Towards Voice Over Ad Hoc Networks: An Adaptive Scheme for Packet Voice Communications Over Wireless Links

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***Abstract*— In a no-guarantees network such as an ad hoc network, nodes are expected to adapt to changes in network and channel conditions. As a step toward supporting voice communications in such environments, we propose an adaptation scheme that takes both voice coding (i.e., compression) and modulation as parameters.**

In such severe environments, the main goal is to maintain a call as opposed to providing excellent quality. To this end, our adaptation scheme demands the least cost in terms of network bandwidth when network conditions are good. As network conditions worsen, more bandwidth is needed so that an acceptable quality can be maintained. This approach is a balance between providing reasonable quality and less resource consumption (with its implication on decreasing network load thus not affecting other connections).

Even though our ultimate goal is supporting voice over ad hoc networks, in this paper we present extensive simulation results over a single hop scenario.

Unlike other adaptation schemes involving source coding and modulation schemes in the literature, our scheme is specific to the nature of voice, thus allowing for better perceived quality without imposing high costs. Our adaptation scheme is investigated under a Rayleigh fading channel modeled as a finite state Markov channel (FSMC). The scheme is generic enough to work for any packet voice system including VoIP over wireless.

Results show that the scheme allows for supporting many more calls when compared to a non-adaptive scheme.

I. Introduction

Packet voice over wireless has received a considerable amount of attention in the last few years. Application domains include wireless LANs, wireless local loop and more recently mobile ad hoc networks (MANETs). Packet voice allows for convergence of different networks (i.e., voice, video, data) which will result in lower costs and will allow for new applications. For example, a hand-held device may have the functionality of a cell phone, a GPS receiver, a PDA and even be integrated with a bio-sensor network implanted in the human body (for medical monitoring).

Supporting real-time applications, such as voice, over wireless data networks faces many challenges. The wireless medium is characterized by high variations resulting from signal's exposure to absorption, interference and multipath and shadow fading. In addition, the spectral bandwidth is scarce. As a result, solutions proposed to wired networks or retrofitted versions of them are not necessarily the best for a wireless context.

For real time applications, quality of service (QoS) parameters' values depend on maintaining the interactivity requirement. For telephony traffic, delays longer than 50 ms require echo cancelation. Long delays affect interactivity and result in cross-talk, a phenomenon that we observe when using Internet telephony. Further, losses cannot be handled by retransmissions as in the case of data traffic, as this will result in excessive end-to-end delays. Nonetheless, voice can afford a small amount of packet loss because the human ear-brain is less sensitive to short dropouts in received speech [15]. In addition, voice is an isochronous application [18], i.e., voice samples are generated at regular intervals. The human ear perceives delay variation, also called jitter, as changes in the pitch. Thus, it is important to keep it as low as possible. On the positive side, voice bandwidth requirements are low. It is known that human voice has a speech activity factor of about 42% [14]. That is, only 42% of the time a human is talking (i.e., generating traffic).

Not only providing the quality needed is important, but also it is important to provide it in the most efficient way minimizing resource usage. By maintaining the resource requirements of a source to the minimum and in turn maintaining the overall real-time traffic in the network to a certain level, voice communications can occur with least disturbance (given non-real-time sources are given less priority).

In this paper, an adaptation scheme that maintains acceptable quality while minimizes bandwidth consumption is proposed. It combines compression and modulation in a way to increase the chances of a connection survival throughout the lifetime of a call as opposed to focusing on short-term quality.

The rest of the paper is organized as follows. Section 2 gives our problem statement and system model. The models used for generating voice traffic and for generating errors over a Rayleigh fading channel are explained in section 3. Section 4 discusses the simulation environment. Section 5 shows our results and we conclude in section 6.

