무선 네트워크에서 QoS 보장을 위한 딜레이 성능 분석

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Analysis of Delay Performance for QoS Support in Wireless Networks

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요 약

무선 링크 상에서의 QoS 제공을 위해서는 트래픽 소스와 무선 채널 그리고 오류 제어 방식간의 상호 작용에 대해 정확히 이해하여야 하고 또한 이를 계량화 하는것이 필요하다. 이 논문에서는 이와 같은 상호작용을 네트워크 계층 관점에서 분석하고 이를 딜레이 성능의 분석에 적용하였다 시간에 따라 전송 용량이 변하는 무선 링크 상에서 on/off 트래픽 소스가 데이타를 전송하는 시나리오를 가정하였다. 패킷 딜레이 분포는 uniformization과 Laplace transform이라는 두가지 방식으로 유도 되었고 두 방식간의 장단점이 비교되었다 수식으로 표현된 딜레이 분포는 다시 딜레이에 대한 서비스 요건이 주어진 경우, 무선 유효 대역폭 (wireless effective bandwidth) 이라는 양을 계산하기 위해 사용되었다 수치적인 분석과 시뮬레이션을 통해서 이 논문에서 제안한 분석의 정확도를 검증하였고 오류 제어와 대역폭 할당이 패킷 딜레이 성능에 미치는 영향을 분석하였다

Key Words Wireless networks, QoS, delay distribution, fluid analysis.

ABSTRACT

Providing quality of service (QoS) guarantees over wireless links requires thorough understanding and quantification of the interactions among the traffic source, the wireless channel, and the underlying error control mechanisms. In this paper, we account for such interactions in a network-layer model that we use to investigate the delay performance of a wireless channel We consider a single ON/OFF traffic stream transported over a wireless link. The capacity of this link fluctuates according to a fluid version of Gilbert-Elliot's model We derive the packet delay distribution via two different approaches. uniformization and Laplace transform. Numerical aspects of both approaches are compared The delay distribution is further used to quantify the wireless effective bandwidth under a given delay guarantee. Numerical results and simulations are used to verify the adequacy of our analysis and to study the impact of error control and bandwidth allocation on the packet delay performance.

I Intorduction

Current trends in wireless networks indicate a desire to provide a flexible broadband wireless infrastructure that can support emerging multimedia services as well as traditional data

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services [1,2,3] In such a multi-service wireless environment, quality-of-service (QoS) guarantees are critical for real-time voice and video. Compared to its wireline counterpart, the provisioning of QoS guarantees over wireless links is a more challenging problem whose difficulty stems from the need to explicitly consider the harsh radio-channel transmission characteristics and the underlying link-layer error control mechanisms This difficulty is further compounded by host mobility and how it impacts the available bandwidth capacity These difficulties indicate a clear need for a general QoS framework in the wireless environment.)

QoS guarantees in wireless networks can be provided through a coordination between connection-level bandwidth reservation and packet-level scheduling. Most previous research on QoS over wireless networks has mainly focused on these issues Reininger et al. identified the high variability of traffic dynamics of mobile multimedia applications as a function of time and space, and proposed a soft QoS control which allows bandwidth renegotiation according to the varying traffic conditions [4] Lu et al proposed a fair scheduling algorithm with adaptation to wireless networks that take into account bursty and location-dependent channel errors [5] Although their work identified many practical issues, it did not address the interaction between packet scheduling and error control. Wu and Negi proposed a link-layer channel model named the effective capacity (EC) model [6] In this approach, a wireless link is modelled by two parameters, namely, the probability of nonempty buffer and QoS exponent Then these are used to derive QoS metrics such as delay bounds.

The primary goal of the underlying work is that we study the delay performance over a wireless link and investigate its implication on optimal bandwidth allocation under delay guarantees Our investigations are carried out for a single stream that is transported over a time-varying wireless link If the link is used to transport more than one connection, then each connection is guaranteed a constant service rate during its active period (i.e., TDMA style). The outcome of packet transmission is determined by the state of the wireless channel and the error control schemes This scenario encompasses point-to-point connections between mobile terminals (MT) and a base station (BS) in the cellular communication systems.

