TCP OVER PACKET RADIO

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Abstract – A wireless packet data communications system utilizing General Packet Radio Service (GPRS) is studied. A traffic model that exhibits self-similarity is considered for wireless internet applications. The radio channel is modeled as a two-state Markov fading channel. The performance of TCP over such a packet radio link is obtained through analysis and simulation. Theoretical bounds are derived for system load and user performance. To gain insight to dimensioning of wireless packet data networks, TCP throughput and delay are studied with respect to the number of users and file size. The impact of radio link retransmission and system load on TCP timeout is also investigated.

I. INTRODUCTION

Transmission Control Protocol (TCP) has been successfully used in data networks to ensure end-to-end performance. The performance of TCP over wireless links can easily degrade due to handoff, high bit error rate and long roundtrip delay on the air interface. The TCP congestion control measures could cause unnecessary reduction of throughput in a wireless network [1].

High-speed wireless packet transmission such as General Packet Radio Service (GPRS) is being developed to accommodate the increasing demand for mobile internet. There are two major issues in deployment of wireless packet data, planning and policy implementation. Planning involves topology design and dimensioning while policy implementation refers to putting in policy and algorithms to ensure optimum performance in real-time. For both issues, it is important to study the user and system performance of TCP in packet radio environment.

Most studies on wireless packet data [1-5] focused on the link level performance. In this paper, we investigate end-to-end TCP performance over packet radio link. We study the TCP user throughput and file transmission delay with a self-similar internet traffic model and dynamic radio channel. The concerns about wireless TCP performance arise primarily because TCP misinterprets wireless loss or delay as result of congestion, causing TCP timeout and packet retransmission. The impact of retransmission over radio link on TCP timeout and retransmission is addressed. The main interest is to study the TCP performance in wireless packet radio such as GPRS with respect to system load and traffic profile. Because the load is a function of traffic pattern and user behavior in packet data system, the study also estimates system load based on the number of users and traffic profile.

The rest of the paper is organized as follows. Section II describes a wireless protocol model, a fading channel model and a traffic model for wireless packet data. Section III presents theoretical analysis of system load, user throughput and delay. Section IV discusses simulation results. Finally, Section V concludes the paper.

II. MODELS

In this section, we briefly describe a protocol model for TCP over packet radio. We also discuss a Markov model for Rayleigh fading channel and an internet traffic model used in the study.

Protocol Model

A simplified model of protocol stack for packet data communications is shown in Fig. 1. In GPRS, TCP packets are first added with Internet Protocol (IP), Subnetwork Dependent Convergence Protocol (SNDCP) and Logical Link Control (LLC) related overhead and then forwarded to Radio Link Control and Medium Access Control (RLC/MAC) layer. RLC/MAC layer adds its own overhead and then forwards it to the physical layer supported by radio frequency. For this model, we assume that the Round Trip Time (RTT) between the base station (BS) and TCP server is negligible compared to the time spent on the radio link. The SNDCP and LLC layers generate relatively fixed overhead, so they are not included in the model. The RLC layer uses selective ARQ, error correction and detection to provide reliable data transmission. When sending data, RLC uses an NAK-based selective repeat to eliminate all detectable errors. The physical layer below RLC uses packet data traffic channel (PDTCH).

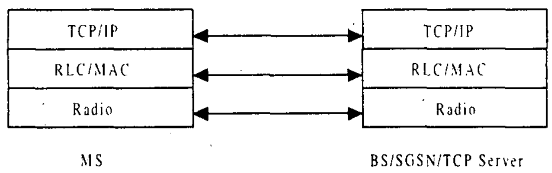

Fig. 1. Model for TCP/IP over GPRS.

Markov Channel Model

We consider a two state Markov model [4] for representing a wireless channel. In Fig. 2, state 0 represents the quiet state in which the BER (ε_0) is low and single bit errors are produced independent of other errors. State 1 represents the noisy or burst state, where the BER (ε_1) is high. The parameters of the model can be derived from the following equations.

$$\varepsilon_0 = d\bar{\varepsilon} \tag{1}$$

$$\varepsilon_1 = \frac{1-d+d^2}{d}\bar{\varepsilon} \tag{2}$$

$$p = \frac{d}{b(1-d)} \tag{3}$$

$$p' = 1 - \frac{1}{b} \tag{4}$$

where, d is the duty cycle of state 1, $\bar{\varepsilon}$ is the average BER, and b is the average burst length in state 1. In this work, we consider a burst channel with a duty cycle of 0.28, an average BER of 1.8% and an average burst length of 25 (500 ms). This translates to $\varepsilon_0 = 0.5\%$ and $\varepsilon_1 = 5\%$. The same analysis can be done for other types of channel.

