COOPERATIVE JAMMING MECHANISM, ROUTING, AND MAC FOR STABILITY AND FAIRNESS IN WIRELESS NETWORKS

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ABSTRACT:

In this text, we describe and examine common programming; Routing and congestion manipulate mechanism for wireless networks. Thus making sure solid balance of buffers equitable distribution of community sources. Tail lengths serve as not unusual statistics for one-of-a-kind layers of network protocol. Our fundamental contribution is to illustrate top-rated variability. From a twin-middle primo controller, called a specific version the transmission manipulate protocol variations are great

INTRODCUTION:

CONSIDER A SET of flows that proportion the belongings of a tough and speedy Wi-Fi network. Each flow is defined by means of manner of its supply-destination node pair, and no longer the use of a priori set up routes. The restricted energy resources and interference among concurrent transmissions necessitate multichip transmission. The nodes that represent the network have to cooperate via forwarding each other's packets towards their places. Thus, each node may want to maintain buffers to maintain packets of those flows different than its very own. For this type of machine, we lay out a joint routing, medium get admission to govern (MAC) and congestion control set of policies that stabilizes the buffers, and drives the mean waft prices to a system-wide sincere allocation component. The query of designing strong scheduling algorithms for Wi-Fi networks became first addressed by using the usage of Tassels and Ephremides [28] underneath the concept that the incoming flows are inelastic, i.e., the go with the glide expenses are constant as for voice or video website online site visitors. They confirmed that scheduling transmissions as a function of the buffer occupancies (queue-lengths) obviously ends in the stability of the buffers. Tassels [26] prolonged this method to derive a joint routing and scheduling set of regulations that ensures the stableness of the queues. These consequences confirmed that the queue-duration-primarily based beneficial aid allocation ensures balance of the buffers as long as the arrival costs lie inside the ability (stability) area of the community. Subsequently, there was a huge body of hard work that extended the equal concept to unique conditions and extra extensive settings [2], [10], [11], [21], [23] [26], [27], [29], [36]. However, the ones works do no longer consider the case of visitors whose fee may be adjusted on line. In the context of wire line networks, the idea of a dispensed float manage primarily based on a system-extensive optimization problem turn out to be advanced in [12], and observed by way of manner of others in [1], [14], [18], [30], and [34]; see [24] for a survey. In those works, the principle contribution turned into the layout of a dispensed congestion manipulates mechanism to strain the charges of elastic flows closer to the gadget-wide closing. In [6] and [35], the authors use this concept to boom congestion manages algorithms for wireless environments by the use of lowering the available functionality location and converting the network into essentially a wire line community. The crucial tendencies of Wi-Fi networks aren't completely addressed there. More currently, the problem of serving elastic visitors over Wi-Fi networks has been investigated in [5], [7], [9], [15], [16], [20], [22], and [25]. Here, the queues and the Wi-Fi characteristics of the network are blanketed inside the machine model. The primary concept in those works has been to combine the outcomes on scheduling inelastic site visitors in wireless networks and dispensed congestion manage in wire line networks to layout joint scheduling-congestion manipulate mechanisms that guarantee maximum useful routes, balance, and foremost charge allocation. These papers prove that a decentralized congestion controller at the shipping layer going for walks on the side of a queue-length-based scheduler at the MAC layer will asymptotically obtain buffer stability, final routing, and truthful price allocation. Moreover, those layers are coupled through commonplace queue-duration statistics. [26], [27], [29], [36]. However, the ones works do not keep in thoughts the case of website site visitors whose rate can be adjusted online. In the context of wire line networks, the concept of a dispensed go with the drift manage primarily based on a device-big optimization hassle become developed in [12], and discovered with the aid of others in [1], [14], [18], [30], and [34]; see [24] for a survey. In the ones works, the principle contribution changed into the format of a distributed congestion manage mechanism to pressure the expenses of elastic flows in the direction of the machine-huge most reliable. In [6] and [35], the authors use this concept to increase congestion control algorithms for wireless environments by using reducing the to be had capacity location and changing the network into essentially a wire line community. The important traits of wireless networks aren't fully addressed there. More recently, the trouble of serving elastic visitors over Wi-Fi networks has been investigated in [5], [7], [9], [15], [16], [20], [22], and [25]. Here, the queues and the wireless characteristics of the network are protected within the tool model. The number one idea in these works has been to mix the consequences on scheduling inelastic web site visitors in Wi-Fi networks and distributed congestion manipulate in wire line networks to layout joint scheduling-congestion manipulate mechanisms that assure maximum beneficial routes, stability, and pleasant fee allocation. These papers show that a decentralized congestion controller at the shipping layer strolling along with a queue-duration-based totally scheduler at the MAC layer will asymptotically obtain buffer stability, pinnacle of the line routing, and honest price allocation. Moreover, those layers are coupled thru not unusual queue-period records. In [9], [15], [20], and [25], the authors suggest and study charge manage algorithms that adapt the waft costs without delay as a function of the get admission to queue-lengths. The price control mechanism studied in all of those works can be categorized because the Dual Congestion Controller when you consider that it could be interpreted as a gradient set of rules for the dual of an optimization trouble. The intrinsic assumption of the twin congestion control mechanism is that the flow prices may be modified right now in reaction to congestion comments within the network. However, it is widely known that adaptive window float manipulate mechanisms together with transmission manage protocol (TCP) reply to congestion remarks not immediately, but gradually. Such a reaction is desired by way of the usage of practitioners due to the truth the rate fluctuations are small. Thus, the take a look at of every other set of guidelines that modifies the go with the glide fees grade by grade is vital. To this cease, we advise and take a look at the so referred to as Primal-Dual Congestion Controller on this artwork. Primal-dual algorithms are broadly recognized within the optimization literature and have been studied substantially in different contexts [1], [17], [24], [30]. Since the response of the primal-dual controller is extra gradual compared to the twin controller, it isn't proper now smooth as to whether or not the buffer balance and charge convergence homes can be maintained. We are conscious that the algorithm considered in [25] updates its charges genuinely in any other manner than the set of policies in [5], [9], [15], [16], and [20]. In [25], the clients' facts prices are nevertheless determined right away as a feature of the buffer occupancies and channel situations; however a median flow rate is maintained for each consumer and used inside the set of rules. On the opposite hand, in our work, we replace the facts quotes to imitate the tendencies of drastically used versions of TCP [24]. Further, the evidence technique used in [25] is quite distinctive from ours, and our set of rules may be without delay interpreted as a gradient set of rules for a primal optimization trouble this is carried out at the assets and a gradient set of guidelines for a twin optimization hassle this is accomplished on the nodes. Here, it have to be confused that in spite of the reality that the congestion manage is shipped, the scheduling continues to be assumed to be centralized on this artwork. In [3], [4], [16], [31], and [32], the effect of decentralized implementations of the scheduler is studied. We are aware that the results of these paintings may be prolonged to allot and asynchronous implementations for a unique class of interference models the usage of the approach in [3]. Finally, we be aware that an associated, however distinct, trouble has been considered in [33], in which a dispensed set of rules has been designed to direction inelastic flows to lower do away with prices in a wireless network

