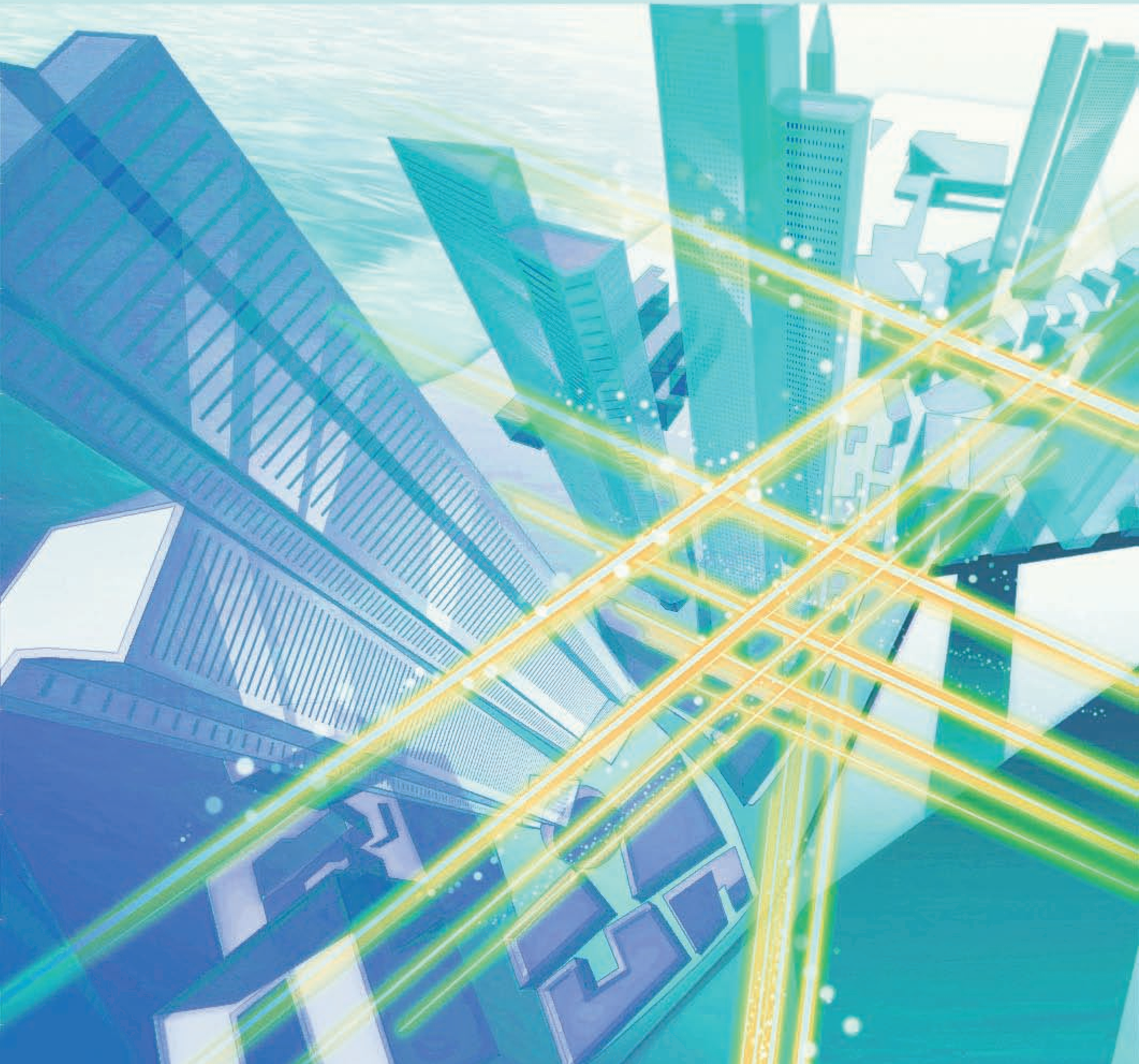
NTT Technical Review

5
2013



May 2013 Vol. 11 No. 5

NTT Technical Review

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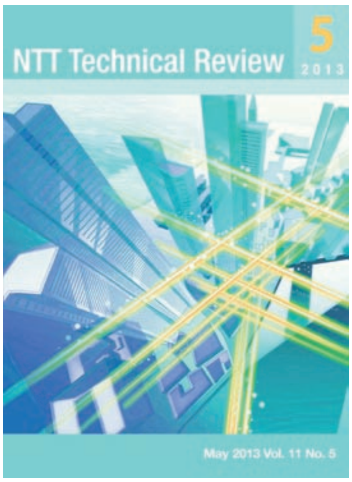


Feature Articles: QoE Estimation Technologies

Regular Articles

Global Standardization Activities

Information

Toward QoE-centric Operation of Telecommunication Services

Akira Takahashi and Takanori Hayashi

Abstract

The rapid spread in the use of smartphones and cloud-based services has accelerated the speed of evolution of telecommunications application software, resulting in a great diversity of such software. While the quality of experience (QoE) of telecommunications services has been directly related to network performance in the past, the current situation is not that simple. This is because the QoE of telecommunications applications is affected not only by network performance, but also by server and terminal performance to a great extent. In this article we propose a framework for improving user QoE, and we introduce recent research and development activities underway at NTT intended to achieve this goal.

1. Quality of experience (QoE) and network performance

All of the factors affecting the end-to-end quality of conventional public switched telephone network (PSTN) services are under the control of telecommunications carriers except for telephone terminals. The characteristics of terminals are clearly defined in standards and in domestic laws, for example, the Regulations for Telecommunications Facilities for Telecommunications Business. Therefore, if the PSTN network is confirmed to be operating normally, it is very rare for an end user to encounter a serious problem. This implies that *QoE* ≈ (is approximately equal to) *network performance* holds in the PSTN world.