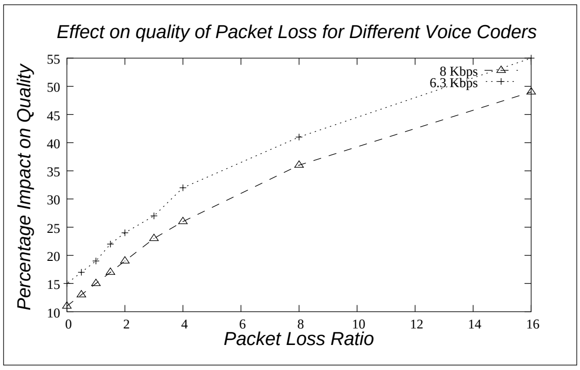

Fig. 1. Effect of Packet Loss on Quality for Two Voice Coders.

II. Problem Statement and System Model

A. The Problem

In a wireless network that does not promise any guarantees, e.g., an ad hoc network, the likelihood of call survival can be increased by decreasing the overall load on the network. This claim stems from the following observations.

On the one hand, using non-compressed voice would mean good quality (if received at the other end). This however would result in more load in the network with its implications on (i) the number of losses (due to full buffers), (ii) the increase in too-late packets (arriving after their deadline) and (iii) the increase of overhead (as the number of packets sent increases, so does the amount of overhead). On the other hand, using highly compressed voice though does not promise as good of a quality, it decreases the pressure on the network. However, compression cannot be used arbitrarily. *The loss of a packet carrying compressed traffic has higher penalty on perceived quality when compared to a packet of the same size carrying non-compressed traffic*. The effect of packet loss on quality is illustrated in figure 1 for two compression schemes. The y-axis shows the loss in quality expressed using the R rating factor of the ITU-T E-model [11][12][13].

In addition to employing compression for load reduction, adaptive modulation can be used. Traditionally, modulation technique is chosen in a way to stand the most severe channel conditions, which results in low spectral efficiency. Adaptive modulation can promise higher spectral efficiency by choosing a modulation scheme based on the current channel status (in terms of signal-to-noise ratio). Thus, when channel is good, a dense modulation is employed, and when it is bad, a sparse modulation is used.

B. Overview of Adaptation Scheme

So far we have established the importance of adaptation and the use of network and channel conditions in choosing compression and modulation schemes. Our adaptation scheme makes the choice of modulation and compression scheme depending on the following guidelines:

1. Adaptive modulation: the better the channel status, the denser the modulation scheme we use. By denser we mean, higher number of bits per symbol. This is justified by increasing spectral efficiency while not sacrificing transmission quality.
2. Adaptive source coding: as we have mentioned, compressed packets' loss has high penalty. Therefore, to decrease compressed packet loss while decrease the amount of real time traffic in the network, we increase the compression ratio as the channel quality gets better. When the channel suffers, we decrease the compression ratio (i.e., increase the data rate requirements per source). Even though it may seem counter-intuitive to increase the data rate when the channel is bad, it resembles an extra step in error control. This way, users can experience more or less a stable quality irrespective of channel effects. We are planning to make the compression ratio aware also of the traffic load in the network in a subsequent work that considers a multi-hop ad hoc network setup.

As we can see, in bad channel conditions, a source's bandwidth requirements increase (less compression) while channel capacity decreases (low density modulation scheme). Obviously, during such conditions, channel may not be able to serve all arriving traffic which would result in a lag stored in a buffer. The idea here is not to compromise call quality (by increasing a call's budget during bad conditions) while increase channel utilization (by taking advantage of good channel times).

C. Evaluation Methodology

In evaluating the performance of our proposed scheme, we try to answer the following question: *how much extra advantage our adaptation scheme gives over a non-adaptive one ?* The advantage is defined in terms of the multiplexing gain. That is, for the same delivered quality, how many extra sources (calls) the adaptive scheme can support when compared to the non-adaptive one. Alternatively, the question can be stated as: *for the same load conditions, how much quality enhancement does the adaptive scheme offer ?*

Performance is quantified in terms of two metrics: the voice quality achieved and multiplexing gain. To quantify the voice quality, we use the degradation in voice quality (DVQ) as a metric. DVQ is defined as:

$$\text{DVQ} = \frac{\text{Nos. packets lost + Nos. packets above delay threshold}}{\text{Total Number of packets}}.$$

Even though DVQ does not explicitly take the delay variations into account, they are being accounted for. It is known that delay variations less than 75 ms give good quality [17]. In our results, we consider 80 ms as the deadline for packet arrivals. Thus, having a low delay deadline eliminates the need to account for delay variations separately.

