Downlink Performance for Mixed Web/VoIP Traffic in 1xEVDO Rev A Networks

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***Abstract*— Current 3G cellular networks, such as those based on 1xEVDO Rev A, have several mechanisms to provide Quality of Service (QoS) to network applications. In this paper, we investigate the performance for mixed Web/VoIP traffic on the downlink in 1xEVDO Rev A networks. The impacts from the mixed traffic are studied by using link-level simulations. Our simulator is built based on a realistic system model that includes physical channel parameters, transmission formats, and channel scheduling algorithms, as well as hierarchical traffic models. User-perceived throughput and packet delay are explored. We present our observations and determine the Erlang capacity of the network. We carry out performance analysis using an equivalent queue approach , and discuss appropriate scheduling algorithms and call admission control mechanisms based on the simulation results.**

I. Introduction

Today's telecommunication networks use statistical multiplexing to share common network resources amongst widely disparate traffic streams with different Quality of Service (QoS) requirements. How efficiently this can be done, and how satisfactorily QoS demands can be met, depends upon the statistical characteristics of the traffic and the resource allocation policies in use.

CDMA2000 1xEVDO networks were originally designed to support high data rate services. With growing demand, these networks are expected to serve other traffic types as well, such as Voice-over-Internet-Protocol (VoIP) from the conversational QoS class. The EVDO Rev A has been standardized with many improvements favorable to VoIP implementation. One of the key considerations is how to overcome the inherent inefficiencies that occur when services with different requirements are combined in the system [1], [2], [3].

On the Internet, the User Datagram Protocol (UDP) has been used for (soft) real-time applications, such as audio and video streaming. However, more than 90 percent of the WAN resources are occupied by Transmission Control Protocol (TCP) traffic. Clearly, the QoS for UDP-based applications is affected by TCP traffic and its flow control mechanism whenever TCP and UDP share a network bottleneck.

In the literature, there are several studies on the performance of Web traffic or VoIP traffic in EVDO systems [6], [7], [10]. Few studies focus on both because Web traffic and VoIP traffic have different stochastic characteristics, and they are handled by different transport protocols. However, both traffic types may share network resources when transmitting on the same base station in a cellular network.

A common method to study mixed traffic is to set a priority for the traffic class of interest, and treat the others as background traffic, for which only "best effort" services are available. For example, Web traffic is considered as a class of traffic that only receives best effort service, while VoIP traffic has priority because of its delay-sensitive nature. However, how these two impact each other is not clear when both require QoS. Therefore, how to do resource allocation and QoS scheduling for mixed traffic in an EVDO network is still under investigation [3], [8].

In this paper, we start our investigation by building a simulation environment to study the interactions between VoIP and Web traffic in the system. We investigate performance of the two services using link-level simulation, to assess user-perceived delay and throughput. We present our observations and determine the Erlang capacity for the network. Then, we do analysis using an equivalent queue approach and discuss appropriate scheduling algorithms and call admission control based on our results.

Our simulation results lead to two counter-intuitive observations related to QoS performance for such mixed traffic. The first observation is that performance of Web traffic and VoIP traffic is not adversely impacted by the other within a certain range of traffic load, implying that effective capacity for different traffic classes is limited mostly by the traffic's own stochastic properties. The second observation is that, to guarantee a certain degree of QoS for Web traffic while the VoIP traffic has reached its own QoS requirements, priority should be given to Web traffic when scheduling. This implies that an appropriate QoS scheduling should exist to support QoS for multiple classes of traffic in the network, which is still an open area for EVDO networks [8].

Our observations benefit from the traffic modeling in our simulation and analysis. We do not adopt the full (saturated) queue method to study the queue delay and throughput. Rather, we use a three layer structure to model the traffic. The session, packet-call, and packet level capture the natural dynamics of the traffic. This dynamic property leaves space for opportunistic access by other classes of traffic when its own traffic capacity reaches the limitation. Meanwhile, we do not ignore transport-layer protocol effects, because (for example) TCP controls traffic arrival rate in the queue at the system based on congestion and dropping in the network, so that the delay and throughput for users are impacted.

