International Research Journal of Engineering and Technology (IRJET) e-ISSN: 2395-0056
Volume: 06 Issue: 03 | Mar 2019 www.irjet.net p-ISSN: 2395-0072

AGGREGATE SIGNATURE SCHEME AND SECURED ID FOR WIRELESS SENSOR NETWORKS

Sathishkumar. S[1], Raghul. S[2], Siva. A[3], Vinoth kumar. V[4]

[1]Assistant Professor, Department of IT, Adhiyamaan College of Engineering (Autonomous), Hosur.
[2,3,4]UG Student Department of IT, Adhiyamaan College of Engineering (Autonomous), Hosur.

Abstract - *A physical-data link cross-layer resource allocation scheme over wireless relay networks for quality-of service (QoS) guarantees. By integrating information theory with the concept of effective capacity, our proposed scheme aims at maximizing the relay network throughput subject to a given delay QoS constraint. This delay constraint is characterized by the so called QoS exponent θ, which is the only requested information exchanged between the physical layer and the data link layer in our cross-layer design based scheme. Over both amplify-and forward (AF) and decode-and-forward (DF) relay networks, we develop the associated dynamic resource allocation algorithms for wireless multimedia communications. Over DF relay network, a fixed power allocation scheme to provide QoS guarantees. 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, 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.*

1. INTRODUCTION

The explosive developments of wireless communications, quality-of-service (QoS) provisioning have become a critically important performance metric for the future wireless networks. Unlike wire line networks, in which QoS can be guaranteed by independent optimization within each layer in the open system interconnection (OSI) model, over wireless networks there is a strong interconnection between layers, which makes the layered design and optimization approach less efficient. the physical layer, a great deal of research focuses on techniques that can enhance the spectral efficiency of wireless systems. The framework used to evaluate these techniques is mainly based on information theory, using the concept of Shannon capacity. However, it is well known that Shannon capacity does not place any restrictions on complexity and delay. As a result, the optimization merely at the physical layer may not lead to the desired delay QoS requested by the services at upper protocol layers.

To deal with this problem, there have been increasing interests in design for wireless networks that relay on interactions between various layers of the protocol stack. This approach, called cross-layer design and optimization, has been widely recognized as a promising solution to provide diverse QoS provisioning in wireless multimedia communications. The cross-layer approach relaxes the layering architecture of the conventional network model, which can result in a significant performance enhancement.

However, such a design principle across different layers usually involves high complexity, which may cause the optimization problem intractable. Consequently, how to develop efficient cross-layer approaches while minimizing the additional requested information exchanged between layers is an important issue from both theoretical and practical point-of-views.

On the other hand, relay communications have recently emerged as a powerful spatial diversity technique that can improve the performance over conventional point-to-point transmissions. The original work on relay communications was initiated by Cover and Gamal. Since then, it has been extensively studied using different performance metrics especially when the concept of user cooperation was proposed. Clearly, combining the idea of cross-layer design with the relay network architecture, it is possible to significantly improve the system QoS provisioning performance.

However, the research on how to efficiently employ the unique nature of relay architecture for designing the cross-layer protocols, and what is the impact of cross-layer resource allocation on supporting diverse QoS requirements over wireless relay networks, are still quite scarce. To remedy the above deficiency, in this paper propose a cross-layer resource allocation scheme for relay networks with the target at delay QoS guarantees for wireless multimedia communications. Our proposed scheme aims at maximizing the relay network throughput subject to a given delay QoS constraint. Our work builds on the integration of information theoretic results with the theory of statistical QoS guarantees, in particular, the recently developed powerful concept termed effective capacity.

The theory of statistical QoS guarantees has been extensively studied in the early 90' with the emphasis on wired asynchronous transfer mode (ATM) networks. This theory enables us to analyze network statistics such as queue distributions, buffer overflow probabilities, and delay-bound violation probabilities, which are all important delay QoS metrics.

