A Framework for Elastic QoS Provisioning in the cdma2000 1xEV-DV Packet Core Network

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ABSTRACT

With the fast proliferation of QoS-enabled wireless packet networks, the need for effective QoS control is increasing. In this article we focus on QoS provisioning in the cdma2000 lx evolution for high-speed integrated data and voice (1xEV-DV) packet core network. We investigate a dynamic bandwidth provisioning method that is able to increase a service provider's revenue. It is achieved by releasing unutilized bandwidth for use by other profitable services. The proposed method is implemented as an SNMP-compliant management information base and deployed at the packet data serving node. The experiments conducted on the LG Telecom 1xEV-DV testbed show that the method can increase the bandwidth for the conversational class and guarantee adequate service quality for the background class as well.

INTRODUCTION

Since the Third-Generation Partnership Project 2 (3GPP2) took up the cdma2000® 1xEV-DV standard development activities, two revisions have been published: cdma2000 Revisions C and D. Revision D, its update completed in February 2004, supports data rates of up to 3.072 Mb/s over the forward link and up to 1.84 Mb/s over the reverse link. It enables wireless networks to support data and voice services by satisfying various quality of service (QoS) requirements. 1xEV-DV is also compatible with the ANSI-41 core network standards.

MOTIVATION

One of the challenges in a multiservice system with heterogeneous types of traffic such as the 1xEV-DV system is that the limited bandwidth has to be efficiently provisioned among multiple traffic types. 1xEV-DV is expected to support diverse real-time applications such as videoconferencing, voice over IP (VoIP), online gaming, and 3G multimedia, demanding different QoS and bandwidth. Both 3GPP and 3GPP2 have proposed four different QoS classes:

- *Conversational*
- *Streaming*
- *Interactive*
- *Background*

Bandwidths for these classes need to be properly provisioned for efficient use of the limited network resources.

In order to meet the push for higher data rates driven by the growing demand for wireless multimedia services, the 1xEV-DV should offer significantly higher data rates both in air and on land. For example, large overhead in VoIP communications requires high bandwidth in the landline. A scenario where a user is using an 8 kb/s codec and sending frames every 20 ms results in voice payloads of 20 bytes for each packet. To transfer these voice payloads over Real-Time Transport Protocol (RTP), the following must be added: an IP header of 20 bytes, a User Datagram Protocol (UDP) header of 8 bytes, and an additional 12 bytes for RTP. Robust header compression (ROHC) reduces the 40-byte header to a minimum of 1 byte. On the landline side, however, the following remains as an overhead: Point-to-Point Protocol (PPP) header of 4 bytes, trailer of 2 bytes, generic routing encapsulation (GRE) header of 16 bytes, IP header of 20 bytes, Ethernet header of 14 bytes, and trailer of 4 bytes. This is a total of 61 bytes overhead to transmit a 20-byte payload. This situation increases the potential for landline 1xEV-DV to be bottlenecked if a proper bandwidth provisioning method is not deployed.

BANDWIDTH PROVISIONING

The typical provisioning mechanism in practice today is to partition the available bandwidth among the different traffic classes. Usually this partitioning is static and accomplished on a long-term basis. Static provisioning is simple to implement but results in poor performance when the traffic patterns do not conform to the partitioning. So far, overprovisioning has been widely used to absorb traffic fluctuations. Landline 3G networks have been overprovisioned as well. However, massive overprovisioning based on the

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amount of traffic measured at peak times is inefficient in terms of resource allocation. Dynamic bandwidth provisioning according to the underlying network condition can increase bandwidth utilization and service provider revenue.

It has been observed that *service degradation* approaches may increase bandwidth utilization and service provider revenue [1, 2]. The common goal of service degradation approaches is to increase bandwidth utilization by lowering the QoS levels of existing users. Reference [2] evaluated the effects of service degradation caused by adaptive bandwidth allocation. Although system performance can be improved by allowing service degradation, the scheme should be used very carefully. First, the impact of degradation on the QoS of individual users should be considered. Second, deploying the schemes to differentiated services (DiffServ)-enabled Internet is complex because the service level agreement (SLA) between users and service providers needs to be renegotiated before the service degradation happens.