The rest of the paper is organized as follows. Section II describes EVDO Rev.A forward channel structure and our traffic modeling. Section III presents the delay and throughput by simulation observations. System performance and delay bound are analyzed in Section IV. Section V discusses the forward link Erlang capacity, the scheduling and the call admission control in the network. Section VI concludes the paper.

II. 1 x EVDO Rev.A Forward Channel and Traffic Modeling

A. Forward channel structure

The EVDO forward link (downlink) is comprised of time-multiplexed user channels that support traffic, pilot, and MAC/control information. On the downlink, each user's data transmission is time-division multiplexed on the data time slot, while the user's MAC-layer transmission is code-division multiplexed on the MAC time slot. The supported modulation formats include QPSK, 8-PSK, and 16-QAM. Synchronous H-ARQ and incremental redundancy are also supported. In each time slot of the forward-link, four information fields are present: preamble, MAC information, pilot, and data. The pilot bits are allocated 192 chips, and the MAC bits occupy 256 chips, while the remaining 1600 chips are allocated to the preamble and data bits.

There is a multi-user packet feature implemented in EVDO Rev A. This feature allows packets from up to 8 users to be combined into a single physical-layer packet. Therefore, it overcomes the shortage of time slots and improves the efficiency of transmitting VoIP with low data rate. The multi-user packetization decision is done dynamically on a packet-by-packet basis [2], [6].

We abstract the properties of the forward link with a queueing model to study performance of different classes of traffic in the system. First, the channel is a single broadband link shared by all users in a cell. Each user transmits data in time slots assigned by the scheduler. Second, adaptive modulation and coding schemes are adapted to support different data rates with reliable transmission. A set C_i of data rates is available based on the estimated channel condition measured by BS from mobile users' feedback. The corresponding transmission formats can be found in [1], [2].

B. Web and VoIP traffic models

1) Web traffic modeling: Our Web (HTTP) traffic model is a hierarchical ON/OFF process based on the proposed model in [4]. In the model, a typical web browsing session is divided into ON/OFF periods representing Web page downloads and the intermediate reading times. The Web page downloads are referred to as packet calls. The packet traffic characteristics within a packet call depend on the version of HTTP used by the Web servers and browsers.

Currently two versions of the protocol, HTTP/1.0 and HTTP/1.1, are widely used by the servers and browsers. These two versions differ in how the transport layer TCP connections are used for the transfer of the main and the embedded objects. To reduce the complexity of our simulation and simplify the analysis, we only implement the HTTP/1.1 persistent connection model in our simulation. Our model characterizes the structure from four levels: user sessions (level 1); TCP connections within sessions (level 2); bursts within a TCP connection (level 3); and packets within bursts (level 4).

2) VoIP traffic modeling: Similar to the Web traffic model, we build a layered model for VoIP traffic. Each voice call initiates one session, with a single packet call. The hierarchical structure of typical voice traffic is built by considering talk spurts within a voice call as the bursts. We assume the ON/OFF periods of this process follow truncated Pareto distributions with 352 ms mean length for talking spurts, and 650 ms mean length for silent periods [11]. The packet characteristic within the talking spurts depends on the physical layer coding and packetization.

In our study, the enhanced variable rate codec (EVRC) speech coder is assumed to be used in the network. We assume 88% full rate and 12% half rate are adopted to generate voice traffic frames. Because voice frames are to be placed into IP packets, there are added protocol overheads for the VoIP packets. For full-rate frames, the VoIP packet consists of 171 bits, with 24 cyclic redundancy bits and 6 bits for coding tail. Half-rate frames use 80 bits, with 24 cyclic redundancy bits and 6 bits coding tail. Assuming that the total IP overhead is compressed into 55 bits, the encoder VoIP packet size is 256 bits for full rate and 165 bits for half rate frame.

At the top level of our models, we assume that sessions arrive according to a Poison process. Because session initiation is related to user actions, it is reasonable to assume the inter-arrival period follows an exponential distribution.