LITERATURE WORK:

"CHEN, C., HEATH, R. W., BOVIK, A. C., AND DE VEC IANA, G. Adaptive policies for real-time video transmission: A markov decision process framework. In IEEE international Conference on Image Processing (Sep 2011)" We study the problem of adaptive video data scheduling over wireless channels. We prove that, under certain assumptions, adaptive video scheduling can be reduced to a Markov decision process over a finite state space. Therefore, the scheduling policy can be optimized via standard stochastic control techniques using a Markov decision formulation. Simulation results show that significant performance improvement can be achieved over heuristic transmission schemes The problem of efficient real-time video transmission over wireless channels is challenging. In the first place, the data throughput is varying over time. In the second place, realtime video delivery can be highly delay-sensitive. To improve the receiver/decoder video quality, the transmitter should optimally allocate bandwidth among current and future frames. In the third place, the video packets are structured. Due to the nature of predictive video coding algorithm, a video frame can be decoded only when its predictor is available. Hence, the prediction structure of the video codec enforces an order on the video packets. A system with finite state space is called a controlled Markovian system if its state transition probability only depends on the current state and the control action taken at the state. If a instantaneous service quality associated with the system is solely determined by the state, this system is called a Markov decision process (MDP) [1]. The average service quality of the system can be maximized by optimizing the control policy. The MDP-based control framework has previously been proposed in the scenario of real-time video transmission. Indeed in [2], a MDP based formulation was introduced for the problem of real-time encoder rate control. The derived optimal control policy operates at the video encoder adapting the video rate according to the channel conditions and video rate-distortion characteristics. In [3], an MDP formulation was proposed for adaptive video play out This research was supported in part by Intel Inc. and Cisco Corp. under the VAWN program and scheduling. The controller controls the play out speed according to the receiver buffer state and channel state to optimize the receiver visual quality. Neither of the above two works considered the adaptive real-time video scheduling. The most closely related work to our paper is by Zhang et al. [4], in which a reinforcement learning framework was studied for adaptive video transmission. The optimal transmission policy is obtained via reinforcement learning rather than MDP-based optimization. Hence, the transmitter need to learn a “good” policy from trying those “bad” policies. For real time video delivery, it will degrade the visual quality until the learning is finished.

