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DYNAMIC VOLTAGE AND FREQUENCY SCALING FOR ENERGY MINIMIZATION IN MANET

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ABSTRACT: **That have defined the way Information Technology based services can be presented. It has transformed the model of storing and managing data for scalable, real time, internet based applications and resources satisfying end users' needs. Additional and new remote crowd equipment is built for obscure services causing extra power dissipation and energy consumption. Over the decades, power consumption has become an important cost factor for computing resources. Here will consider all achievable areas in a characteristic cloud communications that are responsible for considerable capacity of energy consumption and we will address the methodologies by which power utilization can be decreased without compromising Quality of Services (QoS) and overall performance. We also plan to define the scope for further extension of research from the findings. The simulations and statistical results prove that our proposed cross-layer resource allocation can powerfully maintain diverse QoS requirements over wireless relay networks. The scheduling algorithm at the medium access control (MAC) layer for multiple connections with diverse QoS requirements, where each connection employs adaptive modulation and coding (AMC) scheme at the physical (PHY) layer over wireless fading channels. Each connection is assigned a priority, which is updated dynamically based on its channel and service status; the connection with the highest priority is scheduled each time. The wireless bandwidth efficiently, and enjoys flexibility, scalability, and low implementation complexity.**

I. INTRODUCTION

The field of wireless communication is becoming more popular than ever before due to the rapid advancement of wireless technologies and the wide spread of mobile devices. After a natural disaster, such as a fire, flood, or earthquake, Mobile Ad hoc Networks (MANET) are among the limited available options for wireless networks since such a network can be easily configured in a short period of time without the need for a fixed infrastructure network. A Mobile Ad hoc Network (MANET) is a self-organized network with arbitrary distributed nodes. Furthermore, a MANET has the potential to work alongside different networks such as cellular networks and the Internet. Indeed, the set of applications for MANETs is diverse. These applications are not limited to areas such as emergency and crisis management, local-level, commercial and military battlefield applications [1]. Aside from the aforementioned advantages of MANET, the mobile nature of the nodes in such a network imposes a battery lifetime limitation. Furthermore, these nodes may be equipped with different transmission technologies [2] even though the transmission medium is shared. As a result, nodes are competing to send their data, and they must wait a random amount of time. This is known as the Broadcast Storm Problem (BSP).

As a consequence, routing in MANET has become the main challenging issue because a network with a changeable topology leads to frequent path failure. Most available routing protocols have been categorized into three types: proactive, reactive and hybrid. Undoubtedly, a MANET uses reactive routing protocols, which are more practical for such a network due to the low routing overhead that they produce and the low power resources that they need. Recently, several routing protocols have been used in MANET, including the Ad hoc On-demand Distance Vector (AODV) protocol, Dynamic Source Routing (DSR) [6], Location Aided Routing (LAR) [7].

II. EXISTING SYSTEM

The WMRN where a base station (BS) with K first-in first-out (FIFO) data queues transmit to K corresponding users with the aid of an AaF1 relay. In our cross-layer scheduling policy to be described in Section III), a single user with the largest weighted SNR is scheduled for transmission in each scheduling opportunity. We assume independent non identically distributed block

Rayleigh fading in the two hop relay links with a coherence time of Tc seconds.

A. MAC LAYER ARCHITECTURE

The BS has K finite queues with buffer size B, each corresponding to a distinct user. A user's packet is lost if the buffer for the queue is full and a new packet arrives. The arrival process of the packets for each queue is assumed to be a homogeneous Poisson process with rate λk, k = 1. . . K, where each k corresponds to a different queue. The probability that n packets arrive in an interval of time T for the k-th user is then given by

$$\Pr(N_k(T) = n) = \frac{e^{-\lambda_k T}(\lambda_k T)^n}{n!}.$$

Packets can be re-requested with the caveat that the arrival of the re-requested packet is consistent with the Poisson arrival process. The transmission time *T* is the same for all users. Prior to Section VI, we assume that the queues are backlogged such that at least one packet is always available. As a result, the BS is never silent. This assumption is also made in. We relax this restriction in Section VI where we derive the PMF of the buffer state and the PLP due to buffer overflow.

