Content-Aware Multiple Access Protocol for Cooperative Packet Speech Communications

Amr El-Sherif, Andres Kwasinski, Ahmed K. Sadek, and K. J. Ray Liu

***Abstract*—A novel cooperative multiple access protocol for packet speech communications is proposed, where cooperation is achieved through the deployment of a relay node that exploits the silence periods typical of speech communications. The relay forwards speech packets for active calls using a subset of the free time slots left available by users that are silent. No new channel resources are needed for cooperation and the system encounters no bandwidth losses. Because the resources allocated to the relay are not drawn from the pool of reused resources but from those potentially used for random channel access. Thus, the use of cooperation introduces a tradeoff between the amount of help offered to active calls and the probability of a successful contention for channel access. Such cooperation tradeoff is investigated and guidelines for the choice of protocol parameters are developed. The throughput and delay performance of the proposed protocol are characterized and compared to a similar non-cooperative packet speech protocol. Results demonstrate significant gains achieved by the proposed cooperative protocol. Moreover, the speech quality under the cooperative protocol was measured using a perceptual model, and results reveal significant improvement over the non-cooperative protocol especially in the low signal-to-noise ratio regime.**

***Index Terms*—Cooperative communications, Medium Access Control, speech communication, wireless networks.**

I. INTRODUCTION

COOPERATIVE communication is a new paradigm for wireless networks that builds upon the early studies on the relay channel [1]. Cooperation is based on the broadcast nature of wireless channels where a transmitted signal can be overheard by other network nodes and, instead of traditionally discarding it, the overheard signal is processed and relayed to the destination. At the destination, the original and relayed signals are combined to generate a signal with better quality, creating a new form of diversity which can significantly improve the system performance and robustness. The transmission from the source and relay nodes achieves transmit diversity by emulating the transmission from a virtual multi-antenna array (virtual MIMO). Some cooperation schemes, namely, amplify-and-forward, decode-and-forward, selection relaying and incremental relaying, were proposed in [2] and their outage probability performance was analyzed resulting in substantial performance gain compared to non-cooperative strategies. In this paper, the incremental relaying scheme will be adopted as the cooperative protocol. In incremental relaying, the relay utilizes limited feedback from the destination in the form of automatic repeat request to decide whether to forward or not.

While most of the prior works on cooperative communications focused on the physical layer, few have focused on the impact and implementation of cooperation at higher network layers. In [3], the authors proposed a distributed version of the network diversity multiple access (NDMA) protocol [4] and provided pairwise error probability analysis to demonstrate the diversity gain. In [5], the notion of utilizing the spatial separation between users to assign cooperating pairs was presented. In [6], a cognitive multiple access protocol was proposed, the protocol benefits from the source burstiness and enable cooperation by allowing the relay to utilize the terminals' periods of silence. Also [6] characterized the maximum stable throughput region and delay performance of the cognitive protocol.

Speech communication has a distinct characteristic that differentiates it from data communication, which was the main focus of all of the previous work on user cooperation. Speech sources are characterized by periods of silence in between talk spurts. The speech talk-silence patterns could be exploited in statistical multiplexing-like schemes where silent users, release their channel resources, which can then be utilized to admit more users to the network. This comes at the cost of requiring a more sophisticated multiple access protocol. One well-known protocol to address this problem is the Packet Reservation Multiple Access (PRMA) protocol [7], which can be viewed as a combination of TDMA and slotted ALOHA protocols. In PRMA, terminals in talk spurts contend for the channel in empty time slots. If a user is successful, then the slot is reserved for the user. Users with reservations transmit their voice packets in their reserved slots. If a user fails to transmit its packet due to wireless channel errors, the reserved slot becomes free for contention again. One of the weaknesses of this protocol is that channel errors not only lose the damaged packets, but also increase the network traffic and access delay because users with lost packets have to go through the contention process again.

In this paper we propose a different way to exploit the periods of silence during a conversation. We build a cooperative system on top of the PRMA protocol by enabling a relay node to make use of the time slots freed by the silent users to forward talking users' packets. The relay helps improve the communication channel through the spatial diversity it creates. Nevertheless, the improvements in channel quality that can be achieved with the relay come with a cost. This is due to the fact that the time slots the relay uses were previously available for users' channel access contentions. Thus, the use of the relay

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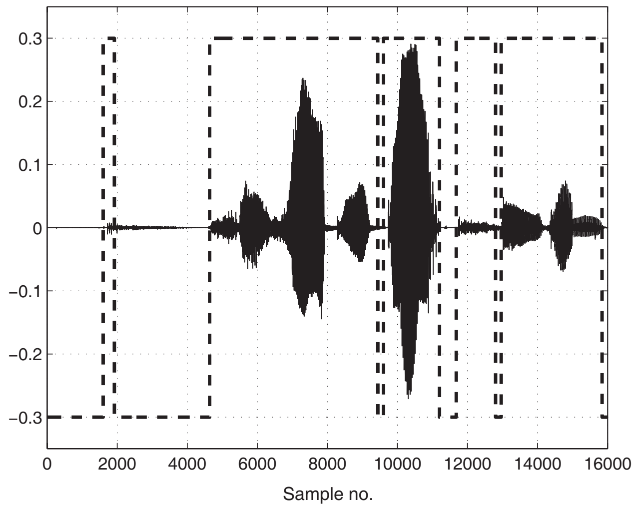

Fig. 1. A typical speech segment illustrating the on/off characteristic of speech. The dashed lines take a value of 0.3 when speech is detected "on" and a value of -0.3 when speech is detected "off".

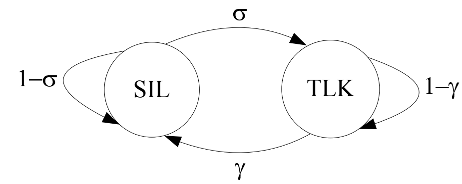

Fig. 2. Speech source model.

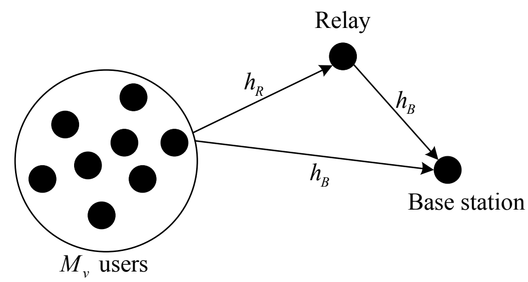

Fig. 3. Network and Channel model.

introduces a tradeoff between the networks ability to admit more users and the amount of help offered by the relay. This kind of tradeoff is, in some sense, similar to the diversity multiplexing tradeoff in MIMO and other user cooperative systems [8]. This tradeoff is studied from the point of view of different performance measures in order to find the best way to share free time slots between the relay and contending users.