“Scheduling Heterogeneous Real-Time Traffic Over Fading Wireless Channels”

We develop a general approach for

designing scheduling policies for real-time traffic over wireless channels. We extend prior work, which characterizes a real-time flow by its traffic pattern, delay bound, timely throughput requirement, and channel reliability, to allow clients to have different deadlines and allow a variety of channel models. In particular, our extended model consider scenarios where channel qualities are time-varying, the access point may not have explicit information on channel qualities, and the access point may or may not employ rate adaptation. Thus, our model allows the treatment of more realistic fading channels as well as scenarios with mobile nodes and the usage of more general transmission strategies. We derive a sufficient condition for a scheduling policy to be feasibility optimal, and thereby establish a class of feasibility optimal policies. We demonstrate the utility of the identified class by deriving a feasibility optimal policy for the scenario with rate adaptation, time-varying channels, and heterogeneous delay bounds. When rate adaptation is not available, we also derive feasibility optimal policies for both scenarios where the access point may or may not have explicit knowledge on channel qualities. For the scenario where rate adaptation is not available but clients have different delay bounds, we describe a heuristic. Simulation results are also presented, which indicate the usefulness of the scheduling policies for more realistic and complex scenario WITH the wide deployment of wireless local area networks (WLANs) and advances in multimedia technology, wireless networks are increasingly being used to carry real-time traffic, such as VoIP and video streaming. These applications usually specify throughput requirement while meeting specified delay bounds. We study the problem of designing scheduling policies for such applications. While there has been much research on scheduling real-time traffic over wire line networks, the results are not directly applicable to wireless networks where channels are unreliable, with qualities that may be time-varying either due to fading or node mobility. Also, individual clients may impose differing delay requirements. These features present new challenges to the scheduling problems We consider the scenario where an access point (AP) is required to serve real-time traffic for a set of clients. A previous work [1] solves the scheduling problem in a restrictive environment and proposes two feasibility optimal policies. In particular, it assumes a fixed transmission rate, a static channel model, and that all clients in the system require the same delay bound. We extend this model to relax these limitations. The extended model considers a variety of scenarios where wireless channel qualities are time-varying. These scenarios include those where the AP may and may not have explicit knowledge on channel qualities, and those where the AP may and may not employ rate adaptation. We establish a sufficient condition for a scheduling policy to be feasibility optimal. Based on this, we describe a class of policies and prove that they are all feasibility optimal. To demonstrate the utility of the class of policies, we study four particular scenarios of interest. Three of these scenarios consider systems without rate adaptation. They address the challenges of time-varying channels, lack of explicit knowledge on channel qualities, and heterogeneous delay bound of clients, respectively. The last scenario employs rate adaptation and treats time-varying channels, as well as allowing different delay bounds for different clients. We derive online scheduling policies for each of them. We have also tested the derived policies using the IEEE 802.11 standard in a simulation environment. The results suggest that the four policies outperform others, including the policies in [1], and a server-centric policy that schedules packets randomly. In particular, since the policies introduced in the previous work fail to provide satisfactory performance in the environments studied here, this suggests that neglecting the facts that the system can apply rate adaptation, that wireless channels are time-varying, and the possibility that clients may require different delay bounds can result in malperformance of the derived policies

