Efficient Resource Allocation for China's 3G/4G Wireless Networks

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ABSTRACT

The all-IP DiffServ model is expected to be the most promising architecture for QoS provisioning in China's next-generation wireless networks, due to its scalability, convenience for mobility support, and capability of interworking heterogeneous radio access networks. This article focuses on efficient resource allocation in a wireless DiffServ architecture. Resource utilization efficiency is particularly important for China's wireless networks as the mobile user density in China is and will continue to be much higher than that in other countries. More specifically, we propose a novel buffer sharing scheme to provide assured service for real-time layer-coded multimedia traffic, which can guarantee the specific packet loss requirement of each layer with UDP as the transport layer protocol. An adaptive optimal buffer configuration can be applied to achieve maximum resource utilization over the time-varying channel. Assured service is also provided to TCP data traffic for guaranteed throughput, where the cross-layer coupling between the TCP layer and link layer is exploited to efficiently utilize the wireless resources.

INTRODUCTION

After many indoor and outdoor tests, extensive upgrade of network infrastructures, and premarket promotion of new services, China will deploy commercial third-generation (3G) mobile/wireless networks very soon. To build up a stable and profitable market for 3G wireless communications, China has to pay particular attention to efficient management of wireless resources, because its 3G wireless networks have to support mobile users with much higher density than that in other countries. The requirement for efficient resource allocation will persist in future-generation wireless networks, due to the limited radio spectrum, huge number of mobile subscribers, and high transmission rate requirements of new multimedia services. In this article we propose two novel quality of service (QoS) techniques with efficient resource utilization to support real-time multimedia (video/audio) services and non-real-time data services, respectively, in 3G/4G wireless networks.

It is very likely that the three 3G air interface models, wideband code-division multiple access (WCDMA), CDMA2000, and the homegrown time-division synchronous CDMA (TD-SCDMA), will all be deployed in China. An all-IP differentiated services (DiffServ) platform [1–3] is the most promising architecture to interwork the heterogeneous wireless access networks and the Internet to provision broadband access, seamless global roaming, Internet/telecommunication services anywhere, and QoS guarantee for various IP multimedia services. The reasons are as follows. First, DiffServ [4] is a scalable class-based traffic management mechanism without using per-flow resource reservation and per-flow signaling in the core routers; second, DiffServ adopts a domain-based resource management model. Each domain can freely choose whatever mechanism is proper for internal resource management as long as its service level agreements (SLAs) with neighboring domains are met. Such a domain-based architecture is very convenient for the interconnection of heterogeneous wireless networks [1]. Lastly, a domain-based architecture can be seamlessly integrated with micromobility protocols to support fast handoff [3]. This advantage can considerably benefit China's 3G/4G wireless networks where micro/picocellular deployment is required for high resource utilization. As a result, this article focuses on efficient resource allocation in an all-IP DiffServ wireless architecture.

The 3G/4G wireless systems are designed for multimedia communications, where broadband mobile video services will become reality for the first time. Recent advances in video coding have made it possible to encode video with a very flexible layering structure, where a *base layer* contains most important features of the video and some *enhancement layers* contain data for refining the reconstructed video quality. The layer coding concept can also be applied to audio traffic. Such a layered structure supports adaptive multimedia services with different bandwidth requirements by adjusting the number of layers delivered. In wireless networks, this type of adaptive services is very important for efficient resource utilization, because wireless resource availability fluctuates due to user mobility and time-varying channel quality. In the following, we propose a novel buffer sharing scheme to provide an assured service to the layer-coded multimedia traffic, which can guarantee the specific packet loss requirement of each layer with User Datagram Protocol (UDP) as the transport layer protocol for real-time services. Adaptive optimal buffer configuration is applied to achieve maximum resource utilization over the time-varying channel. For data traffic with Transmission Control Protocol (TCP) at the transport layer, the assured service is provisioned for a guaranteed throughput. As in wireless networks TCP performance is heavily affected by the loss due to unstable wireless channel quality, we propose a cross-layer model to capture the coupling between the TCP transport layer and the link layer. The target TCP throughput is guaranteed with minimal resources required at the link layer and therefore at the physical layer.

ALL-IP DIFFSERV ARCHITECTURE

The all-IP DiffServ architecture under consideration is shown in Fig. 1, which enables access to Internet/telecommunication services independent of the air interface technique. In the following, we use Internet services to generally represent all the possible Internet/telecommunication services that may appear in the all-IP wireless/wireline networks. In the all-IP DiffServ architecture, a number of nearby radio access networks (RANs) having the same air interface are grouped into a wireless DiffServ domain, and all the domains are connected through the DiffServ Internet backbone to provide end-to-end Internet services to a mobile station (MS). Although wireless local area networks (WLANs) can be seamlessly integrated to the all-IP architecture, here we focus on resource allocation in cellular systems.

