A Framework for Elastic QoS Provisioning in 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 cdma2000 *1x evolution for data and voice* (1xEV-DV) packet core network. We investigate a dynamic bandwidth provisioning method that is able to increase service provider's revenue. It is achieved by releasing unutilized bandwidth for use by other profitable services. The proposed method is implemented as a simple network management protocol (SNMP)-compliant management information base (MIB) and deployed at *packet data serving node* (PDSN). The experiments conducted on 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.

1 Introduction

Since the third-generation partnership project 2 (3GPP2) took up the cdma2000®[5] 1xEV-DV standard development activities, two of the revisions have been published - cdma2000 Revision C and D. The Revision D, completed an update 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 that support data and voice services by satisfying various quality-of-service (QoS) requirements. The 1xEV-DV is also compatible with the ANSI-41 core network standards.

[5] cdma2000 is a registered trademark of the Telecommunications Industry Association (TIA-USA).

1.1 Motivation

One of the challenges in a multiservice system with heterogeneous types of traffic such as 1xEV-DV system is that the limited bandwidth has to be efficiently provisioned among multiple traffic types. The 1xEV-DV is expected to support diverse real-time applications such as video-conferencing, voice over IP (VoIP), online gaming, and 3G-multimedia, demanding different QoS and bandwidth. Both 3GPP and 3GPP2 have proposed four different QoS classes: 1) *conversational*; 2) *streaming*; 3) *interactive*; and 4) *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 rate both in air and land. For example, quite big overhead in VoIP communications requires high bandwidth in landline. When we consider a scenario where a user is using 8 Kb/s codec and sending frames every 20 ms, this 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, 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. In landline side, however, the following still remains as an overhead: 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 of 1xEV-DV to be bottlenecked without deploying proper bandwidth provisioning method.

1.2 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 as long-time scale basis. Static provisioning is simple to implement but will result 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 of 3G networks have been overprovisioned as well. However, massive overprovisioning based on the 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's revenue.

It has been observed that *service degradation* approaches may increase bandwidth utilization and service provider's revenue [1], [2]. The common goal of the service degradation approaches is to increase bandwidth utilization by lowering the QoS levels of existing users. The work [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 the degradation in the quality of service on 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 re-negotiated before the service degradation happens.

Over the last few years, call admission control policies based on *movable-boundaries* [3], [4] have attracted widespread interest to improve total channel utilization. The work [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 work [7], dual threshold bandwidth reservation (DTBR) scheme for a voice and data integrated system was proposed. DTBR scheme enables the complete sharing (CS) of the overall bandwidth, thus leading to efficient usage of the 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 that they assume. Although these schemes were evaluated extensively, the complexities of the algorithms often make them impractical.

The contributions of this article are two-fold: one is to propose a practical bandwidth reprovisioning scheme without any assumption on the statistical properties of the traffic, and the other is to design and implement the proposed scheme as an SNMP MIB for long time scale bandwidth provisioning in 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 landline side of 1xEV-DV system, not to traffic over the air. Our approach has several benefits. First, this approach is simple. Second, the proposed scheme is represented by managed objects (MO's) and implemented as an SNMP-compliant MIB, and therefore, it can be integrated with Internet Engineering Task Force (IETF) network management framework. It enables service providers or network operators to monitor the link utilization and to reprovision the bandwidth. Third, our approach does not use a traffic model. Because traffic from different sources are multiplexed in PDSN node, 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.

2 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. They are the *assured* and *non-assured QoS*. Non-assured QoS is defined to determine a priority level of users that are competing for air interface resources. The 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 QoS block of bits (BLOB). For the detailed introduction to the QoS between mobile users and base stations (BS's), 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 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 rate 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 IntServ). For example, taking 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). The examples of QoS mappings can be found in works [6], [7]. An SLA should be established between the 1xEV-DV networks and the peer networks. The SLA will be enforced at the border router (BR).

