Efficient Bandwidth Utilization Guaranteeing QoS over Adaptive Wireless Links

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Abstract— **Providing guaranteed quality of service (QoS) with efficient bandwidth utilization over wireless fading channels is challenging. In this paper, we derive the throughput, the packet loss rate and the average delay of an end-to-end wireless link, where the transmitter relies on adaptive modulation and coding (AMC) at the physical layer while accounting for finite-length buffer effects at the data link layer. Guided by our cross-layer performance analysis, we develop a simple procedure to determine the minimal required bandwidth, which guarantees the prescribed QoS over the wireless link.**

I. Introduction

Providing QoS (throughput, loss and delay) for multimedia transmissions is of paramount importance in wired-wireless communication networks. The "bottleneck" in such networks is the wireless link, not only because wireless resources (bandwidth and power) are more scarce and expensive, but also because the overall system performance degrades markedly due to multipath fading, Doppler, and time-dispersive effects introduced by the wireless propagation. Unlike wired networks, even if large bandwidth is allocated to a certain connection (flow), the loss and delay requirements may not be satisfied when the wireless channel experiences deep fades. The powerful forward error control (FEC) coding or automatic-repeat-request (ARQ) protocol may reduce the loss rate; however, they both increase bandwidth as well as delay requirements [3]. On the other hand, the reserved bandwidth is not used efficiently, because the queue may be empty from time to time, even though the wireless channel can accommodate transmissions. Therefore, providing *guaranteed* QoS with *efficient* bandwidth utilization is more challenging in wireless than in wired networks [2].

In order to enhance spectral efficiency (bandwidth utilization) while adhering to a target error performance over wireless channels, adaptive modulation and coding (AMC) schemes have been widely used to match transmission parameters to time-varying channel conditions; see e.g., [1], and references therein. However, most existing AMC designs are considered at the physical layer. Their impact on, and interaction with, higher protocol layers remain largely un-explored. We have developed a cross-layer design combining AMC with truncated automatic-repeat-request (ARQ) in [3], investigated the interaction of AMC with finite-length queuing in [4] and studied the effects of AMC on TCP protocol in [5].

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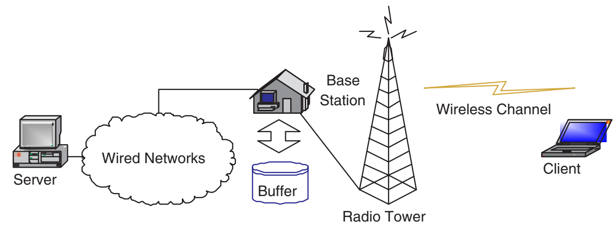

Fig. 1. An end-to-end wired-wireless connection

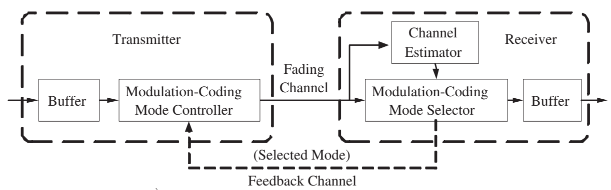

Fig. 2. The wireless link with combined queuing and AMC

In this paper, we study the QoS of an end-to-end connection over a wireless link, where transmitters are equipped with AMC at the physical layer and finite-length buffers at the data link layer. We rely on an appropriate cross-layer model for the wireless link to derive the throughput, the packet loss rate and the average delay analytically, given the allocated wireless bandwidth. Based on the performance analysis, we develop a procedure to determine the minimal required bandwidth, in order to guarantee the prescribed QoS over the wireless link.

II. System Model

A. System Description

Fig. 1 illustrates an end-to-end connection between a server (source) and a client (destination), which includes a wireless link with a single-transmit and a single-receive antenna. We consider here the time division multiplexing (TDM) system. As depicted in Fig. 2, a finite-length queue (buffer) is implemented at the base station of the wireless link, and operates in a first-in-first-out (FIFO) mode. The AMC controller follows the queue at the base station (transmitter), and the AMC selector is implemented at the client (receiver). The layer structure of the system under consideration is shown in Fig. 3. The processing unit at the data link layer is a packet, which comprises multiple information bits. On the other hand,