"Scheduling and Resource Allocation for SVC Streaming over OFDM Downlink Systems" We consider the problem of scheduling and resource allocation for multiuser video streaming over downlink orthogonal frequency division multiplexing (OFDM) channels. The video streams are pre coded using the scalable video coding (SVC) scheme that offers both quality and temporal scalabilities. The OFDM technology provides the flexibility of resource allocation in terms of time, frequency, and power. We propose a gradient based scheduling and resource allocation algorithm, which prioritizes the transmissions of different users by considering video contents, deadline requirements, and transmission history. Simulation results show that the proposed algorithm outperforms the content-blind and deadline-blind algorithms with a gain of as much as 6 dB in terms of average PSNR when the network is congested T HE demand of high-quality video over communication networks exhibits an ever growing trend. However, content distribution and resource allocation are typically studied and optimized separately, which leads to suboptimal network performance. This problem becomes more prominent in wireless networks, since the typically time-varying and limited network resource makes efficient multiuser video streaming particularly challenging. In this letter, we consider the problem of multiuser video streaming over orthogonal frequency division multiplexing (OFDM) networks, where videos are coded in a scalable video coding (SVC) scheme. OFDM is the core technology for a number of wireless data systems (e.g., WiMAX and wireless LANs). The resource allocation in OFDM can be flexibly performed over power, frequency, and time. SVC, on the other hand, allows reconstructing lower quality signals from partially received bitstreams. It provides flexible solutions for transmission over heterogeneous networks and allows easy adaptation to various storage devices and terminals. Our focus in this letter is to design efficient streaming protocols that fully exploit various flexibilities in both OFDM and SVC. There is a growing literature on SVC-based video transmission over wireless networks. Most of them focused on exploiting the scalable feature of SVC to provide QoS quarantee for the end users (e.g., [12], [13]). In [15], the layered bitstream of SVC is exploited together with congestion control algorithm for distributing video to WiMax subscriber stations. In [16], the rate distortion model proposed for H.264/AVC was extended to include the effect of random packet loss on SVC. Reference [2] focused on maximizing the number of admitted users by giving different priorities to different video subflows according to their importance. None of the above results considered power control. An unequal power allocation scheme was proposed in [17] for the transmission of SVC packets in a WiMax system. In [5], a distortion-based gradient scheduling algorithm was proposed. However, they did not consider the influence of video latency on resource allocation. The main contribution of this letter is to provide a framework for efficient multiuser SVC video streaming over OFDM wireless channels. The objective is to maximize the average PSNR of all video users under a total downlink transmission power constraint. The basis of our approach is the stochastic subgradient-based scheduling (e.g., [10]) and our previous work [3] that considers downlink OFDM resource allocation for elastic data traffic.

IMPLEMENTATION:

