in the order of seconds, which will result in long bursts of packet losses. The long down-times suggest that path diversity may be necessary to support voice communications in an ad-hoc network.

When the connection is down, the sender can hold a packet for a certain time, before dropping the packet, to wait for the connection to be setup. We take into account this allowable delay tolerance Δ when estimating an effective packet loss rate, so the probability of packet loss is

$$\begin{aligned} P(packet\ loss) &= P(path\ down)P(down\ time > \Delta) \\ &= (1 - A_s)(\int_{\infty}^{\Delta} \delta_{eq} e^{-\delta_{eq} t}) \\ &= (1 - A_s)(e^{-\delta_{eq}\Delta}) && (16) \end{aligned}$$

where Δ is the allowed delay, assuming that the propagation time is negligible and there is no contention for the channel when the path is active. For an allowed end-to-end delay of $200\ ms$, about $50\ ms$ accounts for codec delay and packetization delay, and about another $50\ ms$ for the jitter buffer at the receiver, so if we allocate about $20\ ms$ for network delays at the router node and the receiver node, we can choose $\Delta = 80\ ms$ as the delay up to which the sender node holds a packet waiting for the connection to be set up.

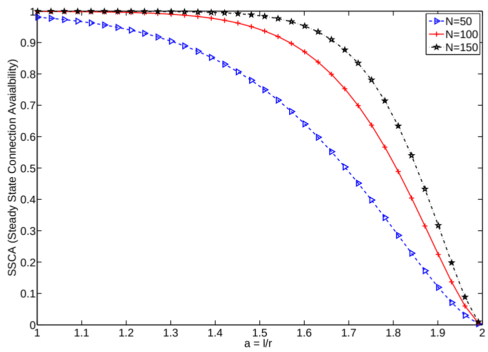


Fig. 2. Path Availability for different transmission radii and changing distance between the nodes

The distortion for each method of communication mentioned in the previous section is calculated using a probability of packet loss calculated as in (16). These burst packet losses are only due to connection failure because of node mobility and not due to the wireless channel. In Fig. 4 we show the effect of burst packet losses only, on the SNR for different communication methods mentioned in Section III, for the number of nodes in the network $N = 100$. When the distance between the sender and the receiver nodes l is small, both SSCA (Fig. 2) and $1/\delta_{eq}$ are small, which results in a small

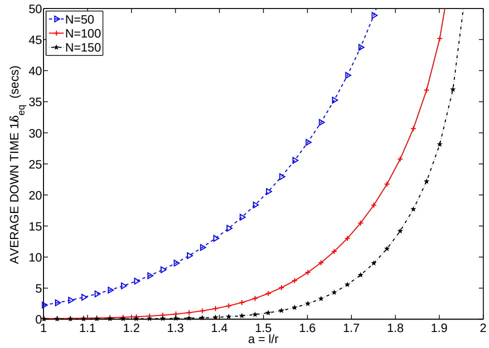


Fig. 3. Average burst lengths due to connection failure for a single path and two paths

packet loss rate. For small values of $a = l/r$, where $r = 250\ m$ is the transmission radius, which corresponds to a small packet loss rate, we see that the no excess joint rate case of MD coding (MD-NJR) and the single description code without path diversity (SD) give high SNRs and the optimal MD coder (MD-OPT) coincides with MD-NJR. As the internode distance l increases, the packet loss rate increases and the SNR for MD-NJR falls below that of MD-NMR, and MD-OPT moves from MD-NJR to MD-NMR. The duplicate full rate method (DFR-PD) consistently has the best performance because path diversity reduces the packet loss rate, and among all the path diversity methods considered, DFR-PD has the least per-symbol distortion due to source coding (2^{-2R}).

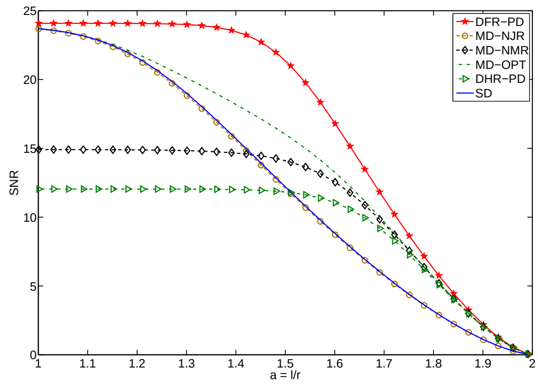


Fig. 4. Effect of burst losses on SNR ($N = 100, r = 250m$)

V. Losses in the Physical Channel

Now we consider packet losses due to the wireless channel when the path is available. We use the Gilbert Elliot model to model the bit errors induced in the physical channel. We use the equations given in [8] to calculate the packet error rate. We do not consider the physical layer preamble bits in our packet error rate calculation because these bits are usually transmitted at the basic rate resulting in the least possible error probability. The time to transmit a voice packet of payload size v and header h is given by $T = \frac{v+h}{R}$ where R is the transmission rate. We choose the $R = 2\ Mbps$ for our calculations.