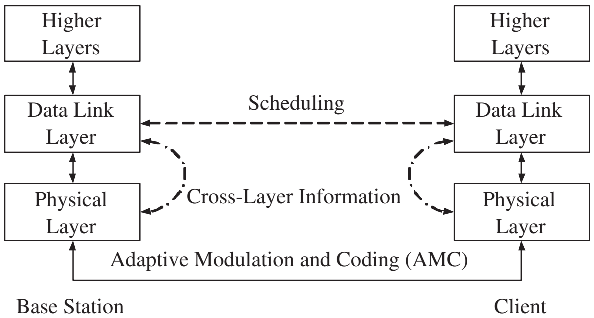

Fig. 3. The cross-layer structure

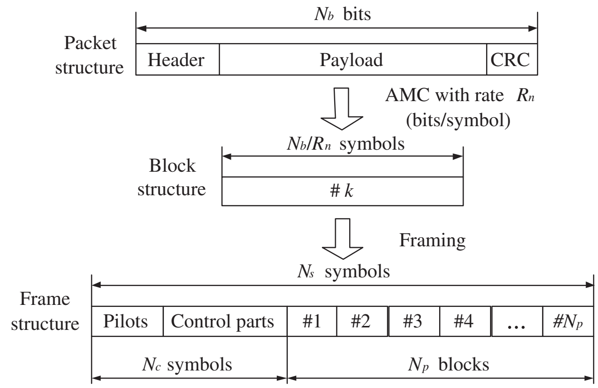

Fig. 4. The processing units at each layer

the processing unit at the physical layer is a frame, which consists of multiple transmitted symbols. The packet and frame structures will be detailed soon.

At the *physical layer* of the wireless link, we assume that multiple transmission modes are available, with each mode representing a pair of a specific modulation format, and a forward error correcting (FEC) code, as in the HIPERLAN/2 and the IEEE 802.11a standards. Based on channel estimation at the receiver, the AMC selector determines the modulation-coding pair (mode), which is sent back to the transmitter through a feedback channel, for the AMC controller to update the transmission mode. Coherent demodulation and maximum-likelihood (ML) decoding are employed at the receiver. The decoded bit streams are mapped to packets, which are pushed upward to the data link layer.

We consider the following group of transmission modes:
TM: Convolutionally coded M_n-ary rectangular/square QAM, adopted from the HIPERLAN/2, or, the IEEE 802.11a standards, which are listed under Table I, in a rate ascending order. Although we focus on TM in this paper, other transmission modes can be similarly constructed [3].

At the *data link layer* of the base station (transmitter), the queue has finite-length (capacity) of K packets. The queue is served by the AMC module at the physical layer. The customers of the queue are packets.

We detail the frame and packet structures, as in Fig. 4:
i) At the *physical layer*, the data are transmitted frame by frame through the wireless link, where each frame contains a fixed number of symbols (N_s). Given a fixed symbol rate, the frame duration (T_f seconds) is constant, and represents the time-unit throughout this paper. Each frame at the physical layer may contain one or more packets coming from the data link layer.
ii) At the *data link layer*, each packet contains a fixed number of bits (N_b), which include packet header, payload, and cyclic redundancy check (CRC) bits. After modulation and coding with mode n of rate R_n (bits/symbol) at the base station, each packet is mapped to a symbol-block containing N_b/R_n symbols. Multiple such blocks, together with N_c pilot symbols and control parts, constitute one frame to be transmitted, as in the HIPERLAN/2 and the IEEE 802.11a standards. If mode n is used, it follows that the number of symbols per frame is $N_s = N_c + N_p N_b / R_n$, which implies that N_p (the number of packets per frame) depends on the chosen mode.

We next list our operating assumptions:
A1: The channel is frequency flat, and remains invariant per frame, but is allowed to vary from frame to frame. This corresponds to a block fading channel model, which is suitable for slowly-varying wireless channels. As a consequence, AMC is adjusted on a frame-by-frame basis.
A2: Perfect channel state information is available at the receiver relying on training-based channel estimation. The corresponding mode selection is fed back to the transmitter without error and latency [1].
A3: If the queue is full, the additional arriving packets will be dropped, so that the overflow content is lost.
A4: Error detection based on CRC is perfect, provided that sufficiently reliable error detection CRC codes are used.
A5: If a packet is received incorrectly at the receiver after error detection, we drop it and declare packet loss.

