

# Framework for Performance Metrics and Service Class for Providing End-to-End Services across Multiple Provider Domains

Chin-Chol Kim<sup>1</sup>, Jaesung Park<sup>2,\*</sup>, and Yujin Lim<sup>3</sup>

<sup>1</sup> Digital Infrastructure Division, National Information Society Agency,  
77 Mugyo-Dong, Jung-Gu, Seoul, 100-775, Korea  
[cckim@nia.or.kr](mailto:cckim@nia.or.kr)

<sup>2</sup> Department of Internet Information Engineering, University of Suwon,  
2-2 San, Wau-ri, Bongdam-eup, Hwaseong-si, Gyeonggi-do, 445-743, Korea

<sup>3</sup> Department of Information Media, University of Suwon,  
2-2 San, Wau-ri, Bongdam-eup, Hwaseong-si, Gyeonggi-do, 445-743, Korea  
[{jaesungpark,yujin}@suwon.ac.kr](mailto:{jaesungpark,yujin}@suwon.ac.kr)

**Abstract.** Developing a unified solution that enables the end-to-end delivery of services over multiple provider domains at a guaranteed quality level is challenging. International standard organizations offer their own definition of performance metrics and service classes. In this paper, we define the unified performance metrics and service classes for interworking of various types of networks.

**Keywords:** Quality-of-service, service-level agreement, performance metric, service class.

## 1 Introduction

The Internet is moving from being a simple monolithic data service network to a ubiquitous multi-service network in which different stakeholders including content providers, service providers, and network providers. They require to co-operate for offering value-added services and applications to content consumers [1]. The problem of how to extend QoS (Quality-of-Service) capabilities across multiple provider domains for providing end-to-end services, has not been solved satisfactorily to-date. Furthermore, developing a unified solution that enables the end-to-end delivery of services over various types of networks at a guaranteed quality level is more challenging. The current practice in service offering is using of Service Level Agreements (SLAs).

This paper presents the solution for end-to-end QoS-enabled service delivery over heterogeneous networks. To do this, we compare the performance metrics defined by international standard organizations and re-define the unified metrics. Then we analyze

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\* Corresponding author.

the service classes presented by the organizations and re-define the unified service class. Besides, we consider how to allocate performance to multiple provider domains.

This paper is organized as follows. Section 2 highlights related works and our proposed performance metrics. Section 3 describes how to achieve the interworking of heterogeneous networks. Finally, conclusions are presented in Section 4.

## 2 Performance Metrics

### 2.1 Related Works

Generally, performance metrics are defined to evaluate the end-to-end QoS provisioning. International standard organizations lay out several criteria for the metrics. We introduce performance metrics defined by three organizations; IETF, ITU-T, and GSMA. First, IETF develops and promotes Internet standards and it proposes 5 metrics to maximize the service quality and reliability of end-to-end path, as below [2-7].

- ◆ Packet Delay

For a real number  $dT$ , “the delay from  $Src$  (source) to  $Dst$  (destination) at  $T$  is  $dT$ ” means that  $Src$  sent the first bit of a packet to  $Dst$  at time  $T$  and that  $Dst$  received the last bit of that packet at time  $T+dT$ .

- ◆ Packet Delay Variation

The variation in packet delay is sometimes called “jitter”. The packet delay variation is defined for two packets from  $Src$  to  $Dst$  as the difference between the value of the delay from  $Src$  to  $Dst$  at  $T_2$  and the value of the delay from  $Src$  to  $Dst$  at  $T_1$ .  $T_1$  is the time at which  $Src$  sent the first bit of the first packet, and  $T_2$  is the time at which  $Src$  sent the first bit of the second packet.

- ◆ Packet Loss

“The loss from  $Src$  to  $Dst$  at  $T$  is 0” means that  $Src$  sent the first bit of a packet to  $Dst$  at time  $T$  and that  $Dst$  received that packet.

“The loss from  $Src$  to  $Dst$  at  $T$  is 1” means that  $Src$  sent the first bit of a packet to  $Dst$  at time  $T$  and that  $Dst$  did not receive that packet.