As mentioned in [8], during the transmission of a packet, the channel state can vary in three different ways; 1) The channel remains in the good state throughout the transmission, 2) the channel remains in the bad state and 3) the channel switches from the initial state to the other state.

The probability of each case is given by Eqs. (17)-(19) [8]

$$p_{case1} = p_G P(G > T) = \frac{\alpha}{\beta + \alpha} e^{-\beta T} \quad (17)$$

$$p_{case2} = p_B P(B > T) = \frac{\beta}{\beta + \alpha} e^{-\alpha T} \quad (18)$$

$$p_{case3} = 1 - p_{case1} - p_{case2}. \quad (19)$$

where β is the the rate at which transitions from the good state to the bad state occur and α is the rate at which transitions from the bad state to the good state occur. Packet error rate in each case is approximated as [8]

$$\epsilon_{case1} = 1 - (1 - BER_G)^{(v+h)} \quad (20)$$

$$\epsilon_{case2} = 1 - (1 - BER_B)^{(v+h)} \quad (21)$$

$$\epsilon_{case3} \leq \epsilon_{case2} \quad (22)$$

Combining the probabilities of the three cases, the total packet error probability is given by

$$p_e = p_{case1}\epsilon_{case1} + p_{case2}\epsilon_{case2} + p_{case3}\epsilon_{case2} \quad (23)$$

where a worst case error rate is assumed for case 3.

For communication over two hops, the packet may be lost in either of the links. When a path is available, the packet is delivered only when both the links successfully deliver the packet. Therefore the effective probability of packet loss is,

$$p = 1 - (1 - (1 - A_s)e^{-\delta_{eq}\Delta})(1 - p_{e1})(1 - p_{e2}) \quad (24)$$

where p_{e1} and p_{e2} are the packet loss rates in the first and the second links respectively.

We consider symmetric paths for our analysis, i.e., both paths have the same availability probability and the model of the physical channel is the same on all the links. We also consider a 30 byte header (typical in IEEE 802.11 MAC with RTP/UDP/IP headers compressed to 2 bytes on average), while the payloads are 20 bytes (full rate) and 10 bytes (MD, half-rate). We use the two channel models given in [8] (also listed in Table I) for our analysis.

TABLE I
Burst Error models

Model	BER_G	BER_B	α	β
1	10^{-10}	10^{-5}	10	30
2	10^{-4}	10^{-2}	20	10

In Fig. 5 we plot the SNRs for each of the methods when the physical channel is modeled using model 1 in Table I. The small bit error rates in this model imply that the packet losses due to the wireless channel are not large. Under these 'good' channel conditions, when the probability of path availability is close to one, we see the ideal behavior expected from each of the source diversity methods. The no-excess joint rate (MD-NJR) method does better than no-excess marginal rate case because when the paths are active, for a majority of the transmission time, both descriptions are received at the decoder and the improvement in distortion achieved for MD-

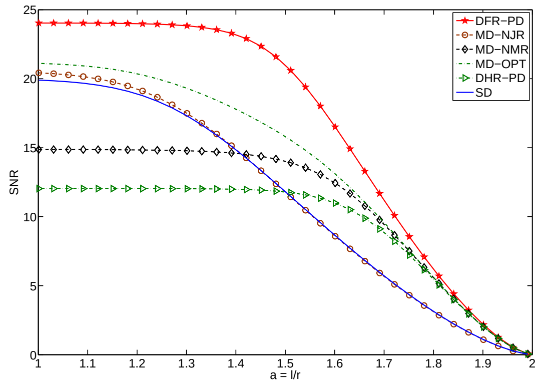


Fig. 5. SNRs for channel model 1 ($N = 100, r = 250m$)

NJR is larger than that of MD-NMR. For these conditions, single description over a single path also does better than MD-NMR, because SD uses the same number of bits per symbol optimally without any redundancy.

In Fig. 6, we plot SNRs when the wireless channel is modeled using the parameters of model 2 (Table I). For this model, the packet loss rates are very high, about 35%. We see that the performance of all the methods degrades significantly at these loss rates. In a classical scenario where source coding methods are compared without including overheads, MD-methods would have performed better because their smaller rate would result in a significantly smaller packet loss rate, unlike here where the large overheads due to the headers mitigate the advantage that a half-rate coder has over a full-rate coder.