III. Delay and Throughput Performance

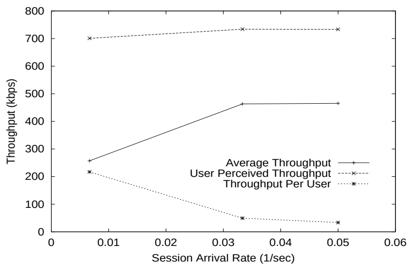
Throughput and delay performance can be approximated with analytical techniques, but many simplifying assumptions are needed. Furthermore, it is difficult to analyze performance of the scheduler algorithm, MAC, and H-ARQ due to the complex interactions between the physical layer user call, and intermediate network protocols (e.g., TCP/IP, RLP, etc.). Therefore, end-to-end system simulation is an essential tool for determining true system performance improvement of each high speed wireless packet data transmission technology.

A. System and simulation setup

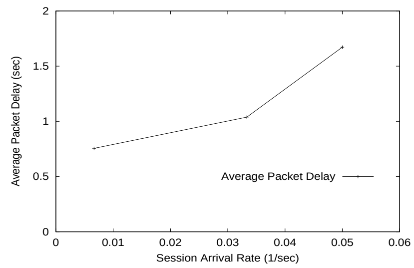
Our simulation models an embedded sector of the central base station in a family of 19 base stations, with three sectors. We developed our simulator using C/C++, combining with simulations in MATLAB to simulte the system. It implements air interface(PHY/MAC), TCP/IP, UDP, HTTP, VoIP etc., as well as various scheduling and call control strategies. In each run of the simulation, a certain number of users are randomly placed into the network with uniform geographic distribution. For Web traffic users, 80% are static during their connections and 20% are mobile. For voice users, 20% are static and 80% are mobile. Each mobile user moves according to "waypoint mobility model" [12]. A multi-path channel model is adopted, assuming pedestrian moving speed of 3 km/h. 4-branch REAK receiver is implemented. We do not simulate handoff in this study, but simply assume conservation of users across sector boundaries. The interference from the 18 surrounding

TABLE I
FACTORS AND PARAMETERS IN SIMULATIONS

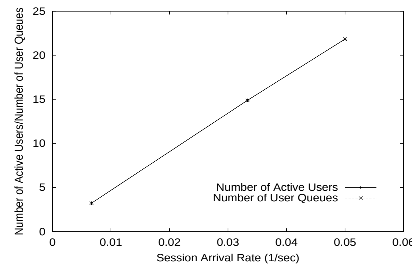
Cellular layout	Cell to cell distance	Pathloss component	Base station power	Std deviation of noise	Average noise power	User mobile speed
19 cells	3Km	4	40dB	6dB	3 dB	3Km/hour



(a) Throughput

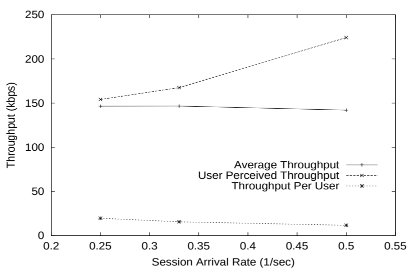


(b) Packet Delay

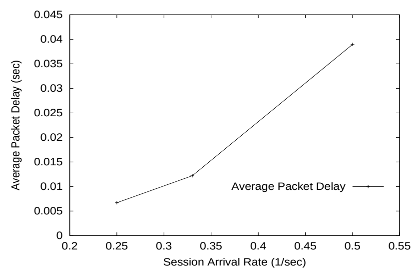


(c) Number of Users

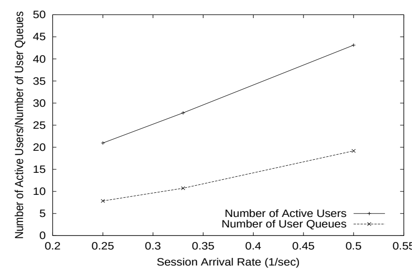
Fig. 1. Overview of Simulation Results for HTTP Traffic



(a) Throughput



(b) Packet Delay



(c) Number of Users

Fig. 2. Overview of Simulation Results for VoIP Traffic

cells is calculated using simulated signal-to-interference-and-noise ratio (SINR) for data rate control (DRC) channel [9]. The frame error rate in the link transmission corresponds to modulation, coding and retransmission schemes. In our simulation, we simply assume a fixed frame error rate (FER) of 2% for all transmission formats. As each packet arrives to the forward link transmission queue of the corresponding access terminal, the transmission format is determined by the scheduling algorithm based on the format table [2]. HARQ scheme is adopted for the fast retransmission. We have several scheduling algorithms implemented in our simulator, including Proportional Fairness (PF) and others. Packet dropping mechanism can be also implemented to guarantee forward link queue delay to a specified class of traffic. Other relevant system simulation parameters are provided in Table I.