For flat fading channels adhering to A1, the channel quality can be captured by a single parameter, namely the received signal-to-noise ratio (SNR) γ. Since the channel varies from frame to frame, we adopt the general Nakagami-m model to describe γ statistically [1]. The received SNR γ per frame is thus a random variable with a Gamma probability density function:

$$p_\gamma(\gamma) = \frac{m^m \gamma^{m-1}}{\bar{\gamma}^m \Gamma(m)} \exp\left(-\frac{m\gamma}{\bar{\gamma}}\right), \qquad (1)$$

where $\bar{\gamma} := E\{\gamma\}$ is the average received SNR, $\Gamma(m) := \int_0^\infty t^{m-1} e^{-t} dt$ is the Gamma function, and m is the Nakagami fading parameter ($m \geq 1/2$).

B. Adaptive Modulation and Coding

The objective of AMC is to maximize the data rate by adjusting transmission parameters to channel variations, while maintaining a prescribed packet error rate P_0. Let N denote the total number of transmission modes available ($N = 5$ for TM). As in [1], we assume constant power transmission, and partition the entire SNR range into $N+1$ non-overlapping consecutive intervals, with boundary points denoted as $\{\gamma_n\}_{n=0}^{N+1}$. In this case,

$$\text{mode } n \text{ is chosen,} \quad \text{when } \gamma \in [\gamma_n, \gamma_{n+1}). \qquad (2)$$

To avoid deep channel fades, no data are sent when $\gamma_0 \leq \gamma < \gamma_1$, which corresponds to the mode $n = 0$ with rate $R_0 = 0$

TABLE I

TRANSMISSION MODES WITH CONVOLUTIONALLY CODED MODULATION

	Mode 1	Mode 2	Mode 3	Mode 4	Mode 5
Modulation	BPSK	QPSK	QPSK	16-QAM	64-QAM
Coding Rate R_c	1/2	1/2	3/4	3/4	3/4
R_n (bits/sym.)	0.50	1.00	1.50	3.00	4.50
a_n	274.7229	90.2514	67.6181	53.3987	35.3508
g_n	7.9932	3.4998	1.6883	0.3756	0.0900
γ_{pn}(dB)	-1.5331	1.0942	3.9722	10.2488	15.9784

(The generator polynomial of the mother code is $g = [133, 171]$. Different coding rates are obtained with puncturing as in HIPERLAN/2.)

(bits/symbol). The design objective for AMC is to determine the boundary points $\{\gamma_n\}_{n=0}^{N+1}$.

For simplicity, we approximate the instantaneous packet error rate (PER) as [3, eq. (5)]:

$$\text{PER}_n(\gamma) \approx \begin{cases} 1, & \text{if} \quad 0 < \gamma < \gamma_{pn}, \\ a_n \exp(-g_n\gamma), & \text{if} \quad \gamma \geq \gamma_{pn}, \end{cases} \tag{3}$$

where n is the mode index, γ is the received SNR, and the mode-dependent parameters a_n, g_n, and γ_{pn} are obtained by fitting (3) to the exact PER [3]. With packet length $N_b = 1,080$, the fitting parameters for TM are provided in Table I [3]. Based on (1) and (2), the mode n will be chosen with probability [1, eq. (34)]:

$$\begin{aligned} \Pr(n) &= \int_{\gamma_n}^{\gamma_{n+1}} p_\gamma(\gamma) d\gamma \\ &= \frac{\Gamma(m, m\gamma_n/\bar{\gamma}) - \Gamma(m, m\gamma_{n+1}/\bar{\gamma})}{\Gamma(m)}, \end{aligned} \tag{4}$$

where $\Gamma(m, x) := \int_x^\infty t^{m-1}e^{-t}dt$ is the complementary incomplete Gamma function. Let $\overline{\text{PER}}_n$ denote the average PER corresponding to mode n. In practice, we have $\gamma_n > \gamma_{pn}$, and thus obtain $\overline{\text{PER}}_n$ in closed-form as (c.f. [1, eq.(37)]):

$$\begin{aligned} \overline{\text{PER}}_n &= \frac{1}{\Pr(n)} \int_{\gamma_n}^{\gamma_{n+1}} a_n \exp(-g_n\gamma) p_\gamma(\gamma) d\gamma \qquad (5) \\ &= \frac{1}{\Pr(n)} \frac{a_n}{\Gamma(m)} \left(\frac{m}{\bar{\gamma}}\right)^m \frac{\Gamma(m, b_n\gamma_n) - \Gamma(m, b_n\gamma_{n+1})}{(b_n)^m}, \end{aligned}$$

where $b_n := m/\bar{\gamma} + g_n$. The average PER of AMC can then be computed as the ratio of the average number of packets in error over the total average number of transmitted packets [1]:

$$\overline{\text{PER}} = \frac{\sum_{n=1}^N R_n \Pr(n) \overline{\text{PER}}_n}{\sum_{n=1}^N R_n \Pr(n)}. \tag{6}$$