Over the last few years, call admission control policies based on *movable boundaries* [3, 4] have attracted widespread interest in improving total channel utilization. Reference [3] proposed a movable boundary scheme that dynamically adjusts the number of channels for voice and data traffic. With this scheme, the bandwidth can be utilized efficiently while satisfying the QoS requirements for voice and data traffic. In [4], a dual threshold bandwidth reservation (DTBR) scheme for an integrated voice and data system was proposed. The DTBR scheme enables complete sharing (CS) of the overall bandwidth, thus leading to efficient usage of wireless resources. However, these approaches require an a priori traffic descriptor in terms of the parameters of a deterministic or stochastic model. One disadvantage is that real traffic does not follow the traffic model they assume. Although these schemes were evaluated extensively, the complexities of the algorithms often make them impractical.

The contributions of this article are twofold: to propose a practical bandwidth reprovisioning scheme without any assumption on the statistical properties of the traffic, and to design and implement the proposed scheme as an SNMP management information base (MIB) for longterm bandwidth provisioning in a 1xEV-DV system. The basic concept of the proposed reprovisioning scheme is to increase the bandwidth for the conversational class appropriately by restricting the bandwidth of the background class. However, we guarantee an upper bound on the packet drop probability of the background class so that it can still enjoy adequate service quality. The reprovisioning scheme is applicable to the landline side of a 1xEV-DV system, not traffic over the air. Our approach has several benefits. First, this approach is simple. Second, the proposed scheme is represented by managed objects (MOs) and implemented as an SNMP-compliant MIB, and therefore, it can be integrated with the Internet Engineering Task Force (IETF) network management framework. It enables service providers or network operators to monitor link utilization and reprovision bandwidth. Third, our approach does not use a traffic model. Because traffic from different sources is multiplexed in packet data serving node (PDSN), the QoS experienced depends on their aggregated behavior. Our approach simply uses *the law of large numbers* to predict aggregate behavior.

In the rest of the article we first present the 1xEV-DV QoS classes. The QoS reprovisioning method, realized as an SNMP-compliant MIB in PDSN, is introduced. The experimental results are shown, presenting the obtained bandwidth gain. Finally, conclusions and future works follow.

1xEV-DV QoS Traffic Classes

The cdma2000 radio bearer services and their associated QoS parameters are defined in [5]. Currently, two types of QoS are defined in the 1xEV-DV air interface: *assured* and *nonassured* QoS. Nonassured QoS is defined to determine a priority level of users that compete for air interface resources. Mobile users with the lowest-grade priority could be denied service. For assured service, there are independently specified QoS parameters for the forward and reverse links. A mobile station may also specify both a requested value and an acceptable value for QoS parameters. Assured QoS is made up of the following set of parameters: priority, minimum requested data rate, minimum acceptable data rate, maximum requested delay, maximum acceptable delay, requested data loss rate, and acceptable data loss rate. A 1xEV-DV mobile user is able to request QoS settings during call setup using a QoS block of bits (BLOB). For a detailed introduction to the QoS between mobile users and base stations (BSs), the reader is directed to [5].

For end-to-end QoS, the 1xEV-DV specifications define four QoS classes: *conversational*, *streaming*, *interactive*, and *background*. The conversational class provides strict delay guarantees, while the background class offers no qualitative or quantitative guarantees. The conversational and streaming classes are intended for real-time traffic such as voice and video applications. The former requires low delay and low data loss, and is sensitive to delay variations. The latter is less sensitive to delay, but may require high bandwidth for supporting one-way bulk streaming data transfer (e.g., live TV applications). The interactive class is suitable for data transfer that has request-response patterns. Example applications are Telnet and FTP. The background class is similar to *best effort* traffic and suitable for bulk and asynchronous traffic flows such as email.

The four QoS classes will be mapped to open IETF QoS architectures (e.g., DiffServ and integrated services, IntServ). For example, taking the 1xEV-DV traffic class model and DiffServ framework into account, the conversational class will be mapped to the expedited forwarding (EF) per-hop behavior (PHB). Examples of QoS mappings can be found in [6, 7]. An SLA should be established between 1xEV-DV networks and peer networks. The SLA will be enforced at the border router (BR).