B. PHYSICAL LAYER ARCHITECTURE

The BS and the relay each transmit for T/2 seconds in half duplex mode such that the total transmission time from the BS to the scheduled user is T seconds, where T ≤ Tc. The transmission time is chosen such that the BS has knowledge of both the BS-relay and relay-user links for scheduling purposes. In the BS-relay link, the received signal at the relay is given by

$$y_R = \sqrt{E_S} h_{SR} x + z_R,$$

where ES is the transmit power at the source, hSR is the Rayleigh fading channel coefficient between the source and the relay, x is the transmitted symbol using binary phaseshift keying (BPSK), quadrature phase-shift keying (QPSK) or M-ary pulse amplitude modulation (M-PAM), and zR is the additive white Gaussian noise (AWGN) with one-sided power spectral density N0.

In the relay-to-user link, the received signal at the scheduled user, denoted by k=∈ {1, . . .,K}, is given by

$$\beta = \sqrt{\frac{1}{E_{SR}|h_{SR}|^2 + cN_0}}.$$

Set *c* = 1 for the case where noise power is included in the relay amplification factor and we set *c* = 0 for the case where the noise power is ignored. The end-to-end SNR of the scheduled user is written as

$$\gamma_{eq} = \frac{\gamma_{SR}\gamma_{Rk^*}}{\gamma_{SR} + \gamma_{Rk^*} + c},$$

where *γSR* is the instantaneous SNR in the source-to-relay link and *γRk∗* is the instantaneous SNR in the relay-touser link. We incorporate the effect of path loss into the instantaneous SNRs such that *γSR* = *d −η S ES*|*hSR*|2/*N*0 and *γRk∗* = *d −η R ER*|*hRk∗* |2/*N*0, where *dS* is the distance between the source and the relay, *dR* is the distance between the relay and the scheduled user, and *η* is the path loss exponent. As both the BS-relay and relay-to-user links experience Rayleigh fading,

C. PROPOSED CROSS-LAYER SCHEDULING POLICY

The scheduling policy selects the user with the largest weighted SNR of the second hop. The weight is a function of the DPS. packet can only be scheduled at the front of a user's queue. As a result, only delays of the packets at the front of each user's queue are required for our scheduler's computations. The header size of each packet can then be significantly reduced in long queues compared with the scheme in as time stamps with a small number of bits are sufficient. The reduction is due to the impact of the large variation in total packet delay on the scheme in caused by the dependence on the number of packets in the queue when the packet arrives. While our approach is heuristic, it is practical and efficient. Moreover, our policy does have a theoretical underpinning as we can guarantee the average SEP and PLP due to buffer overflow. Since the modulation scheme is fixed as described in, the instantaneous SEP only varies when the instantaneous SNR varies. As a result, only the instantaneous SNR is required in the scheduling policy to account for the SEP requirements.

The proposed scheduling policy is given by $k* = \arg\max_{k=1,...,K} \gamma R_k W_k$,

where

(i) $k*$ is the scheduled user;

(ii) γR_k are the instantaneous SNRs of the relay-to-user links, where $k = 1, \ldots, K$;

(iii) W_k is the weight satisfying $W_k(s_k) \geq 0$, $k = 1, \ldots, K$

where $W_k(s_k)$ is an arbitrary function of s_k that is the number of potential scheduling opportunities that have elapsed for the k-th user's packet. We do not account for the BS-relay link in the scheduling policy as it is the same for all users. Users in multimedia applications are often characterized as real time (RT) or best effort (BE). Our scheduling policy can accommodate both types of users by adapting the weight functions W_k. In particular, the weights of the BE users are constant $W_k = 1$, while the RT users weights are functions of the DPS.