To achieve our goals, we follow a fluid-based approach whereby the traffic source is modeled by an on-off fluid process and the channel is modeled by a fluid variant of Gilbert-Elliott's model Using fluid-flow analysis, we compute the delay distribution for a single stream, which is a function of the traffic source, the service rate, the wireless channel, and the error control schemes To obtain the delay distribution at the transmitter buffer, we first evaluate the queue length distribution taking into account the channel behavior and the underlying error control schemes We provide two alternative approaches for obtaining the delay distribution via the uniformization and Laplace transform techniques The two techniques differ in how the accumulative amount of service is determined Numerical aspects of both approaches are compared. In particular, we derive the closed-form expression for the delay distribution using the uniformization technique The analytical results are used to obtain the wireless effective bandwidth under the delay constraints and to investigate the optimal error control strategy that minimizes the use of bandwidth while guaranteeing the QoS Extensive simulations are conducted to verify the goodness of our analytical results.

The rest of the paper is organized as follows In Section 2, we describe the wireless link model. Analysis of the delay performance is provided in Section 3. Numerical results and simulations are reported in Section 4, followed by concluding remarks in Section 5.

II. Wireless Link Model

2.1 Framework

In order to analyze the packet-level performance of a wireless link, we consider a framework in which traffic streams from one or more connections are fed into a finite-size FIFO buffer. A constant service rate c (in packets/second) is assigned to the wireless connection, but the actual drain rate observed at the buffer is reduced due to retransmissions and FEC overhead. In our study, we consider a particular hybrid ARQ/FEC approach in which the cyclic redundancy check (CRC) code is applied first to a packet, followed by FEC. We assume that the CRC code can alone detect almost all bit errors in a packet. In contrast, only a subset of the errors can be corrected by FEC. In addition, we impose a limit on the number of packet transmissions. Imposing such a limit can be used to provide delay guarantees for real-time traffic. Once a packet hits the limit, it will be discarded. We define the packet discard rate (PDR) as the ratio of packets reaching the limit over the total number of transmitted packets (excluding lost packets due to buffer overflow).

The above model has three control parameters: the service rate (or assigned bandwidth), the FEC code rate, and the limit on the number of transmissions. From the network point of view, the selection of these parameters is very crucial and requires thorough understanding of their impact on the packet-level performance. The main theme of this study is to investigate the packet-level performance of a wireless link as a function of the assigned bandwidth, the limit on transmissions, and error control schemes.

2.2 Queueing Model

In this section, we describe the queueing model that is used to analyze the packet-level performance for a single traffic source transported over a wireless link. The source is characterized by an on-off fluid process with peak rate r. Its on and off periods are exponentially distributed with means $1/\alpha$ and $1/\beta$, respectively. The wireless channel is modeled using a fluid version of Gilbert-Elliott (GE) model which is often used to investigate the performance over wireless links [7]. As explained in Figure 1, the GE model is Markovian with two alternating states: Good and Bad. The bit error rates (BER) during the Good and Bad states are given by P_{eg} and P_{eb}, respectively, where $P_{eg} \ll P_{eb}$. The durations of the Good and Bad states are exponentially distributed with means $1/\delta$ and $1/\gamma$, respectively.

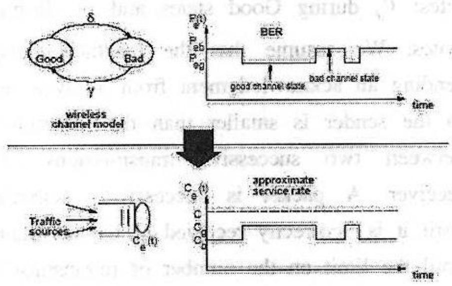

Figure 1: Wireless channel model and corresponding service rate model.