As a part of the statistical QoS theory, effective capacity is particularly convenient for analyzing the statistical QoS performance of wireless multimedia transmissions where the service process is driven by the time-varying wireless channel. Specifically, our resource allocation scheme is across the physical and the data link layers. Applying the effective- focus on simple half-duplex relay protocols, namely, amplify-and-forward (AF) and decode-and forward (DF), and develop the associated dynamic resource allocation algorithms, where the resource allocation policies are functions of both the network channel state information (CSI) and the QoS constraint θ. The resulting resource allocation policy in turn provides a guideline on how to design the relay protocol that can efficiently support stringent QoS constraints. For DF relay networks, we also study a fixed power allocation scheme and investigate its performance. The simulations and numerical results verify that our proposed cross-layer resource allocation can efficiently support diverse QoS requirements over wireless relay networks. Moreover, both AF and DF relays show significant superiorities over direct transmissions when the delay constraints are stringent. On the other hand, our results demonstrate the importance of deploying the dynamic resource.

2. LITERATURE REVIEW

"Cross-Layer Resource Allocation over Wireless Relay Networks for Quality of Service Provisioning, Malcolm Egan, Phee Lep Yeoh, et al, (2013)",

New scheduler for wireless multimedia relay networks (WMRNs). Our scheduler is designed to account for delay, symbol error probability (SEP), and packet loss probability (PLP) due to buffer overflow. We develop a cross layer scheduling approach for the downlink to balance these system metrics. Our scheduler is based on a new metric which is referred to as the delay in packet scheduling (DPS). The user with the largest weighted signal-to-noise ratio is scheduled, where the weight is a function of the DPS. We then derive analytical expressions for the probability mass function (PMF) of the DPS, and the SEP of the scheduled user in Rayleigh fading. We derive an analytical approximation for the PMF of the buffer state. An analytical expression is then derived for the PLP due to buffer overflow.

"A Cross-Layer Scheduling Algorithm with QoS Support in Wireless Networks, Qingwen Liu, Xin Wang, et all (2009)"

Scheduling plays an important role in providing quality of service (QoS) support to multimedia communications in various kinds of wireless networks, including cellular networks, mobile ad hoc networks, and wireless sensor networks. The authors propose a 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 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.

"Cross-Layer Radio Resource Allocation: the Journey so Far and the Road Ahead, Virginia Corvino, Lorenza Giupponi, Ana Perez Neira, et all, (2012)"

The cross-layer concept originated almost ten years ago with the aim of taking the most of the advantage from the difficult wireless media to break the barriers imposed by the layered transmission. One of the domains where cross-layer design has been more investigated is Radio Resource Allocation, since current and future networks need to provide wireless connectivity to heterogeneous users, offering many different data traffic types. Nonetheless, new paradigms are emerging in the field of wireless communications, like cognitive radios, wireless systems with relays and Multiple Input Multiple Output (MIMO) systems, where the potential advantages of cross-layer scheduling are still largely unknown. Moreover, in spite of a large literature on cross-layer, in the most cases different focuses and perspectives, biased by the application(s), are addressed, thus, there is lack of a general framework.

3. 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.

3.1 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.

3.2 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,

3.3 DELAY PERFORMANCE

3.3.1 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**),

3.3.2 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 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 = {**s**i/ a unique k {1, 2, . . .,K} such that **s**i k = 1}. We then define S as S = {**sj** V /**sj** m = **s**i m + 1 **sj** m = **s**i m = d, 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 **P** as pij =_ Pk(**s**i), **sj** S,0, where pij is the (i, j)-th element of **P**.

4) Adjust pi1 such that_ j pij = 1 for all i. This ensures that **P** is a stochastic matrix. Note that for sufficiently large d, this adjustment is small.

2.3.3 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.

3.4 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.