We want to find the thresholds $\{\gamma_n\}_{n=0}^{N+1}$, so that the prescribed P_0 is achieved for each mode: $\overline{\text{PER}}_n = P_0$, which naturally leads to $\overline{\text{PER}} = P_0$ based on (6). Given P_0, $\bar{\gamma}$, and m, the following threshold searching algorithm determines $\{\gamma_n\}_{n=0}^{N+1}$ and guarantees that $\overline{\text{PER}}_n$ is exactly P_0 [4]:

Step 1: Set $n = N$, and $\gamma_{N+1} = +\infty$.

Step 2: Search the unique $\gamma_n \in [0, \gamma_{n+1}]$ that satisfies:

$$\overline{\text{PER}}_n = P_0 \ . \tag{7}$$

Step 3: If $n > 1$, set $n = n - 1$ and go to Step 2; otherwise, go to Step 4.

Step 4: Set $\gamma_0 = 0$.

The SNR region $[\gamma_n, \gamma_{n+1})$ corresponding to transmission mode n constitutes the channel state indexed by n. To describe the transition of these channel states, we rely on a finite state Markov chain (FSMC) model, which we develop next.

C. Finite State Markov Chain Channel Model

As in [4], we adopt an FSMC channel model to analyze the performance of our system. Assuming slow fading conditions so that transition happens only between adjacent states, the probability of transition exceeding two consecutive states is zero; i.e.,

$$P_{l,n} = 0, \qquad |l - n| \geq 2. \tag{8}$$

The adjacent-state transition probability can be determined by:

$$\begin{aligned} P_{n,n+1} &\approx \frac{N_{n+1}T_f}{\Pr(n)}, \quad \text{if } n = 0, \ldots, N-1, \\ P_{n,n-1} &\approx \frac{N_nT_f}{\Pr(n)}, \qquad \text{if } n = 1, \ldots, N, \end{aligned} \tag{9}$$

where N_n is the cross-rate of mode n (either upward or downward), which can be estimated as:

$$N_n = \sqrt{2\pi \frac{m\gamma_n}{\bar{\gamma}}} \frac{f_d}{\Gamma(m)} \left(\frac{m\gamma_n}{\bar{\gamma}}\right)^{m-1} \exp\left(-\frac{m\gamma_n}{\bar{\gamma}}\right), \tag{10}$$

where f_d denotes the mobility-induced Doppler spread. The probability of staying at the same state n is:

$$P_{n,n} = \begin{cases} 1 - P_{n,n+1} - P_{n,n-1}, & \text{if} \quad 0 < n < N, \\ 1 - P_{0,1}, & \text{if} \quad n = 0, \\ 1 - P_{N,N-1}, & \text{if} \quad n = N. \end{cases} \tag{11}$$

In summary, we model the channel as an FSMC with an $(N+1) \times (N+1)$ state transition matrix, as in [4, eq. (12)]:

$$\mathbf{P}_c = [P_{i,j}]_{(N+1)\times(N+1)}. \tag{12}$$

III. QUALITY OF SERVICE OVER WIRELESS LINK

In this section, we will derive the throughput, the packet loss rate and the average delay over the wireless link, which depends on the following modeling and queuing analysis.

1) Model of Queuing Service Process: Different from non-adaptive modulations, AMC dictates a dynamic, rather than deterministic, service process for the queue, with a variable number of packets transmitted per time unit. Let t index the time units, and C_t (packets/time-unit) denote the number of packets transmitted using AMC at time t. Corresponding to each transmission mode n, let c_n (packets/time-unit) denote the number of packets transmitted per time-unit. We then have:

$$C_t \in \mathcal{C}, \qquad \mathcal{C} := \{c_0, c_1, \ldots, c_N\} \ , \tag{13}$$

where c_n takes positive integer values. Suppose that for the rate $R = 1$ transmission mode (e.g., Mode 2 in TM), a total of d packets can be accommodated per frame. We then have $c_n = dR_n$, where d is the bandwidth coefficient determined by the allocated time-slot to this connection.