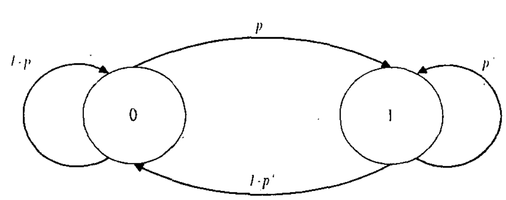

Fig. 2. Two-state Markov channel model.

Traffic Model

Recent research demonstrated that Poisson process and its variations do not capture some of the important characteristic of internet date traffic. For example, the duration of the ON and OFF periods are found to have a heavy-tail distribution [6]. The heavy-tail distribution leads to the self-similar behavior of the aggregated traffic. As the wireless internet becomes more popular, it is important that the underlying traffic model capture the unique characteristic of packet date traffic.

For the analysis and simulation in this paper, a WWW traffic model that exhibits the ON/OFF pattern is considered. A similar model has been described in [7].

The ON period represents the file transmission time on the downlink. The OFF period represents the interval when there is no data transmission. The ON time is a function of the file size, available system bandwidth and the number of simultaneously transmitting users. The file size follows a Pareto distribution of the first kind

$$F(x) = 1 - \left(\frac{k}{x}\right)^{\alpha} \tag{5}$$

where the shape parameter $\alpha = 1.5$ and the minimum file size $k = 4000$ bytes. This leads to an average file size of 12000 bytes. These values are taken from traditional internet traffic and may need adjustment for new wireless internet technologies.

The OFF period follows an exponential distribution with a mean idle time of 90 seconds. The default parameters used in this paper are summarized in Table 1.

Table 1 Traffic Parameters

Number of PDTCHs	8
Max. number of PDTCHs a terminal can access	8
WWW file size	Pareto with $\alpha = 1.5$ and $k = 4000$
Idle time	Exponential with mean 90 seconds

III. ANALYSIS

In this paper, user performance is represented by throughput and delay. Assuming volume-based charging, an appropriate indicator of system performance is its load. The objective of a system designer is to maximize system performance while maintaining a level of user performance. User and system performance relies heavily on user behavior, specifically, file or traffic arrival pattern. We consider two file arrival scenarios, corresponding to the highest and lowest possible system load for a given user application traffic profile.

In general, after downloading a file a user will stay idle for a period of time before initiating another download. Following assumptions on mobile station (MS)'s access to a radio channel are made in the analysis. An MS can access any number of (up to eight for GPRS) time slots of a downlink channel and MS are all in the same class. That is, if only one MS is receiving at a time, the MS will occupy all eight time slots. If more than one MS are receiving at a time, they will share the time slots in a round-robin manner.

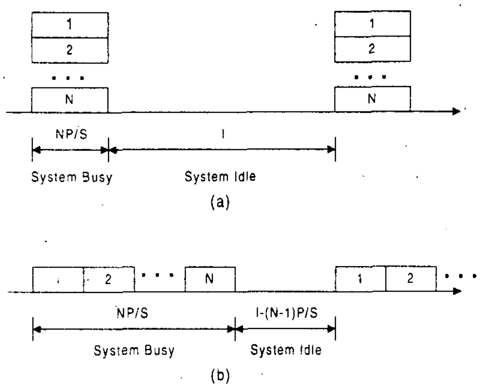

Fig. 3. File arrival scenarios.

Let us consider a file arrival scenario shown in Fig. 3(a). The files from N users arrive at the base station's transmitter simultaneously with same average file size of P (bits). The system transmits these files to MS's at the same rate and all files are delivered at the same time. After delivering the files, the system stays idle until the next batch of file arrivals. Assuming a system throughput of S (bps) and an idle interval of I (sec), the system will be busy for an interval of $B = NP/S$ and idle for $T_1 = I$. The system load U_1, user throughput S_1 and file transmission delay D_1 can be expressed as

$$U_1 = \frac{B}{T_1 + B} = \frac{NP}{SI + NP} \tag{6}$$

$$S_1 = \frac{S}{N} \tag{7}$$

$$D_1 = \frac{NP}{S} \tag{8}$$

In the other scenario, file arrivals from the N users are sequential and do not overlap with each other during packet transmission. It means that at most one file is served by the system at any time as shown in Fig. 3(b). The system busy time (B) in the cycle remains as NP/S. Between ending of first user's file and beginning of its next arrival, there are (N-1) files transmitted separately. The system idle time in this cycle is $T_2 = I - (N-1)P/S$. The system load, user throughput and transmission delay can be expressed as

$$U_2 = \frac{B}{T_2 + B} = \frac{NP}{SI + P} \tag{9}$$

$$S_2 = S \tag{10}$$

$$D_2 = \frac{P}{S} \tag{11}$$

The two scenarios represent the worst and best cases. In the first scenario (worst), files from all users arrive at the same time thus have lowest throughput and longest delay. In the second scenario (best), all users have file arrivals at different times thus have highest throughput and shortest delay. From the system point of view, the first scenario has the maximum amount of idle time and carries the minimum load. In contrast, the second scenario carries the maximum load.