The FEC capability in the underlying hybrid ARQ/FEC mechanism is characterized by three parameters: the number of bits in a code block (n), the number of payload bits (k), and the maximum number of correctable bits in a code block (τ). Note that n consists of the k payload bits and the extra parity bits. The FEC code rate $e(\tau)$ is defined as

$$e(\tau) = \frac{k}{n(\tau)}.$$

Assuming that a FEC code can correct up to τ bits and that bit errors during a given channel state are independent, the probability that a packet contains a non-correctable error is given by:

$$P_c(p_b, \tau) = \sum_{j=\tau+1}^{n(\tau)} \binom{n(\tau)}{j} p_b^j (1-p_b)^{n(\tau)-j} \quad (1)$$

where p_b is the bit error probability; $p_b \in \{P_{eg}, P_{eb}\}$ To account for the FEC overhead, we obtain the actual service rate c_e observed at the output of the buffer.

$$c_e = c \cdot e(\tau) \quad (2)$$

where c is the bandwidth assigned to the connection

The exact behavior of ARQ and FEC in the underlying queueing model is difficult to analyze. To obtain analytically tractable results, we assume that the packet departure process follows a fluid process with a service rate that is modulated by the channel state (see Fig.1). This approximation implies that there are two deterministic service rates c_g during Good states and c_b during Bad states We assume that the feedback delay for sending an acknowledgment from a given receiver to the sender is smaller than the minimum time between two successive transmissions to that receiver A packet is successively retransmitted until it is correctly received at the destination or until the limit on the number of retransmissions is reached. In this scenario, the total time needed to successfully deliver a packet conditioned on the channel state follows a truncated geometric distribution Let N_{tr} denote the number of retransmissions until a packet is successfully received or is discarded because it reached the limit on retransmissions For a given packet error probability P_c and a limit on transmissions N_l, the expected value of N_{tr} is given by:

$$E[N_{tr}] = \frac{1-P_c^{N_l}}{1-P_c} \quad (3)$$

Thus, c_g and c_b correspond to them ean transmission rates of the truncated geometric trial with parameters $(P_{c,g}, N_l)$ and $(P_{c,b}, N_l)$, respectively, where $P_{c,g}$ and $P'_{c,b}$ are the packet error probabilities in Good and Bad states, respectively, given by (1) Formally,

$$c_g = \frac{c \cdot e(\tau) \cdot (1-P_{c,g})}{1-P_{c,g}^{N_l}} \quad (4)$$

$$c_b = \frac{c \cdot e(\tau) \cdot (1-P_{c,b})}{1-P_{c,b}^{N_l}} \quad (5)$$

where
$P_{c,g} = P_c(P_{e,g}, \tau)$ and $P_{c,b} = P_c(P_{e,b}, \tau)$

Ⅲ. Analysis of Delay Performance

Following the discussion in the previous section, we construct the Markovian queueing system with four states as shown in Figure 3. Let S denote the state space. Thus,

$$S = \{(0,g),(0,b),(1,g),(1,b)\} \quad (6)$$

where 0 and 1 denote the on and off states of a traffic source, respectively, and g and b denote Good and Bad channel states, respectively

Following a standard fluid approach (see [8], for example), the evolution of the buffer content can be described by the following differential equation.

$$\frac{d\Pi(x)}{dx} D = \Pi(x) M \quad (7)$$

where

$$D \triangleq diag[-c_g, -c_b, r-c_g, r-c_b],$$

$$\Pi(x) \triangleq \{\Pi_{0,g}(x)\Pi_{0,b}(x)\Pi_{1,g}(x)\Pi_{1,b}(x)\},$$

$$\Pi_s(x) \triangleq Pr\{buffer\ content \le x\ and\ the\ systems \in S\},$$

and M is the generator matrix of the underlying Markov chain

$$M = \begin{bmatrix} -(\beta+\alpha) & \delta & \beta & 0 \\ \gamma & -\beta+\gamma & 0 & \beta \\ \alpha & 0 & -(\alpha+\delta) & \delta \\ 0 & \alpha & \gamma & -(\alpha+\gamma) \end{bmatrix}$$