In addition, we study the performance of the proposed protocol by developing a Markov chain model that describes the network evolution in the presence of the cooperative relay. Through the analysis of this Markov model, we gain insights into the dynamics of the network and the tradeoffs associated with the presence of the relay. We characterize the throughput and delay performance of our proposed protocol, and compare them with a network that does not implement cooperation. Furthermore, we study the packet dropping probability and, in the application layer, the speech quality. Our results show how the deployment of a single relay node can lead to significant improvements in network performance with significant gains of the proposed cooperative protocol over the non-cooperative one.

The rest of the paper is organized as follows. In section II, we describe the system model. Our new cooperative multiple access protocol is presented in section III. In section IV the analytic model for the study of our protocol is developed, and different performance measures are studied in section V. Numerical results are presented and discussed in section VI. Finally, this work is concluded in section VII.

II. System Model

A. Speech Source Model

Speech sources are characterized by periods of silence in between talk spurts that account for roughly 60 % of the conversation time [9]. This key property could be exploited to significantly improve the utilization of channel resources but with the cost of requiring a more sophisticated multiple access protocol. Figure 1 shows the voice signal amplitude for a speech sequence and illustrates the alternation of speech between talking and silence periods. To model this alternation, each speech source in a conversation is modeled as a Markov chain as shown in Fig. 2 with two states: talk (TLK) and silence (SIL) states. In packet speech communications scenarios, it is convenient to choose the same basic time unit for the Markov chain as the one used for channel access [10]. As will be discussed in the next section, the channel is divided into TDMA time frames, each of duration T seconds. Hence, it is suitable to choose the basic time unit for the Markov chain equal modeling speech to T seconds also. This means that state transitions are only allowed at the frame boundaries.

With the assumption that the waiting time in any state has an exponential distribution [9], the state transition probabilities for the Markov chain can be calculated as follows: The transition probability from the talking state to the silence state is the probability that a talk spurt with mean duration t_1 ends in a frame of duration T and is given by

$$\gamma = 1 - e^{-T/t_1}. \tag{1}$$

Similarly, the probability that a silence gap of mean duration t_2 ends during a frame of duration T is the transition probability from the silence state to the talking state

$$\sigma = 1 - e^{-T/t_2}. \tag{2}$$

It should be noted that the statistics in 1 and 2 are at the output of the speech encoder.

B. Network Model

We consider the uplink channel of a speech network employing the packet reservation multiple access (PRMA) protocol [7] as a medium access protocol (see Fig. 3). PRMA can be viewed as a combination of TDMA and slotted ALOHA protocols where the channel is subdivided into time frames and each frame is in turn subdivided into N time slots. Users in talk spurts contend independently for the channel in empty time slots with access probability p_v. If a user is successful in the contention process, then the slot is reserved for that user; otherwise, the base station feeds back a NULL message to make the slot available for contention in the next time frame. Users with reserved slots use them to transmit their corresponding speech packets. Upon ending a talk spurt, a user enters a silence state where it is not generating or transmitting any packets. In this case, the base station feeds back a NULL message to declare that the previously reserved time slot is free again for other users to use (we assume immediate feedback of the NULL message). Note that the PRMA protocol exploits the on-off nature of speech to improve the utilization of the channel by reserving slots only to calls in a talk spurt.

The state of every time slot (free or reserved) in the current frame is determined by the base station feedback at the end of each time slot in the previous frame. It is assumed that the feedback channel is error free, thus there is no uncertainty in the state of any time slot. Moreover, it is assumed that the base station will also feed back a NULL message in response to errors due to contention and due to wireless channel impairments. This means that a user will lose its reservation if it faces a channel error while transmitting its packet. Here, we ideally assume that the feedback message is immediate.

Because speech communication is very sensitive to delay, speech packets require prompt delivery. In PRMA, the voice packets from calls that fail the contention to access the channel are placed in a waiting queue. If a packet remains undelivered for a pre-specified maximum delay of D_{max} frames, the packet is dropped from the user's queue.

C. Channel Model

The received signal at the base station can be written as

$$y_B = \sqrt{G r_B^{-\alpha}} h_B x + \eta_B; \tag{3}$$

similarly, the received signal at the relay

$$y_R = \sqrt{G r_R^{-\alpha}} h_R x + \eta_R; \tag{4}$$

where x is the transmitted signal, G the transmission power, assumed to be the same for all users and the relay, r_B and r_R denotes the distance from any user to the base station and to the relay, respectively, α is the path loss exponent, and h_B and h_R are the channel fading coefficients for the user-base station and user-relay links, respectively, which are modeled as zero-mean complex Gaussian random variables with unit variance. The additive noise terms η_B and η_R are modeled as zero-mean complex Gaussian random variables with variance N_0. We assume that the channel coefficients are constant for the transmission duration of one packet. In this work, we only considered the case of a symmetric network, where all the inter-users channels are assumed to be statistically identical.

We characterize the success and failure of packet reception by outage events and outage probabilities. The outage probability is defined as the probability that the Signal-to-Noise Ratio (SNR) at the receiver is less than a given SNR threshold β, called *outage SNR* [11]. For the channel model in (3) and (4), the received SNR of a signal transmitted between any user and the base station can be specified as follows

$$\mathrm{SNR}_B = \frac{| h_B |^2 r_B^{-\alpha} G}{N_0}, \tag{5}$$

where $| h_B |^2$ is the random channel gain magnitude squared, which has an exponential distribution with unit mean. The outage event for an outage SNR β is equivalent to

$$\{h_B : \mathrm{SNR}_B < \beta\} = \left\{h_B :| h_B |^2 < \frac{\beta N_0 r_B^{\alpha}}{G}\right\}. \tag{6}$$

Accordingly, and from the exponential distribution of the received SNR, the outage probability is

$$P_{OB} = \Pr\left\{| h_B |^2 < \frac{\beta N_0 r_B^{\alpha}}{G}\right\} = 1 - \exp\left(-\frac{\beta N_0 r_B^{\alpha}}{G}\right). \tag{7}$$

Similar relations hold for the outage probability between any user and the relay (P_{OR}), and between relay and base station (P_{ORB}).

III. Content-Aware Cooperative Multiple Access Protocol

Transmission errors inherent to wireless communication channels impact significantly the performance of the network described in Section II-B [12]. On one side, a user that experiences an error while contending for access to a time slot, fails on the try and has to contend again in another free slot. Moreover, a user that already holds a reserved slot has to give it up and go through the contention process again because when receiving a packet with errors the base station sends a NULL feedback that indicates that the reserved slot is now free. These effects translate into an increase in contending users and, thus, a significant increase in network traffic and in delay to gain a slot reservation, which ultimately severely degrades the speech quality. In fact, the congestion may reach a level where all users experience reduced speech quality due to packets dropped due to excessive delay [7].