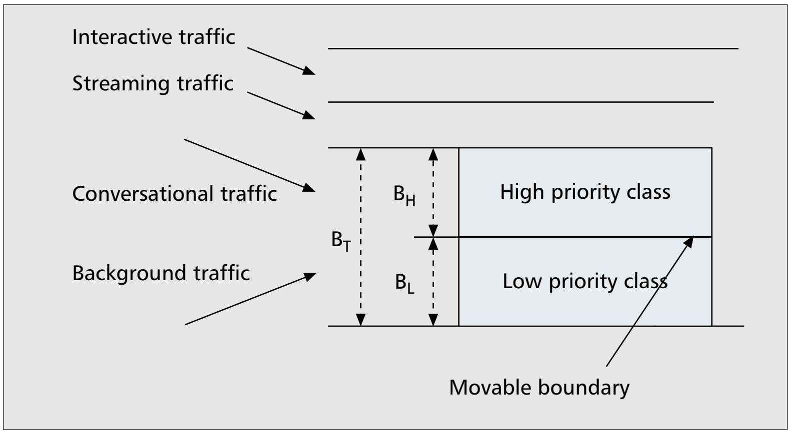

Figure 1. *Logical structure of the provisioned bandwidth.*

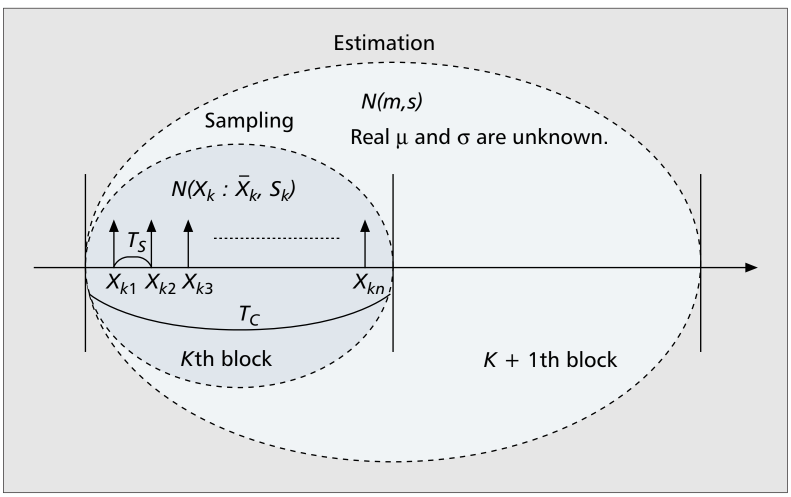

Figure 2. *Bandwidth reprovisioning process.*

Description of the Proposed Method

This article focuses on appropriate QoS provisioning between four different QoS classes (especially conversational and background). The system under consideration is an integrated voice/data mobile network with supporting real-time multimedia applications. 1xEV-DV (especially, A8 and A10 interfaces) is the target environment for which we conducted several experiments. The network comprises a number of BSs connected by a wireline backbone network through PDSN equipment. The proposed reprovisioning method is designed and implemented as an SNMP MIB and runs in a PDSN with an SNMP agent.

There is a movable boundary between high priority class (e.g., conversational class) and low priority class (e.g., background class). The total bandwidth of both classes is B_T b/s. The bandwidths provisioned for low and high priority classes are B_L and B_H b/s, respectively. The assumption is $B_T = B_L + B_H$, and throughput loss in B_L directly translates into gains in B_H. The logical structure of the link provisioned for four different classes is shown in Fig. 1.

The goal of the proposed reprovisioning scheme is to give as much bandwidth as possible to the conversational class by restricting the amount of bandwidth of the background class. In our scheme, a boundary between two classes moves according to the link utilization of the low priority traffic, while guaranteeing an upper bound on the packet drop probability of the background class so that it can still enjoy adequate service quality.

The bandwidth control has two stages: bandwidth estimation and bandwidth reprovisioning. Let us consider Fig. 2. The estimation and reprovisioning are done by each block. The bandwidth estimation is to estimate the unknown real link utilization and variance of the (k and $k + 1$)th block with measured samples in the kth block, assuming that the traffic characteristics of two blocks are the same. The bandwidth reprovisioning is to reallocate the bandwidths for the $k + 1$)th block with guaranteeing an upper bound on the packet drop probability of the background class. The block size should be large enough so that the number of samples will be sufficiently large, and small enough such that two consecutive blocks are strongly correlated.