Multiplexing gain is quantified in terms of number of calls (sources) that can be supported.

D. Prior Work

The use of adaptive schemes is not new. In [2] adaptive voice compression over IP is studied. In [20], different adaptation schemes are studied. It is shown that rate adaptive (that is, compression-level) applications (as opposed to delay adaptive applications which change their delay requirements) react better to network congestion. Many studies have used adaptive modulation in areas ranging from wireless LANs to Cellular networks [10][1][25].

Adaptive Multi Rate (AMR) voice has been proposed for the Global System for Mobile Communications (GSM) along with fixed error control. AMR sacrifices the quality by sending at lower rate in bad conditions in order to add more channel coding [9]. Unlike the AMR scheme, which tries to divide the gross rate between source and channel coding, our scheme does not necessitate such a constraint as we propose it for packet wireless networks where the bandwidth budget for a node is not fixed.

Non of the aforementioned schemes was designed with ad hoc networks in mind. Our scheme is one step towards voice communications in ad hoc networks. By being aware of the severe conditions of the network (i.e., lack of central entity, lack of CAC, channel changes over time, etc.) sources' expectations are kept to the minimum thus not over-burdening the network with realtime traffic.

III. Source and Wireless Channel Models

This section explains the models we used for voice traffic generation and for the error behavior of a Rayleigh fading channel. We present the models in generic terms and in the simulation section we give the parameters we used in our experiments.

A. Voice Traffic Model

Modeling of voice traffic has been studied extensively in the literature, which resulted in several models. For a detailed explanation of voice models see [16]. In general, these models represent speech as a Markov chain with different number of states. The more the number of states in the model, the more complicated it is. Among the simplest models for voice traffic is the On-Off model first proposed by Brady [3], where a two-state chain is assumed. This is the most widely used model for its simplicity and highly acceptable results. The two states correspond to the talk spurt and silence periods. In this model, the activities are exclusive, i.e. while A is talking B is silent, and vise versa. The same applies to the silence state. This means that the model does not account for double talk, and mutual silence. Hence, it does not model two-way conversations accurately. The holding time in the talk state and the silence state is assumed to be exponentially distributed with mean times of 352 ms and 650 ms, respectively [4]. These values are highly acceptable in the published literature.

B. Wireless Channel Error Model Using an N-State FSMC

1) Finite state Markovian Model of the Wireless Channel Errors: In a previous work [8], we have used a two-state Gilbert-Elliot model to capture the error behavior of a wireless link experiencing Rayleigh fading. The 2-state model is not highly responsive to the wide-range of changes in the channel. Thus, a more elaborate k-state Markov model is used in this paper. Not only will this capture the variations in the wireless channel in a better way, it allows for using more compression and modulation levels.