When the Internet era arrived, the variation in terminals drastically increased. At that time, terminals consisted mainly of personal computers (PCs). However, in the 1990s, the primary factor affecting the QoE of web-browsing was network speed, referred to as *throughput*, since network capacity was limited. The relationship *QoE* ≈ *network performance* was still maintained.

With the arrival of the broadband era in the late 1990s, the behavior of telecommunications application software had become increasingly complex. This resulted in a change in the dominant factors of QoE; that is, the server and terminal performance seriously affects QoE. For example, the video quality of Internet protocol television (IPTV) is dependent on terminal technologies such as forward error correction. In cloud-based applications, the server mainly transmits executable programs, and the terminal executes them in order to create and render the content. This means that PC performance is more important than network speed. Moreover, the use of wireless local area networks and power-line communication in home networks has brought new quality degradation factors that cannot be monitored or controlled by network carriers.

The above-mentioned examples reveal that the relationship *QoE* ≈ *network performance* no longer holds. What makes the situation more complicated is that even network operation has become difficult in an end-to-end sense due to the multi-provider environment (**Fig. 1**).

If we take all of these emerging factors into account, we can conclude that the first step towards maximizing user QoE is to quantify and visualize it.

2. QoE-centric operation

As a telecommunications carrier, NTT has made every effort to maintain and improve network performance through daily network operations. However, due to changes in the balance of quality factors, there are quite a few problems that cannot be found and