3.4.1 BUFFER STATE

We first obtain the PMF of the buffer state that gives the probability that the buffer has l, $0 \le l \le B$ 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

$$Pk = \infty\_sn=1\ n=1,...,K\ Pk(\mathbf{s})\pi\mathbf{s}$$

where Pk(**s**) is the probability user k is scheduled in DPS state **s** and π**s** is the probability that the DPS state vector is **s**. An accurate approximation of can be obtained using the stationary distribution arising from the truncated multidimensional Markov chain given in. To obtain the PMF of the buffer state, we require the stationary distribution of the associated Markov chain. Since the buffer state of each user is only dependent on the individual user's statistics, the Markov chain is one-dimensional and the stationary distribution can be obtained by explicitly constructing the transition matrix. In particular, the construction of the transition matrix for user k, **T**k , tij is the (i, j)-th element of **T**k where $1 \le i, j \le B+1$. The (i, j)-th element represents the transition from the buffer storing $i-1$ packets to the buffer storing $j-1$ packets. We note that since the buffer is finite, the corresponding irreducible and ergodic Markov chain is also finite. As a result, it has a unique stationary distribution. In order to analyze the PMF of the buffer state, we assume that it is possible for a user to be scheduled without a packet. This is necessary as the weight in our scheduling policy does not account for the buffer state.

3.4.2 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 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 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

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

Where $E[Wk]$ is the expected total packet delay for user k, Lk is the expected number of packets in the queue for user k, which can be obtained using and $\lambda e,k$ is the effective arrival rate for user k given by

$$\lambda e,k = \lambda k(1 - PL,k),$$

Where λk is the actual packet arrival rate and PL,k is the PLP of the user given by Surprisingly, this means that we can quantify the expected total packet delay, despite only exploiting the DPS in our scheme. In the next section, we use numerical and simulation results to evaluate the effect of scheduling policy and examine the accuracy of the approximation when the weights are not fixed.

3.4.3 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 = $(1 - PE,coded)(1 - PL,ave)$, where $PE,coded$ is the coded SEP of the scheduled user and PL,ave is the average PLP over all queues. The average PLP Throughput R−1 = T×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 $Tunc$, arrival rate λk = 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

$$PL,ave = 1\ K\_K\ i=1\ PL,i.$$

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 $Tunc$. Here $T = TuncR-1$, where R is the normalized rate of the coded signal and the rate of the uncoded signal is $R = 1$. The data is coded using punctured convolutional codes. The distances and transmit powers are $dS = dR = 1$ and $ES = ER = 1$. In the simulation, the buffer size is $B = 20$, the arrival rate is λk = 0.1 ms−1, k = 1, 2, 3, the number of transmitted QPSK data symbols is 100, and the scheduling policy is $W1(s1) = e0.2s1$, $W2(s2) = e0.1s2$, $W3 = 1$.We observe in the figure that the throughput increases with increasing transmission time, T, when $Tunc$ = 0.5 ms. In contrast, the throughput decreases with increasing T when $Tunc$ = 2ms. Of particular interest is the scenario where $Tunc$ is between 0.5 ms and 2 ms. From the figure we see that when $Tunc$ = 1 ms, the throughput does not vary monotonically with T . This suggests that an efficient tradeoff between the PLP and the coded SEP of the scheduled user.

4. 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

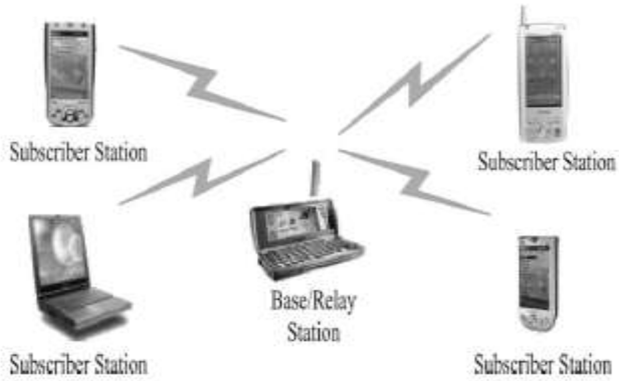

Figure 4.1 Network topology

4.1 SYSTEM ARCHITECTURE

4.1.1 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.