In each DiffServ wireless domain, the RAN architecture is the same as that defined in the 3G specifications, but all the network elements are enhanced to fulfill the functions of a DiffServ IP router. The gateway and base stations are edge routers of the domain and connected through core routers. The gateway is the interface to the DiffServ Internet backbone. For example, the gateway General Packet Radio Service (GPRS) support node (GGSN) in the WCDMA domain is the gateway of the domain to the external DiffServ Internet. In the gateway, SLAs are negotiated to specify the resources allocated by the Internet service provider to serve the aggregate traffic flowing from/into the domain. The gateway conditions the aggregate traffic for each service class according to the SLA resource commitments. The base stations provide MSs with access points to the Internet, and perform per-flow traffic conditioning and marking for uplink transmission. All DiffServ routers use several separate queues, controlled by certain scheduling algorithms, to provide differentiated classes of services.

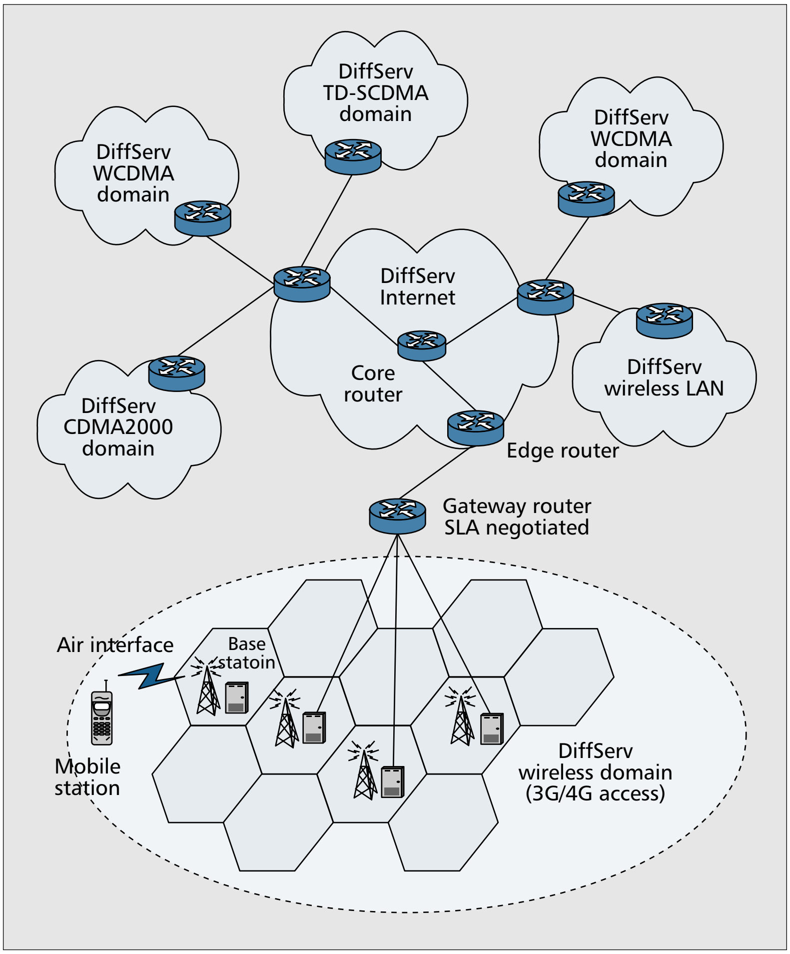

Figure 1. *An all-IP DiffServ architecture for 3G/4G wireless communications.*

SERVICE CLASS MAPPING IN WIRELESS DIFFSERV

Among the 3G standards, the Universal Mobile Telecommunications System (UMTS) based on WCDMA technologies defines four QoS classes: conversational, streaming, interactive, and background [2]. Conversational and streaming classes are intended for real-time traffic. They both preserve time relations between adjacent information elements of the stream, but conversational class has stricter delay requirements. For the interactive and background classes, transfer delay is not a main concern, but error-free delivery of data should be guaranteed. The UMTS QoS class definitions can also be extended to other 3G/4G wireless domains. On the other hand, DiffServ defines the expedited forwarding (EF) [5] per-hop behavior (PHB) for premium service and the assured forwarding (AF) PHB [6] for assured service, in addition to the classic best effort service. To extend IP services to the wireless domain, the UMTS QoS classes must be mapped to the DiffServ classes [2].