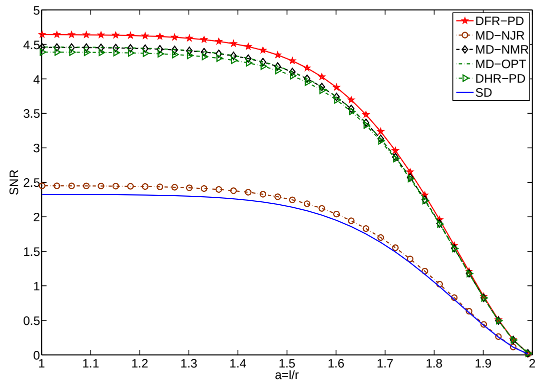


Fig. 6. SNRs for channel model 2 ($N = 100, r = 250m$)

Voice communications cannot usually tolerate large packet loss rates in the range of 35%. One way to counter these physical layer losses is to use retransmissions. Some researchers have suggested using a smaller number of maximum retransmissions for voice, so that the delay introduced by the retransmissions is not large. If we allow two retransmissions per packet, then the effective packet error rate is

$$p_e^{New} = p_e^3 \qquad (25)$$

where p_e is the packet error rate given by (23). When we use (25) to calculate the physical layer packet error rate for model 2, the packet error rates are reduced to about 4.3%. The SNRs when two retransmissions are allowed are plotted in Fig. 7. Observe that there is a significant improvement in the SNRs for all the methods when just two retransmissions are allowed.

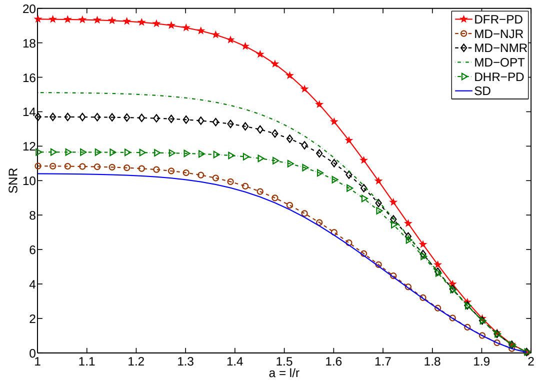


Fig. 7. SNRs for channel model 2 when two retransmissions are allowed

VI. Conclusions

We see that for small packet loss rates, the MD methods do not offer much advantage over SD in terms of SNR. However, note that the connection down times obtained using the path availability model can be large for each path resulting in long bursts of losses. Such large burst losses result in clipping of speech and large perceptual distortions that can be avoided by using path diversity methods. However, if the down-times are small, SD can be used avoiding path diversity when the physical channel is good. Also, when the node density is small, path diversity may be necessary to reduce the burst lengths. When multiple independent paths are established for communication, the probability that all the paths breakdown simultaneously is small leading to a smaller down time.

Performance of MD coding falls between the performance of DHR-PD and DFR-PD. Our results also demonstrate the difficulty in designing an MD coder that is suitable for all packet loss rates. If speech quality is the decisive factor, then DFR-PD is the best choice, since for a small increase (about 15.6% here), in the bits transmitted (when overheads are considered), compared to half rate methods it gives considerable improvement in SNR.

References

[1] J. Balam and J. D. Gibson, “Path diversity and multiple descriptions with rate dependent packet losses,” *Information Theory and Applications Workshop, University of California, San Diego, La Jolla, CA*, 2006.
[2] M. Alasti, K. Sayrafian-Pour, A. Ephremides, and N. Farvardin, “Multiple description coding in networks with congestion problem,” *IEEE Transactions on Information Theory*, p. 47(3):891902, Mar. 2001.
[3] D. Trajanov, S. Filiposka, M. Efnuseva, and A. Grnarov, “Ad hoc networks connection availability modelings,” *PE-WASUN ’04: Proceedings of the 1st ACM international workshop on Performance evaluation of wireless ad hoc, sensor, and ubiquitous networks*, 2004.
[4] M. Effros, R. Koetter, A. J. Goldsmith, and M. Medard, “On source and channel codes for multiple inputs and outputs: Does multiple description beat space time?” *Information Theory Workshop 2004*, Oct. 2004.
[5] L. Ozarow, “On a source-coding problem with two channels and three receivers,” *Bell Syst. Tech. J.*, vol. 59-10, pp. 1909–1921, 1980.
[6] *Coding of Speech at 8 kbit/s Using Conjugate-Structure Algebraic-Code-Excited Linear-Prediction (CS-ACELP)*, ITU-T G.729 Recommendation.
[7] D. Chen, S. Garg, and K.S. Trivedi, “Network Survivability Performance Evaluation: A Quantitative Approach with Applications in Wireless Ad-hoc Networks,” *Proceedings of the 5th ACM international workshop on Modeling analysis and simulation of wireless and mobile systems*, 2002.
[8] M. Veeraraghavan, N. Cocker, and T. Moors, “Support of voice services in IEEE 802.11 wireless LANs,” *IEEE Infocomm*, 2001.