D. NORMALIZED SERVICE RATE

First derive the average normalized service rate for the k-th user, i.e., the probability that the k-th user is scheduled. Denote $P_k(\mathbf{s})$ as the normalized service rate when the users' queue states are the elements of the state vector $\mathbf{s} = [s_1 \ldots s_K]^T$, where each s_k, $k = 1, \ldots, K$ denotes the number of scheduling opportunities that the packet for user k has been waiting at the front of the queue. The normalized service rate for user k in state $\mathbf{s}$, $P_k(\mathbf{s})$,

E. DELAY IN PACKET SCHEDULING

Next, we derive the statistics of the DPS. We require the probability that the current state vector is s. Denote s(n) as the state vector after n transmission slots. The state vectors then form a Markov chain as $\Pr(s(n)|s(1), \ldots, s(n-1)) = \Pr(s(n)|s(n-1))$. We note that the transition probability from state $s(n-1)$ to state $s(n)$ when user k is scheduled is given by $P_k(s(n-1))$. Hence, the scheduler forms a K-dimensional Markov chain with a countably infinite state space. In general, the required eigen value equation is intractable and it is not possible to obtain closed form expressions. the steady state characteristics by truncating the Markov chain and forming a 1-dimensional Markov chain with an augmented transition matrix. This technique for approximating the *K*-dimensional Markov chain is known as generating the augmented Markov chain. It has been well-studied and used in several applications such as.That the approximation is accurate. The approximation proceeds as follows:

1) Determine the required maximum DPS for each user to achieve a given accuracy of the approximation. Denote the largest of these as d.

2) Enumerate in lexicographic order all possible state vectors with integer elements greater than or equal to one, with each element less than or equal to d.

3) Let V be the set of states that contain a single element $\mathbf{s}_i^k = 1$, where $\mathbf{s}_i^k$ is the k-th element of the i-th state vector in the lexicographic enumeration. We note that the set V can be written as $V = \{\mathbf{s}_i | \exists$ a unique $k \in \{1, 2, \ldots, K\}$ such that $\mathbf{s}_i^k = 1\}$. We then define S as $S = \{\mathbf{s}_j \in V \,|\mathbf{s}_j^m = \mathbf{s}_i^m + 1 \vee \mathbf{s}_j^m = \mathbf{s}_i^m = d, \forall m \_= k\}$,

where i is the index of the enumerated state for the current state vector and j is the index of the enumerated state for the future state vector. We then construct the transition probability matrix $\mathbf{P}$ as $p_{ij} =\_ P_k(\mathbf{s}_i),\ \mathbf{s}_j \in S, 0$, where p_{ij} is the (i, j)-th element of $\mathbf{P}$.

4) Adjust p_{i1} such that$_ j\ p_{ij} = 1$ for all i. This ensures that $\mathbf{P}$ is a stochastic matrix. Note that for sufficiently large d, this adjustment is small.

F. SYMBOL ERROR PROBABILITY

The SEP of the scheduled user for different modulation formats can be evaluated according to PS = a 2_bπ_ ∞ 0 γ −1 2 e –bγ Fγeq (γ) dγ. The constants a and b are modulation-specific with a = 1, b = 1 for BPSK, a = 1, b = 0.5 for QPSK, and a = 2(M − 1)/M, b = 3/_M2 − 1_ for M-PAM. We note that is absolutely convergent. As such, we can swap the sum in Theorem 1 and the integral in applying the dominated convergence theorem. This ensures that the infinite sum converges. The integral can then be evaluated efficiently using numerical integration, leading to reduced evaluation time compared with Monte Carlo simulation.

G. PACKET LOSS PERFORMANCE

In this section, we analyze the PLP of each queue using the proposed scheduling policy. This is achieved by constructing a new Markov chain for the buffer states for each queue with transition probabilities dependent on the scheduling policy, arrival rate, and transmission time.

H. BUFFER STATE

We first obtain the PMF of the buffer state that gives the probability that the buffer has l, $0 \leq l \leq B$ packets. We note that the buffer state is measured at the beginning of a scheduling slot, after a packet is scheduled in the current slot, and before new arrivals. This is important as the time when the buffer state is measured affects the PMF of the buffer state and subsequently the PLP. We also note that the buffer state is independent of the DPS.