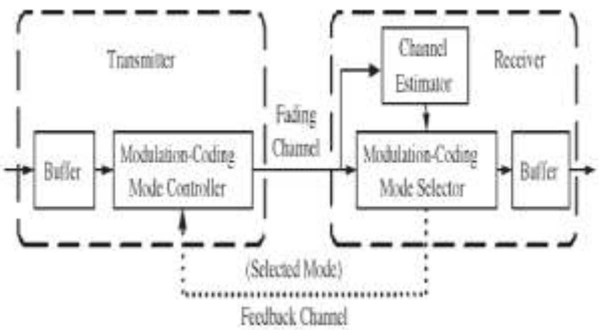

Figure 4.2 Wireless links from BS to SS.

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. We consider the following group of transmission modes as in the IEEE 802.16 standard. Transmission modes (TM): The modulations are Mn-ary rectangular/square quadrature amplitude modulators (QAMs), and the FEC codes are Reed–Solomon (RS) concatenated with convolutional codes (CC) (see Table I). Although we focus on this TM, other transmission modes can be similarly constructed. At the PHY, the processing unit is a frame consisting of multiple transmitted symbols. At the MAC, the processing unit is a packet comprising multiple information bits.

4.1.2 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.

4.1.3 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) Non real-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.

Although no delay and rate is specified for BE connections, a prescribed PER should be maintained over wireless channels. The signalling and procedure for the service setup and maintenance of each connection are defined as in the IEEE 802.16 standard. However, the standard does not define the scheduling mechanism or the admission control and traffic policing processes. The signalling overhead is not included in our design and analysis.

5. SIMULATION RESULTS

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.