Multi-user packets are modeled in the simulator. A user's data rate needs to be at least 153.6 kbps to qualify for multi-user packet. A multi-user packet of 1024 bits can practically accommodate up to four user packets while large size multi-user packet can support maximum of eight user packets[6].

Voice and Web traffic are generated based on the models proposed in section II-B, and transmitted according to UDP and TCP, respectively.

B. Traffic performance criteria and simulation assumptions

Although there are acceptable delay guidelines for voice service and Web traffic [6], [10], [5], it is not clear what will be appropriate for mixed services in EVDO networks. To proceed with our investigation, we assume that at least 98% VoIP user and 80% HTTP users in the network should meet their delay criterions, the threshold of maximum delay user packet can tolerate. Erlang capacity is defined as the average number of users the system can accommodate under given performance criteria.

Packet latency on the forward link consists of two parts: the queue delay at scheduler and transmission latency over air link. Our study concerns queue delay. We vary VoIP and HTTP packet call arrival rates to run experiments for different scenarios. Multi-user packet mechanism is enabled. Packet loss due to channel error and excessive delay is set as option. To see the overall impact of traffic characteristics, following simulation results are obtained without enabling packet dropping mechanism. Our experiments simulate about 2 hours of the transmission with at least 500,000 packets transmitted through the scheduler.

C. Simulation results

Figure 1 and Figure 2 display overview performance of HTTP traffic and VoIP traffic with various session initiation rates. These two traffics are examined in simulation experiments separately. (a) of the figures compares diferent throughputs in the downlink, where link average throughput, user perceived throughput and the average throughput per user are compared. (b) of the figures shows packet average delay,

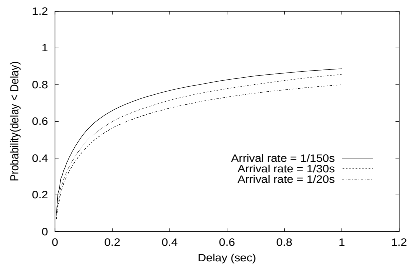


Fig. 3. Cumulative Delay Distribution for HTTP Packets

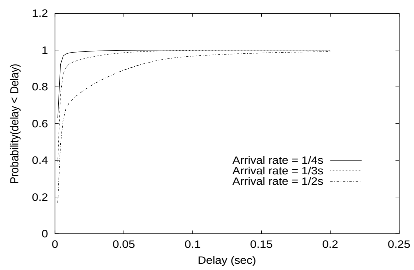


Fig. 4. Cumulative Delay Distribution for VoIP Packets

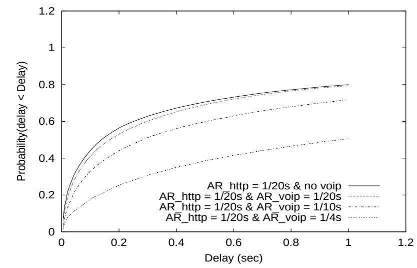


Fig. 5. Comparison for Pure HTTP with Mixed HTTP and VoIP Packets

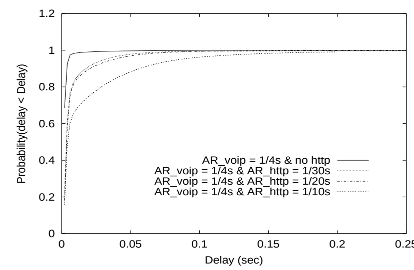


Fig. 6. Comparison for Pure VoIP with Mixed VoIP and HTTP Packets

and (c) of the figures presents mean number of user queues and mean number of active users.