By enabling cooperation in the voice network, one can benefit from the spatial diversity offered by cooperation to mitigate the wireless channel impairments. Here we propose the deployment of a single relay node into the network. This node will have the task of helping users holding slot reservations to forward their packets by operating in an incremental decode-and-forward mode [2]. In this mode, the relay first decodes the received packet, and then re-encodes and forwards a regenerated version of the packet to the base station. In order to decide whether to forward the packet or not, the relay utilizes limited feedback from the base station in the form of automatic repeat request (here we consider the NULL feedback as the repeat request message to the relay). This means that the relay will only forward the packets that were not successfully received by the base station. If the relay successfully forwards the packet to the base station then the user owning that packet will not lose its reservation and will continue sending new packet in the upcoming frames. If on the other hand, the relay fails to forward that packet then the user will lose its reserved slot and will have to go through the contention process again. It is clear that the use of the relay results in a more reliable end-to-end link and, hence, a reduction in the number of users losing their reserved time slots. This leads to a further reduction in the average number of contending users, and therefore, much lower access delay and packet dropping probability, which ultimately improves speech quality.

To incorporate the relay operation into the network, we propose the following frame structure. The first M_T slots create a variable size compartment (from frame to frame) of slots reserved for the talking users. Of the remaining $(N - M_T)$ free slots, a fraction p_r is assigned to the relay and the remaining free slots are made available for contention. The ordering of slots in a frame is first the M_T slots reserved for the talking users, followed by M_R slots assigned to the relay, and the remaining slots are used in the contention process. When a user gives up its reservation or gains a new reservation, the slots are rearranged in order to maintain this frame structure.

Through the base station ACK to a successful contention, all network members will know that a new user has gained a reservation. This user's reserved slot will be appended to the end of the reserved slots compartment. When a user gives up its reservation, the base station feedback will inform all users that this time slot will be free in the next frame. To keep the reserved slots compartment contiguous, any user whose reserved slot is after the freed up slot will shift its transmission one slot earlier in the next frame. It should be noted that rearrangement of slots reserved slots is achieved through the information provided by the base station feedback, and no extra scheduling is required.

In any specific frame, a fraction p_r of the non-reserved slots is assigned to the relay. In some frames, when the number of failed packets is smaller then the number of relay slots, the relay might not use all its assigned slots to correct all the failed packets. In such a case, when the base station detects the correction of all failed packets, it sends a special feed back message declaring the end of the relay slots and the start of the contention slots. Moreover, since the relay is helping talking users only, no slots are assigned to the relay when there is no users with reserved slots. The maximum number of time slots assigned to the relay is then,

$$M_R = \begin{cases} 0 & M_T = 0 \\ \text{round}(p_r(N - M_T)) & M_T > 0 \end{cases} \tag{8}$$

It is clear that the value of p_r determines how much help the relay will offer to talking users, also it determines the reduction in the number of free time slots available for contention. Therefore, the introduction of cooperation poses a tradeoff between the amount of help the relay offers to existing users and the ability of the network to admit new users because of the reduction in the number of contention slots. Since such tradeoff is governed by p_r, the choice of the value of this parameter is crucial for the optimal performance of the system. In what follows, we will study this tradeoff by investigating the effect of p_r on different performance measures, and guidelines for the choice of p_r will be drawn.

IV. Dynamic State Model

In this section, we develop an analytical model to study and measure the network performance. Based on the models discussed above, a user can be in one of three states: "SIL" when in a silence period, "CON" when contending for channel access, and "TLK" when holding a reserved slot. The dynamics of user transitions between these three states can be described by the Markov chain of Fig. 4 [10]. A user in SIL state moves to CON state when a new talk spurt begins. When there is an available slot, with probability p_v, a user in CON state will send the packet at the head of its queue. If contention succeeds, a user in CON state transits to TLK state, where it will have the slot reserved in subsequent frames. A user moves from CON state to SIL state if its talk spurt ends before gaining access to the channel. A user in TLK state transits to SIL state when its talk spurt ends, and transits to CON state if its packet is not received correctly by the base station and on the first try and the relay is unable to help. Again, we will consider one complete frame as the time step for the Markov chain. Although the actions of different users are independent, the transition probabilities between different states for a given user are in general dependent on the number of users in CON and TLK states. These numbers will affect the probability with which a user succeeds in contention. Moreover, the number of users in TLK state will determine the number of slots assigned to the relay, and hence the relay's ability to help users.

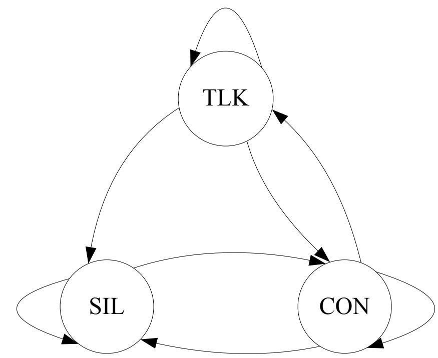

Fig. 4. User's terminal model.

In order to take these dependencies into consideration, the whole network will be modeled as the two-dimensional Markov chain (M_C, M_T), where M_C and M_T are random variables denoting the number of users in CON and TLK states, respectively. Assuming there are M_v users in the network, then the number of users in the SIL state is $M_S = M_v - M_C - M_T$. In what follows, we will analyze this Markov chain and calculate its stationary distribution, then based on this analysis, we will derive different performance measures for the cooperative protocol.

Let $S_1 = (M_{C_1}, M_{T_1})$, and $S_2 = (M_{C_2}, M_{T_2})$ be the system states at two consecutive frames. Then,

$$M_{C_2} = M_{C_1} + m_{SC} + m_{TC} - m_{CS} - m_{CT}, \tag{9}$$
$$M_{T_2} = M_{T_1} + m_{CT} - m_{TS} - m_{TC}, \tag{10}$$

where m_{ij} denotes the number of users departing from state $i \in \{S, C, T\}$ to state $j \in \{S, C, T\}$, for example, m_{SC} is the number of users departing from SIL state to CON state. This implies that the transition probability between any two states can be determined in terms of the distributions of m_{SC}, m_{CS}, m_{CT}, m_{TS}, and m_{TC}. Next we will calculate these distributions.

A. Distribution of m_{SC}

From the speech source of Fig. 2, and since all users are independent, the number of users making a transition from the SIL state to the CON state, m_{SC}, follows a binomial distribution with parameter σ, where σ is defined in (2). Then,

$$\Pr(m_{SC}=i)=\binom{M_S}{i}\sigma^i(1-\sigma)^{M_S-i}, \quad i=0,...,M_S. \tag{11}$$

B. Distribution of m_{TS}

It follows from the speech source model of Fig. 2, and from users independence, that the number of users making a transition from the TLK state to the SIL state, m_{TS}, is binomially distributed with parameter γ, where γ is defined in (1). Then,

$$\Pr(m_{TS}=i)=\binom{M_T}{i}\gamma^i(1-\gamma)^{M_T-i}, \quad i=0,...,M_T. \tag{12}$$

C. Distribution of m_{TC}

A user leaves the TLK state to the CON state if its transmitted packet fails to reach the base station successfully, and if the relay did not help that user. Also, a user in TLK state will leave to SIL state if its talk spurt ends in the current frame irrespective of the reception state of its last transmitted packet. This means that this user will not attempt to retransmit its last packet in the talk spurt and the relay will not try to help this user.