Bandwidth Estimation

We monitor the utilization of the bandwidth provisioned for the background class at regular intervals T_S. Suppose that $X_{k1}, X_{k2}, \ldots, X_{kn}$ is a sample from a large population having unknown real mean link utilization of the background class μ and variance σ^2. Let us consider an interval estimate of μ. When n is sufficiently large, we can establish a confidence interval for μ by considering the sampling distribution of $\bar{X}_k$. According to the central limit theorem, we can expect that the sampling distribution of $\bar{X}_k$ to be approximately normally distributed with mean $\mu_{\bar{X}_k} = \mu$ and standard deviation $\sigma_{\bar{X}_k} = \sigma/\sqrt{n}$.

When σ is unknown and $n \geq 30$, S_k can replace σ. Here, S_k is the sample standard deviation of the kth block. A (1 – α) 100 percent confidence interval for μ is given by

$$\left(\bar{X}_k - z_{\alpha/2}\frac{S_k}{\sqrt{n}},\ \bar{X}_k + z_{\alpha/2}\frac{S_k}{\sqrt{n}}\right), \qquad (1)$$

where $z_{\alpha/2}$ is the z-value leaving an area of α/2 to the right in standard normal distribution probability density function (pdf). To find 95 percent confidence interval, we get $z_{0.025} = 1.96$; to find 99 percent confidence interval, we get $z_{0.005} = 2.58$.

We denote the term $\bar{X}_k + z_{\alpha/2}\, S_k/\sqrt{n}$ as $\theta_{k,k+1}$. It is called the *upper confidence limit*, and can be interpreted as the safety margin left to serve the burstiness of traffic. Also, bandwidth $\theta_{k,k+1}$ implies that the background class is guaranteed to have (1 – α) 100 percent link utilization, and therefore, (1 – α) 100 percent traffic will be delivered to the next hop with this bandwidth.

Bandwidth Reprovisioning

The upper confidence limit $\theta_{k,k+1}$ is the bandwidth to guarantee (1 – α) 100 percent throughput of background class in the kth and ($k + 1$)th block. Since the initially provisioned bandwidth for the background class is B_L, from Eq. 1 we reprovision the bandwidth for the background class in the ($k + 1$)th block (B_{Lk+1}):

$$B_{L_{k+1}} = Min(B_L, \theta_{k,k+1}). \quad (2)$$

Note that when $B_L \le \theta_{k,k+1}$, bandwidth reprovisioning does not happen; the throughput of the background class is not guaranteed to have $(1 - \alpha)$ 100 percent. When $B_L > \theta_{k,k+1}$, unutilized bandwidth $B_L - \theta_{k,k+1}$ is released for use by conversational class with guaranteeing $(1 - \alpha)$ 100 percent utilization of background class. We reprovision the bandwidth for the conversational class in the $k + 1$th block ($B_{H_{k+1}}$) as

$$B_{H_{k+1}} = Max(B_H, B_H + B_L - \theta_{k,k+1}). \quad (3)$$

Equation 3 is the consequence of giving the unutilized bandwidth of the background class to the conversational class. Note that when $B_L \le \theta_{k,k+1}$, the conversational class bandwidth for the $(k + 1)$th block is the same as the previous (kth) block. However, when $B_L > \theta_{k,k+1}$, we can take advantage of bandwidth reprovisioning, since the bandwidth of the conversational class for the $(k + 1)$th block increases by $B_L - \theta_{k,k+1}$ This bandwidth reprovisioning guarantees $(1 - \alpha)$ 100 percent throughput of the background class. Thus, the background class can still enjoy the adequate service quality while lending its unutilized bandwidth to the conversational class.

Elastic Provisioning MIB

The aim of this section is to give a brief introduction of the structure of the elastic provisioning (EP) MIB. The design objectives of the MIB are to integrate the QoS provisioning scheme with the IETF standard management framework and provide a network operator's interface. Although the MIB was implemented in PDSN, deploying the MIB to a BS controller (BSC) is also possible.

Structure of the EP MIB

The EP MIB is an SNMP-compliant MIB for elastic bandwidth reprovisioning as well as utilization monitoring. Figure 3 shows the logical structure of the EP MIB. It consists of three tables, represented by table entries in Fig. 3.