It is observed that HTTP users receive mean of 720 kbps user perceived throughput, while link average throughput is 400 kbps with each user receives about 80 kbps. For VoIP users, the mean user perceived throughput is 184 kps while link average throughput is 144 kbps with each user received about 16 kbps.

From Figure 1 (a), it can be seen that link average throughput for HTTP traffic increases when session initiation rate increases, while user perceived throughput keeps almost the same. This is reasonable because increasing session initiation rate means more traffic flows accepted into the system. Therefore, link utilization is higher. Meanwhile, Higher number of flows evens the occupations of channel transmission slots by deducing burstness caused by the ON/OFF pattern of traffic. Therefore, link average throughput increases with increasing of session initiation rate until saturated. The user perceived throughput are almost same because , due to TCP's dynamic control, mean traffic arrival amount and channel service rate saturate with the session initiation rates examined.

In Figure 1 (a), it shows average throughput per user decreases due to increasing of number of users in the system . However, VoIP packets arrive into the queues according to VoIP stochastic pattern because it is transmitted using UDP. Higher numbers of VoIP users in the system means higher percent of multi-user packets transmitted. Therefore, in Figure 2 (a), we observe that user perceived throughput increases due to increasing of mean number of users, while the link average throughput has almost no changes with changes of VoIP packet call initiation rate.

Increasing session initiation rate for HTTP traffic or packet call initiation rate for VoIP traffic induces more traffic load in the system. Therefore, It is expected that packet delay increases when traffic load increases, shown in Figure 1 (b) and Figure 2 (b).

For HTTP traffic, number of user queues is always equal to number of active users because, within a connection, an ACK always bring other arrivals so that queue has no chance to be empty. For VoIP traffic, queue number is always smaller than number of active users. From Figure 1 (c) and Figure 2 (c), we find that numbers of queues and active users increase almost linearly with session or packet call initiation rate.

Figure 4 and Figure 3 give observations to packet cumulative delay distributions of VoIP and HTTP flows. We see that, with packet call arrival rate less than 0.5 per second, 98% VoIP user packet has queue delay less than 0.2 seconds. For HTTP traffic, 80% user packet delay is less than 1 second if session arrival rate is less than 0.05 per second.

By understanding individual performance of HTTP and VoIP in the system, we are ready to study performance of their mixed traffic. We investigate several combined cases where HTTP and VoIP have different arrival rates . We also compare these cases to the single class of traffic case. Figure 5 compares the cumulative delay distribution of pure HTTP traffic with that of mixed traffic where HTTP session arrival rate keeps the same. It can be seen that packet delay performance of HTTP traffic is dramatically decreased from 80% packet delay less than 1 second to 50%, when packet call arrival rate of VoIP traffic increases. This also happens to VoIP packet delay performance when we vary HTTP session arrival rate, while the VoIP packet call arrival rate keeps constant in Figure 6. It shows that increasing HTTP session arrival rate does not hurt a lot to the VoIP delay performance because there are still 98% VoIP packets' delay are less than 0.2 second. This implies that VoIP traffic accepts better delay performance overall than that of HTTP traffic due to VoIP's small packet size and light traffic load.

It can be noticed that the HTTP packet delay performance does not change a lot within a certain range of variation of