Throughout the paper, matrices and vectors are boldfaced

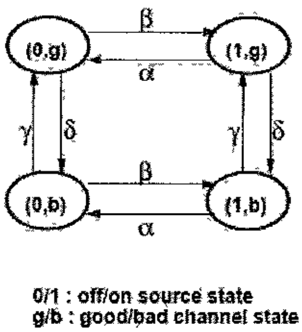

Figure 2. State transition diagram

The solution of (7) corresponds to the solution of the eigenvalue/eigenvector problem

$$z\phi D = \phi M \tag{8}$$

which is generally given by

$$\Pi(x) = \sum_{z_i \le 0} a_i exp(z_i x)\phi_i \tag{9}$$

where $a_i's$ are constant coefficients and the pairs $(z_i, \phi_i), i = 1, 2, \cdots,$ are the eigenvalues and the right eigenvectors of the matrix MD^{-1} [8,9]. Let w denote the stationary probability vector of the Markov chain; w satisfies $wM = 0$ and $w1 = 1$, where 1 is a column vector of ones. Then w is given by

$$w = \frac{1}{(\alpha+\beta)(\delta+\gamma)}[\alpha\gamma \ \alpha\delta \ \beta\gamma \ \beta\delta]. \tag{10}$$

In order to solve (8), we follow the approach used in [9]. The four-state Markov process is decomposed into two processes; one describes the on-off source and the other describes the state of the channel.

After obtaining the eigenvalues, the eigenvectors, and the coefficients, we can construct the stationary buffer content distribution $\Pi(x)$. Consequently, the packet loss rate due to buffer overflow G(x) is given by

$$G(x) = 1 - 1\Pi(x) \tag{11}$$

In the following section, $\Pi(x)$ is used to obtain the delay distribution.

3.2 Delay distribution

In fluid queueing models with an error-free channel and constant service rate, e.g., ATM link, the packet delay distribution can be directly obtained from the queue length distribution [10]. However, the scenario we consider in this study includes a time-varying wireless channel that is being approximated by a two-state Markov modulated fluid process.

We assume an infinite-capacity buffer. Let D denote the delay experienced by an arriving packet. Let C(t) denote the accumulative amount of service during a period of length t.

$$C(t) = \int_0^t c(s)ds$$

where c(s) is the service rate at the time s. The channel state at time t is denoted by $h(t) \in \{g, b\}$, where g and b denote Good and Bad states, respectively. The probability that the delay seen by a packet is less than or equal to t is equal to the probability that C(t) is greater than or equal to the queue length Q_0 at the instant of the packet arrival. Thus, we have

$$Pr[D \le t] = Pr[C(t) \ge Q_0]$$

$$= \sum_{i \in S} \int_{0^-}^{\infty} Pr[C(t) \ge x | i, Q_0 = x]\pi_i(x)dx$$

$$= \frac{r}{T}\int_{0^-}^{\infty} Pr[C_g(t) \ge x]\pi_{1,g}(x) + Pr[C_b(t) \ge x]\pi_{1,b}(x)dx \tag{12}$$

where T is the throughput, π_i is the pdf of the queue length in a state i, $i \in S$, and

$$C_i(t) \triangleq \int_0^t c(s)ds \text{ given } h(0) = i, \text{ for } i \in g, b$$

The quantity $r\pi(x)/T$ represents the fraction of carried flow that arrives at the queue when its content is x. For an infinite-capacity buffer, the throughput T is given by:

$$T = r(w_{1,g} + w_{1,b}). \tag{13}$$

In order to obtain $Pr[C_i(t) \le x], i \in \{g, b\}$, we provide two methods. direct calculation using Laplace transform and uniformization The equivalence of these approaches will be verified using numerical examples.