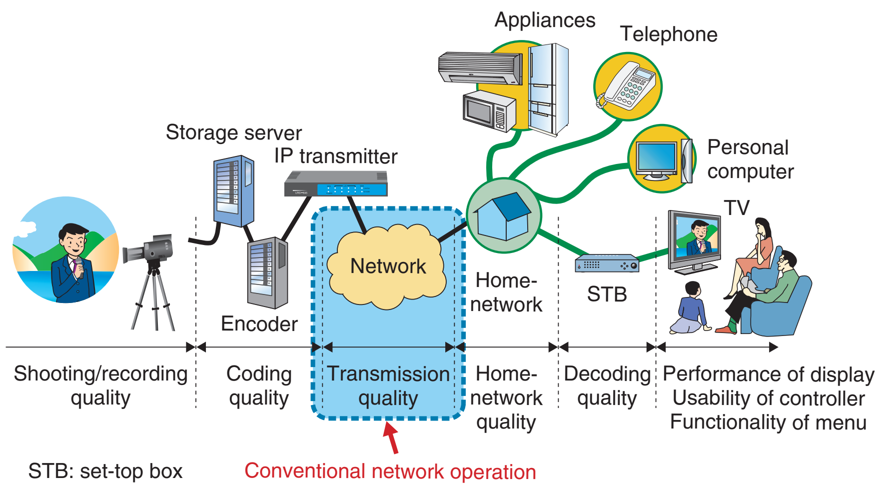


Fig. 1. Factors affecting QoE of IPTV.

solved through network operation only. How can a network carrier solve the problems in the following examples?

2.1 Example 1

Bob likes watching movies. He was interested in the video on demand menu of an IPTV service. He decided to subscribe to the IPTV service and to a Next Generation Network (NGN) service as well. He placed the set-top box (STB) in his living room, although the home gateway was placed in the pantry, where the former PSTN line was installed. It was difficult to lay Ethernet cable from there to the living room, so he bought a Wi-Fi router, on whose box was the claim "300 Mbit/s Wi-Fi! No degradation even for HD (high definition) videos!"

In the beginning, he enjoyed watching movies via the NGN. However, he began experiencing serious video degradation that continued to increase, and the video sometimes froze. He did not notice—which was not surprising—that the Wi-Fi router and the digital cordless telephone that he had recently bought used the same frequency range and were interfering with each other.

He was too angry about the quality to think of calling the customer support center. Rather, he remembered the flyer about the cable TV that the neighborhood self-governing body had recommended, and he immediately called the cable company. Then, he cancelled the IPTV service and at the same time cancelled the NGN service because the cable company salesman had suggested that it would be much cheaper to use the cable network for his telephone and Internet service.

The cable company laid the cable directly to the STB in the living room. Thus, Bob did not use the Wi-Fi router. Naturally, he experienced no degradation and therefore believed that cable TV was much better than IPTV. Since then, Bob talks about his experience to his friends whenever he has a chance and stresses the superiority of cable TV over IPTV.

2.2 Example 2

A corporate customer (Company A), was planning to replace the business applications they were using. At the same time, they were considering the possibility of using the cloud environment provided by a telecom operator (Operator B) in order to reduce costs. However, they were worried about quality degradation such as response latency in using the application software. Therefore, they asked a sales engineer of Operator B about this and were told that Operator B had carried out a complete performance evaluation and found no problems.

After a one-month trial by the telecommunications service division of Company A, they decided to adopt the proposal by Operator B since they had found no performance problems. They then extended the trial to all their divisions.

A couple of weeks later, the telecommunications service division started receiving many complaints from several locations about the very slow response of the application software. The person in charge of the telecommunications service division in Company

A did not experience any degradation, so he asked Operator B to check the transmission performance of the virtual private network (VPN) provided by Operator B. Operator B enhanced the bandwidth of the VPN, and the performance improved in some locations. However, most users still experienced serious delays in using the application software.

Finally, because of this problem, Company A decided not to accept Operator B's proposal when the trial was finished. Nobody knew that many users in Company A had installed a certain kind of software in the background, and this software wasted CPU (central processing unit) power, resulting in poor rendering performance for browser-based applications.

Although the above examples are fictitious, they could actually occur. The issue here is that it is becoming increasingly difficult to understand user dissatisfaction based only on conventional network management. We often take a questionnaire approach to determine QoE problems. However, this takes a lot of time to solve the problems. Thus, we have been studying methodologies for estimating user QoE based on parameters that can be observed by service/network providers. These technologies will contribute to the operation of telecommunications services based on user QoE, which we call *QoE-centric operation.*

3. QoE estimation technologies

These Feature Articles introduce QoE estimation technologies with which we can estimate user QoE based only on observable parameters such as packet-header information. Conventionally, such technologies were assumed to be applied to off-line quality measurement, and therefore, the media-signal-based approach was taken to analyze speech waveforms or pixel data of videos. This approach requires an algorithm capable of handling a very heavy and complex computational load. However, to enable online real-time monitoring of QoE, passive measurement and a light-weight computational load are necessary for estimating QoE.