The `epProvTable` provides the ability to provision bandwidth via SNMP SET operation. Bandwidth can be provisioned by writing an `epProvEntry`. The `epProvDataSource` object identifies network interface. The `epProvServiceClass` object specifies the service class (i.e., conversational (1), streaming (2), interactive (3), and background (4)). The `epProvBandwidth` object specifies the bandwidth to be allocated. The `epProvTable` is useful for initial bandwidth provisioning of each class. For example, the bandwidth of a conversational class can be reprovisioned to 15 Mb/s by `snrnpset` (…, `epProvServiceClass` = 1, `epProvBandwidth` = 1,500,0000, …).

The `epCtrlTable` is used to define reprovisioning functions for one or more of the network interfaces. The details are two traffic classes separated by movable boundary (i.e., `epCtrlLowPriorityClass` and `epCtrlHighPriorityClass`), and the control parameters $z_{\alpha/2}$, T_s, n, and T_c, which are `epCtrlGuaranteedDropRatio`, `epCtrlSampleInterval`, `epCtrlWindowSize`, and `epCtrlActionInterval`, respectively. A network operator can activate the reprovisioning algorithm by creating a new row of `epCtrlTable`. If the operator creates a row with `epCtrlSampleInterval` = 0, statistics is recorded to a stat table without reprovisioning.

Figure 3. *Logical structure of the elastic provisioning MIB.*

The `epStatTable` records the results of each reprovisioning. The stat table is indexed by `epCtrlIndex` and then `epStatIndex`. As each reprovisioning interval occurs, a new row is added to `epStatTable` with the same `epCtrlIndex` as the other rows for this reprovisioning and with an `epStatIndex` of one more than the value for the row corresponding to the previous reprovisioning interval. For each of the M rows of `epCtrlTable`, there is a set of rows of `epStatTable`. For information on the activated reprovisioning algorithm specified by the row in the `epCtrlTable`, the stat table contains one row for each bandwidth reprovisioning. Thus, as long as the `epCtrlTable` information is not changed, one row is added to the stat table each time reprovisioning happens. A network operator can obtain the results through SNMP with periodic polling of `epStatTable`.

The `epCtrlMaxRows` object limits the size of

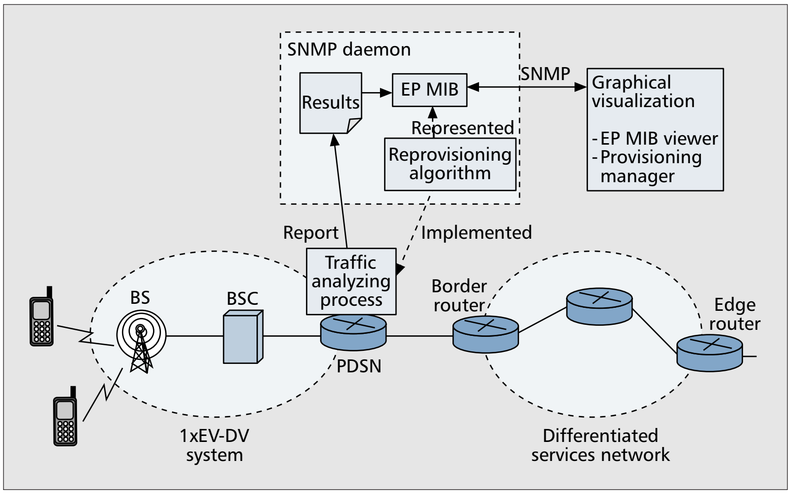

Figure 4. *MIB implementation environment.*

Parameters	Values	Remarks
`epCtrlLowPriorityClass`	4	Background class
`epCtrlHighPriorityClass`	1	Conversational class
`epCtrlGuaranteedDropRatio`	5,1	Guaranteeing drop rate below 5% ($z_{0.025}$ = 1.96). Guaranteeing drop rate below 1 percent ($z_{0.005}$ = 2.58).
`epCtrlSampleInterval`	2	2 s sampling interval
`epCtrlwindowSize`	300	300 samples will be used for calculating upper confidence limit ($\theta_{k,k+1}$)
`epCtrlActionInterval`	600	10 min (600 s) for reprovisioning interval.