Laplace Transform Approach

For numerical convenience, we transform the random variable $C_i(t)$ to $\overline{C}_i(t)$ defined as:

$$\overline{C}_i(t) \triangleq C_i(t) - c_b t, \quad i \in g, b.$$

By this transformation, $\overline{C}_i(t)$ is the accumulative service resulting from a "normalized" channel with service rates $c_g - c_b$ (during Good states) and 0 (during Bad states) Note that the minimum amount of accumulative service in a period of length t is $c_b t$. Thus,

$$Pr[C_i(t) \geq x] = \begin{cases} 1, & if\ x < c_b t \\ 1 - Pr[\overline{C_i(t)} \leq x - c_b t], & if\ x \geq c_b t \end{cases}$$

The following proposition gives the probabilities $Pr[C_g(t) \geq x]$ and $Pr[C_b(t) \geq x]$ by solving the partial differential equations (PDE) for the consumption rate C(t).

Proposition 3.1

The probabilities $Pr[C_i(t) \geq x]$, $i \in \{g, b\}$ when $x \geq c_b t$, are given by

$$\Pr[C_g(t) \geq x] = e^{-\delta x}\Big(e^{-\gamma(t-\hat{x})} J_0(2\sqrt{-\delta\gamma\hat{x}(t-\hat{x})}) + \sum_{n=0}^{\infty} \frac{\delta\hat{x}}{(n!)^2}\Gamma(n+1, \gamma(t-\hat{x}))\Big) \quad (14)$$

$$Pr[C_b(t) \geq x] = e^{-\delta x}\sum_{n=0}^{\infty} \frac{\delta\hat{x}}{(n!)^2}\Gamma(n+1, \gamma(t-\hat{x})) \quad (15)$$

where $\hat{x} = (x - c_b t)/(c_g - c_b)$, $J_0(z)$ is the Bessel function of order 0 given by

$$J_0(z) = \sum_{n=0}^{\infty} \frac{(-1)^n (z/2)^{2n}}{(n!)^2},$$

and

$\Gamma(n, z)$ is the incomplete gamma function given by.

$$\Gamma(n, z) = \int_0^z e^{-x} x^{n-1} dx$$

Equation (15) in the Proposition 3.1 is substituted into (12) to evaluate the delay distribution Numerical complexity of Equation (14) and (15) is associated with the infinite sum in them We observe that the value of the infinite sum converges fast with moderate iteration, e.g., n=20.

Uniformization Approach

As a second approach to obtaining $Pr[C_i(t) \leq x], i \in \{g, b\},$, we use the uniformization approach.

Let t_g and t_b denote the accumulative sojourn times of Good and Bad channel states during an interval of length t, respectively. That is,

$$t_g \triangleq \int_0^t 1_{\{h(s)=g\}} ds$$

$$t_b \triangleq \int_0^t 1_{\{h(s)=b\}} ds$$

Then, the accumulative service C(t) is given by

$$C(t) = c_g t_g + c_b t_b, \quad 0 \leq t_g,\ t_b \leq t. \quad (16)$$

Since $t = t_{g+t_b}$, C(t) can be expressed as

$$C(t) = c_g t_g + c_b(t - t_g) \quad \text{or}$$
$$C(t) = c_g(t - t_b) + c_b t_b \quad (17)$$

Proposition 3.2

The probabilities $Pr[C_g(t) \geq x]$ and $Pr[C_b(t) \geq x]$ are given by

$$Pr[C_g(t) \geq x] = 1 - e^{-(\delta+\gamma)t}\sum_{n=1}^{\infty} \frac{(\delta t)^n}{n!}\sum_{k=1}^{n}\binom{n}{k-1}\left(\frac{\gamma}{\delta}\right)^{k-1}\sum_{i=k}^{n}\binom{n}{i}\chi^i(1-\chi)^{n-i} \quad (18)$$

and

$$Pr[C_b(t) \geq x] = e^{-(\delta+\gamma)t}\sum_{n=1}^{\infty} \frac{(\gamma t)^n}{n!}\sum_{k=1}^{n}\binom{n}{k-1}\left(\frac{\delta}{\gamma}\right)^{k-1}\sum_{i=k}^{n}\binom{n}{i}\chi^{n-i}(1-\chi)^i \quad (19)$$

where $\chi = \frac{x - c_b t}{(c_g - c_b)t}$.