This paper presents a new simple algorithm for minimizing sub modular functions. For integer valued sub modular functions, the algorithm runs in $O(n6\mathrm{EO}\log nM)$ time, where n is the cardinality of the ground set, M is the maximum absolute value of the function value, and EO is the time for function evaluation. The algorithm can be improved to run in $O((n4\mathrm{EO}+n5) \log nM)$ time. The strongly polynomial version of this faster algorithm runs in $O((n5\mathrm{EO} + n6) \log n)$ time for real valued general sub modular functions. These are comparable to the best known running time bounds for sub modular function minimization. The algorithm can also be implemented in strongly polynomial time using only additions, subtractions, comparisons, and the oracle calls for function evaluation. This is the first fully combinatorial sub modular function minimization algorithm that does not rely on the scaling method. This paper presents a new simple algorithm for minimizing sub modular functions. For integer valued sub modular functions, the algorithm runs in $O(n6\mathrm{EO}\log nM)$ time, where n is the cardinality of the ground set, M is the maximum absolute value of the function value, and EO is the time for function evaluation. The algorithm can be improved to run in $O((n4\mathrm{EO}+n5) \log nM)$ time. The strongly polynomial version of this faster algorithm runs in $O((n5\mathrm{EO} + n6) \log n)$ time for real valued general sub modular functions. These are comparable to the best known running time bounds for sub modular function minimization. The algorithm can also be implemented in strongly polynomial time using only additions, subtractions, comparisons, and the oracle calls for function evaluation. This is the first fully combinatorial sub modular function minimization algorithm that does not rely on the scaling method. We consider the problem of simultaneous on-demand streaming of stored video to multiple users in a multi-cell wireless network where multiple unicast streaming sessions are run in parallel and share the same frequency band. Each streaming session is formed by the sequential transmission of video "chunks", such that each chunk arrives into the corresponding user playback buffer within its playback deadline. We model the wireless network with a graph G(N , E), where N represents the set of N = |N | wireless nodes and E denotes the set of directed wireless links. Link e = (m, n) ∈ E connects two nodes m, n ∈ N if and only if node n is in the transmission range of node m. We use the notations e and (m, n) interchangeably. The set of data flows is denoted by F and the number of data flows is denoted by F = |F|. The set of source nodes is denoted by S. Data transmission between a source sf ∈ S and the destination df of flow f ∈ F can be relayed through multiple hops. We use multipath routing for data transmission. The set Kf contains Kf = |Kf | routing paths for flow f ∈ F. For each link e ∈ E, path k ∈ Kf , and flow f ∈ F, we define a fk e = 1 if link e belongs to the k th routing path for flow f, and a fk e = 0, otherwise. For any node n ∈ N , each data flow f ∈ F, and any path k ∈ Kf , let i fk n and o fk n ∈ E be the input and output links to and from node n on path k of flow f, respectively. Whenever the context is clear, we remove the indices n, f, k and denote the input and output links with i and o, respectively (see Fig. 1). A slotted notion of time is used with time slots t ∈ {1, 2, . . .}. We denote the value of time-varying parameters at the beginning of each time slot t with the index t. We use the same parameter without the index t to denote its average value over all time slots. At each intermediate node n ∈ N , we assume a separate queue for any path k ∈ Kf of flow

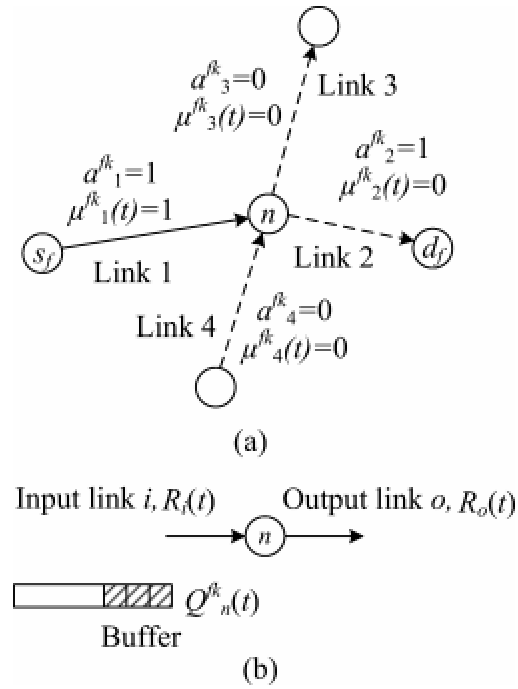


Fig. 1. (a) Path k of flow f from node sf to node df which uses node n as a relay node is shown. It is shown that solid link 1 is active (μ fk 1 (t) = 1) while dotted links 2 and 3 are not active (μ fk 2 (t) = μ fk 3 (t) = 0) in that particular time slot t. Note that links 1 and 2 belong to the mentioned path (a fk 1 = a fk 2 = 1) but link 3 does not (a fk 3 = 0). (b) A relay node n is shown with its input link i and its output link o corresponding to the k th path of flow f. The corresponding packets are stored in Q fk n before they are sent