The first article in this issue, which is entitled "Monitoring the Quality of IPTV Services" [1], introduces a QoE estimation methodology for UDP (user datagram protocol)-based IPTV. The validity of this technology was thoroughly evaluated by ITU-T (International Telecommunication Union, Telecommunication Standardization Sector) and was standardized as ITU-T Recommendation P.1201.1 in 2012.

The second article, entitled "Playback State Estimation of Progressive Download-based Video Services" [2], introduces a similar technology for TCP (transmission control protocol)-based progressive-download applications such as BeeTV, which is provided by NTT DOCOMO, and YouTube. The method developed at NTT intelligently exploits the mechanism of TCP and achieves QoE monitoring at any point between a server and terminal.

The third article, entitled "Performance Estimation Techniques for Browser-based Applications" [3], describes techniques that enable QoE monitoring of browser-based applications, which are widely used in cloud-based environments. These techniques make it possible to estimate the waiting time for complex content that uses recent web technologies such as Ajax and Flash.

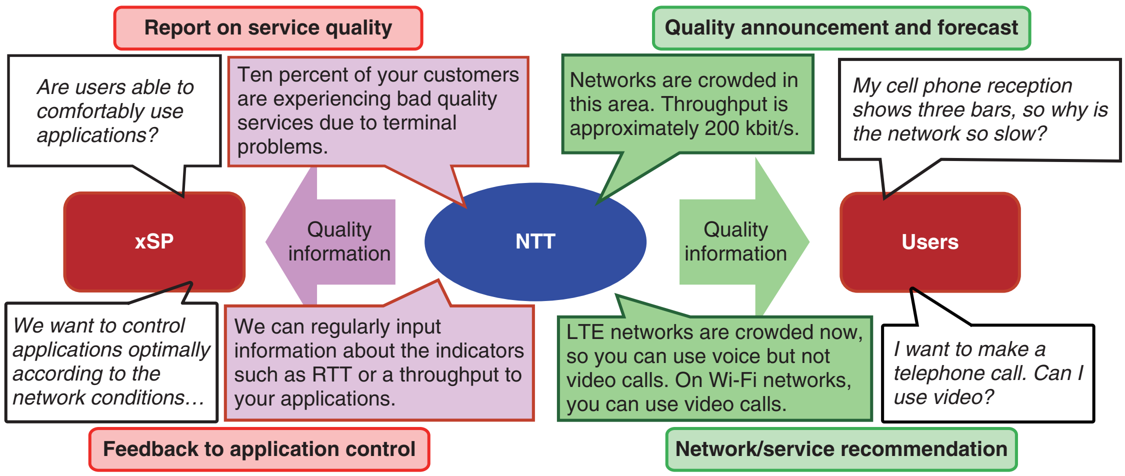
Finally, in the fourth article, "QoE Assessment Methodologies for 3D Video Services" [4], we introduce techniques to evaluate the quality of 3D video and also describe their objective estimation methodology. In evaluating 3D video, it is important to quantify the degree of fatigue, for example, in addition to the conventional picture clarity.

4. Expanding horizons of service-quality research

Up to this point, we have emphasized the importance of evaluating/estimating QoE. It is clear that evaluation/estimation is the first step toward realizing QoE-centric operation, for which we also need to develop methodologies for mapping network, server, and terminal performance onto QoE. In addition, it is important to develop appropriate customer relations management policies.

The concept of QoE-centric operation is a framework for providing and maintaining good telecommunications quality. We also propose a framework in which network providers improve user QoE in collaboration with service providers and end users. This is not the conventional one-way quality given by network providers, but a mutual effort with service providers and end users (**Fig. 2**). We call this *Co-creation Quality.*

Obtaining information on the current quality and possible alternatives is essential in order to encourage users to take action to improve their QoE. For example, users of public transportation such as trains expect the trains to run exactly on schedule. By contrast, people who drive their own cars need information on traffic jams and weather forecasts to help



LTE: Long Term Evolution
RTT: round trip time
xSP: any type of service provider

Fig. 2. Concept of Co-creation Quality.

