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PERFORMANCE EVALUATION OF DATA OFFLOADING APPROACHES FOR MOBILE SOCIAL NETWORKS

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Abstract: **Mobile networks are in front of strict traffic overloads suitable to the propagation of tidy handheld devices and transfer ravenous applications. This tries to offload the cellular network traffic through Opportunistic Communications and Social Participation formed by the short-range communication technologies in the smart phones (e.g., Wi-Fi, Bluetooth). We plan an improved Greedy algorithm used for object locate choice Problem. Our design enables to select the most active, fixed located and having more energy mobile nodes into the Target-set, which affect the more number of infected users. Our simulation grades demonstrate to facilitate quantity of node enclosed by improved Greedy algorithm is 89% and Greedy algorithm is 76%. We are able to achieve the rate of success information delivery over the networks in improved Greedy and Greedy algorithms are 71% and 53% respectively. The simulations and numerical results verify that our proposed cross-layer resource allocation can efficiently support diverse QoS requirements over wireless relay networks.the scheduling algorithm at the medium access control (MAC) layer for multiple connections with diverse QoS requirements.**

I. INTRODUCTION

In recent years, the demand for mobile broadband services in cellular networks has increased rapidly, especially in video streaming and content sharing. The EU FP7 project METIS has provided several key quantitative performance indicators for 5G networks, 1,000 times higher mobile data volume per area and 10 Gbps peak data rate are included. In addition, the new wireless broadband communication services, including e-banking, e-health and e-learning will be integrated in future everyday life. Therefore, the future 5G network should be designed towards a highly integrated system to meet the predicted data traffic growth and these various requirements. Not only traditional network optimization technologies should be utilized, such as interference management and cooperative communication, but also new upcoming solutions will be of great importance, e.g., network densification, user-behaviour study cognitive networks. Software-defined networking (SDN), and intelligent wireless backhauling. Caching at the network edge has become an important means of offloading the traffic and tackling the backhaul bottleneck in order to reduce the latency of services and the cost of the cellular network. In single-tier networks it has been shown that the backhaul capacity and the size of cache have a significant impact on the energy efficiency. In heterogeneous networks, much work has been done on content caching using various algorithms and schemes. A framework to model heterogeneous cellular networks has been built with the aid of a factor graph, where distributed caching optimization algorithms are designed and compared in. In a coded caching scheme is proposed to enable content pre-fetching prior to knowing user demands based on the YouTube dataset. Proactively pushes popular content to the relays and users with caching ability via broadcasting. However, the power consumption and the backhaul limitation of the network are not negligible for future 5G.

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,

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.

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 Pk(**s**) as the normalized service rate when the users' queue states are the elements of the state vector **s** = [s1. . . sK]T , where each sk, k = 1, . . . , 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 **s**, Pk(**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), . . . , 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 Pk(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.

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 corresponding to user *k*. To calculate the PMF of the buffer state, we require the average probability that user *k* is scheduled, which is given by *Pk* =∞_*sn*=1 *n*=1,...,*K Pk*(**s**)*π***s**

I. PACKET LOSS PROBABILITY

The PLP is the probability that a packet is lost due to buffer overflow. Before evaluating the PLP for a given packet, The PLP for each user can now be obtained for a given buffer size by considering the probability that the buffer is full at time $0 < t < T$ after a scheduling opportunity. Here, *t* is the the time of the new packet arrival. Theorem 2 gives an approximation of the PLP. The approximation arises due to dependence on the stationary distribution and is exact when the scheduling policy weights are fixed constants.

The PLP approximation shows the clear dependence on the the transmission time and arrival rates for the user under consideration. Intuitively, if the arrival rate is high or the transmission time long, the PLP due to buffer overflow is large. We will see in Section VII-C that a consequence of this is that additional redundancy through channel coding does not always improve the throughput.

We note that the expected total packet delay can be obtained via Little's law from the buffer state distribution and the PLP. In particular, we have

$$E[Wk] = Lk/\lambda e,k,$$

J. TRANSMISSION TIME

That increasing the transmission time impacts on the PLP. To determine the optimal transmission time, the effect of channel coding must be accounted for. Of course, when employing coding, a longer transmission time is required to account for the redundancy in the signal. To examine the trade off between the coded SEP of the scheduled user and the PLP for each queue we consider the throughput given by

Throughput = $(1 - P_{E,coded})(1 - P_{L,ave})$,

where $P_{E,coded}$ is the coded SEP of the scheduled user and $P_{L,ave}$ is the average PLP over all queues. The average PLP Throughput $R-1 = T \times T$ unc-1 Tunc= 2 ms, Tunc= 1 ms Tunc=0.5 ms