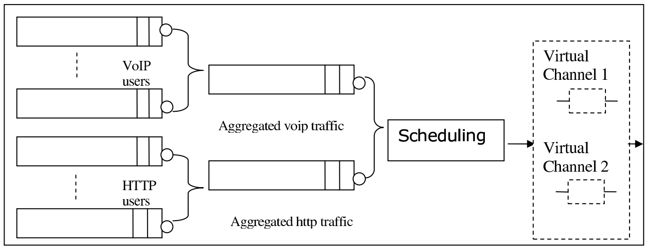


Fig. 7. Diagram of the Queues in System

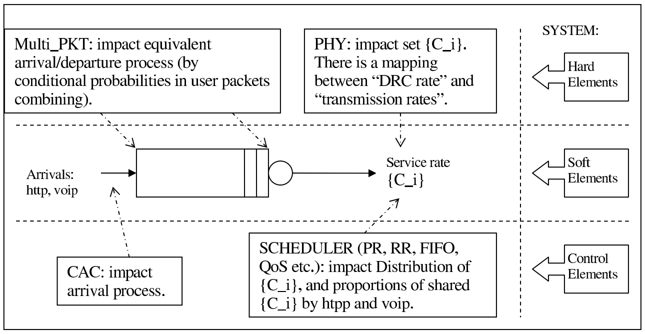


Fig. 8. Performance Analysis Framework

VoIP packet call arrival rate ($\leq 1/20$), so does VoIP packet delay performance within a certain range of variation of HTTP session arrival rate ($\leq 1/10$). This means that Erlang capacity of HTTP and VoIP traffic is not impact by each other within certain range of traffic load, impalying that effective capacity for different traffic classes is limited mostly by the traffic's own stochastic properties if system transmission mechanism is given. This is because the ON/OFF stochastic characteristics of the traffic flows leave opportunistic access chances for the resource sharing by other classes of traffic. This is an interesting observation because it will have no chances to see it if using full queue analysis.

IV. Performance Analysis

To understand performance of the network, we chose an approximate approach (than the accurate queue analysis) because complexity of the system makes analysis of the queue model intractable. We derive the delay performance for an equivalent single service queue fed by aggregated traffic flows. Main factors in the forward link are considered in the derivation by factored as equivalent parameters in the queue process. Figure 7 gives the diagram of the equivalent queue concept, and Figure 8 shows the framework of our analysis metholodgy.

By using the virtual service channel concept, we can analysis the queue system with multiple classes of traffic. Each class of traffic corresponds to one subqueue system with its own condiational probability of transmission rate. This probability is impacted by the scheduling algorithm implemented in the network. It can be calculated or measured from the system.

Figure 9 and Figure 10 show the examples of modeling the subqueue systems for VoIP and HTTP traffic, respectively. Since the Markov chain described by Figure 10 can be solved easily, we only display solving process for Figure 9.

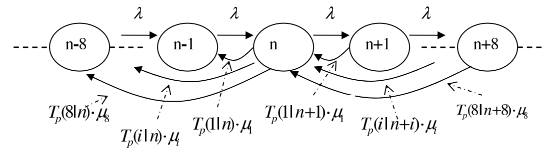


Fig. 9. VoIP Markov Chain

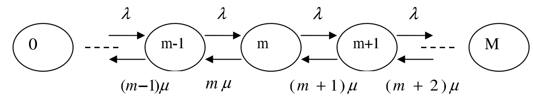


Fig. 10. HTTP Markov Chain

Defining a set of q_n as the probability mass function, it satisfies following equilibrium equations [6]:

$$q_n \cdot (\lambda + \sum_{i=1}^{min(n,8)} T_p(i|n) \cdot \mu_i) =$$

$$q_{n-1} \cdot \lambda + \sum_{i=1}^{8} q_{n+i} \cdot T_p(i|n+i) \cdot \mu_i \quad (1)$$

$$q_0 \cdot \lambda = \sum_{i=1}^{8} q_i \cdot T_p(i|i) \cdot \mu_i \quad (2)$$

where, λ is the aggregated packet arrival rate, μ_i is service rate to i-user packet and $T_p(i|n)$ is the probability the i users are severed by scheduling a i-user packet when there are n user packet queued. Therefore, the distribution $f(t)$ of user waiting time can be evaluated from:

$$W(s) = \sum_{n=1}^{\infty} q_n \cdot W_n(s) \quad (3)$$

where $W_n(s) = \sum_{j=1}^{min(n,8)} T_p(j|n) \cdot W_{n-j}(s) \cdot S_j(s)$. $W(s)$ and $W_n(s)$ are the Laplace transform of $f(t)$ and $f(t|n)$, respectively. $S_j(s)$ represents the Laplace transform of the service time distribution when j users are served.