III. PROPOSED SYSTEM

Traditional schedulers for wire line networks only consider traffic and queuing status; however, channel capacity in wireless networks is time varying due to multipath fading and Doppler effects. Even if large bandwidth is allocated to a certain connection, the prescribed delay or throughput performance may not be satisfied, and the allocated bandwidth is wasted when the wireless channel experiences deep fades. An overview of scheduling techniques for wireless networking can be found in where a number of desirable features have been summarized, and many classes of schedulers have been compared on the basis of these features. To schedule wireless resources (such as bandwidth and power) efficiently for diverse QoS guarantees, the interactive queuing behaviour induced by heterogeneous traffic as well as the dynamic variation of wireless channel should be considered in scheduler design

Figure 1. Network topology

A. NETWORK CONFIGURATION

Figure 1 illustrates the wireless network topology under consideration. Multiple subscriber stations (SS) are connected to the base station (BS) or relay station over wireless channels, where multiple connections (sessions, flows) can be supported by each SS.

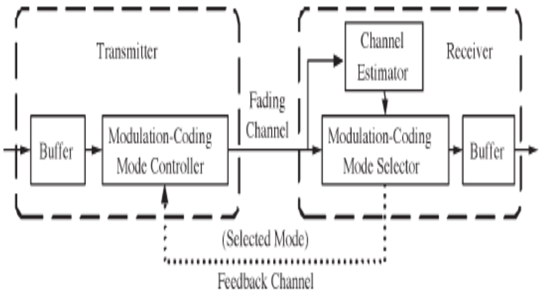

Figure 2. Wireless links from BS to SS.

This kind of star topology is not only applicable to cellular networks but is also used to describe the connections between each relay station and multiple SS in mobile ad hoc networks and wireless sensor networks. All connections communicate with the BS using time division multiplexing/time-division multiple access (TDM/ TDMA). We will focus on the downlink here, although our results can be extended to the uplink as well. The wireless link of each connection from the BS to each SS is depicted in Fig. 2. A buffer is implemented at the BS for each connection and operates in a first-input-first-output (FIFO) mode. The AMC controller follows the buffer at the BS (transmitter), and the AMC selector is implemented at the SS (receiver). At the PHY, multiple transmission modes are available to each user, with each mode representing a pair of a specific modulation format and a forward error control (FEC) code, as in IEEE 802.11/15/16, 3GPP, and 3GPP2 standards. Based on channel estimates obtained at the receiver, the AMC selector determines the modulation-coding pair (mode or burst profile), whose index is sent back to the transmitter through a feedback channel, for the AMC controller to update the transmission mode. Coherent demodulation and soft-decision Viterbi decoding are employed at the receiver. The decoded bit streams are mapped to packets, which are pushed upward to the MAC.

B. THE PACKET AND FRAME STRUCTURES.

1) At the MAC, each packet contains a fixed number of bits Nb, which include packet header, payload, and cyclic redundancy check (CRC) bits. After modulation and coding with mode n of rate Rn as in Table I, each packet is mapped to a symbol block containing Nb/Rn symbols.

2) At the PHY, the data are transmitted frame by frame through the wireless channel, with each frame containing a fixed number of symbols Ns. Given a fixed symbol rate, the frame duration Tf (in seconds) is constant and represents the time unit throughout this paper. With TDM, each frame is divided into $Nc + Nd$ time slots, where for convenience we let each time slot contain a fixed number of $2Nb/R1$ symbols. As a result, each time slot can transmit exactly $2Rn/R1$ packets with transmission mode n. For the TM in particular, one time slot can accommodate $2R1/R1 = 2$ packets with mode $n = 1$, $2R2/R1 = 3$ packets with mode $n = 2$, and so on. The Nc time slots contain control information and pilots. The Nd time slots convey data, which are scheduled to different connections dynamically. Each connection is allocated a certain number of time slots during each frame.