As the number of users and file size increase, the system load increases approximately linearly and the user throughput degrades from S to S/N. The transmission delay increases linearly with file size.

IV. SIMULATION RESULTS

The simulation is based on the model in Fig. 1. The TCP algorithms by Jacobson [8] and Karn [9] are implemented. The TCP behavior in dynamic radio environment is simulated. Throughput and delay are evaluated for the TCP link in an established state. The GPRS coding scheme 3 (CS-3 or 3/4 convolutional code) is used. The maximum throughput on radio link is 112 kbps for CS-3. Note that adaptive channel coding may be used to improve performance [5] but is not considered here. The channel fading rate is represented by the average burst length in the 2-state Markov channel model. The average burst length in the simulation is 500 ms. Other parameters for the channel model are specified in Section II. Note that we assume all mobile stations are able to access eight time slots. This is not going to be case in real networks during initial deployment of GPRS. However, the same model can be used to simulate mobile stations that can only access one, two or four time slots.

Fig. 4 shows the user throughput with respect to the number of users. The average file size is 12 kbytes (kB). The throughput degrades approximately linearly with increasing number of users. Note that the user throughput is much less than the system throughput because of TCP slow start, radio link retransmission, and TCP/IP protocol overhead.

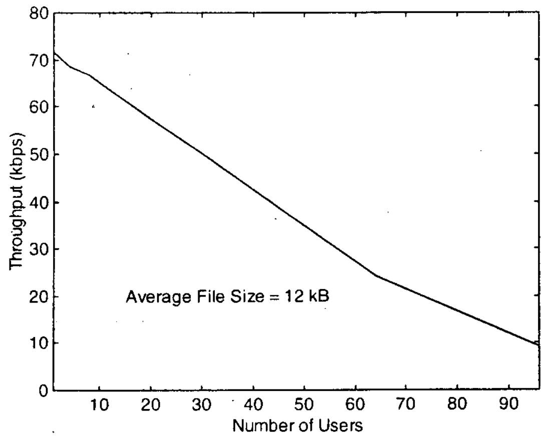

Fig. 4. User throughput with respect to number of users.

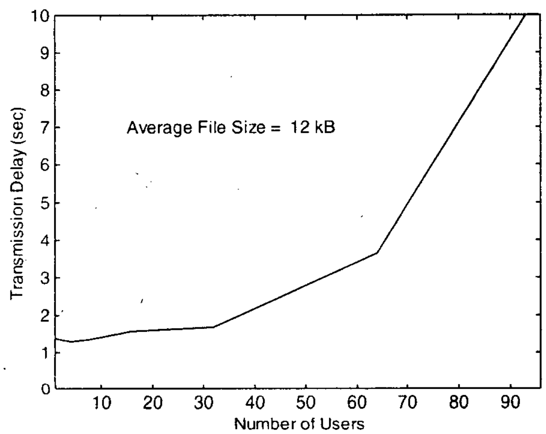

Fig. 5. Delay with respect to number of users.

Fig. 5 shows the file transmission delay with respect to the number of users. It is seen that when the number of users is small, corresponding to a light system load, the delay is small and almost constant. In fact the delay is mainly dependent on the file size in this case. When the load increases, however, the delay increases rapidly because multiple users compete for transmission resource.

Now let us consider the case with a fixed number of users and variable file size. Figs. 6 and 7 show the user throughput and delay with respect to the file size for 16 users. For small file sizes, the user throughput is relatively constant. The throughput is reduced for very small file size because of the slow start nature of TCP and relatively large overhead. As the file size increases, the throughput decreases because overlapping transmissions increase and multiple users share the radio channel. If the file size continues to grow, the throughput approaches the minimum given by (7).

It is seen in Fig. 7 that the delay increases approximately linearly with the file size. This is also due to increasing overlapping transmissions for large file sizes. The delay is linearly bounded by (8) and (11).

More simulation was performed to investigate the tradeoff between the number of users and file size while keeping the load constant [5]. The result indicates that as expected the user throughput is in favor of small number of users and large file size while the delay is exactly the opposite. Note that under light to medium system load the throughput would decrease and delay would increase if the mobile station cannot access all eight time slots. The difference diminishes for heavy load.