3 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. The 1xEV-DV (especially, A8 and A10 interfaces) is the target environment that we conducted several experiments. The network comprises a number of base stations connected by a wireline backbone network through PDSN equipment. The proposed reprovisioning method is designed and implemented as an SNMP MIB and runs in PDSN with 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 class 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 bandwidth as much 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: 1) bandwidth estimation; and 2) bandwidth reprovisioning. Let us consider the 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 k^{th} 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.

3.1 Bandwidth Estimation

We monitor the utilization of the bandwidth provisioned for the background class at regular intervals T_S. Suppose that $X_{k_1}, X_{k_2}, \cdots, X_{k_n}$ 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 the interval estimate of μ. When n is sufficiently large, we can establish a confidence interval for μ by considering the sampling distribution of $\overline{X}_k$. According to the central limit theorem, we can expect that the sampling distribution of $\overline{X}_k$ to be approximately normally distributed with mean $\mu_{\overline{X}_k} = \mu$ and standard deviation $\sigma_{\overline{X}_k} = \sigma/\sqrt{n}$.

When σ is unknown, and $n \geq 30$, S_k can replace σ. Here, S_k is the sample standard deviation of the k^{th} block. A $(1-\alpha)100$ % confidence interval for μ is given by

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

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

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

3.2 Bandwidth Reprovisioning

The upper confidence limit $\theta_{k,k+1}$ is the bandwidth to guarantee $(1-\alpha)100$ % throughput of background class in the k^{th} and $k+1^{th}$ block. Since the initially provisioned bandwidth for the background class is B_L, from (1) we reprovision the bandwidth for the background class in the $k+1^{th}$ block ($B_{L_{k+1}}$) as:

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

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

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

Equation (3) is the consequence of the fact that we give the unutilized bandwidth of the background class to the conversational class. Note that when $B_L \leq \theta_{k,k+1}$, the conversational class bandwidth for the $(k+1)^{th}$ block is the same as the previous (k^{th}) block. However, when $B_L > \theta_{k,k+1}$, we can take advantage of the 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 $(1-\alpha)100$ % 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.

4 Elastic Provisioning MIB

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

4.1 Structure of the EP MIB

The EP MIB is an SNMP-compliant MIB for elastic bandwidth reprovisioning as well as utilization monitoring. Fig. 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, a bandwidth of a conversational class can be reprovisioned to 15 Mb/s by `snmpset(..., epProvServiceClass = 1, epProvBandwidth = 15000000, ...)`.

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 reprovisioning algorithm by creating a new row of `epCtrlTable`. If the operator creates a row with `epCtrlSampleInterval` = 0, then statistics is recorded to stat table without reprovisioning.

The `epStatTable` records the results each time reprovisioning happens. 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 of 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 the stat table. The number of rows in the stat table can be expressed as $\sum_{i=0}^{M} \texttt{epCtrlMaxRows}(i)$, 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 circular buffer. As each new row is added to the set, the oldest row associated with this control is deleted.

4.2 Implementation

The network management system is composed of a variety of modules running on UNIX system (Solaris 2.8). The modules include *EP MIB Viewer* and *Provisioning Manager*. Unfortunately, since our current PDSN equipment does not support 3GPP2 IOS 4.3 A10/A11 interface specification for 1xEV-DV, we replaced PDSN with an emulation device named 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 at 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 voice over IP (VoIP) calls that PDSN serves on a per class basis. It runs on vxWorks 5.4 operating system with 128 MB of main memory. It arranges raw data to a regular format and writes it into a binary file and reports the collected information to 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 the 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 *Traffic Analyzing Process* using TCP/IP socket connection. The experiment results as well as the measured data are constructed as EP MIB.

5 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 1xEV-DV system that is connected to 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 that were used in the experiments are summarized in Table 1.

5.1 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 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.

5.2 Experiments

We performed a reprovisioning experiment with Auckland-IV data set [11], which is a continuous 6 1/2 week 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 KBytes/s 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 experiment 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 minutes guaranteeing packet drop rate of the background class below 5% or 1% using previous 300 measured samples. The results are recorded to stat table every 10 min. We obtained 138 ($10min \times 6 \times 23h$) rows of `epStatTable` as results.