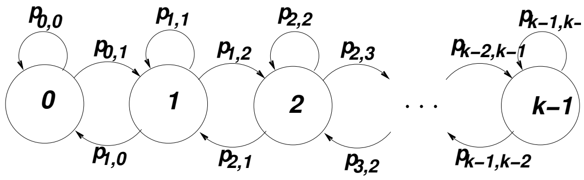

Fig. 2. Generic Model of the Wireless Channel

The use of a Markov process to model the Rayleigh fading channel was proposed by Wang et al [26] as an extension of the simpler two-state Gilbert-Elliot model. Finite State Markov Channels (FSMCs) are defined by constant Markov processes. A constant Markov process has the property of being stationary. That is, the transition probabilities are independent of the time index. They are defined by a finite number of states $\{s_0, s_1, ..., s_{k-1}\}$. A state i resembles a SNR that falls in the range $[s_i, s_{i+1}]$. SNR characterizes the channel quality. A high SNR would mean a lower BER and a low SNR results in a higher BER. In [21], it is shown that the probability density function (pdf) of the SNR in a Rayleigh fading environment is exponentially distributed. Figure 2 illustrates the FSMC. The first state starts at $SNR = 0$, and the last includes all SNR values greater than some threshold s_{k-1}.

The FSMC can be described mathematically by the following three elements (for a k-state chain):

1. The transition probability matrix (p): A matrix of elements P_{ij} representing the transition probability from state i to state j. P_{ij} is between 0 and 1, and each row adds up to 1.
2. Steady State Probability (π): A vector representing the steady state probabilities. That is, the probability of being in some state i, $i = 0, 1, ..., k-1$.
3. Cross Over Probability (e): A vector representing the cross-over probabilities of k Binary Symmetric Channels (BSCs). A BSC represents how a transmitted bit can be received in error. It depends on the SNR range the state is representing.

2) Parameter Calculation: The steady state probability is the probability that the SNR falls in the range that state is representing. Given that SNR is represented by the random variable S, π_i, the probability that the SNR falls in the interval $[s_i, s_{i+1}]$, can be calculated by:

$$\pi_k = \int_{s_k}^{s_{k+1}} p(s)ds = \exp^{\frac{s_k}{E[S]}} - \exp^{\frac{s_{k+1}}{E[S]}} \qquad (1)$$

where $p(s)$ is the exponential pdf of the SNR and is given by:

$$p(s) = \frac{\exp^{\frac{-s}{E[S]}}}{E[S]} \qquad (2)$$

where $E[S]$ is the average SNR. Bit errors depend on two factors. The SNR and the modulation scheme used. Since we are considering an adaptive modulation scheme, the calculation of the BER in a given state will depend on the modulation method we are planning to use when the channel is in that state. The probability of error in state i is:

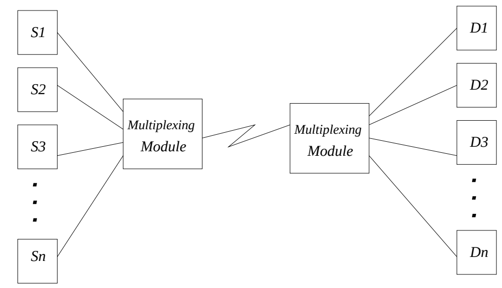

Fig. 3. Source Configuration

$$BER_i = \frac{1}{\pi_i} \int_{s_i}^{s_{i+1}} Q(.)p(s)ds \tag{3}$$

where $Q(.)$ is the Q-function of the particular modulation scheme, and is equal to [19]:

$$Q(z) = \frac{1}{\sqrt{2\pi}} \int_{z}^{\infty} \exp^{\frac{-y^2}{2}} dy \tag{4}$$

The calculation of the transition probability is based on the assumption that the fading is slow enough that the received SNR remains constant during a channel symbol. Given this, the number of times the SNR passes downward across a certain level (N_a) can be calculated. It is shown in [26] that the transition probability from state k to state $k-1$ is equal to: $\frac{N_k}{R^k}$ where R^k is the number of symbols transmitted in that state. It is assumed that the channel can move from one state only to its neighbors. This is a valid assumption in a slow, Rayleigh fading channel.

IV. Simulations

Figure 3 shows the source configuration. Voice sources send their traffic to a multiplexing module, which buffers packets till it can forward them. Packets are served in a first-in first-out (FIFO) manner. When the buffer is full, tail drop is applied. Upon receiving a packet, the destination module forwards it to the respective destination node. Even though this setup may seem like a wireless local loop , it is not necessarily the case. Since a major goal of this work is to quantify multiplexing gain, and since voice sources' rate is not an arbitrary parameter (i.e., unlike data traffic which can be increased/decreased as necessary), we needed to experiment with the number of sources that can be supported. Further, the number of sources can be thought of as the number of calls generated by a single source turning this into a single source-destination communication.