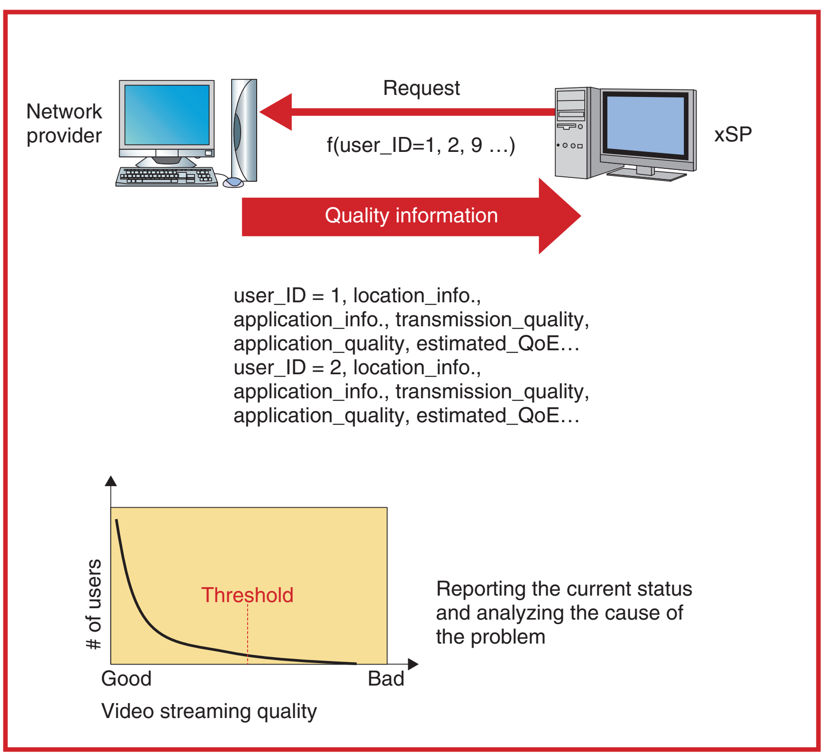


Fig. 3. Quality API.

them decide their route and schedule. Similarly, we believe network providers should provide more information on quality to users and/or service providers. To do this, technologies for monitoring, predicting, and visualizing information on quality are necessary.

Such information could be used to improve customer support by service providers. For example, if a service provider can obtain the current QoE information of their end users from a network provider simply by requesting it with a user ID, they can recommend an alternative to the end user when quality was low before receiving a complaint from the user (**Fig. 3**). It may also be possible to optimize the behavior of application software so that it adjusts to current and future quality conditions of networks. We call this kind of network function a Quality API (application programming interface) and consider it part of the network API.

The technologies introduced in this article are fundamental to achieving QoE-centric operation, Co-creation Quality, and Quality APIs in the future. NTT R&D (research and development) will continue such studies in the future.

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Monitoring the Quality of IPTV Services

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Abstract

This article introduces a quality-monitoring model that can be used to estimate end-users' quality of experience of Internet protocol television services using packet header information. This model can be used for in-service quality monitoring.

1. Introduction

Internet protocol television (IPTV) services are now widely provided. The quality of experience (QoE) of IPTV services is affected by the audiovisual content, encoding and decoding techniques, network performance, and display technology. It is therefore important to ensure that end-users receive high-quality IPTV content. The ideal way to do that would be for the service provider to identify the quality degradation factors, monitor QoE by taking the main quality degradation factors into account, and finally, to conduct a thorough investigation to determine the cause of quality degradation and then address the problems.