Bandwidth Estimation Fig. 5-(a) shows that the actual traffic load and the estimated bandwidths that guarantee the packet drop rate of the background class below a given threshold (1 or 5%). As shown in Fig. 5-(a), when $0:00 \leq t \leq 8:30$, $8:50 \leq t \leq 9:00$, $t = 9:20$, $t = 11:00$, $t = 14:50$, $15:20 \leq t \leq 16:00$, $t = 16:30$, $17:20 \leq t \leq 19:20$, and $t \geq 19:40$, the upper confidence limits of $z_{0.005}$ are smaller than the initially provisioned bandwidth 200 KBytes/s (i.e., $\theta_{k,k+1} < B_L$), and therefore, bandwidth reprovisioning happens. During these periods, from (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%. When the reprovisioning algorithm runs with the parameter of `epCtrlGuaranteedDropRatio` $= 5$, reprovisioning does not happen at $t = 8:30$, $t = 10:10$, $10:30 \leq t \leq 10:50$, 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 service provider's revenue. Fig. 5-(b) 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 $11:10 \leq t \leq 14:40$, 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 $16:40 \leq t \leq 20:00$, 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 $18:40 \leq t \leq 18:50$, the number of accepted users increased from 309 without reprovisioning to 330 with reprovisioning. Call blocking probability decreased from 9% without reprovisioning to 3% with reprovisioning.

Packet Drops Fig. 5-(c) 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% packet drops were monitored between $t = 11:10$ and $t = 14:40$. When reprovisioning happens, the background class traffic experiences 0–3.931% 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% for without reprovisioning, reprovisioning with guaranteeing the packet drop probability of the background class below 1 and 5%, respectively. In most cases, the packet drop probabilities that we monitored in this experiment were bounded below a given threshold.

6 Conclusions

The 1xEV-DV is expected to support diverse real-time applications such as video-conferencing, voice over IP, and 3G-multimedia, demanding different QoS and bandwidth. Bandwidths for 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's 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 equation (2) and (3) are reprovisioned for the new bandwidths of the next block. Thus, unutilized bandwidth of background class is reprovisioned for the conversational class. The experiments performed on 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%.

The proposed scheme has two special features. First, the reprovisioning scheme is represented as a set of managed objects and implemented as EP MIB. It conforms to the IETF network and system management framework, and therefore, 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 where the control algorithm runs on - 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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Table 1. Summary of experiment conditions

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% ($z_{0.005} = 2.58$).
`epCtrlSampleInterval`	2	2 second sampling interval
`epCtrlWindowSize`	300	300 samples will be used for calculating upper confidence limit ($\theta_{k,k+1}$).
`epCtrlActionInterval`	600	10 minutes (600 seconds) for reprovisioning interval.

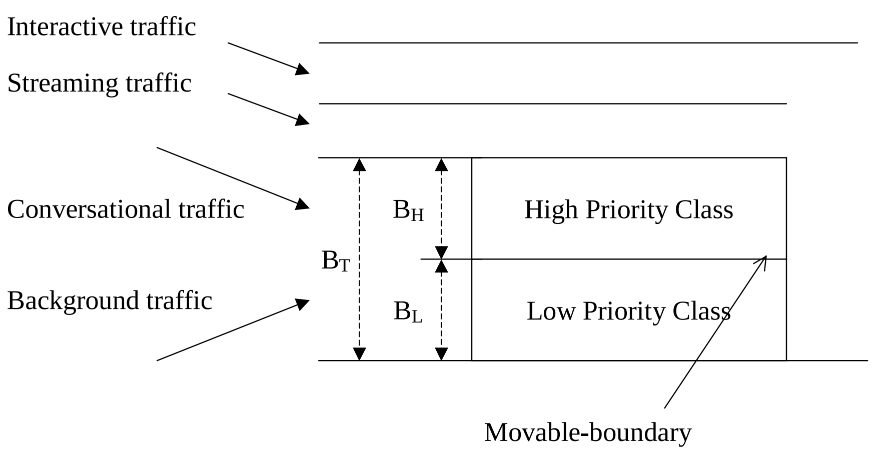


Fig. 1. Logical structure of the provisioned bandwidth.