For the sake of our simulations, we discretized the channel into six states. The choice was thought to capture the behavior of the channel in an ample manner and allows the use of a whole spectrum of modulation schemes and voice coding algorithms. Nonetheless, the adopted channel model is generic and any number of states can be used. Figure 4 shows the 6-state FSMC we used in our simulations along with the respective modulation scheme and the voice compression rate applied at each state.

Using Eq. 3, the following formulas show how we can calculate the error rate in each state. The Q-functions for the different

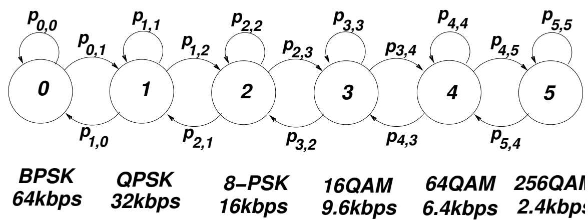

Fig. 4. A 6-state FSMC model of the Wireless Channel

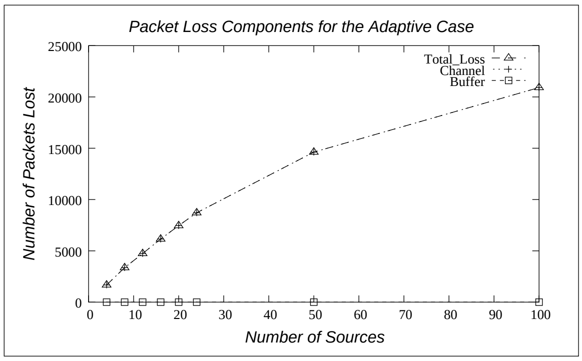

Fig. 5. Packet Loss Components for the adaptive both case (i.e., Mod + Enc)

modulation schemes are well known in the literature and can be found for example in [6].

$$BER_{e,BPSK} = \frac{1}{\pi_0} \int_{s_0}^{s_1} 0.5erfc(\sqrt{s}).p(s)ds \tag{5}$$

$$BER_{e,QPSK} = \frac{1}{\pi_1} \int_{s_1}^{s_2} 0.5erfc(\sqrt{s}).p(s)ds \tag{6}$$

$$BER_{e,8-PSK} = \frac{1}{\pi_2} \int_{s_2}^{s_3} \frac{1}{3}erfc(\sqrt{3s}\sin\frac{\pi}{8}).p(s)ds \tag{7}$$

$$BER_{e,16QAM} = \frac{1}{\pi_3} \int_{s_3}^{s_4} 0.375erfc(\sqrt{0.4s}).p(s)ds \tag{8}$$

$$BER_{e,64QAM} = \frac{1}{\pi_4} \int_{s_4}^{s_5} 0.292erfc(\sqrt{\frac{18}{126}s}).p(s)ds \tag{9}$$

$$BER_{e,256QAM} = \frac{1}{\pi_5} \int_{s_5}^{s_6} 0.234erfc(\sqrt{\frac{12}{255}s}).p(s)ds \tag{10}$$

where $erfc$ is the complementary error function and is defined by:

$$erfc(z) = \frac{2}{\sqrt{\pi}} \int_{z}^{\infty} \exp^{-x^2} dx \tag{11}$$

The above integrals were solved numerically using Matlab. We partitioned the SNR in a way to give us the following BER vector $[10^{-2}, 10^{-3}, ..., 10^{-7}]$. Having the SNR partitioning, we can get the transition probabilities and the steady state probabilities.