The throughput of an equivalent single user network versus the inverse of the code rate $R-1$ for varying uncoded transmission times T_{unc}, arrival rate $\lambda k = 0.1$ ms-1, $k = 1, 2, 3$, and scheduling policy ($W1 = e0.2s1$, $W2 = e0.1s2$, $W3 = 1$) over all the queues is given by

$$P_{L,ave} = 1\ K\_K\ i=1\ P_{L,i}.$$

The throughput expression in approximates the WMRN as a single point-to-point link using a single queue with a PLP given by the average over all queues. As a result, gives a simple characterization of a WMRN as transmission times are varied. The throughput is compared to the inverse of the code rate $R-1$, for varying uncoded transmission time T_{unc}. Here $T = T_{unc}R-1$, where R is the normalized rate.

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 heterogenous traffic as well as the dynamic variation of wireless channel should be considered in scheduler design

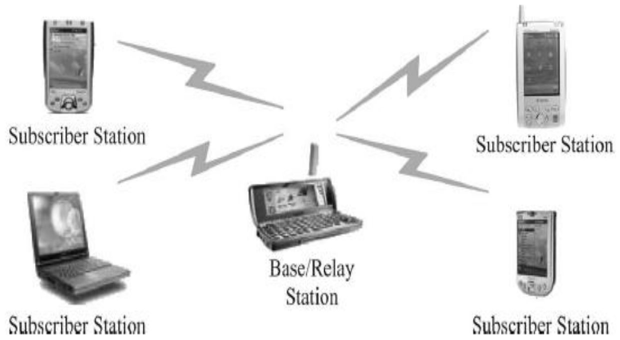

Figure 1 Network topology

A. NETWORK CONFIGURATION

Figure 2.1 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.

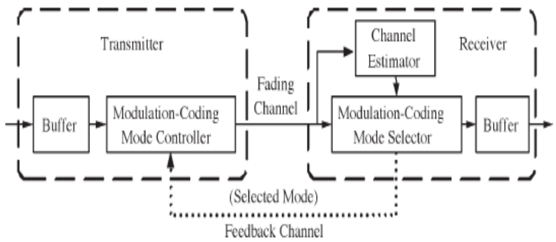

Figure 2 Wireless links from BS to SS.

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.

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 2*Nb/R*1 symbols. As a result, each time slot can transmit exactly 2*Rn/R*1 packets with transmission mode *n*. For the TM in particular, one time slot can accommodate 2*R*1*/R*1 = 2 packets with mode *n* = 1, 2*R*2*/R*1 = 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.

D. AMC DESIGN AT THE PHY

Efficient bandwidth utilization for a prescribed PER performance at the PHY can be accomplished with AMC schemes, which match transmission parameters to the time-varying wireless channel conditions adaptively and have been adopted by many standard wireless networks, such as IEEE 802.11/15/16 and 3GPP/3GPP2. Each connection with rtPS, nrtPS, and BE services relies on AMC at the PHY. The objective of AMC is to maximize the data rate by adjusting transmission modes to channel variations while maintaining a prescribed PER P0, and the design procedure is similar to that proposed.

Let N denote the total number of transmission modes available (N = 6 for TM). As in we assume constant power transmission and partition the entire signal-to-noise ratio (SNR) range in N + 1 nonoverlapping consecutive intervals, with boundary points denoted as $\{\gamma n\}N+1\ n=0$. In this case mode n is chosen when $\gamma \in [\gamma n, \gamma n+1)$, for n = 1, . . . , N.

E. ROUTER STRUCTURE

Every node has a fixed number of transmitter-receiver pairs (transceivers) used to set up communication links with other nodes. The number of transceivers that are in an IDLE state, varies dynamically with time depending on the traffic load and the average session duration. A necessary condition for a new connection to be established is that at least one transceiver is available at every node in the path to the destination. New connection requests are blocked when one or more nodes along the path do not have any of their transceivers idle. A pure first-come first-serve policy is assumed without considering any preemptive policies in which a high priority session can preempt an ongoing session of lower priority. For simplicity no priority is given to hand-off requests which have to compete against new connection requests in search of communication paths.

F. ALGORITHM EXECUTION

Algorithm execution can be viewed as occurring in three logical phases, the "Construction phase", the "Maintenance phase" and the "Termination Phase", which execute simultaneously in a dynamic topology. A source node which desires a connection to the DEST transmits a "Connection-Request" (CR) packet along one of the existing DN links. If multiple DN links exist a decision over which link to transmit is made either upon information on the resources available along the existing paths or randomly, if no such information has been obtained. In particular the parameter for the selection of the DN link is the available number of transceivers along the outgoing paths and such information is collected during the algorithm construction phase by messages piggybacked in the transmitted acknowledgments.