Normally, the conversational class can be mapped to the EF PHB for very low-delay and

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low-loss service, streaming and interactive traffic to the AF PHB, and background traffic to best-effort service [2]. However, the peak rate bandwidth allocation for EF services is not suitable for wireless communications. The constant (peak rate) bandwidth requirement during each session lifetime is very hard to guarantee due to the time-varying wireless channel capacity. Furthermore, Internet traffic often has a bursty characteristic; inefficient peak rate allocation is particularly undesirable for wireless transmission with scarce resources. In DiffServ, AF PHBs aim at providing throughput guaranteed services with in-flow loss differentiation. When combined with admission control, AF PHBs can be extended to a very flexible service architecture. Taking into account that current video/audio codecs can work properly with a small packet loss probability, we propose that all the real-time classes (i.e., conversational and streaming classes) and non-real-time class (i.e., interactive class) are mapped into AF PHBs to guarantee statistical QoS requirements and improve resource utilization. Specifically, we propose using a partitioned first-in-first-out (FIFO) buffer [7] to provide an AF PHB for real-time layer-coded multimedia traffic under UDP. For non-real-time TCP data traffic, an AF PHB is provisioned to guarantee a specified throughput, and a cross-layer model is proposed to capture the coupling between the transport and link layers to achieve efficient wireless resource utilization. QoS provisioning for real-time and non-real-time AF PHBs will be discussed in separate sections, after some general principles for wireless resource allocation over IP-based 3G networks are described.

Wireless Resource Allocation

Based on CDMA technology, all three 3G standards are likely to be deployed in China. The all-IP architecture provides an efficient way to serve multimedia traffic over heterogeneous wireless domains. For each wireless domain, the bandwidth requirement from the network layer will eventually be mapped to a wireless resource requirement at the link layer.

For circuit-switched voice traffic the resource allocation is relatively simple, and the 2G mobile/wireless systems have shown success in serving voice applications over wireless channels. However, for IP-based multimedia traffic, the case is quite different where different applications have different QoS (e.g., transmission rate and accuracy) requirements, and packet switching should be used to efficiently utilize the scarce radio resources. Generally, orthogonal variable spreading factor (OVSF) codes or multicode transmission can be employed to support high-rate applications; and high transmission accuracy can be achieved by assigning a high signal-to-noise (plus interference)-ratio (SINR) and/or a powerful forward error correction (FEC) code, or by means of automatic repeat request (ARQ). It is shown in [8] that, in the uplink, the smaller bit error rate an MS requires, the higher SINR the MS should receive, and the more resources should be allocated to the MS to transfer a link layer packet. As CDMA systems have interference limited capacity, a properly designed power allocation and access control scheme can efficiently reduce interference and improve the successful transmission rate of data packets. In such a scheme, the SINR value for each MS should be designed carefully.

For real-time multimedia traffic, UDP is usually used in the transport layer to avoid retransmission delay, while the link and physical layers try to provide stable capacity to upper layers by assigning a large SINR and/or powerful FEC for reliable transmission. For non-real-time data traffic, TCP is usually adopted for an error free connection. TCP dynamically adjusts its congestion window according to the network congestion status (e.g., the packet loss event rate and round-trip delay) and retransmits the lost TCP segments. Over a wireless link, the link layer resource allocation ultimately determines the transmission delay and packet loss event rate, and therefore affects TCP performance. As discussed below, in the DiffServ architecture a target TCP throughput is guaranteed for non-real-time data traffic, and a cross-layer optimization between the transport and link layers is proposed for efficient resource utilization over the wireless link.

Buffer Sharing for Layer-Coded Multimedia Traffic

Providing real-time multimedia services to mobile users is one of the prominent features of 3G/4G wireless communications. Layer-coded video/audio traffic is very suitable for transmission over the wireless link. We propose a novel buffer-sharing approach to serve the layered multimedia traffic. Consider video services as an example. Suppose a video source generates traffic having J ($J \geq 1$) layers, referred to as J classes. The QoS requirement is specified by a packet loss probability (PLP) ε_j for class j, $j \in \{1, 2, \ldots, J\}$. Letting class J represent the base layer information and have the most stringent QoS requirement, we have $\varepsilon_1 > \varepsilon_2 > \cdots > \varepsilon_J > 0$. A buffer of size B serves traffic with a channel capacity c provided by the link layer, where parameters B and c determine the queuing delay bound at the buffer. The traffic admission policy is based upon a buffer space reservation scheme, using a buffer partition vector $\mathbf{B}_t = (B_1, B_2, \cdots, B_{J-1})$ to provide J loss priorities, where $0 = B_0 < B_1 < B_2 < \cdots < B_{J-1} < B_J = B$. Let X be the number of packets of all the classes in the buffer at time t. When $B_{j-1} \leq X < B_J$ $(1 \leq j \leq J)$, only traffic of classes $\{j, j + 1, \cdots, J\}$ is admitted into the buffer. Therefore, higher classes of traffic are served with higher priority by access to a larger buffer space. The *partitioned buffer* scheme is an implementation of the AF PHB defined in the DiffServ architecture.