Given the number of users making transitions from TLK state to SIL state (their talk spurt ended and have no packets to transmit), m_{TS}, the number of erroneous packets from the remaining users in the TLK state, ε, follows a binomial distribution with parameter P_{OB}, the outage probability of the link between any user and the base station as defined in (7). Therefore,

$$\Pr(\varepsilon=i|m_{TS})=\binom{M_T'}{i}P_{OB}^{\,i}(1-P_{OB})^{M_T'-i}, \quad i=0,...,M_T', \tag{13}$$

where $M_T'=M_T-m_{TS}$, the number of remaining users in the TLK state. Assume that the relay can successfully receive ε_R packets of the ε erroneous packets. Then, conditioned on ε, the number of successfully received packets by the relay, ε_R, is also binomially distributed but with parameter P_{OR}, the outage probability of the link from any user to the relay,

$$\Pr(\varepsilon_R=i|\varepsilon)=\binom{\varepsilon}{i}(1-P_{OR})^iP_{OR}^{\,\varepsilon-i}, \quad i=0,...,\varepsilon. \tag{14}$$

For each of the slots assigned to the relay, a packet among the ε_R packets in the relay's queue is selected at random and forwarded. It follows that for $\varepsilon_R\geq M_R$, the number of successfully forwarded packets ε_F is binomially distributed with parameter P_{ORB}, the outage probability of the link from relay to base station,

$$\Pr(\varepsilon_F=i|\varepsilon_R)=\binom{M_R}{i}(1-P_{ORB})^iP_{ORB}^{\,M_R-i}, \quad i=0,...,M_R, \tag{15}$$

where M_R is the max. number of time slots assigned to the relay. For $\varepsilon_R<M_R$, the distribution of the number of successfully forwarded packets follows (15) for $i=0,...,\varepsilon_R-1$, and the probability that $\varepsilon_F=\varepsilon_R$ is

$$\Pr(\varepsilon_F=\varepsilon_R|\varepsilon_R)=\sum_{i=0}^{M_R-\varepsilon_R}\binom{\varepsilon_R+i-1}{i}\times P_{ORB}^{\,i}(1-P_{ORB})^{\varepsilon_R}, \tag{16}$$

which accounts for all the possible combinations of successful and failed packet transmissions before the ε_Rth successful packet.

Now, the probability that i users make the transition from TLK state to CON state is the probability that from the ε erroneous packets, the relay successfully forwards ($\varepsilon_F=\varepsilon_R-i$) packets. Then the distribution of m_{TC} is given by

$$\Pr(m_{TC}=i|m_{TS})=\sum_{k=0}^{M_T-m_{TS}}\sum_{l=0}^{k}\Pr(\varepsilon_F=\varepsilon_R-i|\varepsilon_R=l)\times\Pr(\varepsilon_R=l|\varepsilon=k)\Pr(\varepsilon=k|m_{TS}). \tag{17}$$

D. Distribution of m_{CT}

Upon a successful contention, a user transits from the CON to the TLK state. This transition occurs at the end of each free slot where contention can take place. Thus, the number of contending users will vary from slot to slot. Suppose there are M_T reserved slots and the relays uses m_R slots in a given frame, then there is $(N-M_T-m_R)$ free slots for contention. We want to calculate the distribution of the number of users that moved from CON state to TLK state at the end of the last free slot. This distribution could be calculated using the following recurrence model. Let $q(M_C')$ be the probability that a user succeeds in contention when there are M_C' contending users, and p_v the users' channel access probability, then

$$q(M_C')=M_C'p_v(1-p_v)^{M_C'-1}(1-P_{OB}), \tag{18}$$

which is the probability that only one user has permission to transmit and the channel was not in outage during packet transmission.

Define $R_k(M_C')$ as the probability that M_C' terminals remain in the CON state at the end of the kth available slot, $(k=0,1,2,...,N-M_T-m_R)$. Conditioning on the outcome of the $(k-1)st$ time slot, it follows that

$$R_k(M_C')=R_{k-1}(M_C')[1-q(M_C')]+R_{k-1}(M_C'+1)q(M_C'+1), \quad M_C'=0,1,...,M_C, \tag{19}$$

where M_C is the number of users in the CON state at the beginning of the frame. The initial condition for this recursion is

$$R_0(M_C')=\begin{cases}1 & M_C'=M_C\\ 0 & M_C'\neq M_C\end{cases} \tag{20}$$

and the boundary condition $q(M_C+1)=0$, which follows from the fact that the total number of contending users is M_C. Finally, the distribution of m_{CT} is

$$\Pr(m_{CT}=i|m_R)=R_{N-M_T-m_R}(M_C-i), \quad i=0,...,M_C, \tag{21}$$

i.e., the probability that i users succeed in contention in the current frame is equal to the probability that (M_C-i) users remain in the contention state at the end of that frame.

To compute the distribution of m_R, the number of time slots the relay actually uses, we need to condition on ε_R and ε_F, the number of packets in the relay queue and the number of successfully forwarded packets by the relay, respectively. For $\varepsilon_R \geq M_R$, i.e., the relay has more packets to forward than it has assigned time slots , and for $\varepsilon_R < M_R$ with $0 \leq \varepsilon_F < \varepsilon_R$, where the relay uses all the M_R slots to successfully forward ε_F packets, we have $\Pr(m_R = M_R|\varepsilon_R, \varepsilon_F) = 1$. For the remaining case where $\varepsilon_R < M_R$, it follows from (16) by a simple change of variables that

$$\Pr(m_R = i|\varepsilon_R, \varepsilon_F) = \binom{i-1}{i-\varepsilon_R} Po_{RB}^{i-\varepsilon_R} \quad i = \varepsilon_R, ..., M_R \tag{22}$$

which is the probability that there are i failed transmissions before the ε_Rth successful transmission.

E. Distribution of m_{CS}

A user makes a transition from the CON state to the SIL state if its talk spurt ends before gaining access to the channel. If we condition on the number of users that successfully accessed the channel, m_{CT}, and through the same argument as in IV-B, we have

$$\Pr(m_{CS} = i|m_{CT}) = \binom{M_C - m_{CT}}{i} \gamma^i (1-\gamma)^{M_C - m_{CT} - i}, \quad i = 0, ..., M_C - m_{CT}. \tag{23}$$

We will keep the distribution conditioned on m_{CT} because this is the form we will be interested in when calculating the state transition matrix.

Remark: All the distributions calculated above are state dependent because they generally depend on M_C and M_T. This means that we have to calculate a different set of distributions for each possible state of the system.