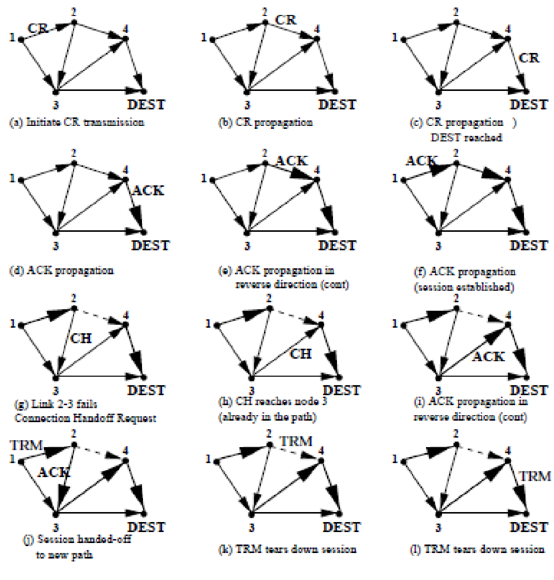

Figure 3. Example of algorithm execution

If a CR reaches the DEST and the request is admitted, the destination node updates the corresponding entry in its connectivity table and transmits backwards to the source an "Acknowledgment" (ACK) control message (node DEST in the example of figure 1(d)). Otherwise if the request cannot be serviced by the DEST , a NAK is transmitted back to the link over which the CR was received. ACK messages are generated and transmitted by destination nodes, are destined to the source node of the CR and must follow the same path of the CR in the reverse direction.

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 modeled with devices, traffic etc and the performance is analyzed. Typically, users can then customize the simulator to fulfill 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.

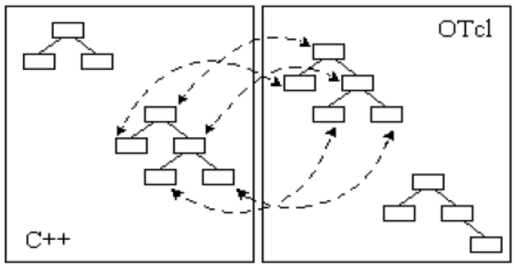

Figure 4. Cooperation of C++ and Otcl language

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 user"s 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. NETWORK SIMULATOR 2 (NS2)

NS2 is an open- source simulation tool that runs on Linux. It is a discreet event simulator targeted at networking research and provides substantial support for simulation of routing, multicast protocols and IP protocols, such as UDP, TCP over wired and wireless (local and satellite) networks. It has many advantages that make it useful tool, such as support for multiple protocols and the capability of graphically detailing network traffic. Additionally, NS2 supports several algorithms in routing and queuing. Queuing algorithms include fair queuing, deficit round-robin and FIFO. REAL is a network simulator originally intended for studying the dynamic behavior of flow and congestion control schemes in packet switched data network. NS2 is available on several platforms such as FreeBSD, Linux, Sim OS and Solaris. NS2 also builds and runs under Windows.

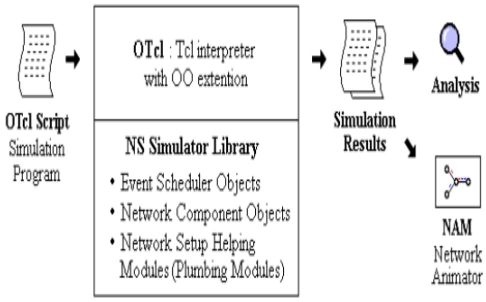

Figure 5.Simplified user's view of NS

B. NODE BASICS

The basic primitive for creating a node is
set ns[new Simulator]
$ns node

The instance procedure node constructs a node out of simpler classifier objects. All nodes contain the following components:

1. An address or id_, monotonically increasing by 1 (from initial value o) across the simulation namespace as nodes are created
2. A list of neighbors (neighbor_)
3. A list of agents (agent_)
4. A node type identifier (node type_)

C. NODE METHODS: CONFIGURING THE NODE

Procedures to configure an individual node can be classified into:

1. Control functions
2. Address and Port number management, unicast routing functions
3. Agent management
4. Adding neighbours.