Calculations based on the above partitioning resulted in the following steady state probability vector:

$$\pi = [0.080152,\ 0.273014,\ 0.048657,\\ 0.493896,\ 0.104279,\ 0.000002]$$

The link capacity is equal to $256KHz$. Simulation time is 120 sec, which is the typical holding time of a call. Every simulation is run 20 times, giving a 95% confidence interval. The Doppler frequency shift: $100Hz$, resembling high mobility conditions which means it induces more fading (as the ISI becomes higher due to the increase in frequency). Hence, better results can be expected with a smaller Doppler value. Packet size is 30 bytes. Average SNR, $E[S]$ = 15dB. In our results, the main and only delay component that is captured is the queueing delay. Thus, transmission delay is not taken into account as its very small (a maximum of $0.9ms$). Simulations were implemented using C++.

V. Results and Analysis

Figure 5 shows the packet loss components (buffer and channel losses) for the adaptive both (i.e., encoding and modulation) case. As can be seen, even though in a non-adaptive context the link capacity allows only for four sources, we have plotted up to 100 sources, and yet buffer losses are equal to 0. Given the highly dense modulation schemes considered along with the high compression ratios, we succeeded to achieve high multiplexing gain. Losses are solely channel's. With the high savings from adaptation, part of the load saving can be used to add error correction to the payload.

To have a closer look at the loss behavior, we show the packet loss ratio for the given case in figure 6. Even though the behavior of this graph seems counterintuitive, it does make sense. As the load increases, the idle time of the channel becomes less, and given that different states have different probabilities, it turns out that the load increase translates into serving more in good states than in bad ones, and hence with load increase, packet drop even though increases but not as much as throughput increase. That is why the packet loss ratio decreases as the load (represented by the number of sources) increases. This can be seen in figure 7. As we can see, with load increase, state 4 serves more of the traffic than the rest (it has a steady state prob = 0.49, see the steady state vector). Since this state has a low BER (= 10^{-5}), with load increase overall packet loss ratio decreases.

To have better understanding, we plotted the loss behavior of a single source under light, medium and heavy load conditions represented by 4, 24, and 100 sources, respectively. As we can see from figure 8, which shows the number of packets lost per second for a source over a duration of 120 sec, the heavier the load, the less the number of packets lost. This means, not only does the adaptation scheme give better loss performance for the aggregate traffic, but also for individual connections.

To verify that this claim is accurate, we also plotted the distribution of the loss bursts for a source under light, medium and heavy load conditions. As we can see from figure 9, the heavier the load is, the less the probability of a bigger burst.

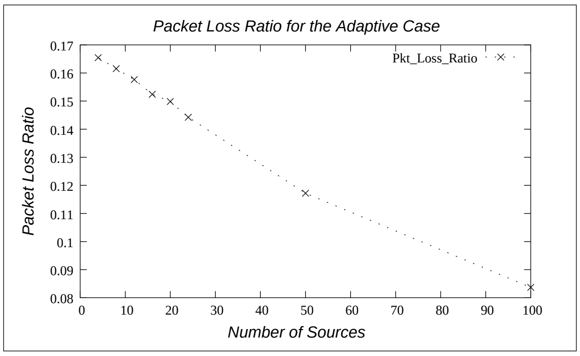

Fig. 6. Packet Loss Ratio for the adaptive both case (i.e., Mod + Enc)