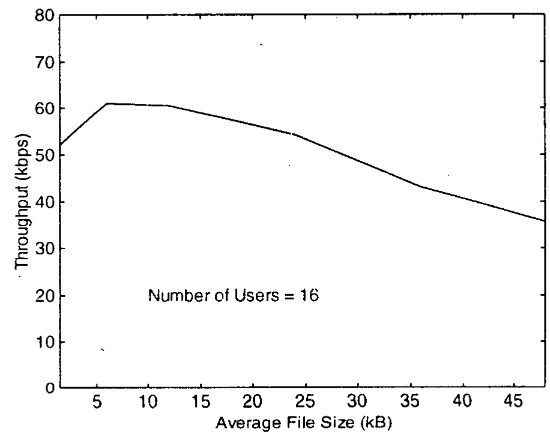

Fig. 6. User throughput with respect to file size.

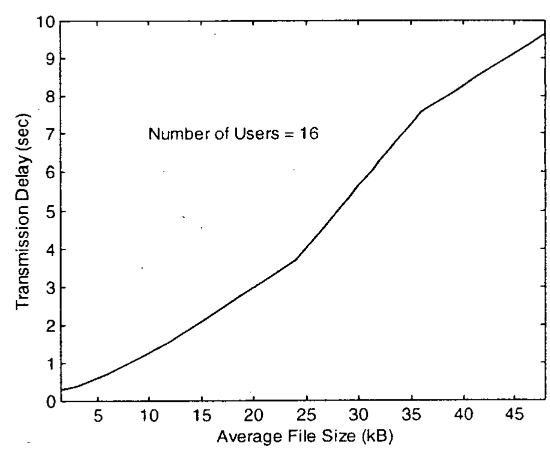

Fig. 7. Delay with respect to file size.

The wireless link-related TCP timeouts and the resulting packet retransmissions have been among the main concerns for wireless internet performance. TCP protocols use a timer at the sender side to decide if the round trip delay between the transmitter and receiver exceeds a threshold. If the delay exceeds the threshold, TCP will timeout and retransmit the packet. Since the threshold takes into account the average and some deviation of the current estimate of round-trip delay, timeout will only occur when there is a large delay variation.

In wireless packet data environment, the following two scenarios may create significant packet delay fluctuation. In the first scenario, the number of users in the system varies greatly and quickly, so the time required to deliver a TCP packet may be significantly different depending on the available radio resource allocated to a particular user. A sudden increase of the number of users may cause TCP timeout. In the second scenario, radio channel stays in the noisy state for an extensive period of time so continuous retransmissions are necessary at the radio link layer. The round trip delay could vary from relatively short in the quiet state to very long in the noisy state. In short, TCP timeouts may be caused by sudden increase of system load or continuous radio retransmissions in the noisy state.

Let us analyze the two effects in detail. It is assumed that TCP uses the default packet size of 576 bytes (including 40 bytes TCP/IP header) and the data rate of eight time slots with GPRS CS-3 is 112 kbps. In the simulation, the average and minimum deviation of RTT are about 50 ms and 100 ms. The Round Trip Timeout (RTO) threshold is the sum of average and four times deviation of RTT, or 450 ms, in the quiet state. A TCP timeout may occur if the number of users in the system changes suddenly from 1 to 10 because RTT will change from 50 ms for single user to 500 ms for ten users. In reality, this type of fast change is very unlikely. Even if it occurs, the problem can be solved with proper scheduling mechanism such as gradually reducing the radio resource for the existing users.

Let us consider the impact of radio link retransmission on TCP timeouts. The frame error rate (FER) is about 0.68 in the noisy state, so the average time to transfer a radio frame successfully in the noisy state is about 3 times the time in the quiet state. The average RTT is 150 ms in the noisy state and 50 ms in the quiet state, respectively. Therefore, with the current model timeouts would not normally occur due to radio link retransmission in the noisy state.

Indeed, TCP timeouts are very rare in the simulation. Furthermore, the user throughput recovers very fast from slow start activated by TCP timeout if the RTT delay is mainly decided on the radio link. The time to deliver a TCP packet from BS to MS is about 50 ms and the TCP acknowledgment from MS to BS will take another 50 ms. At slow start, the throughput is about 46 kbps since it takes 100 ms to deliver a 576-byte packet. It will reach the maximum rate of 112 kbps once the TCP window size reaches two. The TCP sender could then transmit packets continuously after the first acknowledgment is received. Therefore the impact of slow start on the user throughput is very little if the radio link delay is the dominant factor in the entire system.

V. CONCLUSIONS

The performance of TCP over packet radio has been studied. Using a Markov channel model and an internet traffic model, user throughput and delay with different load conditions and traffic profile have been simulated. The results indicate that the user throughput is more sensitive to the number of users while the delay is more sensitive to the file size. TCP timeouts associated with system load and radio retransmission have been analyzed. Packet data transmission that incorporates quality of services is currently under study.

VI. REFERENCES

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