The loss calculation and admission control in the partitioned buffer system are required to serve multiplexed layer-coded video sources with QoS guarantee; that is, the actual PLP experienced by class j traffic (PLP_j) should satisfy $PLP_j \leq \varepsilon_j$ $(1 \leq j \leq J)$. The loss analysis is relatively simple when the buffer has a large size (the delay requirement is not strict). For a large buffer, it is widely agreed that the packet loss probability can be well approximated by the *overflow proba-*

bility in an infinite buffer system, that is, $\text{PLP}_j \approx \text{P}\{X > B_j\}$ ($1 \le j \le J$). Furthermore, when each partition region $B_j - B_{j-1}$ is large enough, the queuing process in neighboring regions can be approximately considered independent of each other, if the input traffic is short-range dependent (SRD).[1] Based on the above facts, PLPs of the J traffic classes in the partitioned buffer can be approximated by a two-step algorithm:

1) Calculate $\text{PLP}_1 \approx \text{P}\{X > B_1\}$, using the large buffer overflow approximation.
2) Calculate PLP for classes 2 to J iteratively, as $\text{PLP}_j \approx \text{P}\{X > B_j\} \approx \text{P}\{X > B_{j-1}\} \cdot \text{P}\{X > B_j - B_{j-1}\}$ with overflow approximation applied in each region $B_j - B_{j-1}$. The iterative calculation has an explicit physical meaning, (i.e., the overload at B_j happens in two steps): the first is that the queue occupancy goes into the jth partition region, and the second is that an overload event happens in that region. Note that the traffic arrival process in each region is different due to the buffer sharing policy.

Details of the large buffer analysis can be found in our study [7]. When the traffic statistical parameters and buffer size are fixed, the choice of $\boldsymbol{B}_t$ determines the channel capacity required to guarantee the QoS for all J classes. The selection of an *optimal partition vector* $\boldsymbol{B}_t^*$ results in the minimal channel capacity requirement c^*, in other words, the maximum resource utilization. An algorithm to jointly solve $\boldsymbol{B}_t^*$ and c^* is given in [7]. The capacity c^* can be defined as the *minimal effective bandwidth* and used to achieve linear admission control. Linear admission control is desired for online operation. It avoids directly solving the optimization problem for the aggregate traffic, by approximating the minimal resource requirement of the aggregate traffic with the summation of the minimal effective bandwidths. The optimal partition vector for the admitted traffic can be heuristically approximated by a weighted summation of the single-source optimization results.

To serve the real-time conversational/streaming multimedia traffic, the buffer size should be small to achieve a low queuing delay. The large buffer technique performs poorly in this case. Furthermore, it is quite possible that the input traffic may be long-range dependent (LRD). To address these two problems, we develop a loss calculation technique for a finite-size partitioned buffer with fractional Brownian motion (FBM) input. FBM is a Gaussian process with stationary increments, characterized by three parameters: the mean arrival rate λ, the variance of traffic in a time unit σ^2, and the *Hurst parameter H* describing the correlation in the traffic arrival process. FBM is a model originally used to characterize LRD (self-similar) Internet traffic. However, recent studies show that the aggregate of SRD sources can be equivalently substituted by an FBM process, from the perspective that either the buffer overflow probability or available capacity is preserved after substitution. Therefore, the loss calculation technique based on FBM traffic is a general technique that can be applied to both LRD and SRD input sources.

One foundation of the proposed loss calculation technique is the recent observation that the loss probability in a finite buffer system with size B can be accurately estimated from the overflow probability $\text{P}\{X > B\}$ of an infinite buffer system, according to a simple mapping relation $\text{PLP} \approx \alpha\text{P}\{X > B\}$, where $\alpha < 1$ is a constant. We extend the mapping relation to the partitioned buffer (with finite size) case by proving that $\text{PLP}_j \approx \alpha_j\text{P}\{X > B_j\}$ for all J classes of traffic. Another foundation of the proposed technique is the calculation technique of overflow probability for FBM traffic, based on which we find the relation $\text{P}\{X > B_j\} \approx f(\text{P}\{X > B_{j-1}\}, \text{P}\{X > B_j - B_{j-1}\}, H)$, where the function $f(\cdot)$ captures the correlation between the queuing processes of neighboring partition regions by taking the Hurst parameter H into account. All the derivations and related references are given in [9]. By adopting the new techniques, the loss calculation in a finite-size partitioned buffer is as follows:

1. Model or substitute the input traffic with an FBM process.
2. Calculate $\text{P}\{X > B_1\}$ using the overflow calculation technique for the FBM process.
3. Iteratively calculate $\text{P}\{X > B_j\} \approx f(\text{P}\{X > B_{j-1}\}, \text{P}\{X > B_j - B_{j-1}\}, H)$ for $2 \le j \le J$, where the overflow calculation technique for FBM is applied in each region $B_j - B_{j-1}$.
4. Find the PLP according to the mapping relation as $\text{PLP}_j \approx \alpha_j\text{P}\{X > B_j\}$.