Table 1. *Summary of experiment conditions.*

the stat table. The number of rows in the stat table can be expressed as $\Sigma_{i=0}^{M}$ `epCtrlMaxRows(i)`, where `epCtrlMaxRows(i)` is value of `epCtrlMaxRows` for row i of the `epCtrlTable`, and M is the number of rows in the `epCtrlTable`. Once the number of rows for a stat table becomes equal to `epCtrlMaxRows`, the set of rows for that reprovisioning functions as a circular buffer. As each new row is added to the set, the oldest row associated with this control is deleted.

Implementation

The network management system is composed of a variety of modules running on a UNIX system (Solaris 2.8). The modules include *EP MIB Viewer* and *Provisioning Manager*. Unfortunately, since our current PDSN equipment does not support the 3GPP2 IOS 4.3 A10/A11 interface specification for 1xEV-DV, we replaced the PDSN with an emulation device called a packet interface processor (PIP, developed by LG Electronics Inc.), which is an implementation of packet control function (PCF). We have implemented the proposed reprovisioning function in PIP. Our system consists of the following components shown in Fig. 4.

Traffic Analyzing Process: *The traffic analyzing process* gathers performance information such as number of inbound/outbound traffic, number of packet drops, and number of VoIP calls PDSN serves on a per class basis. It runs on the vxWorks 5.4 operating system with 128 Mbytes of main memory. It arranges raw data to a regular format, writes it into a binary file, and reports the collected information to the SNMP agent periodically. Bandwidth control functions are included in this operation and maintenance (OAM) process as well.

The EP MIB with Reprovisioning Algorithm: In order to develop the prototype quickly and be able to focus on MIB implementation, we decided to use a publicly available SNMP platform. Our prototype uses the ucd-snmp-4.2.6 package. In our implementation EP MIB communicates with the *traffic analyzing process* using TCP/IP socket connection. The experiment results as well as the measured data are constructed as an EP MIB.

Experimental Results and Discussions

In this section the performance evaluation of the proposed reprovisioning scheme is presented in terms of bandwidth gain of conversational class. We have performed a number of experiments on a 1xEV-DV system connected to an IP core network through PDSN equipment. We varied traffic sources, sampling/action interval, and guaranteed packet drop probability of the background class (determined by $z_{\alpha/2}$). The system parameters used in the experiments are summarized in Table 1.

QoS Traffic Models for 1xEV-DV

3GPP2 has proposed system evaluation methodology for simulating 1xEV-DV performance. The proposal [8] includes data traffic models such as HTTP, FTP, WAP, and video streaming. With significantly increased data rates, 1xEV-DV enables new mobile applications such as online gaming. Recently, mobile gaming models are drawing attention in research for traffic source models [9, 10]. For a traffic model in an access link such as A8 and A10, a model considering aggregation of various types of traffic is reasonable; real traces measured at an Internet access link are useful for background class traffic models.

Experiments

We performed a reprovisioning experiment with the Auckland-IV data set [11], which is a continuous 6-1/2-week Global Positioning System (GPS)-synchronized IP header trace taken at the University of Auckland Internet access link. The reprovisioning experiment was performed over a period of 23 h. Initially we provisioned 200 and 300 kbytes/s for the background and conversational class, respectively.

The reprovisioning experiment was performed with the following parameters: sampling interval $T_s = 2$ s, window size $n = 300$, control interval $T_c = 10$ min, and $z_{0.025}(1.96)$. The second experiment was performed with the same parameters as the first except using $z_{0.005}(2.58)$ instead of $z_{0.025}(1.96)$. Thus, the bandwidth for the conversational class is reprovisioned every 10 min, guaranteeing a packet drop rate of the background class below 5 or 1 percent using the previous 300 measured samples. The results are recorded to the stat table every 10 min. We obtained 138 (10 min × 6 × 23 h) rows of `epStatTable` as results.