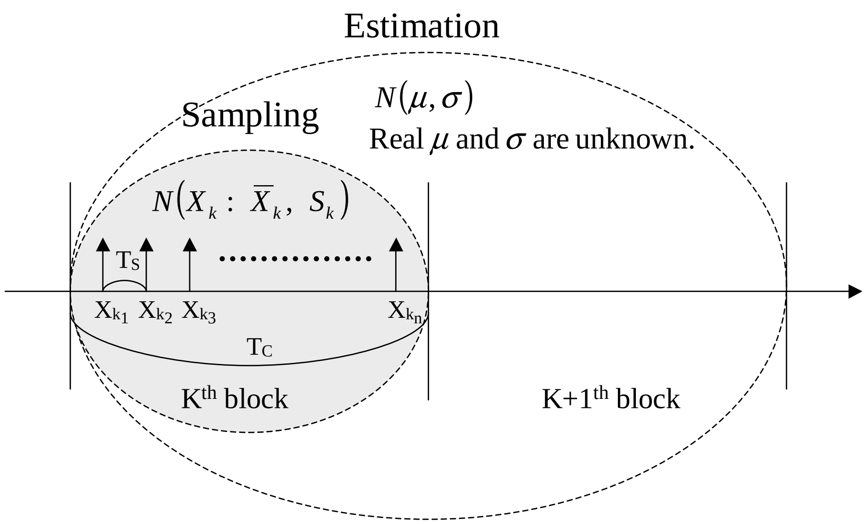


Fig. 2. Bandwidth reprovisioning process

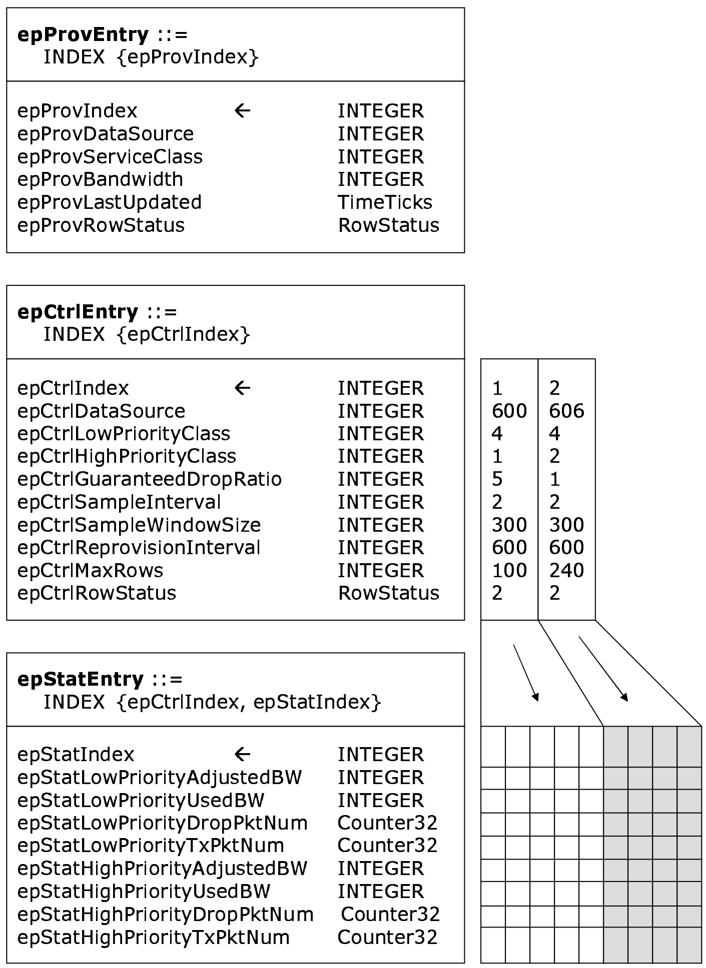


Fig. 3. Logical structure of Elastic Provisioning MIB.