f ∈ F. The number of data bits corresponding to path k for flow f, stored in node n is denoted as $Q_{fk}^{n}(t)$. We assume $Q^{fk}_{df}(t) = 0$, $\forall t, k \in K_f, f \in F$ since the received bits are transfered to the upper layers at the destination node d_f. We incorporate all the queue backlogs in the vector $Q(t) = (Q_{fk}^{n}(t), \forall n \in N, k \in K_f, f \in F)$. We use link-dependent channel code rates to counter channel variations and improve network reliability. Each source node or intermediate node $n \in N$ for any flow encodes data bits by adding redundant bits and transmitting the resultant codeword of length g. Hereafter, we assume each packet consists of one codeword and we use the terms packet and codeword interchangeably. We define the code rate $R_e(t)$ as the ratio of the data bits to the total transmitted bits (data plus redundant bits) on link $e \in E$. We concatenate code rates for all links $e \in E$ in vector $R(t) = (R_e(t), \forall e \in E)$. The smaller the code rate $R_e(t)$, the greater number of redundant bits is added, and the higher the reliability is. The reliability is gained at the cost of increased network traffic. When $R_e(t)$ is equal to one, channel coding is not used on link e. We use $R_{0e} \leq 1$ to denote the cut-off rate of wireless link $e \in E$. The cut-off rate is a channel parameter to which the rate of the adopted coding scheme is always limited [22]. In general, R_{0e} depends on the particular modulation scheme which is being used and also the signal-to-noise ratio (SNR) in the receiver node. For example, for a binary phase shift keying (BPSK) waveform [22], we have $R_{0e} = 1 - \log_2(1 + e^{-\gamma_e})$, where γ_e denotes the SNR at the receiver node of wireless link $e \in E$. When γ_e is relatively large, R_{0e} is close to 1. Given $R_e(t) \leq R_{0e}$ for $e \in E$, we have $P_e(t) \geq 1 - 2^{-g(R_{0e}-R_e(t))}$, (1) where $P_e(t)$ is the probability that a codeword of length g is received correctly on link e with rate $R_e(t)$ [22, pp. 392- 397]. The vector $P(t) = (P_e(t), \forall e \in E)$ represents the successful probabilities on all links $e \in E$. For the rest of this paper, we consider the worst case in which inequality (1) is satisfied with equality. For each transmission on link $e \in E$, we define $\rho_e(t) = 1$ if the packet is transmitted successfully and $\rho_e(t) = 0$ otherwise. We have $\rho_e(t) = 1$ with the probability of $P_e(t)$. We define $\rho(t) = (\rho_e(t), e \in E)$ as the channel state at time slot t. As mentioned above, a codeword may be corrupted with probability $1 - P_e(t)$ through a transmission on link $e \in E$. The receiver at link e sends a link-level acknowledgement (ACK) to the transmitter if the packet is received correctly. The transmitter retransmits the packet if no ACK is received within a predefined time period.

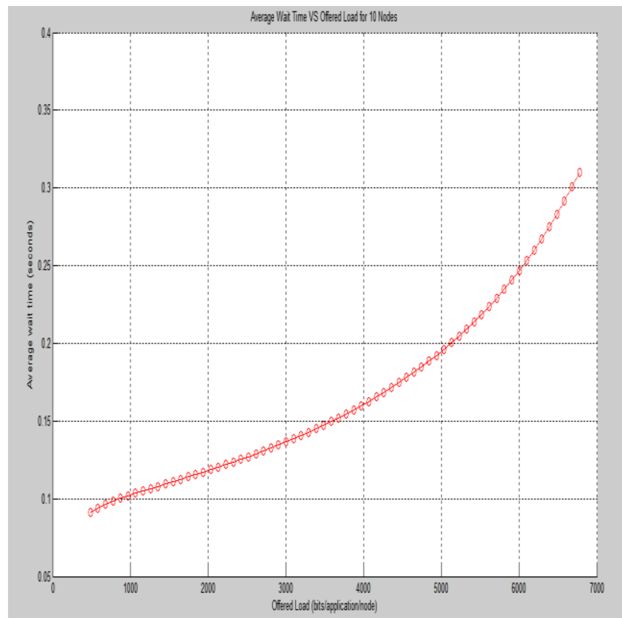
Retransmissions ensure that packets admitted to the network will be received at their corresponding destination nodes. This is at the cost of increased network load. We denote the number of data bits which are admitted to the path k ∈ Kf of flow f ∈ F at the beginning of time slot t as α k f (t). The vector α(t) = (α k f (t), ∀ k ∈ Kf , f ∈ F). Suppose all admissions are upper bounded (i.e., α k f (t) ≤ αmax). We assume that all source nodes are backlogged (i.e., each source node has at least αmax data bits available to send over each of its routing paths at any time slot). We define the capacity region Λ as the closure of the set of all sending rate vectors α (considering all possible routing and scheduling policies), for which the network is stable, that is