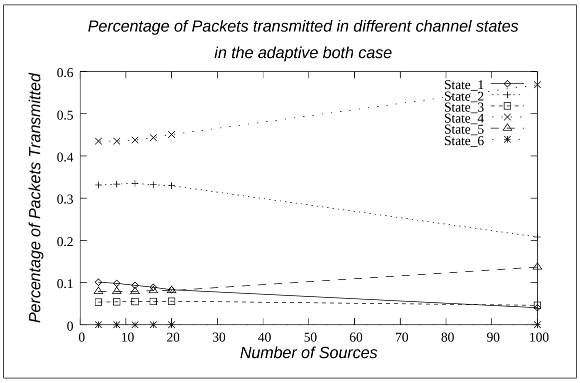

Fig. 7. Packet transmission distribution in different channel states

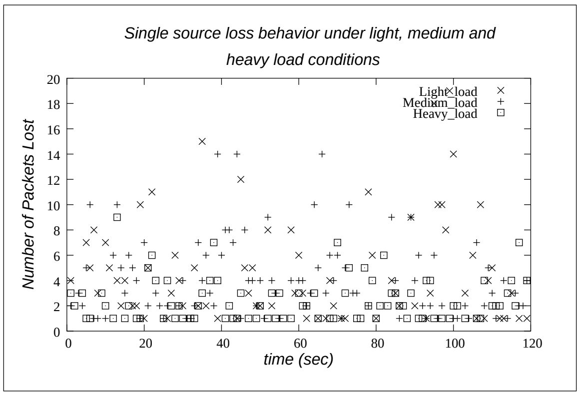

Fig. 8. Single source loss behavior under light, medium, and heavy load conditions

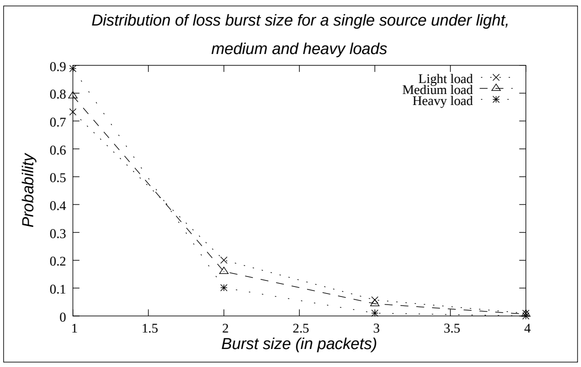

Fig. 9. Distribution of loss burst size for a source under light, medium and heavy loads

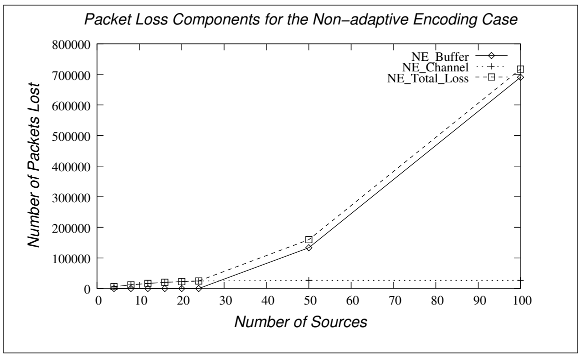

Fig. 10. Packet Loss Components for the Non-adaptive Enc. Case

Figure 10 shows the loss components for the adaptive-modulation-only encoding case (that is, source coding is non-adaptive). The number of packets lost at a number of sources equal to 16 (around 20000 packets), where the total traffic load is 180000 packets (not shown), is the same as the number of packets lost in the adaptive case when having 100 sources where the traffic load is 250000. Even though the gap between the two loads is big, the number of losses is the same. This can be explained by the fact that in the non-adaptive case the amount of traffic served in every state is the same. Put another way, each state serves an equal share of the traffic, whereas in the adaptive case, good states take higher share of the traffic (resembled by more compressed traffic which means more sources).

Figure 11 shows the DVQ (defined earlier) for the three cases (i.e., adaptive both, adaptive-encoding-only and adaptive-modulation-only). As can be seen, the DVQ decreases till it reaches a particular load where the adaptive case stabilizes and the two non-adaptive ones start to increase. The decrease, as we mentioned earlier, is due to better use of the channel (less idle, more time in good states since they have higher probabilities). The increase, on the other hand, is due to the increase of the load beyond the channel's capacity at which point buffering starts to take place, where some packets are so late and others are being dropped due to a full buffer.

For comparison purposes, we also show packet loss for the three cases in figure 12. As we can see, the absence of adaptive modulation did not seem to affect the packet loss in figure 12 as much as the absence of adaptive source coding, which explains why it turned out to give better overall result when looking at figure 11.