- ◆ Packet Reordering

If a packet  $s$  is found to be reordered by comparison with the  $NextExp$  value, its “Packet-Reordered” = True; otherwise, “Packet-Reordered” = False

if  $s \geq NextExp$  then /\*  $s$  is in-order \*/

$NextExp = s + 1;$

    Packet-Reordered = False;

else /\* when  $s < NextExp$  \*/

    Packet-Reordered = True;

- ◆ Packet Duplication

The packet duplication is a positive integer number indicating the number of (uncorrupted and identical) copies received by  $Dst$  in the interval  $[T, T+T_0]$  for a packet sent by  $Src$  at time  $T$ .

Second, ITU-T coordinates standards for telecommunications. It provides 8 metrics for QoS monitoring across heterogeneous provider domains to provide end-to-end services as below [8].

- ◆ IP Packet Transfer Delay (IPTD)

IPTD is the time,  $(t_2 - t_1)$  between the occurrence of two corresponding IP packet reference events, ingress event at time  $t_1$  and egress event at time  $t_2$ , where  $(t_2 > t_1)$  and  $(t_2 - t_1) \leq T_{max}$ .

- ◆ IP Packet Error Rate (IPER)

IPER is the ratio of total errored IP packet outcomes to the total of successful IP packet transfer outcomes plus errored IP packet outcomes.

- ◆ IP Packet Loss Rate (IPLR)

IPLR is the ratio of total lost IP packet outcomes to total transmitted IP packets.

- ◆ Spurious IP Packet Rate

Spurious IP Packet Rate is the total number of spurious IP packets observed at that egress point during a specified time interval.

- ◆ IP Packet Reordered Ratio (IPRR)

IPRR is the ratio of the total reordered packet outcomes to the total of successful IP packet transfer outcomes.

- ◆ IP Packet Severe Loss Block Ratio (IPSLBR)

IPSLBR is the ratio of the IP packet severe loss block outcomes to total blocks.

- ◆ IP Packet Duplicate Ratio (IPDR)

IPDR is the ratio of total duplicate IP packet outcomes to the total of successful IP packet transfer outcomes minus the duplicate IP packet outcomes.

- ◆ Replicated IP Packet Ratio (RIPR)

RIPR is the ratio of total replicated IP packet outcomes to the total of successful IP packet transfer outcomes minus the duplicate IP packet outcomes.

Third, GSMA is an association of mobile operators and related companies devoted to supporting the standardizing, deployment and promotion of the GSM mobile telephone system. GSMA defines the metrics requested to IPX providers by wireless service providers as below [9].

- ◆ Max Delay

Max Delay is the maximum value of the one-way transit delays across an IP transport network.

- ◆ Max Jitter

Max Jitter is the maximum value of delay variations across an IP transport network.

◆ **Packet Loss**

Packet Loss is the ratio of total lost packets to total transmitted packets via an IP transport network.

◆ **SDU Error Ratio**

SDU Error Ratio is the ratio of total errored packets to total transmitted packets via an IP transport network.

◆ **Service Availability**

Service availability is a proportion of the time that the service is considered available to service providers on a monthly average basis.

ITU-T			IETF
<ul style="list-style-type: none"> <li>• Packet Error</li> <li>• Spurious Packet Rate</li> <li>• Packet Reordered Ratio</li> <li>• Replicated Packet Ratio</li> <li>• IPSLBR (IP Packet Severe Loss Block Ratio)</li> </ul>	Delay	The one-way packet transit delays from source to destination	<ul style="list-style-type: none"> <li>• Packet Reordering</li> <li>• Packet Duplication</li> </ul>
	Jitter	The one-way packet delay variation from source to destination	
	Packet Loss	The one-way packet loss from source to destination	
<ul style="list-style-type: none"> <li>• Service Availability</li> <li>• Packet Error</li> </ul>			GSMA

**Fig. 1.** The comparison of performance metrics

## 2.2 Definition of Performance Metrics

We summarize performance metrics mentioned in the previous subsection, as shown in Fig 1. We select the common metrics among the metrics defined by the standard organizations, such as delay, jitter, and packet loss. The delay and jitter seriously affect the quality of real-time streaming multimedia applications such as voice over IP, online games and IPTV. In some cases, excessive delay can render the application unusable. Some network transport protocols such as TCP provide for reliable delivery of packets. In the event of packet loss, the receiver asks for retransmission or the sender automatically resends any segments. Although TCP can recover from packet loss, retransmitting missing packets causes the throughput of the connection to decrease. In addition, the retransmission possibly causes severe delay in the overall transmission. Thus, when the end-to-end QoS is evaluated across multiple providers, at least three metrics should be considered. However, the definitions of the metrics are different among the organizations. We re-define the metrics consistently as below.