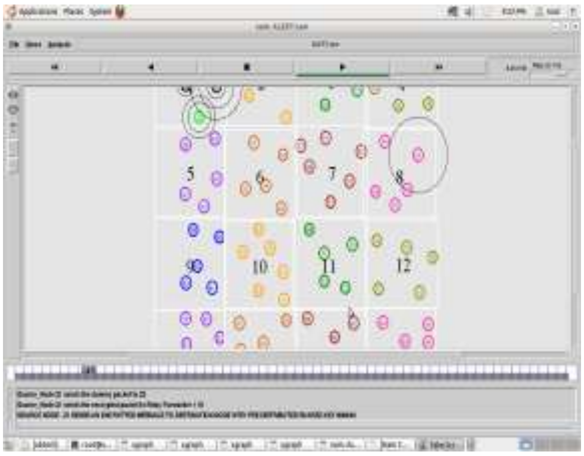

Figure 5.1 Communicate Different Types of Cluster Head

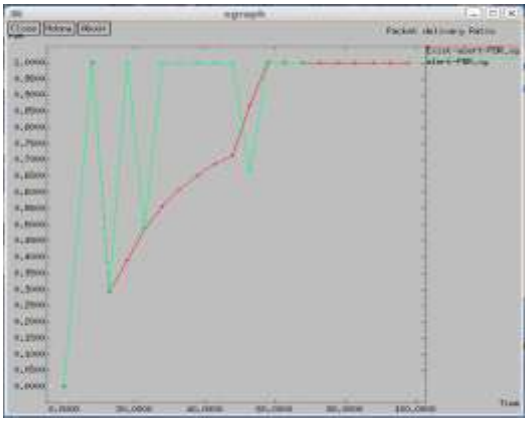

Figure 5.2 Packet Delivery Ratio

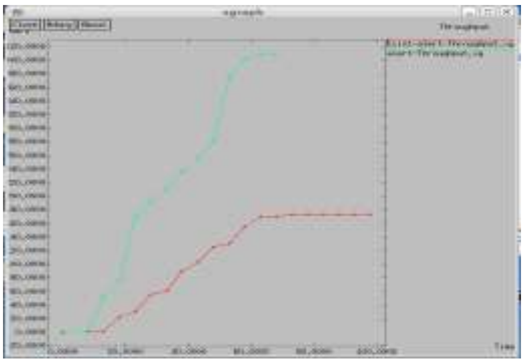

Figure 5.3 Throughput

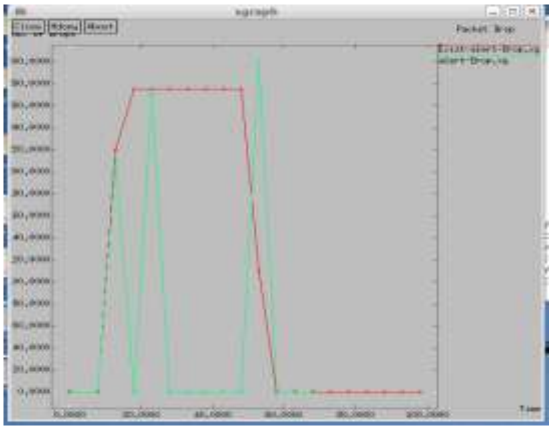

Figure 5.4 Packet Drop

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.

6. CONCLUSION AND FUTURE DIRECTIONS

A cross-layer scheduling algorithm is developed 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 real time 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.

REFERENCES

1. I. Paik, T. Tanaka, H. Ohashi and W. Chen, "Big Data Infrastructure for Active Situation Awareness on Social Network Services," *Big Data (BigData Congress), 2013 IEEE International Congress on.* IEEE, pp. 411-412, 2013.
2. E. Hargittai, "Is Bigger Always Better? Potential Biases of Big Data Derived from Social Network Sites," *Annals of the American Academy of Political & Social Science,* vol. 659, no. 1, pp. 63-76, 2015.
3. Z. Fu, X. Sun, Q. Liu, L. Zhou, J. Shu, "Achieving Efficient Cloud Search Services: Multi-keyword Ranked Search over Encrypted Cloud Data Supporting Parallel Computing," *IEICE Transactions on Communications,*vol. E98-B, no. 1, pp.190-200, 2015.
4. I. Hashem, I. Yaqoob, N. Anuar, et al., "The rise of "big data" on cloud computing: Review and open research issues," *Information Systems,* vol. 47, no. 47, pp. 98-115, 2015.
5. H. Li, Y. Yang, T. Luan, X. Liang, L. Zhou and X. Shen, "Enabling Finegrained Multi-keyword Search Supporting Classified Sub-dictionaries over Encrypted Cloud Data," *IEEE Transactions on Dependable and Secure Computing,* DOI10.1109/TDSC.2015.2406704, 2015.
6. H. Li, D. Liu, Y. Dai and T. Luan, "Engineering Searchable Encryption of Mobile Cloud Networks: When QoE Meets QoP," *IEEE Wireless Communications,* vol. 22, no. 4, pp. 74-80, 2015.
7. X. Liu, B. Qin, R. Deng, Y. Li, "An Efficient Privacy-Preserving Outsourced Computation over Public Data," *IEEE Transactions on Services Computing,* 2015, doi: 10.1109/TSC.2015.2511008.
8. X. Liu, R. Choo, R. Deng, R. Lu, "Efficient and privacy-preserving outsourced calculation of rational numbers," *IEEE Transactions on Dependable and Secure Computing*, 2016, doi: 10.1109/TDSC.2016.2536601.
9. H. Li, X. Lin, H. Yang, X. Liang, R. Lu, and X. Shen, "EPPDR: An Efficient Privacy-Preserving Demand Response Scheme with Adaptive Key Evolution in Smart Grid," *IEEE Transactions on Parallel and Distributed Systems,* vol. 25, no.8, pp. 2053-2064, 2014.
10. H. Li, R. Lu, L. Zhou, B. Yang, X. Shen, "An Efficient Merkle Tree Based Authentication Scheme for Smart Grid," *IEEE SYSTEMS Journal,* vol. 8, no.2, pp. 655-663, 2014.
11. C. Chen, C. Zhang, "Data-intensive applications, challenges, techniques and technologies: A survey on Big Data," *Information Sciences,* vol. 275, no. 11, pp. 314-347, 2014.