Bandwidth Estimation — Figure 5a shows the actual traffic load and estimated bandwidths that guarantee the packet drop rate of the background class below a given threshold (1 or 5 percent). As shown in Fig. 5a, when $0{:}00 \le t \le 8{:}30$, $8{:}50 \le t \le 9{:}00$, $t = 9{:}20$, $t = 11{:}00$, $t = 14{:}50$, $15{:}20 < t < 16{:}00$, $t = 16{:}30$, $17{:}20 \le t \le 19{:}20$, and $t \ge 19{:}40$, the upper confidence limits of $z_{0.005}$ are smaller than the initially provisioned bandwidth 200 kbytes/s (i.e., $\theta_{k,k+1} < B_L$); therefore, bandwidth reprovisioning happens. During these periods, from Eq. 3, additional $200 \times 8 - \theta_{k,k+1}$ Mb/s bandwidth is reprovisioned to the conversational class while guaranteeing the packet drop rate of the background class below 1 percent. When the reprovisioning algorithm runs with the parameter of `epCtrlGuaranteedDropRatio` = 5, reprovisioning does not happen at $t = 8{:}30$, $t = 10{:}10$, $10{:}30 \le t \le 10{:}50$, and $t = 16{:}30$ because the upper confidence limits of $z_{0.025}$ are greater than the initially provisioned bandwidth 200 kbytes/s (i.e., $\theta_{k,k+1} > B_L$).

Bandwidth Gain — Let us consider the bandwidth gain of the conversational class. The bandwidth gain is an important measure because it has implications not only for reducing call blocking probability of the conversational class but also increasing a service provider's revenue. Figure 5b shows the bandwidth gain of the conversational class at each reprovisioning interval, 10 min. Obviously, we get a gain when the upper confidence limit $\theta_{k,k+1}$ is smaller than 200 kbytes/s. The maximum gain with $z_{0.025}$ is 173.957 kbytes/s at $t = 23{:}00$. We do not have any gain for $11{:}10 \le t \le 14{:}40$, because the upper confidence limits are greater than the originally provisioned bandwidth, 200 kbytes/s.

In order to find the increased number of VoIP calls that can be admitted to the system, we conducted the following experiment. For $16{:}40 \le t \le 20:00$, we offered load of 510 Erlang (average call arrival rate of 170 calls/min and call holding time of 3 min). VoIP call arrives at PDSN with the rate of 32.4 kb/s including transmit overhead (8 kb/s codec rate + 24.4 kb/s overhead). We observed that an additional 0–31 users were accepted to the system for each control block, 10 min. For example, for $18{:}40 \le t \le 18{:}50$, the number of accepted users increased from 309 without reprovisioning to 330 with reprovisioning. Call blocking probability decreased from 9 percent without reprovisioning to 3 percent with reprovisioning.