Up to this point, we have discussed the alternative approaches to obtaining the probability $Pr[C_i(t) \geq x], i \in \{g, b\}$. In the following, the results from Proposition 3.2 are used to obtain the delay distribution The following Proposition gives the closed-form expression for delay distribution by substituting (18) and (19) into (20)

Proposion 3.3

$$\Pr[D \leq t] = \frac{r}{T}(\Pi_{1,g}(c_g t) + \Pi_{1,b}(c_b t))$$
$$- \frac{r}{T} e^{-(\delta+\gamma)t} \sum_{n=1}^{\infty} \frac{(\delta t)^n}{(n+1)!} \sum_{k=1}^{n} \binom{n}{k-1} \left(\frac{\gamma}{\delta}\right)^{k-1}$$
$$\cdot \sum_{i=k}^{n} (c_g - c_b) t \sum_l a_l e^{z_l c_b t} \Phi(i+1, n+2, z_l(c_g - c_b)t)$$
$$+ \frac{r}{T} e^{-(\delta+\gamma)t} \sum_{n=1}^{\infty} \frac{(\gamma t)^n}{(n+1)!} \sum_{k=1}^{n} \binom{n}{k-1} \left(\frac{\delta}{\gamma}\right)^{k-1}$$
$$\cdot \sum_{i=k}^{n} (c_g - c_b) t \sum_m a_m e^{z_m c_b t} \Phi(n-i+1, n+2, z_m(c_g - c_b)t) \quad (20)$$

where $\pi_{1,g}(x) = \sum_l a_l e^{z_l x}$, $\pi_{1,b}(x) = \sum_m a_m e^{z_m x}$, the indexes l, m are used to index the negative eigenvalues, and

$$\Phi(x, y; z) \triangleq \sum_{k=0}^{\infty} \frac{(x)_k}{(y)_k} \frac{z^k}{k!}$$

with $(a)_n \triangleq a(a+1)\cdots(a+n-1)$

Wireless Effective Bandwidth

The notion of effective bandwidth has been employed to achieve efficient provisioning of QoS guarantees in ATM networks [12]. In this study, we extend the notion for a wireless connection under probabilistic delay constraints

We defined the wireless effective bandwidth c_{eb} under the delay constraint $Pr[delay > t] = \epsilon$ by

$$c_{eb} \triangleq \min\{c \mid c \text{ satisfies } \Pr[delay > t] = \epsilon\} \quad (21)$$

where c is the service rate. In contrast to wireline effective bandwidth, the wireless effective bandwidth is configured along with the optimal number of correctable bits which minimizes the use of bandwidth while providing the requested reliability at the physical link In Section 4, we provide the numerical examples on this issue and investigate the characteristic of the pair of QoS parameters (c_{eb}, τ) in detail.

IV. Numerical Results and Discussion

In this section, we present numerical examples of our analytical results. We verify the adequacy of these results by contrasting them against more realistic simulations.

Similar to the analysis, the simulation results are obtained using on-off traffic sources with exponentially distributed on and off periods The ARQ retransmission process is simulated in a more realistic manner, whereby a packet is transmitted repeatedly until it is received with no errors or until it reaches the limit on the number of transmissions The probability of a packet error is computed from (1) for both channel states Transitions between Good and Bad states are assumed to occur only at the beginning of a packet transmission slot. A packet is retransmitted if it has uncorrectable errors It is assumed that the propagation delay is small, so that the ACK/NAK message for a packet is received at the sender before the next attempt of transmission Finally, we use an infinite-capacity buffer in our simulations.