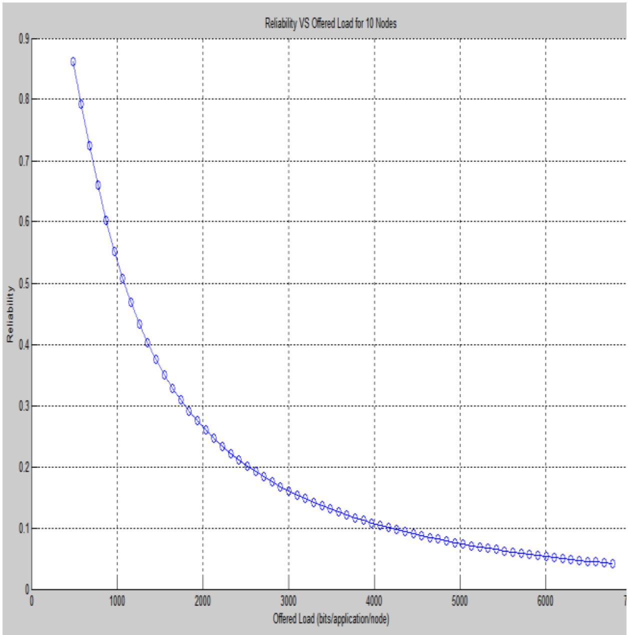
$$\mathbf{A=\{}\boldsymbol{\alpha}\,|\boldsymbol{\alpha} \geq \mathbf{0,}\lim_{t\to\infty} sup\frac{1}{t}\sum_{t=0}^{t-1}\sum\sum_{k\in kf,f\in F} E\{Q_n^{fk}(\tau)\}<\mathbf{M}$$

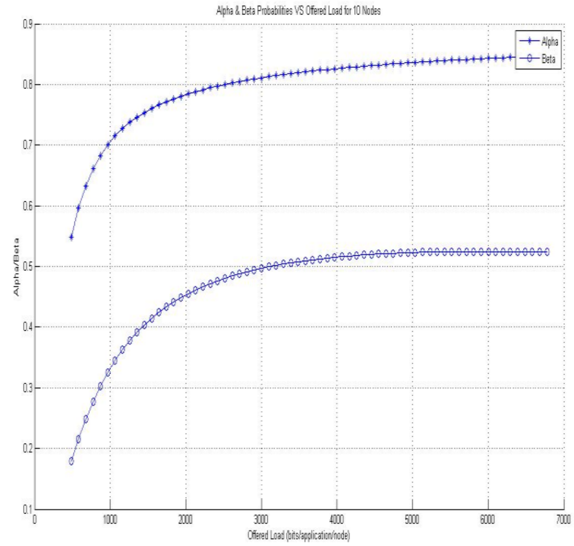
$$\mathbf{n} \in \boldsymbol{N}\mathbf{n}$$

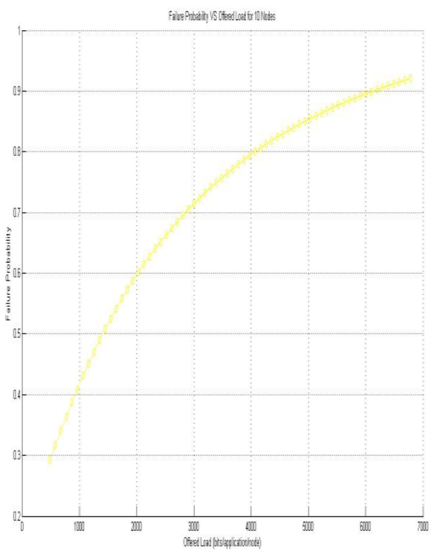
where M is a finite number. Note that α = limt→∞ 1 t Pt−1 τ=0 α(τ) is the time average value of α(t). Two links e1, e2 ∈ E mutually interfere with each other if and only if the receiver of one link is in the transmission range of the sender of the other. At each time slot t, only one wireless link may be active among those wireless links which are in mutual interference with each other. We define μ fk e (t) = 1 if link e is active in data transmission for the k th routing path of flow f at time slot t, and μ fk e (t) = 0 otherwise. We define ce as the number of bits that can be transmitted by link e ∈ E in each time slot t. ce contains data bits as well as redundant bits due to channel coding