As specified in (13), the AMC module yields a queue server with a total of $N+1$ states $\{c_n\}_{n=0}^{N}$, with the service process C_t representing the evolution of server states. Since the AMC mode n is chosen when the channel enters the state n, we model the service process C_t as an FSMC with transition matrix given by (12).

2) Queuing Analysis: Having modeled the queuing service process, we now focus on the queue itself. Let U_t denote the queue state (the number of packets in the queue) at the end of time-unit t, or, at the beginning of time-unit $t+1$. Let A_t denote the number of packets arriving at time t. It is clear that $U_t \in \mathcal{U} := \{0, 1, \ldots, K\}$, and $A_t \in \mathcal{A} := \{0, 1, \ldots, \infty\}$. Here, A_t only needs to be stationary and independent of U_t and C_t. For convenience, we assume that A_t is Poisson distributed with parameter λ:

$$P(A_t = a) = \frac{(\lambda T_f)^a \exp(-\lambda T_f)}{a!}, \qquad a \geq 0, \tag{14}$$

where the ensemble-average $E\{A_t\} = \lambda T_f$.

Let (U_{t-1}, C_t) denote the pair of queue and server states, whose variation is modeled as an augmented FSMC [4]. We have proved that the stationary distribution of (U_{t-1}, C_t) exists and is unique; see [4, eq. (24)] for the calculation of the stationary distribution denoted as:

$$P(U = u, C = c) := \lim_{t \to \infty} P(U_{t-1} = u, C_t = c). \tag{15}$$

3) QoS over Wireless Link: We are now ready to evaluate the packet loss rate ξ, the throughput η and the average delay T_{wl} over the wireless link.

Let P_d denote the packet dropping (overflow or blocking) probability upon the queue. Based on $P(A_t = a)$ in (14) and $P(U = u, C = c)$ in (15), we can readily compute P_d, as illustrated in [4, eq. (31)]. A packet is correctly received by the client, only if it is not dropped from the queue (with probability $1 - P_d$), and is correctly received through the wireless channel (with probability $1 - P_0$). Hence, we can obtain the packet loss rate, as in [4, eq. (13)]:

$$\xi = 1 - (1 - P_d)(1 - P_0). \tag{16}$$

The throughput can then be evaluated as:

$$\eta = \lambda T_f (1 - \xi). \tag{17}$$

We now derive the average delay over the wireless link T_{wl}. With the stationary distribution $P(U = u, C = c)$ in (15), we can derive the average number of packets in the wireless link (both in the queue and in transmission), as in [5, eq. (21)]:

$$\begin{aligned} N_{wl} = &\sum_{u \in \mathcal{U}, c \in \mathcal{C}} uP(U = u, C = c) \\ &+ \sum_{u \in \mathcal{U}, c \in \mathcal{C}} \min\{u, c\} P(U = u, C = c). \end{aligned} \tag{18}$$

Based on Little's Theorem, the average delay per packet through the wireless link can be calculated, as in [5, eq. (22)]:

$$T_{wl} = \frac{N_{wl}}{\lambda T_f (1 - P_d)}. \tag{19}$$

In summary, given the bandwidth coefficient d, target packet error rate P_0, Doppler spread f_d, average SNR $\bar{\gamma}$, Nakagami parameter m, queue length K and data arriving rate λT_f, we can obtain the QoS over the wireless link analytically through the following steps:
1) Determine the boundary points of AMC $\{\gamma_n\}_{n=0}^{N+1}$ by the threshold searching algorithm.
2) Build the transition matrix $\mathbf{P}_c$ in (12) for the channel as well as the queue server states.
3) Compute the stationary distribution $P(U = u, C = c)$ as in (15).
4) Calculate the packet loss rate ξ from (16), the throughput η from (17) and the average delay T_{wl} from (19).

IV. Bandwidth Optimization

The parameters d, P_0, f_d, $\bar{\gamma}$, m, K, λT_f, can be divided in two categories: i) the channel condition parameters, which include f_d, $\bar{\gamma}$ and m and the QoS requirement parameter λT_f; and ii) the resource management parameters, which include d, P_0 and K. The parameters in the first category are decided by the application scenario, while the parameters in the second category are controllable via radio resource management (RRM).