◆ **Delay**

The arithmetic mean of one-way packet transit delays between source and destination.

- ◆ Jitter  
The arithmetic mean of differences between successive packet delays.
- ◆ Packet Loss  
The ratio of total lost packets to total transmitted packets.

### 3 Interworking of Multiple Providers

#### 3.1 Definition of Service Class

Service providers offer their own service class and the quality of these services are different. When the traffic enters into the provider domain and the traffic is mapped into the higher level of service class than the requested level, the network resources are wasted. Whereas, when the traffic is mapped into the lower level of service class than the requested level, the service quality is not guaranteed. Thus we define the unified service classes to solve the mapping problem between service classes of different providers.

GSMA	IETF	ITU-T
Conversational	Telephony	Class 0 / 1
	Real-Time Interactive	
Interactive	Multimedia Conferencing	
	Signaling	Class 2
	Low-Latency Data	Class 3
	High-Throughput Data	
	Broadcast Video	Class 4
	Multimedia Streaming	
Background	Low-Priority Data	Class 5
	Standard	

**Fig. 2.** The comparison of service classes

Class	Service	Characteristics	
		Applications	Priority for resource allocation
CT3	Emergency	Emergent communication	High
CT2	Real-time/guaranteed media traffic	VoIP, Video conference, IPTV	High
CT1	Premium data traffic	Premium data	Medium
CT0	Best-effort data traffic	Best-effort data	Low

**Fig. 3.** Definition of service classes

Fig. 2 summarizes the service classes proposed by IETF, ITU-T, and GSMA [10-12]. IETF divides the service classes based on the services offered. ITU-T divides the classes based on the service characteristics and GSMA focuses on the services used. We re-define the service class for interworking of heterogeneous networks, as shown in Fig. 3.

### 3.2 Allocation of Performance

Even though the performance metrics and service classes are re-defined consistently, another challenge to achieve the end-to-end QoS is presented: how can QoS classes, e.g., network performance, be assured for users? There are two basic approaches to solve the problem [13]. One involves allocating performance to a limited number of network segments, which allows operators to contribute known levels of impairments per segment, but restricts the number of operators that can participate in the path. The other approach is impairment accumulation, which allows any number of operators to participate in a path. On the surface, this may appear too relaxed, but assuming operators in a competitive environment will actively manage and improve performance.

The static allocation approach divides the end-to-end path into a fixed number of segments and budgets the impairments such that the total objective is met in principle. It requires that individual segments have knowledge of the distance and traffic characteristics between the edges of their domains, as these properties of the segment affect the resulting allocations. For example, the delay budget allocated to a network segment depends on whether it is access or transit, and whether the transit distance is metro or regional. Similarly, packet loss and delay variation will have to be allocated according to whether the segment is access or transit, as the traffic aspects can differ significantly. An important aspect of the static allocation is its dependence on the number of providers, as the allocation has to be done accordingly. This can result in undershooting or overshooting the objective because paths can have a different number of network segments than designed for.

Accumulation approach is defined as those that include requests of what performance level each provider can offer, followed by decisions based on the calculated estimate of end-to-end performance. The requester may be the customer-facing provider only or include all the providers along a path. The responder may be a provider or their proxy. However, there are several weaknesses of the approach. First, users' segment impairments are not taken into account. Second, if the initial process fails, multiple passes of request/estimation cycle may be required. Third, it requires customer or customer proxy involvement. Finally, commitments for each network segment must be pre-calculated taking distance into account.

## 4 Conclusions

In this paper, we focus on how to extend QoS capabilities across multiple provider domains for providing end-to-end services. To solve the problem, we define the unified performance metrics and the service classes for interworking of multiple provider domains. For the future work, we need to solve another problem of how to allocate performance to the domains, by using IETF PDB (Per-Domain Behavior) concept.

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