V. SIMULATION RESULTS

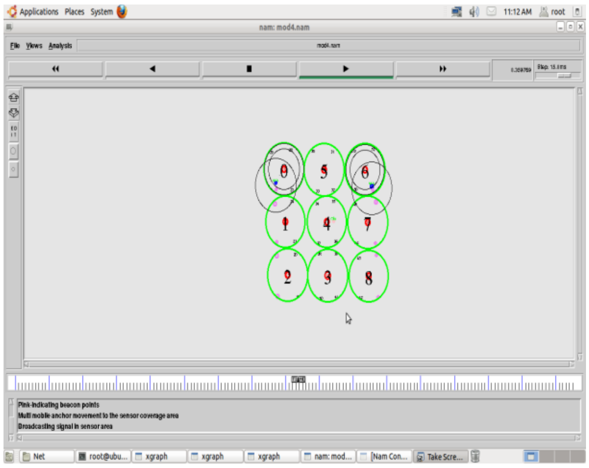

Figure 6. Communicate Different Types of Cluster Head

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.

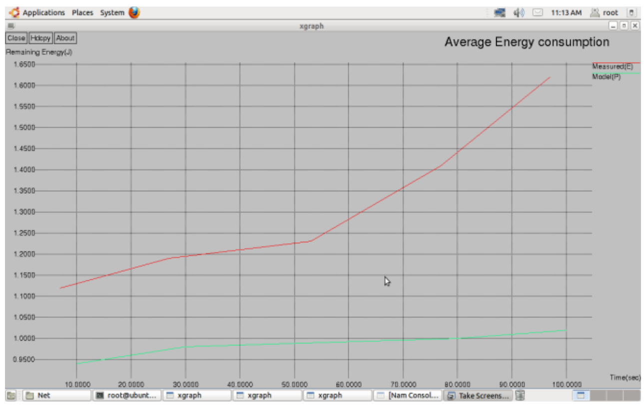

Figure 7 Comparison between and proposed and existing system in Energy value

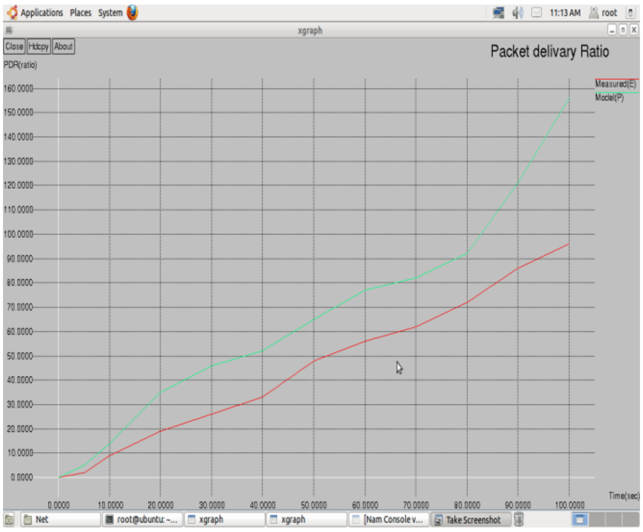

Figure 8 Comparison between and proposed and existing system in PDR

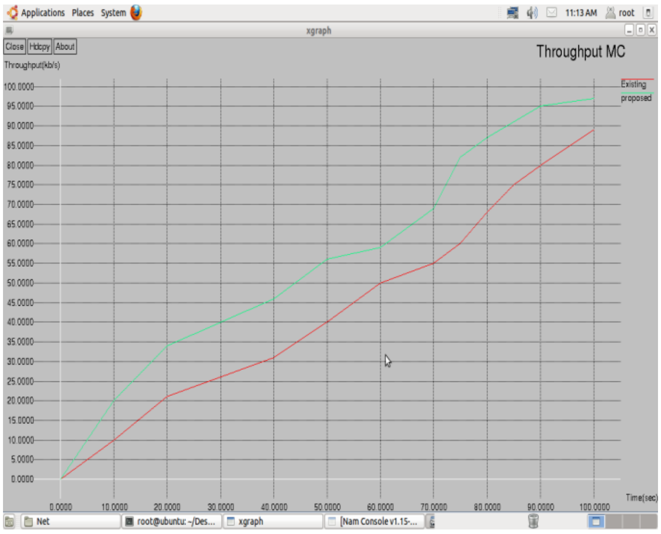

Figure 9 Comparison between and proposed and existing system in Throughput

VI. CONCLUSION AND FUTURE DIRECTIONS

Developed a cross-layer scheduling algorithm at the MAC layer for multiple connections with different QoS needs, which can be used in mobile 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 execution difficulty. 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.

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