If the QoS requirements of the connection over the wireless link are requested by signaling through the resource reservation protocol (RSVP) for instance, how can we determine, and even minimize, the required radio resources, such as d and K, in order to guarantee the prescribed QoS? We will investigate this problem next, using numerical examples.

For simplicity, we fix the buffer-length K and focus on the effects of the bandwidth coefficient d on QoS. We consider $T_f = 2$ (ms), packet arriving rate $\lambda T_f = 2.5$ (packets/time-unit), average SNR $\bar{\gamma} = 12$ (dB), normalized Doppler frequency $f_d T_f = 0.01$, Nakagami parameter $m = 1.0$ and buffer length $K = 100$. The target packet error rate P_0 varies from 10^{-4} to 10^{-1}. The bandwidth coefficient d takes values $\{2, 4, 6, 8, 10, 12\}$, in order to keep the integer values of packets transported by one frame for TM. We plot the curves of the packet loss rate ξ versus P_0 in Fig. 5; and the average delay T_{wl} versus P_0 in Fig. 6.

From Fig. 6, we notice that T_{wl} decreases with P_0 increasing, because a high value of P_0 leads to high probability of selecting high rate modes in TM, i.e., high service rate for the queue. However, the packet loss rate ξ depends on both P_0 and P_d [c.f. (16)]; their joint effects on ξ are shown by Fig. 5. From the shapes of these plots, we infer that P_0 dominates ξ when P_0 has large values, while P_d dominates ξ when P_0 has small values. This observation agrees with the intuition behind (16). Increasing the value of bandwidth coefficient d results in the service rate increasing upon the queue, so that P_d, and thus ξ, decreases, as in Fig. 5; and T_{wl} decreases, as in Fig. 6.

Suppose that the prescribed QoS requirements are: i) $\eta \geq \eta_0 = 2.5$ (packets/time-unit) $= 1.35$ (Mbps); ii) $\xi \leq \xi_0 =$

10^{-2}; and iii) $T_{wl} \leq T_{wl0} = 4T_f = 8$ (ms) (delay over the wireless link, not the end-to-end delay), which are typical QoS requirements for real-time video transmissions [3]. From Fig. 5, $\xi \leq \xi_0$ can be guaranteed, only for $\{d = 10, 3\times 10^{-4} \leq P_0 \leq 4\times 10^{-3}\}$ and $\{d = 12, 1\times 10^{-4} \leq P_0 \leq 5\times 10^{-3}\}$. The union of these sets is denoted as *the loss-guaranteed region*, which is illustrated by the dashed lines in Fig. 7. From Fig. 6, $T_{wl} \leq T_{wl0}$ can be guaranteed, only for $\{d = 6, P_0 \geq 1\times 10^{-2}\}$, $\{d = 8, P_0 \geq 2\times 10^{-3}\}$, $\{d = 10, P_0 \geq 5\times 10^{-4}\}$ and $\{d = 12, P_0 \geq 2\times 10^{-4}\}$. The union of these sets is denoted as *the delay-guaranteed region*, which is illustrated by the dotted lines in Fig. 7.

In the loss-guaranteed region, the $\xi \leq \xi_0 = 10^{-2}$ is guaranteed, i.e., more than 99% packets pass through the wireless link, so that the throughput requirement is practically guaranteed. Therefore, the prescribed QoS can be guaranteed, only if the RRM selects d and P_0 in the intersection of the loss-guaranteed and delay-guaranteed regions. This intersection is denoted as *the QoS-guaranteed region*: $\{d = 10, 5\times 10^{-4} \leq P_0 \leq 4\times 10^{-3}\}$ and $\{d = 12, 2\times 10^{-4} \leq P_0 \leq 5\times 10^{-3}\}$, which is indicated by the solid lines in Fig. 7. It is clear that selecting the minimal value of d in the QoS-guaranteed region leads to the best bandwidth utilization. In this case, $d = 10$ is the minimal required bandwidth coefficient, in order to guarantee the prescribed QoS.

In summary, given the prescribed QoS requirements η_0, ξ_0 and T_{wl0}; and the system parameters: f_d, $\bar{\gamma}$, m, K and $\lambda T_f = \eta_0$; we can obtain the minimal required bandwidth coefficient analytically through the following procedure:

Step 1: Initialization: let $\mathcal{P}$ be the set of possible target PER values; and let $\mathcal{D}$ be the set of possible values of the bandwidth coefficient, in an increasing order, i.e., $d_k < d_{k+1}, \forall k, d_k \in \mathcal{D}$. Set the initial index: $k = 1$.