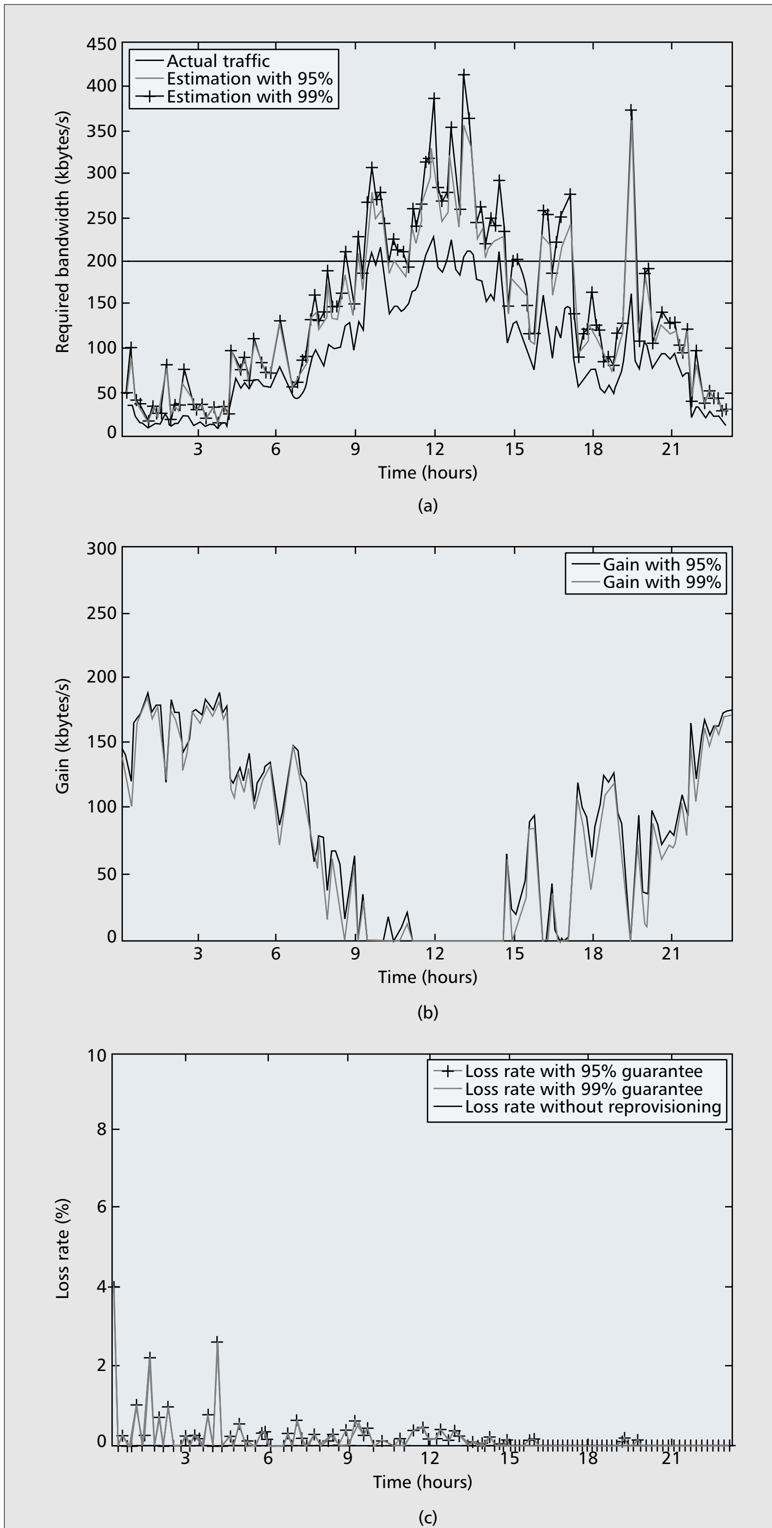

Figure 5. *The Auckland-TV real trace example: a) the actual traffic load and the upper confidence limits that guarantee the packet drop probability below given thresholds (1 and 5 percent); b) bandwidth gains; c) packet loss probability.* T_C *= 10 min.*

Packet Drops — Figure 5c shows the packet loss probability of the background class. For the period of no reprovisioning, the packet loss probability depends on the offered traffic load. The 0–0.408 percent packet drops were monitored between $t = 11{:}10$ and $t = 14{:}40$. When reprovisioning happens, the background class traffic experiences 0-3.931 percent packet drops with $z_{0.025}(1.96)$. For example, at $t = 7{:}50$, the packet loss probabilities of the background class are 0.036, 0.157, and 0.236 percent without reprovisioning, and reprovisioning with guaranteeing the packet drop probability of the background class below 1 and 5 percent, respectively. In most cases, the packet drop probabilities we monitored in this experiment were bounded below a given threshold.

CONCLUSIONS

The 1xEV-DV is expected to support diverse real-time applications such as videoconferencing, voice over IP, and 3G multimedia, demanding different QoS and bandwidth. Bandwidths for a 1xEV-DV system need to be properly provisioned for efficient use of the limited network resources.

Releasing unutilized bandwidth for use by other services is important for increasing service provider revenue. We have investigated how to increase the available bandwidth for high priority class (e.g., conversational) traffic while keeping the packet loss probability of low priority class (e.g., background) traffic below a given threshold. The bandwidths obtained from Eqs. 2 and 3 are reprovisioned for the new bandwidths of the next block. Thus, unutilized bandwidth of the background class is reprovisioned for the conversational class. The experiments performed on the LG Telecom 1xEV-DV testbed using a real link trace showed that the proposed reprovisioning scheme enables the bandwidth of the conversational class to be increased by 0–57.7 percent.

The proposed scheme has two special features. First, the reprovisioning scheme is represented as a set of managed objects and implemented as an EP MIB. It conforms to the IETF network and system management framework, so SNMP protocol is used for reprovisioning. Second, our scheme does not need to have any assumption on the statistical properties of individual sources, because traffic from different sources are multiplexed in PDSN on which the control algorithm runs; the QoS experienced generally depends on their aggregated behavior.

For more precise control, we are currently extending our study by combining the proposed reprovisioning scheme with a traffic forecasting method.

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