The optimal buffer partitioning concept can also be applied to the finite buffer FBM case for maximum resource utilization. The loss analysis based on FBM can not only deal with the finite-size effect, but also achieve a statistical multiplexing gain by considering the aggregate traffic directly (to be demonstrated by numerical examples).

The proposed partitioned buffer technique can be applied to all layer-coded multimedia traffic. When the wireless channel capacity varies with time, a layer-coded source can adjust the bandwidth requirement correspondingly by increasing or decreasing the number of layers transmitted for a better trade-off between efficient resource utilization and QoS satisfaction. The buffer configuration can be adjusted correspondingly to serve the traffic. An example of such adaptive service and the procedure to dynamically configure the partitioned buffer can be found in our study given in [3].

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Cross-Layer Optimization for TCP Data Traffic

While UDP is often used as the transport layer protocol for real-time multimedia service, TCP should be used for error-free delivery of data traffic. Over an air interface, the coupling between the TCP and link layers should be considered in order to efficiently utilize the scarce wireless resources as well as satisfy the throughput requirement. In our system, a TCP segment consists of a number of link layer packets to be ready for transmission over the wireless link.

TCP was originally developed for wireline networks with a reliable physical layer, where packet loss mainly results from network congestion. In a wireless environment, TCP perfor-

[1] *The Markov modulated fluid source or Markov modulated Poisson process are two classic short-range dependent models of video traffic [7].*

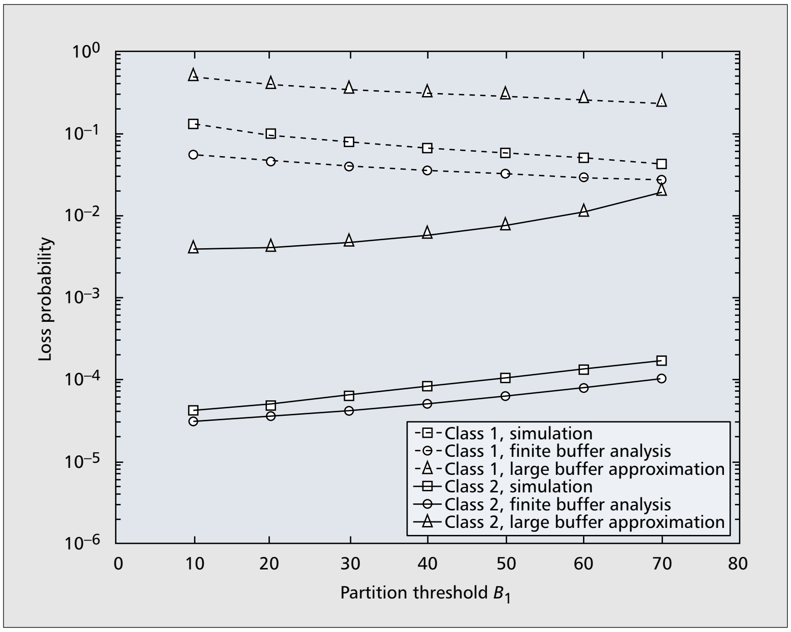

Figure 2. *Loss probabilities of two-layer coded traffic with* B = *80 packets and* c = *340 packets/s.*

mance can degrade severely due to unreliable link layer transmission. In order to reduce the link layer packet loss rate seen by TCP, the base station employs a radio link protocol (RLP) which uses a selective repeat automatic repeat request (SR-ARQ) error recovery mechanism to retransmit link layer packets not received successfully in the previous frames. Current RLP protocols normally have a constraint on the number of retransmissions (i.e., if a link layer packet cannot be received successfully after a given number of retransmissions, the sender will discard this packet and the subsequent link layer packets belonging to the same TCP segment not transmitted yet). This is important to control the transmission latency in a correlative fading channel where bursty packet losses may be induced due to inaccurate power control. We have shown that in the link layer a delayed multiple copy retransmission (DMCR) scheme under a maximum number of retransmissions can benefit higher-layer TCP performance [10]. However, as a TCP segment normally consists of many link layer packets, the bandwidth is not utilized efficiently if such a packet discarding event occurs after the main part of the TCP segment has been transmitted successfully. Hence, here we propose unlimited retransmissions (until successful receipt) to improve resource utilization for delay-tolerant data traffic. As retransmissions use part of the bandwidth, TCP will then see a reliable link with larger delay and smaller bandwidth.