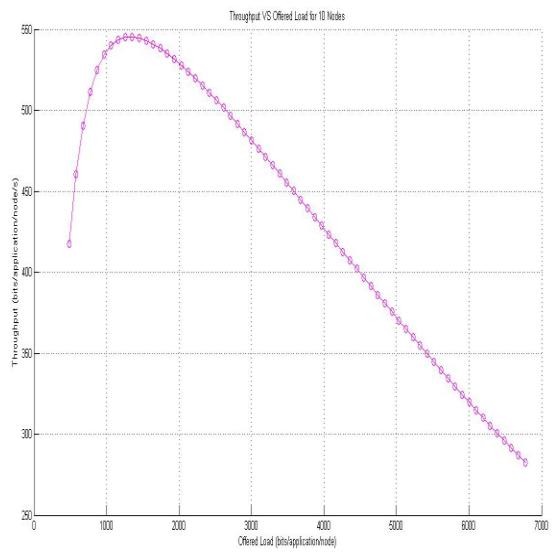
SIMULATION RESULTS:

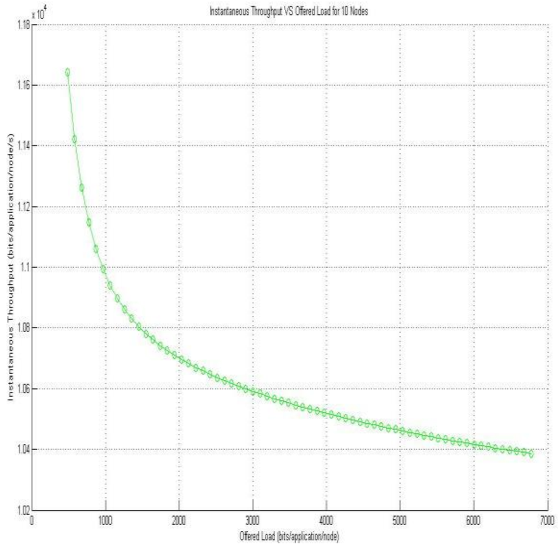


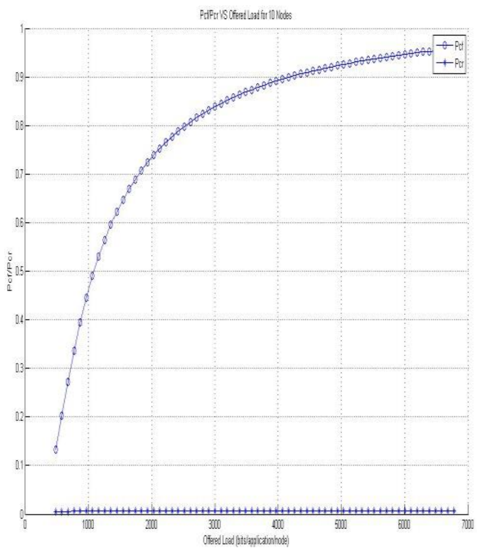

Reliability VS Offered Load for 10 Nodes
Reliability
Offered Load (bits/application/node)


Alpha & Beta Probabilities VS Offered Load for 10 Nodes
Alpha
Beta
Alpha/Beta
Offered Load (bits/application/node)


Failure Probability VS Offered Load for 10 Nodes
Failure Probability
Offered Load (bits/application/node)


Throughput VS Offered Load for 10 Nodes
Throughput (bits/application/node/s)
Offered Load (bits/application/node)





CONCLUSION:

In this paper, we recommend and have a look at a go-layer scheduling routing-congestion manage mechanism for wi-fi networks. We model many current variations of TCP/AQM schemes the use of the primal-twin congestion controller. It is shown that this controller, in conjunction with suitable MAC/routing protocol achieves fairness and stability. Architecturally, we keep the traditional protocol stack, however couple them thru the usage of queue-duration facts.

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