F. State transition probabilities

Here we consider the state transition matrix $\mathbf{P}$. This matrix is square, and it can be shown that its dimension M is

$$M = \begin{cases} \frac{(N+1)(N+2)}{2} & \text{if } M_v \leq N \\ (N+1)(M_v - \frac{N}{2} + 1) & \text{if } M_v > N \end{cases}$$

An element $P(S_1, S_2)$ of this matrix is the transition probability from state $S_1 = (M_{C_1}, M_{T_1})$ to state $S_2 = (M_{C_2}, M_{T_2})$. If $M_{T_2} > \min(M_{T_1} + M_{C_1}, N)$, then $P(S_1, S_2) = 0$ because the number of terminals in TLK state in the next frame cannot exceed the total number of time slots in a frame or the number of terminals in TLK and CON states in the current frame. It easily follows from (9) and (10), and the distributions developed above that the transition probability $P(S_1, S_2)$ is given by

$$\begin{aligned} P(S_1, S_2) = &\sum_{x=0}^{M_{C_1}} \sum_{y=0}^{M'} \sum_{z=0}^{M_{T_1}} \Pr(m_{CS} = x|m_{CT} = y, S_1) \\ &\times \Pr(m_{TC} = M_{T_1} - M_{T_2} + y - z|m_{TS} = z, S_1) \\ &\times \Pr(m_{SC} = M_{C_2} - M_{C_1} + x + y - z|S_1) \\ &\times \Pr(m_{CT} = y|S_1) Pr(m_{TS} = z|S_1), \end{aligned} \tag{24}$$

where $M' = \min(M_{C_1} - x, N - M_{T_1} - M_{R_1})$. It should be noted that $\Pr(m_{TC} = M_{T_1} - M_{T_2} + y - z|S_1) = 0$ if $M_{T_1} - M_{T_2} + y - z > M_{T_1} - z$, because the number of users transiting from TLK state to CON state cannot exceed the difference between the number of users initially in the TLK state and the number of users leaving the TLK state to the SIL state. Also, $\Pr(m_{SC} = M_{C_2} - M_{C_1} + x + y - z|S_1) = 0$ if $M_{C_2} - M_{C_1} + x + y - z > M_{S_1}$, because the number of users leaving the SIL state cannot be larger than the number of users initially in this state.

Finally, the stationary distribution vector π can be calculated as the left eigenvector of the minimum eigenvalue of the matrix $\mathbf{P}$.

V. Performance Analysis

To assess the performance of the voice network under our proposed cooperative protocol, four measures will be considered: network throughput, multiple access delay, packet dropping probability and speech quality.

A. Network Throughput

The throughput can be defined as the aggregate average amount of data transported through the channel in a unit time. In our case, the number of packets successfully transmitted in a given frame can be decomposed into two components linked by the tradeoff between the use of cooperation and reduction in the number of contention slots. The first one comes from the contending users who succeed in gaining access to the channel. And the second component comes from the talking users who succeed in transmitting their packets to the base station, either by themselves or through the help of the relay. Thus, the throughput can be expressed as

$$Th = \frac{E\{E\{m_{CT}|S_1\} + M_T - E\{m_{TC}|S_1\}\}}{N}, \tag{25}$$

where $E\{\cdot\}$ is the expectation operator. The term $E\{m_{CT}|S_1\}$ corresponds to the successful contention, whereas the number of successfully transmitted packets is expressed as $(M_T - E\{m_{TC}|S_1\})$, the number of users in TLK state minus the expected number of users leaving the TLK state to the CON state, which are the users with failures in their transmissions. Finally, the outermost expectation is with respect to the stationary distribution of the system's Markov chain.

B. Multiple Access Delay

The delay is the number of frames a user remains in the CON state before gaining access to the channel. This delay is a function of the probability with which a user succeeds in contending during a given frame. This success probability depends on the network state at the instant the user enters the CON state, and will differ from frame to frame according to the path the network follows in the state space. Therefore, for exact evaluation of the multiple access delay, one should condition on the state at which our user of interest enters the CON state for the first time. Starting from this state, the delay is obtained from the calculation of the statistics of all possible paths the network follows in the state space till the user succeeds in the contention process. It is possible to show that for a network with N time slots per frame and M_v users, the total number of states is given by $(M_v - N/2+1)(N+1)$ for $M_v \geq N$. For a network with $M_v = N = 10$, the number of states is 66. With such large number of states, finding an exact expression of the multiple access delay becomes prohibitively complex. To get an approximate expression for the delay, we will assume that when the user enters the contention state the system state will not change until that user succeeds in contention. Thus, the success probability will be constant throughout the whole contention process, and the delay at any given state will follow a geometric distribution with parameter $p_s(i)$, the success probability at any state i. The approximate average delay is given by

$$D_{avg} = \sum_{i \in \Omega} \frac{\pi(i)}{p_s(i)}, \tag{26}$$

where Ω is the set of states where $M_C \neq 0$ and $\pi(i)$ is the ith element of the stationary distribution vector π.

The last step is to calculate the success probability $p_s(i)$. Given the assumption that all users are statistically identical, the probability that a user succeeds during contention in a given frame is equal to the probability that at least one user succeeds during contention in that frame, which can be easily computed using the recursion of (19).

C. Packet Dropping Probability

Speech communication is delay sensitive and requires prompt delivery of speech packets. Because of this, in the proposed scheme packets start to be dropped if they are delayed in the network for more than a maximum allowable delay of D_{max} frames. Based on the assumption that the speech coder generates exactly one speech packet per frame, every user will maintain a buffer of length D_{max}. Whenever the buffer is full at the start of a frame, the oldest packet is dropped until the user succeeds in reserving a time slot. If the talk spurt ends before getting a slot reservation, all the packets in the buffer are dropped. Because of channel errors, a user with a reserved time slot may lose its reservation and return to the group of contending users, thus risking further packet dropping.

To analyze the packet dropping probability, we adopt the method developed in [13] and [14] for the analysis of the PRMA protocol. First, we will consider the case when a user is trying to access the channel for the first time. Given that the system is at state i with M_C contending user and M_T users holding slot reservations, consider a contending user whose talk spurt started at the current frame. The talk spurt consists of L packets, where L is a random variable. The user will start to contend for a time slot in the current frame and continue in subsequent frames until it succeeds or the talk spurt ends. The user waits in the CON state for D frames to obtain reservation. Using the assumption developed in section V-B that delay D is geometrically distributed, the probability that a user waits for d frames given the system is in state i is

$$P_D(d,i) = (1-P_s(i))(P_s(i))^d, \quad d = 0, 1, \ldots \tag{27}$$

We need to distinguish between two different cases relating the length of the talk spurt L and the maximum allowable delay D_{max}. Also we should note the assumption that when a terminal transits to the silence state all remaining packets in the buffer are dropped.