A concern about unlimited retransmissions is possible performance degradation due to competing retransmissions between TCP and the link layer ARQ. TCP Reno and its variants usually employ coarse granularities of retransmission timeout value (typically multiples of 500 ms). Hence, the impact of competing retransmissions on TCP performance is not significant [11]. It is also shown in [12] that an unlimited retransmission mechanism is effective in terms of TCP throughput performance and wireless energy savings. Furthermore, a random early detection (RED) buffer rather than a drop-tail buffer is employed at the wireless sender side to control the possible delay inflation caused by unlimited retransmissions. RED has the ability to control the queuing delay and prevent correlation of packet dropping events (i.e., consecutive packet losses). Each TCP flow has its own RED buffer for the wireless link. As the per-flow scheme applies only at the edge of the DiffServ domain, it will not incur any scalability problem. In addition to the RED buffer, each flow (say, flow i) over the wireless link also keeps a link layer buffer for transmission and retransmission of link layer packets. The two buffers can be viewed as a virtual RED buffer at the link layer. To serve this virtual buffer, up to M_i link layer packets from TCP flow i can be scheduled (by multicode CDMA transmissions) in each radio frame (i.e., link layer time frame) with SINR value Γ_i, where (M_i, Γ_i) is referred to as the resource vector.

It has been shown that TCP throughput T is determined by the packet loss event rate and round-trip delay [13]. Assuming that a TCP flow i experiences a fixed delay and a fixed packet loss event rate in the wireline part, its throughput is determined mainly by the performance of the wireless link, which is further determined by the wireless resource vector (M_i, Γ_i). To achieve

(K_5, K_6)		(20, 0)	(16, 2)	(12, 4)	(8, 6)	(4, 8)	(0, 10)
Optimization with	B_1^*	821	801	779	754	727	696
large buffer	c^*	1369.2	1369.2	1369.2	1369.2	1369.2	1369.2
approximation	PLP_1	2.04×10^{-3}	1.35×10^{-3}	1.07×10^{-3}	9.97×10^{-4}	9.20×10^{-4}	8.09×10^{-4}
and simulated PLP	PLP_2	1.10×10^{-6}	8.90×10^{-7}	8.33×10^{-7}	5.47×10^{-7}	4.67×10^{-7}	2.19×10^{-6}
Optimization with	B_1^*	946	912	878	848	823	805
finite buffer	c^*	1312.9	1279.1	1248.4	1220.0	1193.3	1167.8
analysis and	PLP_1	5.43×10^{-3}	5.68×10^{-3}	5.29×10^{-3}	4.71×10^{-3}	4.88×10^{-3}	5.05×10^{-3}
simulated PLP	PLP_2	6.25×10^{-5}	3.49×10^{-5}	3.87×10^{-5}	4.14×10^{-5}	7.10×10^{-5}	6.71×10^{-5}

Table 1. *Optimal buffer configurations and simulated packet loss probabilities: large buffer approximation vs. finite buffer analysis.*

the target TCP throughput, for each possible M_i value the Γ_i value is determined iteratively. For an intermediate Γ_i value, we calculate the wireless link delay and packet loss event rate, and further estimate the achieved TCP throughput T. If T is greater (less) than the required throughput, we decrease (increase) the Γ_i value. This procedure is repeated until the achieved T converges to the target throughput, and we call the obtained (M_i, Γ_i) a *feasible resource vector*. There may exist many feasible resource vectors for a target throughput. We define the *normalized resource amount* required by flow i in a radio frame as $m_i\Gamma_i/(G + m_i\Gamma_i)$ [8], where m_i ($\leq M_i$) is the number of actually scheduled link layer packets in the radio frame and G the CDMA processing gain. The average normalized resource amount over all the radio frames is termed the *equivalent resource amount*. Among all the feasible resource vectors, we can get the optimal one that minimizes the equivalent resource requirement at the link layer, therefore achieving the maximal resource utilization. The mapping between the required throughput and the optimal resource vector is obtained at the connection setup via a table lookup. More detailed discussion of the cross-layer optimization is given in [8].