In our experiments, we vary the BER during the Bad state (P_{eb}), and we fix the BER during the Good state at $P_{eg} = 10^{-6}$. We set the mean of the off period to ten times that of the on period. In addition, we take the parameters related to the wireless channel from [7]

We adopt Bose-Chaudhuri-Hocquenghem (BCH) code [13] for FEC We consider fixed packet sizes, e.g., ATM cell Since we treat the CRC code as part of the payload, the FEC code is applied to 424-bit blocks (i.e., k = 424 bits). All simulation results are reported with 95% confidence intervals. For the delay distribution, 10^7 to 4×10^7 samples were needed in the simulations. Table 1 summarizes the values of the various parameters in the simulations and numerical examples. For the parameters c, P_{eb}, τ, and N_l, the values in the parenthesis are assumed unless specified otherwise.

Figure 3 depicts the complementary cumulative distribution for delay $\Pr[delay > t]$. We vary the service rates (c) from 800 to 1200 (packets/sec) while fixing other parameters, $P_{eb} = 10^{-2}$, $\tau = 7$, and $N_l = \infty$. The difference between the analytical and simulation results is negligible for all service rates.

Table 1: Parameter values used in the simulations and numerical results.

Parameter	Symbol	Value
source peak rate	τ	1 Mbps (or 2604.1667 packets/sec)
service rate	c	100 - 8000$ (packets/sec) (1000)
mean on period	$1/\alpha$	0.02304 sec
mean off period	$1/\beta$	0.2304 sec
mean Good channel period	$1/\delta$	0.1 sec
mean Bad channel period	$1/\gamma$	0.0333 sec
BER in Good channel state	P_{eg}	10^{-6}
BER in Bad channel state	P_{eb}	$10^{-2} - 10^{-5} (10^{-2})$
number of correctable bits	τ	0 - 20 (7)
limit on transmissions	N_l	$1 - \infty (\infty)$

We observe a slight deviation at the tail part of the distribution. However, it is associated with the number of samples taken from the simulation. For c=1200, we generated 4×10^7 packets to obtain the shown results. Note that the simulation is based on the realistic scenario in which the packet is transmitted until it is successfully transmitted or until it reaches the limit on the retransmission, whereas the analytical results are based on the fluid analysis.

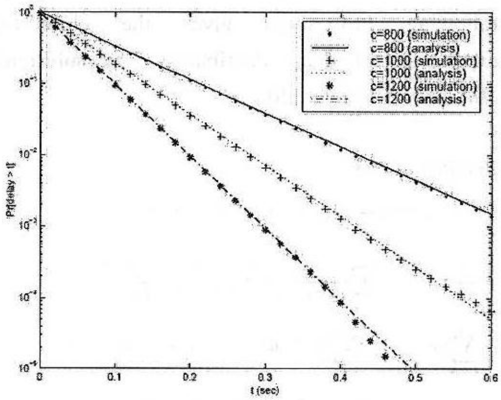

Figure 3: Complementary delay distribution for different service rates.

Figure 4 shows the effective bandwidth as a function of the number of correctable bits (τ) for the three target delay constraints Pr[delay>0.01]=0.01, 0.05, 0.1. Expectedly, more bandwidth is needed as a higher quality is required. In particular, we observe that much higher bandwidth is required when only ARQ ($\tau = 0$) is used as an error control scheme. Thus, the appropriate use of FEC is essential in an efficient use of scarce wireless bandwidth. In addition, this figure clearly indicates that there is an optimal