C. QOS ARCHITECTURE AT THE MAC

At the MAC, each connection belongs to a single service class and is associated with a set of QoS parameters that quantify its characteristics. Four QoS classes are provided by the MAC in the IEEE 802.16 standard. 1) Unsolicited grant service (UGS) supports constant bit rate (CBR) or fixed throughput connections such as E1/T1 lines and voice over IP (VoIP). This service provides guarantees on throughput, latency, and jitter to the necessary levels as TDM services. The QoS metrics here are the packet error rate (PER) and the service rate. 2) Real-time polling service (rtPS) provides guarantees on throughput and latency, but with greater tolerance on latency relative to UGS, e.g., MPEG video conferencing and video streaming. The delayed packets are useless and will be dropped. The QoS metrics are the PER and the maximum delay (or the maximum delay for a given outage probability). 3) Nonreal-time polling service (nrtPS) provides guarantees in terms of throughput only and is therefore suitable for mission critical data applications, such as File Transfer Protocol (FTP). These applications are time-insensitive and require minimum throughput. For example, an FTP file can be downloaded within a bounded waiting time if the minimum reserved rate is guaranteed. The QoS metrics are the PER and the minimum reserved rate. 4) Best effort (BE) service provides no guarantees on delay or throughput and is used for Hypertext Transport Protocol (HTTP) and electronic mail (e-mail), for example. BE applications receive the residual bandwidth after the bandwidth is allocated to the connections of the previous three service classes.

D. TERMINATION PHASE

During the termination phase, nodes clear entries in their connectivity tables that are no longer needed either because a connection was terminated or because it was interrupted due to a link failure or was handed–off to a new path. When a session is completed the source node which initiated the call tears it down by generating a TRM control message and broadcasting it along the session path to the DEST (see example in figure 1(k-l)). Another situation where a TRM packet is needed is when a link experiences a hard failure.

E. PERFORMANCE ANALYSIS

We have simulated the system in NS2 simulator, a discrete–event simulator with the required features in modelling a distributed algorithm. In this section we highlight the main properties of the simulation model and present some initial simulation results.

F. MOBILE CALL MODEL

New call arrivals to the network are assumed to be Poisson with rate _ = 1 request/min. The call holding time is exponentially distributed with mean _ = 30 sec. Each new call arrival is equally likely to arrive at any node as its source, and any of the remaining nodes is equally likely to be its destination.

G. CONNECTIVITY MODEL

To capture node mobility and physical layer impairment, we employ a model of fixed-node topology but with dynamic link status. In particular, the source and destination of each call remain fixed as long as the call is in progress. Instead, the status of each link is assumed to be subject to dynamic (perhaps random) changes throughout the call duration. A simple three–state probabilistic link status model is assumed. In particular, the possible states of the link status are “FULL”, “HALF”, or “ZERO”, corresponding to the status of the link being able to support transmissions at the full rate, half rate, or link out-of-service, respectively. Specifically, the transition from FULL to HALF state models a soft link failure, which parallels a link quality degradation situation, whereas the FULL to ZERO state transition corresponds to a hard link failure implying complete loss of connectivity. This model allows a unified treatment of node mobility and physical impairment in a flexible fashion.

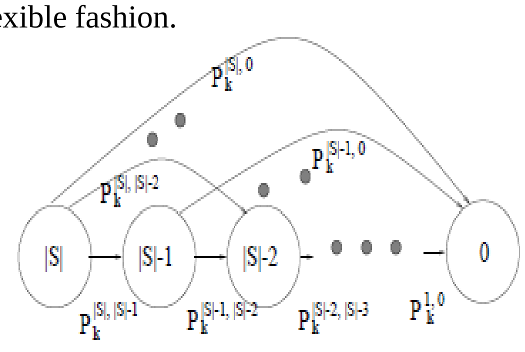

Figure 3. The Markov Chain

We run simulations for different values of average holding times for each state and steady state distribution. The steady state distribution gives an indication of how much time on average each link spends at each case, while the holding times determine the average.