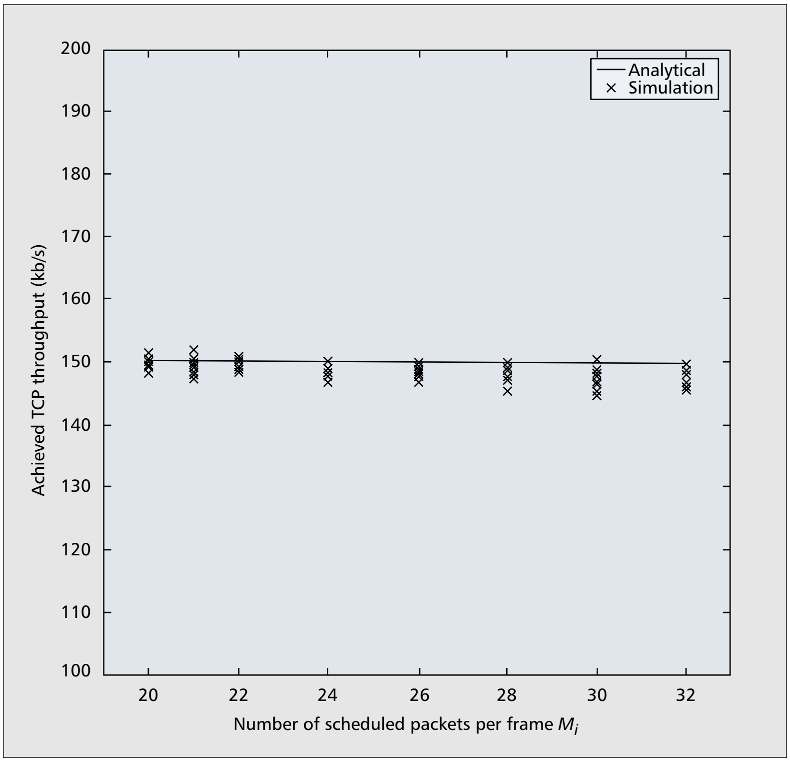

Figure 3. *Achieved TCP throughput vs.* M_i *for a data flow with a throughput requirement of 150 kb/s.*

Performance Evaluation

In this section analysis and computer simulation results are presented to demonstrate the performance of the proposed resource allocation techniques. We first show the resource utilization improvement by optimally configuring a partitioned buffer for real-time traffic, and then demonstrate the performance of the cross-layer optimization for non-real-time data traffic.

Optimal Buffer Configuration with Accurate Loss Calculation

For a large buffer, the effectiveness of the optimal buffer partitioning and minimal effective bandwidth to improve resource utilization has been demonstrated in [7]. Here, we show that resource utilization can be further improved by accurately calculating the PLP instead of using overflow approximation. The units of traffic arrival rate and channel capacity are packet per second, and the unit of buffer size is packet.

The first example demonstrates the accuracy of the proposed loss calculation technique for a partitioned buffer of finite size. Consider a two-class FBM input served with a partitioned buffer of size 80. The channel capacity c is 340. In this setting, the worst case delay is 80/340 ≈ 235 ms, small enough for the streaming class services. The FBM source consists of a class-1 (enhancement layer) FBM with $\lambda_1 = 20$, $\sigma_1^2 = 50$, and a class-2 (base layer) FBM with $\lambda_2 = 300$, $\sigma_2^2 = 500$, with the Hurst parameter $H = 0.83$ for both classes. The buffer is partitioned into two regions to differentiate the loss behaviors of class 1 and class 2. Figure 2 shows the loss probabilities of two classes vs. the partition threshold B_1, obtained from analysis and computer simulation. It is observed that in this small buffer example, the simulation and finite buffer analysis results are in a very close match, while the large buffer approximation deviates far from the simulation results, especially for the class 2 traffic where the finite buffer effect is more obvious. Accurate loss analysis in the small buffer case is particularly useful for real-time traffic having both delay and loss requirements.

The accurate loss calculation technique can be used in admission control to further improve resource utilization. Here we re-examine the admission control example considered in [7] by substituting the input traffic with the FBM process and applying the accurate loss calculation technique instead of the overflow approximation. Heterogeneous two-class on/off sources (type-5 and type-6 sources in [7]) are considered, with the PLP requirement of 10^{-2} and 10^{-4} for class 1 and class 2, respectively. Specifically, type-5 sources with the minimal effective bandwidth of $c_5^* = 68.46$ and type-6 sources with $c_6^* = 2c_5^*$ are multiplexed in a partitioned buffer of size $B = 1000$ and served with channel capacity $c = 20c_5^*$. With K_5 (K_6) denoting the number of accepted type-5 (type-6) sources, Table 1 lists the optimal partition threshold B_1^* and the minimal channel capacity to guarantee QoS, obtained from large buffer approximation and FBM finite buffer analysis, respectively. The simulation results for loss probability corresponding to the two configurations are also given in Table 1. It can be observed that the minimal effective bandwidth and linear admission control from large buffer approximation are a conservative resource allocation approach, over-satisfying the QoS; on the other hand, the FBM finite buffer analysis technique clearly