1) $L \leq D_{max}$: In this case, the buffer is long enough to store the whole talk spurt. If reservation is obtained before the talk spurt ends, j packets are lost if the transition from TLK to SIL occurred after the $(L-j)th$ transmission which has a probability of $\gamma(1-\gamma)^{L-j}$. Otherwise, all the talk spurt packets are discarded. As a function of the waiting time d, the number of dropped packets is

$$n_d(d) = \begin{cases} j, & 0 \leq d < L \\ L, & d \geq L \end{cases} \tag{28}$$

and the distribution of the number of dropped packets is given by

$$\Pr\{n_d | L \leq D_{max}, i\} = \begin{cases} \gamma(1-\gamma)^{L-j} \sum_{d=0}^{L} P_D(d,i), & n_d = 0 \\ \sum_{d=L+1}^{\infty} P_D(d,i), & n_d = L \end{cases} \tag{29}$$

2) $L > D_{max}$: In this case, after waiting D_{max} frames, one packet is dropped per frame until being able to reserve a slot. The dropped packet is the oldest in the queue with an associated delay of D_{max}. The number of dropped packets as a function of the delay is given by

$$n_d(d) = \begin{cases} j, & 0 \leq d \leq D_{max} - 1 \\ k, & d = D_{max} + k - 1, \\ & k = 1, 2, \ldots, (L - D_{max}) \\ L, & d \geq L \end{cases} \tag{30}$$

and its distribution

$$\Pr\{n_d | L > D_{max}, i\} = \begin{cases} \gamma(1-\gamma)^{L-j} \sum_{d=0}^{D_{max}-1} P_D(d,i), & n_d = 0 \\ (1-P_s(i))(P_s(i))^{n_d}, & n_d = 1, 2, \ldots, (L - D_{max}) \\ \sum_{d=L}^{\infty} P_D(d,i), & n_d = L \end{cases} \tag{31}$$

We note here that although all the summations mentioned above have closed form expressions, they tend to become complex and lengthy. Therefore, we avoid writing them here, so as to keep the presentation compact.

The expected number of dropped packets for the above two cases, namely $E\{n_d | L \leq D_{max}, i\}$ and $E\{n_d | L > D_{max}, i\}$, can be easily calculated using the corresponding distributions and, then, combined to get the total expected number of dropped packets as

$$E\{n_d | i\} = \sum_{l=1}^{D_{max}} E\{n_d | L \leq D_{max}, i\} P_L(l) + \sum_{l=D_{max}+1}^{\infty} E\{n_d | L > D_{max}, i\} P_L(l), \tag{32}$$

where $P_L(l)$ is the probability mass function of the length of the talk spurt. From the speech source model of Fig. 2, the talk spurt duration, L, is geometrically distributed with parameter γ, i.e.,

$$P_L(l) = \gamma(1-\gamma)^{l-1}, \quad l = 1, 2, \ldots \tag{33}$$

Finally, the packet dropping probability is the ratio between the average number of dropped packets per talk spurt to the average number of packets generated per talk spurt, i.e,

$$P_{d_0} = \frac{1}{\gamma} \sum_{i \in \Omega} E\{n_d | i\} \pi(i), \tag{34}$$

where the sum is over Ω, the set of states with $M_C \neq 0$ (because packets are dropped only when the user is in the CON state).

Next we consider the packet dropping probability due to the first transition from the TLK state to the CON state, which is caused by channel errors. First, we need to make the following assumptions:

- Any user in TLK state has obtained its reservation with the first packet in the talk spurt. This means no packets were dropped in the first contention process. Furthermore, this packet is delayed by $D_0 = D_{avg}$ frames, i.e., this packet is delayed by the average multiple access delay calculated in the last section.
- The first channel error occurs while transmitting the jth packet of the talk spurt. Since the first packet was delayed by D_0 frames, the remaining maximum delay for the subsequent packets in the talk spurt is $D_1 = D_{max} - D_0$ frames.
- There are L packets in the talk spurt, and L_1 packets following and including the jth packet which encountered a channel error.

Based on the time instant when the user left TLK state to CON state, we need to analyze three cases:

1) Transmission instant of the jth packet is after the end of the talk spurt. This means, $D_0 + (j-1) \geq L$, or $L_1 \leq D_0$. In this case all the remaining L_1 packets are discarded without any contentions and $E\{n_d | L_1 \leq D_0, i\} = L_1$.
2) Transmission instant of the jth packet is before the end of the talk spurt and the remaining time till the end of the talk spurt is less than the maximum remaining delay D_1. That is, $0 < L - D_0 - (j-1) \leq D_1$, or $D_0 < L_1 \leq D_{max}$. In this case, no packets are dropped if the user gets a reserved slot before the end of the talk spurt. Otherwise, all L_1 are discarded.
3) $L - D_0 - (j-1) > D_1$, or $L_1 > D_{max}$. In this case, the jth packet is dropped after waiting for D_1 frames and a packet will be dropped every frame till the user gets access to the channel. If the talk spurt ends before accessing the channel, all the packets in the buffer are discarded.

In each one of these cases, we calculate the distribution of the number of dropped packets, from with we get the expected number of dropped packets. The detailed derivations follow directly form the analysis provided in [13] and [14], and will be omitted for the sake of compactness.

D. Speech quality measure

We will base the voice quality assessment of our protocol on the predictive model developed in [15]. This model uses source codec parameters, end-to-end delay and packet dropping probability to predict the value of the conversational Mean Opinion Score (MOS_c) [16], a perceptual voice quality measure based on the ITU-T PESQ quality measure standard [17, p. 862] that takes values in the range from 1 (bad quality) to 5 (excellent quality). For the GSM AMR 12.2 kbps voice codec [18], the (MOS_c) can be estimated using [15] as,

$$\begin{aligned} MOS_c = {} & 3.91 - 0.17P_d + 1.57 \cdot 10^{-3}D + 6.51 \cdot 10^{-3}P_d^2 \\ & - 2.40 \cdot 10^{-5}D^2 - 7.53 \cdot 10^{-6}P_dD - 10^{-4}P_d^3 \\ & + 2.62 \cdot 10^{-8}D^3 + 1.38 \cdot 10^{-7}P_dD^2 \\ & - 5.51 \cdot 10^{-8}P_d^2D, \end{aligned} \tag{35}$$

where P_d is the packet dropping probability and D is the average delay we calculated earlier.

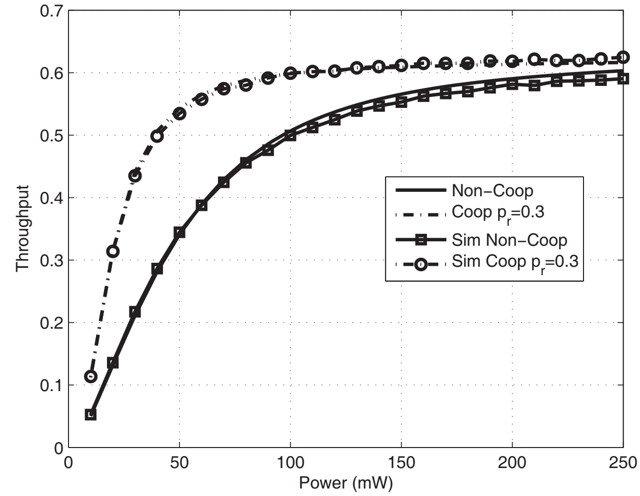

Fig. 5. Network performance measures for 15 users and transmission power varying from 10mW to 250mW: throughput.