To target at ξ percentage of user packet delay less than a required delay threshold under the given frame error rate, we obtain the system delay bound T_D from:

$$\int_0^{T_D} f(t) \cdot dt = \xi \quad (4)$$

V. Forward Link Erlang Capacity, QoS Scheduling and Call Admission Control

Figure 11 shows the capacity estimate from simulations under the quality criterion of 98% VoIP packet delay less than 200 ms and 80% HTTP packet delay less than 1000 ms. One curve describes the upbound of VoIP users limited by its corresponding HTTP conditions, and the other presents upbound of HTTP users with regarding to the VoIP limitations. Obviously, VoIP traffic obtains much better delay performance and it leaves a large space for putting HTTP traffic into the system if only providing best service to HTTP traffic. It is

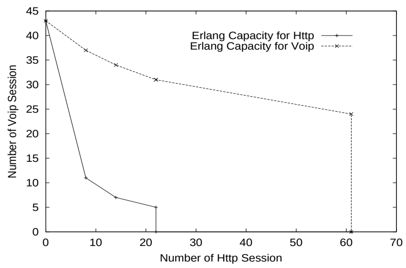


Fig. 11. Erlang Capacity for Mixed HTTP and VoIP Traffic

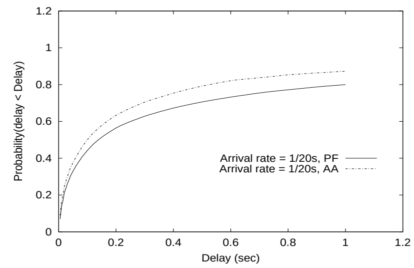


Fig. 12. Delay Performance for HTTP Traffic with PF and AA Scheduling

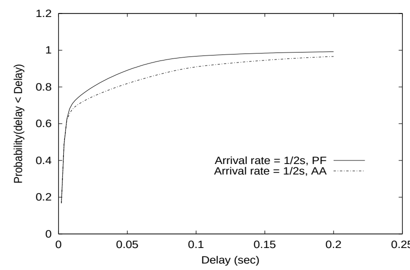


Fig. 13. Delay Performance for VoIP Traffic with PF and AA Scheduling

also seen that HTTP packet delay constraint limits the system capacity even if relaxing VoIP service quality requirement. Therefore, against to the intuition, HTTP flow may need to be given priority under circumstances for obtaining higher packet delay performance, while VoIP packet delay performance is over qualified. From this observation, we can conclude that, to provide QoS to multiple classes of traffic, scheduling and access control has to be carefully designed based on detailed QoS requirements to enhance effective capacity of the network. A good scheduling or CAC can move the HTTP capacity curve up and VoIP capacity curve down to balance the effective capacity for different services.

To illustrate the impact of scheduling in the multiple services network, we compare proportional fairness (PF) scheduling with an arrival-aware (AA) scheduling in which arrival rate per time unit to user queues is considered. From Figure 12 and Figure 13, it can be seen that AA scheduling works for HTTP traffic since it improves packet delay performance while it hurts a bit on VoIP's packet delay performance. The reason being that is the AA balances the queue lengths of HTTP users but reduces the efficiency of VoIP packet transmission comparing to PF scheduling.

A connection admission control policy can be design to guarantee QoS requirements for different classes of traffics by limitting concurrent sessions in the system following the dynamic of the Erlang capacity curve. The maximum number of sessions initiated in the system can be estimated from $Max\{Max(V+H), Max(H)+Max(V_0)\}$. The pair of H and V represents one solution for numbers of HTTP users and VoIP users on the capacity curve in Figure 11. V_0 is the maxmum number of VoIP users under which Erlang capacity of HTTP traffic is not affected by VoIP traffic load.

VI. Conclusion

This paper studies interactions and impacts from Web traffic and VoIP traffic in 1XEVDO Rev.A network, focusing on the forward link. We explore QoS performance for both applications. An equivalent queue system is built for performance analysis where realistic system factors, such as physical channel parameters, transmission formats and protocols, traffic statistical parameters, scheduling algorithms etc., are factored in the model. we found that effective capacitys for different applications are mainly impacted by the stochastic characteristics of the applications under certain range of traffic load. QoS scheduling and call admission control has to be careful designed for the mixed traffic when there are delay-sensitive and throughput-sensitive traffics in the network.

Our future work will focus on the impact of scheduing algorithms in the system with multiple QoS applications. Efficient QoS provisionning techniques will be investigated.

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