It can be deduced from these two figures that increasing the channel capacity alone (represented by employing adaptive modulation) does not promise good quality. Adaptive coding, where more weight of the traffic is served during good channel times (represented by more compressed traffic) turned out to promise better overall quality.

The number of packets arriving after the delay threshold is shown in figure 13. As we can see, adaptive modulation resulted in much less stale packets as it speeds up serving packets from the buffer.

From these results we can conclude that adaptive modulation if combined with adaptive coding can promise much better performance.

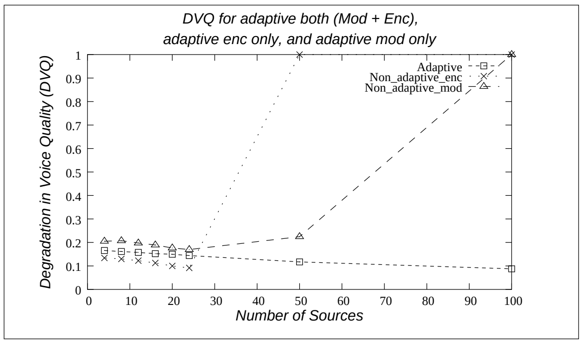

Fig. 11. DVQ for adaptive both (Mod + Enc), adaptive-encoding-only and adaptive-modulation-only

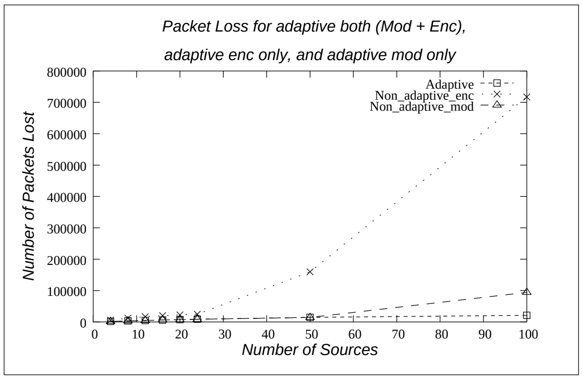

Fig. 12. Packet Loss for adaptive both (Mod + Enc), non-adaptive enc only and non-adaptive mod only

VI. Conclusions

An adaptation scheme targeted for voice over ad hoc networks was proposed. The scheme takes into account variations in channel condition trying to minimize the bandwidth cost of a call without sacrificing the quality perceived. As a starting point, we showed the results for a one-hop scenario which we are currently extending to multiple hops. A FSMC model was used to capture the error behavior of a wireless link suffering Rayleigh fading. Results show that the multiplexing gain is high compared to a non-adaptive scheme while maintaining a reasonable quality.

We are planning to extend this work by considering a multi-hop ad hoc network setup, where voice over IP (VoIP) over 802.11 is transmitted. We will also include data traffic and limit the number of voice calls to a certain level of the overall load. In addition, the aforementioned adaptation parameters will depend on network load in addition to channel state.

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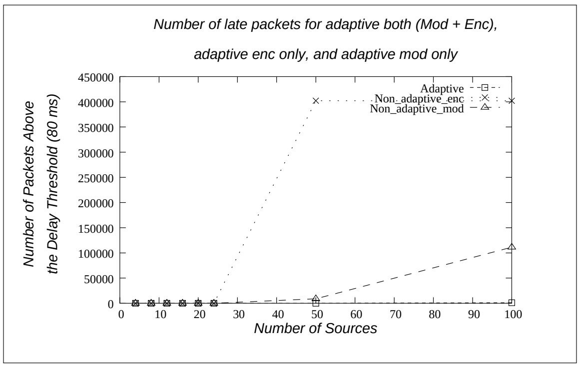

Fig. 13. Number of Packets Beyond a delay threshold of 80 ms for adaptive both (Mod + Enc), adaptive enc only, and adaptive mod only

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