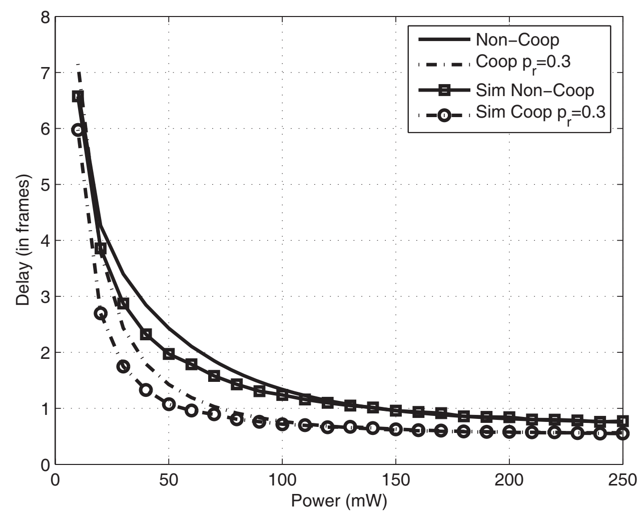

Fig. 6. Network performance measures for 15 users and transmission power varying from 10mW to 250mW: approximate delay of (26).

VI. Numerical Results and Discussions

We compare the performance of our cooperative multiple access protocol and the PRMA protocol without cooperation. The parameters settings are as follows. The speech source model has a mean talk spurt and a mean silence period duration of $t_1 = 1$ and $t_2 = 1.35$ seconds, respectively. The

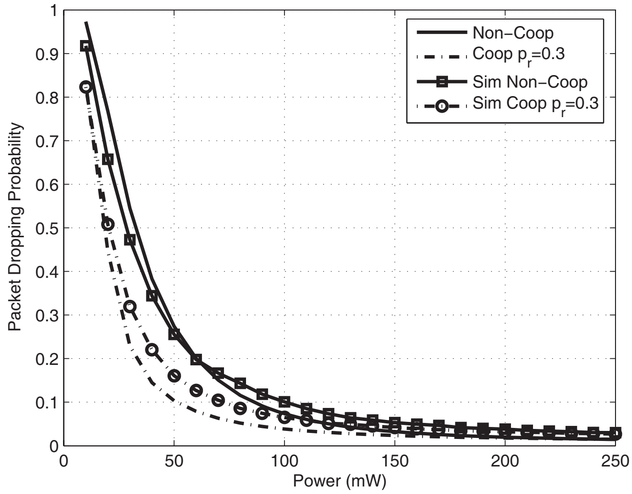

Fig. 7. Network performance measures for 15 users and transmission power varying from 10mW to 250mW: Packet dropping probability.

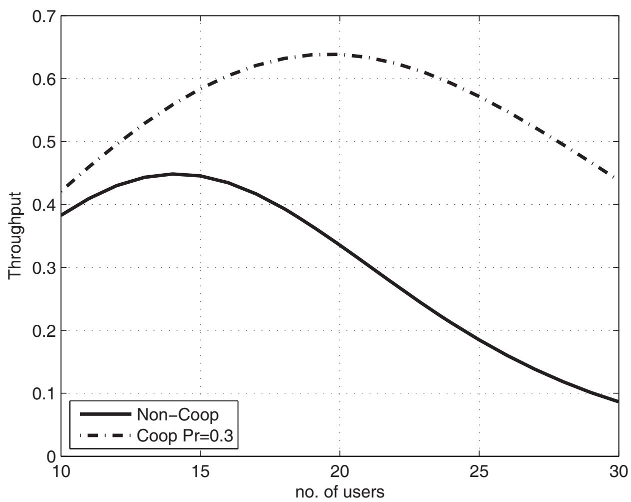

Fig. 8. Throughput vs. number of users.

speech encoder has a 12.2 kbps data rate, and we assume each packet carries 114 bits of speech data as in the GSM system. Therefore, for each use to send a single packet per frame, frame duration is 9.35 ms. The maximum allowable delay is 20 ms, i.e., $D_{max} = 2$ frames, this value of 20 ms is chosen based on the acceptable delay for conversational interactive speech. For this setup it is possible to accept delays of up to 100-150 ms [19]. However, there are other sources of delay such as coding delay (typically 20 ms), network delay, delay at other transcoders in the network, echo cancelers, etc.; so a value of 20 ms is a safe choice to ensure good end-to-end delay behavior. Each frame is divided into $N = 10$ time slots, contention permission probability $p_v = 0.3$, SNR threshold $\beta = 15$ dB and path loss exponent $\alpha = 3.7$. The distance between any user and the base station is 100 m, between any user and the relay is 50 m, and between the relay and base station is 100 m

Figures 5-7 show the different performance measures vs. transmit power for a fixed number of users $M_v = 15$. It is noted from the figures that there is a good match between the analytical and simulation results, which validates our derived analytical expressions. Fig. 5 depicts the gain in throughput due to cooperations. For example, at a low power level of 50 mW, the non-cooperative throughput is around 0.35 while the cooperative throughput with $p_r = 0.3$ is around 0.53, which amounts to an 80% increase. The gains in delay and packet dropping probability are depicted in Figures 6 and 7, respectively, where again we can see significant decrease in delay and packet dropping probability in the low power region. It is noted that increasing the power decreases cooperation gains, which is due to the fact that at low power levels the performance is limited by outage events, which is where the relay plays a role in reducing the probability of such events. On the other hand, at high power levels outage probability is low and the performance is limited by packet collisions.

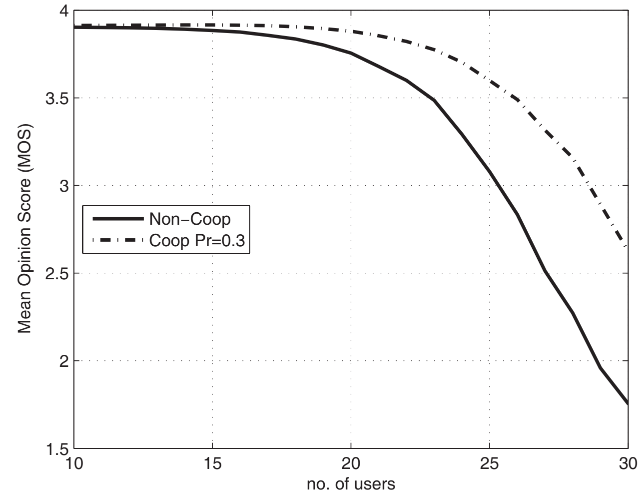

Fig. 9. Speech quality vs. number of users.