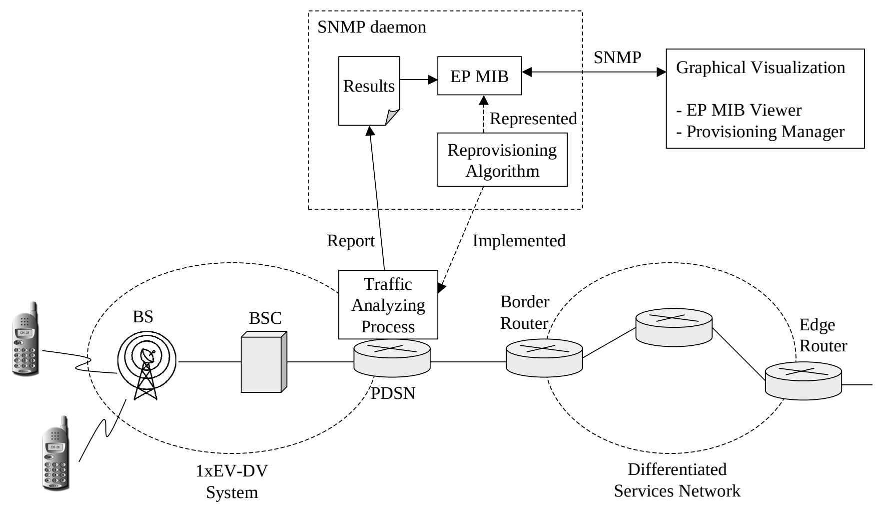
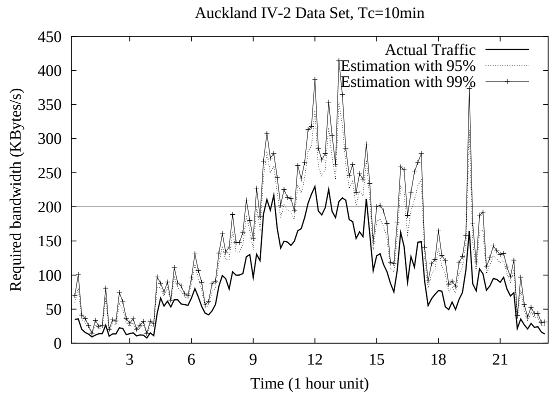
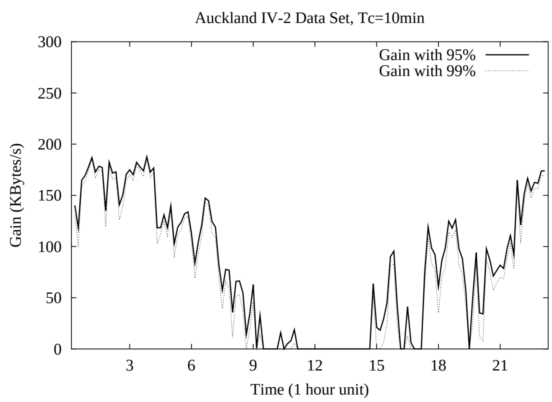


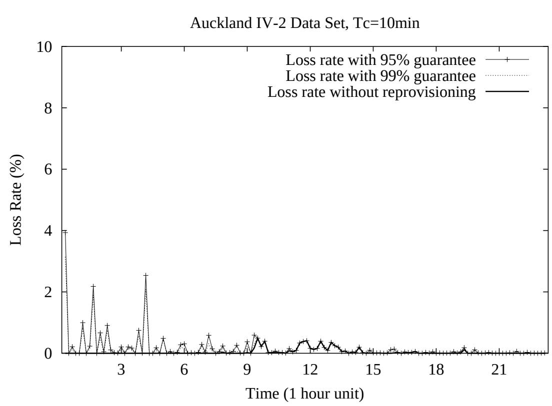
Fig. 4. MIB implementation environment.



(a) The estimated bandwidth, $T_C = 10$ min



(b) The bandwidth gain, $T_C = 10$ min



(c) The packet loss probability, $T_C = 10$ min

Fig. 5. The Auckland-IV 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%); (b) bandwidth gains; (c) packet loss probability.