IV. NETWORK SIMULATOR

A network simulator is a software program that imitates the working of a computer network. In simulators, the computer network is typically modelled with devices, traffic etc and the performance is analyzed. Typically, users can then customize the simulator to fulfil their specific analysis needs. Simulators typically come with support for the most popular protocols in the use today, such as Wireless LAN, Wi-Max, UDP, and TCP. A network simulator is a piece of software or hardware that predicts the behaviour of a network, without an actual network being present. NS is an object oriented simulator, written in C++, with an OTcl interpreter as a frontend.

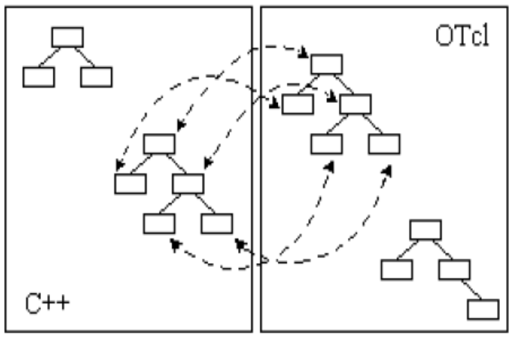

Figure 6. Flow chart for C++ and OTcl

The simulator supports a class hierarchy in C++ and a similar class hierarchy within the OTcl interpreter. The two hierarchies are closely related to each other; from the users perspective, there is one-to-one correspondence between a class in the interpreted hierarchy and one in the compiled hierarchy. The root of this hierarchy is the class Tcl object. Users create a new simulator objects through the interpreter; these objects are instantiated within the hierarchy. The interpreted class hierarchy is automatically established through methods defined in the class Tcl object. There are other hierarchies in the C++ code and OTcl scripts;

these other hierarchies are not mirrored in the manner of Tcl object.

A. USES OF NETWORK SIMULATORS

Network simulators serve a variety of needs. Compared to the cost and time involved in setting up an entire test bed containing multiple networked computers, routers and data links, network simulators are relatively fast and inexpensive. They allow engineers to test scenarios that might be particularly difficult or expensive to emulate using real hardware- for instance, simulating the effects of sudden bursts in the traffic or a Dos attack on a network service. Networking simulators are particularly useful in allowing designers to test new networking protocols or changed to existing protocols in a controlled and reproducible environment. Typical network simulators encompasses a wide range of networking technologies and help the users to build complex networks from basic building blocks like variety of nodes and links. With the help of simulators one can design hierarchical networks using various types of nodes like computers, hubs, bridges, routers, optical cross-connects, multicast routers, mobile units, etc. various types of Wide Area Network (WAN) technologies like TCP, ATM, IP etc and Local Area Network (LAN) technologies like Ethernet, token rings etc, can all be simulated with the typical simulator and the user can test, analyze various routing etc.

B. SIMULATION AND RESULTS

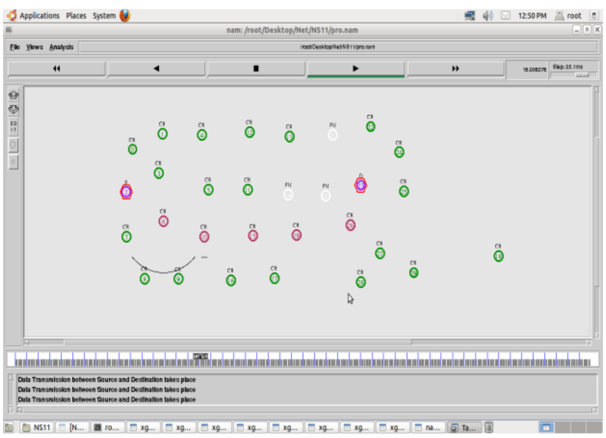

Figure 4. Communicate Different Types of Cluster Head

To calculate the SEP of the scheduled user, we require the cumulative distribution function (CDF) of the end-to-end SNR. As the CDF is dependent on the state vector **s**, we first calculate the conditional CDF. The expression for the conditional CDF of the SNR of the relay-user link of the scheduled user $k*$ is given in Lemma 2. To simplify the notation, we write $\gamma 2 = \gamma Rk*$ for the SNR of the scheduled user's relay-to-user link.