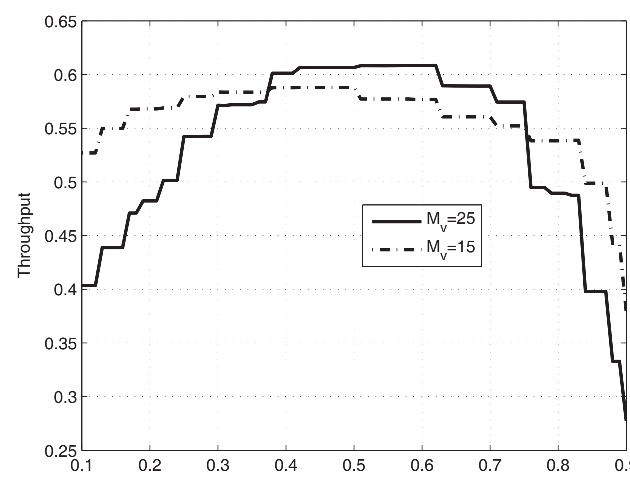

Fig. 10. Throughput as a function of p_r for 75mW transmission power.

For 75 mW transmission power, Fig. 8 shows the throughput against number of users. A significant gain is achieved in throughput and in the number of users maximizing this throughput. We see a 45 % increase in throughput and the number of users maximizing the throughput increase from 14 to 20 users. But in a speech network the maximum number of supported users should be defined by the speech quality

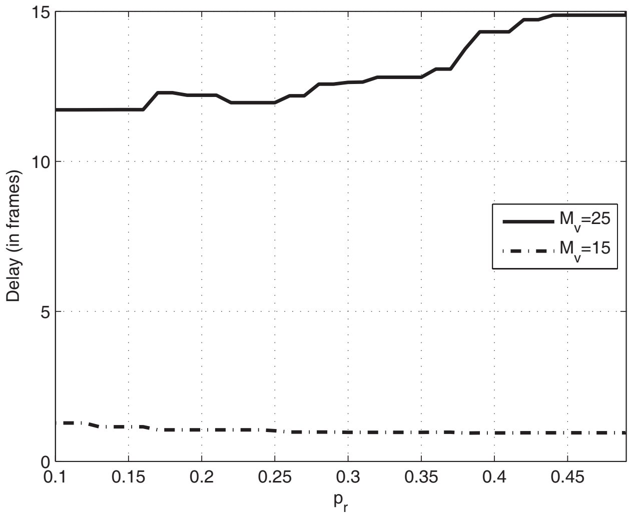

Fig. 11. Delay as a function of p_r for 75mW transmission power .

and not only the network throughput. Fig. 9 depicts the mean opinion score (MOS) speech quality measure against the number of users. At MOS of 3.5 for example, which is an acceptable quality, we see that our protocol increases the number of supported users from 23 to 26, or a 13 % increase in the number of users.

To study the effect of the amount of resources assigned to the relay, Fig. 10 depicts the throughput as a function of p_r for the case of a congested network ($M_v = 25$) and a network with moderate number of users ($M_v = 15$). It is noted that throughput is initially increasing with p_r, since increasing the p_r increases the relay's ability to help more users combat channel fading, hence decreasing outage probability and increasing the average number of successful packets per frames. Then throughput starts to decrease as p_r increases because of the network's inability to accept new users since the relay is occupying a larger portion of the contention slots, thus leading in a reduction in the average number of TLK users and a reduction in throughput. Delay performance as a function of p_r is shown in Fig. 11. While for a moderately loaded network the delay decreases with increasing p_r (up to the value of $p_r = 0.5$ after which delay increases dramatically), a congested network suffers from an increase in delay, which is associated with increased packet dropping probability and decreased speech quality. This is mainly due to the reduction in the number of contention slots in favor of the relay. This effect appears in the congested network only because of the larger average number of contending users compared to the moderately loaded case. Therefore, the introduction of cooperation introduces a tradeoff between the amount of help provided by the relay, and the network's ability to serve users starting a talk spurt. From Figures 10 and 11 we can see that by assigning about 30% to 50% of the free resources to the relay good throughput performance is achieved while the delay is kept at an acceptable level.

It should be noted that in this work (due to space limitations) we assumed perfect feedback channel. However, situations may arise where the feedback message could take longer then expected or it is received in error by the relay or network users. Since it is through feedback messages that users and relay determine the state of each slot in the upcoming frame and whether a packet needs retransmission or not, delays or errors in these messages lead to ambiguity in the state of different time slots and packet transmissions. One possible solution to deal with such imperfect feedback channel is to make different nodes take different actions in response to lost or delayed feedaback. For instance, a delayed or lost feedback after a packet transmission by a user with reserved slot can be considered by that user and the relay as a NACK message, thus this packet will be considered for help by the relay or retransmission by the user in the next frame, while for other users this should be considered as an ACK message so the time slot involved will still be considered as reserved in the next frame and no collisions occur with the original user's transmissions. For a contention slot, the situation should be different. All users shall assume the delayed or lost feedback as a NACK message, so no slot reservation in made and any user involved in the contention process will retry in the next time slot.

VII. Conclusions

In this paper, we have proposed a new cognitive multiple access protocol for cooperative packet speech communications over wireless networks. Through cooperation, the proposed protocol addresses one important limitation of PRMA-like protocols, namely that wireless channel errors leads to active calls loosing their channel reservation, which leads to an increase in medium access contention and a reduction in system capacity. Cooperation is implemented through a relay that efficiently helps active calls by using resources released by those users undergoing a period of silence in the speech conversations. Because the network resources vacated by users in a period of silence are also used for medium access contention by those calls starting a talk spurt, there is a tradeoff between the resources used by the relay to help other calls and the network's ability to serve users at the start of a talk spurt. As studied, this tradeoff influences the protocol performance and is an important design parameter.

In order to characterize the performance of the proposed protocol, we developed and analyzed a Markov model describing the network dynamics in the presence of a relay node. We studied the throughput, delay performance, packet dropping probability and perceptual speech quality of the proposed protocol and compared it to the non-cooperative PRMA protocol. Our results show around 80% increase in throughput and a significant decrease in delay in the low SNR regime. The decrease in delay is translated into around 50% decrease in packet dropping probability, which in turn is translated into an improved speech quality. Furthermore, we studied the tradeoff between resources allocated to the relay and to medium access contention, and we showed that the protocol parameters can be chosen considering this tradeoff so that the proposed cooperative protocol outperforms the non-cooperative in all performance measures considered.

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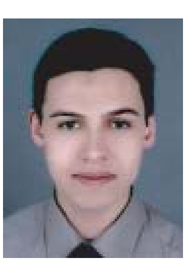

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