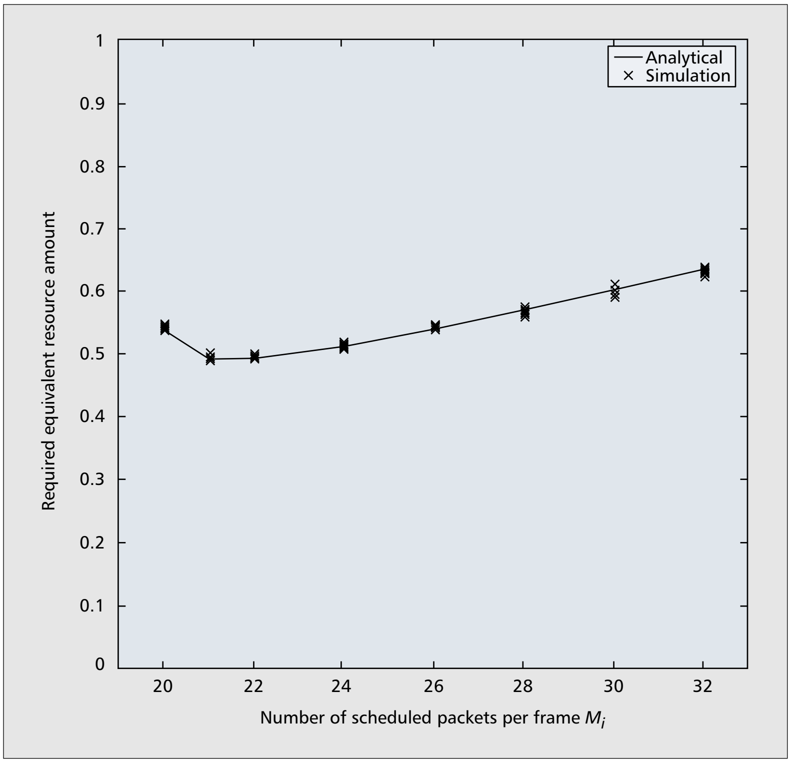

Figure 4. *Equivalent resource requirement vs.* M_i *for a data flow with a throughput requirement of 150 kb/s.*

improves resource utilization by more accurate calculation of the loss probability in a finite buffer and better exploitation of statistical multiplexing among multiple flows.

Performance of Cross-Layer Optimization

Consider a single data flow initiated from an MS with a target throughput requirement of 150 kb/s in a single-cell CDMA system. The radio frame length is set to be 20 ms. Each link layer packet has 192 bits, of which 160 bits are payload. The processing gain G for each link layer packet is 128. In the transport layer, a TCP segment has a fixed size of 8000 bits. To achieve the target throughput requirement, we first obtain the feasible resource vectors (M_i, Γ_i), calculate the equivalent resource amount required by each vector, and then run computer simulations to examine the selected feasible resource vectors. For each selected feasible resource vector, 10 independent simulations are carried out. Each simulation runs for 30, 000 radio frames, and the statistics are collected in the last 24, 000 frames. In the simulations, the achieved throughput is traced through the accumulative Acknowledgments (ACKs) for TCP segments from the correspondence node.

Figures 3 and 4 show the achieved TCP throughput and required equivalent resource amount for different feasible resource vectors, respectively. As the feasible resource vector has only one degree of freedom under the constraint of the target throughput, only values of M_i are shown in the horizontal axis. It can be seen that the simulation results closely match the numerical analysis results. Each feasible resource vector can guarantee the throughput requirement of a TCP flow. An optimal point ($M_i = 21$ in this example) can be found among the feasible resource vectors for the minimal equivalent resource requirement, thus achieving maximal resource utilization.

Conclusion

The prominent characteristics of the 3G/4G wireless communications in China are coexistence of heterogeneous access technologies and high distribution density of mobile users. This article addresses efficient resource management of such a 3G/4G wireless network by suggesting an all-IP DiffServ wireless architecture. In particular, we present two novel QoS techniques for efficient resource allocation. One contribution is a partitioned buffer scheme that provisions UDP layer-coded multimedia traffic via an AF PHB with delay and packet loss guarantee. Maximum resource utilization is achieved by accurate loss analysis in the partitioned buffer, exploitation of statistical multiplexing through FBM modeling, and adaptive optimal buffer configuration for the time-varying wireless channel. The other contribution is the implementation of an AF PHB to serve TCP data traffic with a guaranteed throughput. To improve TCP performance over the air interface, the coupling between the TCP and link layers is considered, and a cross-layer optimization technique is proposed for efficient utilization of wireless resources. Although this study is motivated by the stringent requirement for efficient resource allocation in China's 3G/4G wireless communications, the proposed QoS and resource allocation techniques can be applied to general 3G/4G wireless networks.

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The average normalized resource amount over all the radio frames is termed the equivalent resource amount. Among all the feasible resource vectors, we can get the optimal one that minimizes the equivalent resource requirement at the link layer, therefore achieving the maximal resource utilization.