Step 2: Calculate

$$J = \sum_{P_0 \in \mathcal{P}} \Big[1\{\xi(P_0, d_k) \leq \xi_0\} \times 1\{\tau(P_0, d_k) \leq \tau_0\} \Big], \quad (20)$$

where $1\{\cdot\}$ is the indicator function, based on (16) and (19).

Step 3: If $J > 0$, $d^* = d_k$ is the optimal bandwidth coefficient and we stop searching. If $J = 0$ and $k < |\mathcal{D}|$, set $k = k + 1$ and go to Step 2. Otherwise, no bandwidth coefficient in $\mathcal{D}$ can afford the prescribed QoS and stop searching.

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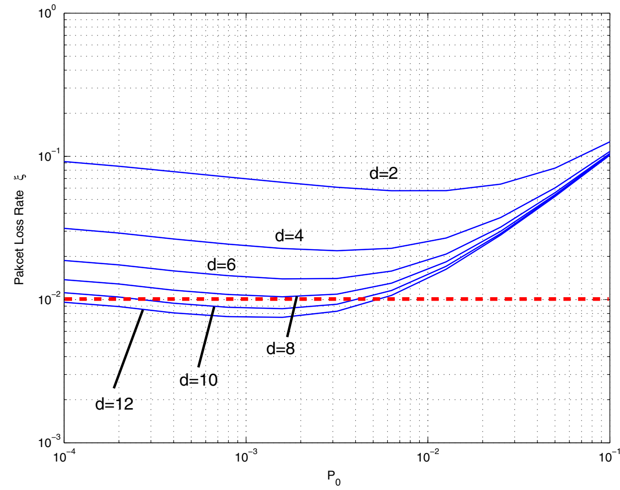

Fig. 5. Packet loss rate vs. target PER

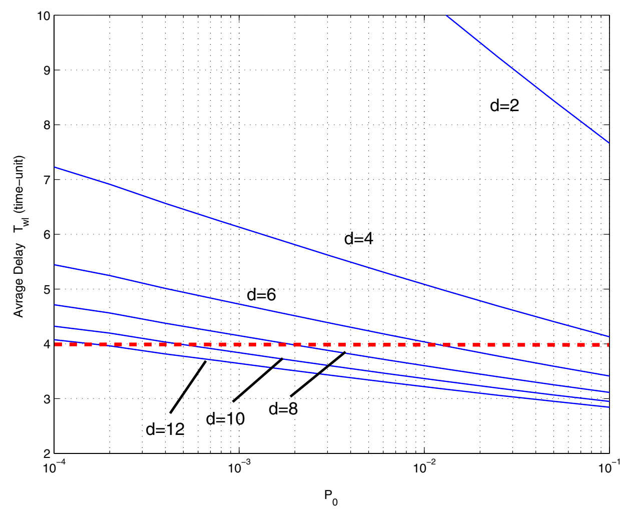

Fig. 6. Average delay vs. target PER

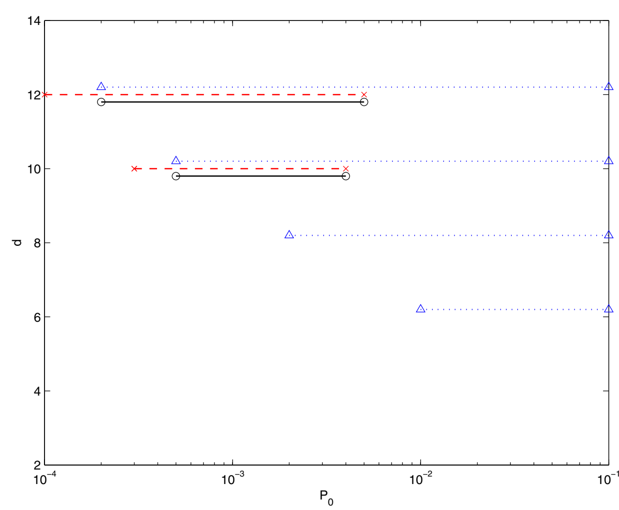

Fig. 7. Loss-guaranteed (dashed), delay-guaranteed (dotted) and QoS-guaranteed (solid) regions