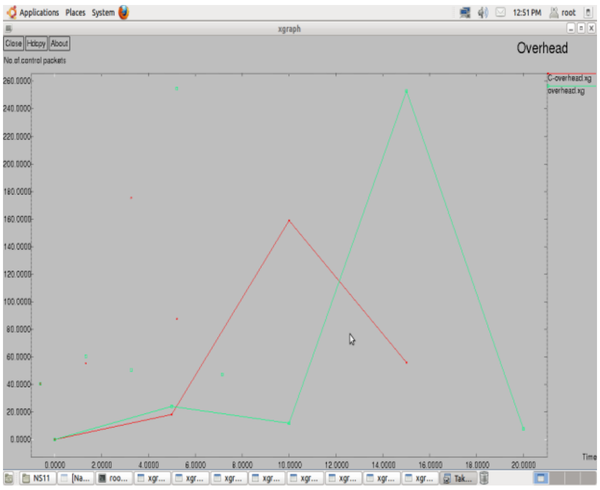

Figure 5. SEP schedule the overhead value

In Figure 5 the SEP of the scheduled user at $\gamma 1 = 10$ dB is compared with the scheduling policy index in (25) for $K = 3$. We observe that an increase in the scheduling policy index causes an increase in the SEP of the scheduled user. Moreover, we observe that the SEP for the adaptive policies of $j \leq 6$ is less than 4% greater than the SEP for the equal weight policy of $j = 1$.

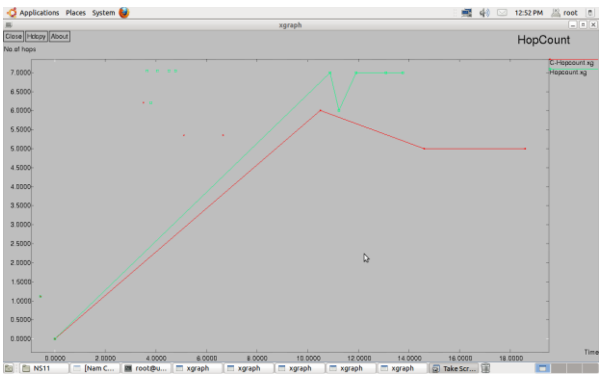

Figure 6. Various Network Levels for Probability of hopcount

In Figure 6. the probability that the DPS target of Pr $(s1 \leq 3)$ is satisfied for the RT user is plotted against the scheduling policy index. We point out the accuracy of the approximation in (15) by noting that the simulations curves are consistent with the analytical curves. The figure shows that the probability of satisfying the hop count target is greater than 90% for $j \geq 5$.

V. CONCLUSION AND FUTURE DIRECTIONS

Developed a cross-layer scheduling algorithm at the MAC layer for multiple connections with diverse QoS requirements, which can be used in cellular networks, mobile ad hoc networks, and wireless sensor networks. Each connection admitted in the system is assigned a priority, which is updated dynamically depending on its channel quality, QoS satisfaction, and service priority; thus, the connection with the highest priority is scheduled first each time. Our proposed scheduler offers prescribed delay, and rate guarantees for rea ltime and non real-time traffic; at the same time, it uses the wireless bandwidth efficiently by exploiting multiuser diversity among connections with different kinds of services.

Furthermore, our scheduler enjoys flexibility, scalability, and low implementation complexity. Performance of our scheduler was evaluated via simulations in the IEEE 802.16 standard setting, where the upper-bound βrtPS, βnrtPS, βBE, and the delay guard time ΔTi were set heuristically. Their effects on performance are worthy of further research. Furthermore, our scheduler allocates all Nr time slots to one connection each time for simplicity; however, scheduling multiple connections each time may lead to better performance, which is under current investigation. The fairness issue for the users in the same service class is another topic in our research agenda. The effects of imperfect channel state information due to estimation error and feedback latency.

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