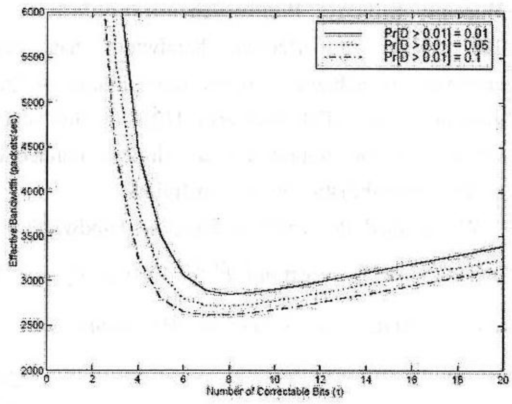

Figure 4: Effective bandwidth versus τ for target delay constraints Pr[delay >0.01].

τ ($\tau = 7$ in this example) for a given BER $P_{eb} = 0.01$, which satisfies a delay QoS constraint while minimizing the use of bandwidth.

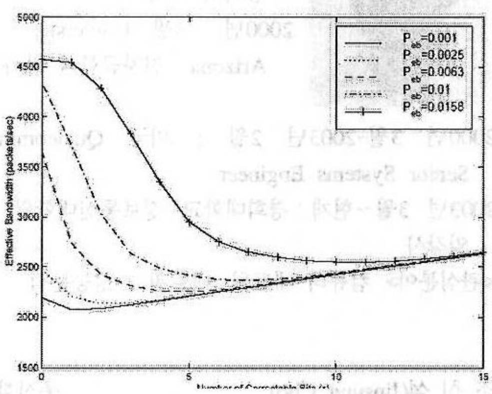

Figure 5: Effective bandwidth versus τ for different BER's

The optimal number of correctable bits as a function of the BER in Bad channel state is shown in Fig. 5. The target delay constraint is fixed to $\Pr[delay > 0.01] = 0.25$. We vary the BER from $P_{e,b} = 0.001$ to $P_{e,b} = 0.0158$. For each BER, we observe the optimal number of correctable bits exists. To better pictorial view, we provide the effective bandwidth as a function of τ's and BER's ($P_{e,b}$) altogether in Fig. 6.

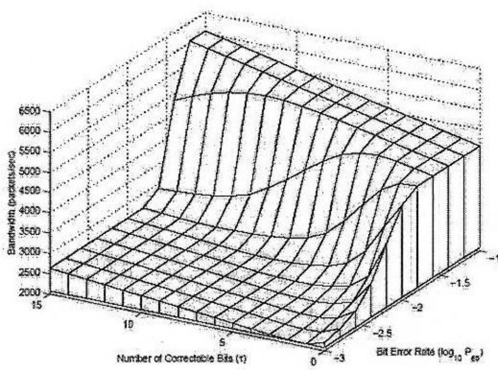

Figure 6: Effective bandwidth versus BER's (P_{eb}) and τ's

V. Conclusions

In this paper, we investigated the packet delay performance for an on/off source transported over a wireless channel. Accurate yet simple fluid models were used to capture the bursty nature of the arriving traffic and the channel's time-varying error characteristics. Error control schemes (ARQ and FEC), which are essential elements of any wireless packet network, were incorporated. We obtained the delay distribution using two alternative approaches: Laplace transform and uniformization. The solution was then used to obtain the wireless effective bandwidth (defined here as the minimum amount of bandwidth required to satisfy a given probabilistic delay constraint), which can be used as a valuable tool in resource allocation and admission control in wireless networks. Our analytical results were validated by contrasting them with simulations. It was observed that the analytically obtained delay distribution is quite accurate over a wide range of parameters. In a future work, we plan to investigate delay performance of the multimedia connections which share the wireless link according to "opportunistic scheduling."

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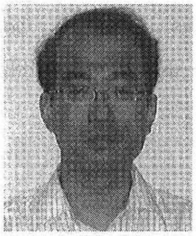

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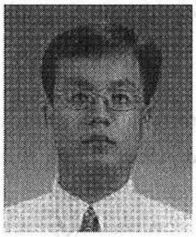

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