2. Activities of ITU-T SG12 in quality monitoring

An IPTV processing chain is shown in **Fig. 1**. This processing chain consists of video acquisition and editing, encoding, network transmission, decoding, and display, as mentioned in section 1. All of these elements can affect end-users' QoE. ITU-T (International Telecommunication Union, Telecommunication Standardization Sector) Recommendation G.1081 [1] defines performance monitoring points for IPTV services that will enable the service provider and/or network operator to monitor the performance of the entire IPTV service delivery process. There are five points in the processing chain where performance quality is monitored: the source media and metadata are monitored at point 1; encoded and packetized source media are monitored at point 2; packet transmission characteristics are monitored at point 3; received packets are monitored at point 4 to determine whether they can adequately provide the required QoE at the client terminal, and the displayed media are monitored at the client terminal at point 5. If all the information obtained at these monitoring

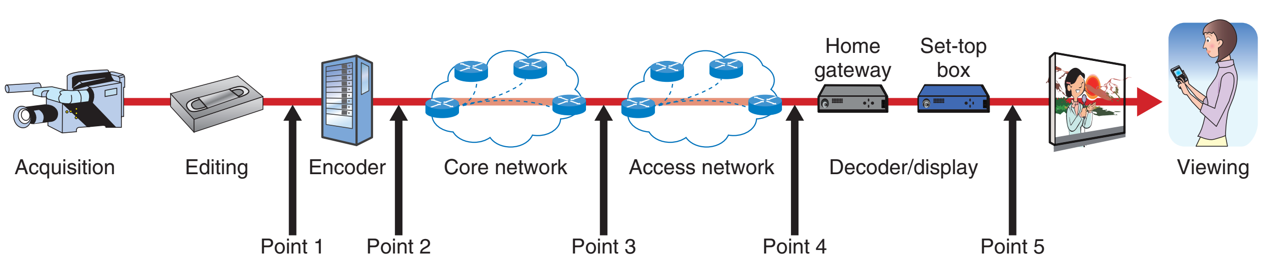

Fig. 1. IPTV processing chain.

points is integrated, the locations where quality has degraded can be determined.

ITU-T Recommendation G.1081 does not define how objective quality assessment models should be applied at each performance monitoring point. Monitoring the QoE at the head end is important because quality degradation at points 1 and 2 influences the QoE of all users. In these cases, full reference (FR) media-layer models are suitable for monitoring quality. At points 3 and 4, it is preferable to analyze packets using a method with low computational load because the analysis needs to be implemented in terminals such as mobile terminals, home gateways, and set-top boxes (STBs). In these cases, packet-layer models are suitable for estimating QoE from IP packet header information. At point 5, it is essential to monitor the QoE by using the no-reference (NR) media-layer model, which estimates QoE from media signals received at the client terminal. Because FR media-layer models [2]–[4] that can be applied to head-end QoE monitoring have already been standardized, NTT has focused on developing packet-layer models that can be used for end-user QoE monitoring using packet header information [5]. Application areas can be categorized into lower resolution (LR: quarter common intermediate format (QCIF), quarter video graphics array (QVGA), and half VGA (HVGA)) and higher resolution (HR: standard definition (SD) and high definition (HD)) areas. The LR application area can be used for mobile IPTV services, and the HR application area can be used for STB based IPTV services. In 2006, ITU-T study group 12 (SG12) also launched a project that aims to develop quality monitoring models for LR and HR application areas. In October 2012, ITU-T standardized two Recommendations; ITU-T Recommendation P.1201.1 [6] can be used for the LR application area, and P.1201.2 [7] can be used for the HR application area. This article introduces ITU-T Recommendation P.1201.1 as it is most relevant because of the recent rapid growth of Mobile IPTV services. The P.1201.1 model was developed by integrating the NTT and Huawei models and was verified in an ITU-T performance evaluation contest.

3. P.1201.1 model

A block diagram of the P.1201.1 model is shown in **Fig. 2**. This model consists of three modules: parameter-extraction, parameter-calculation, and quality-estimation modules as follows.

1) Parameter-extraction module

This module extracts a Real-time Transport Protocol (RTP) timestamp, sequence number, and marker

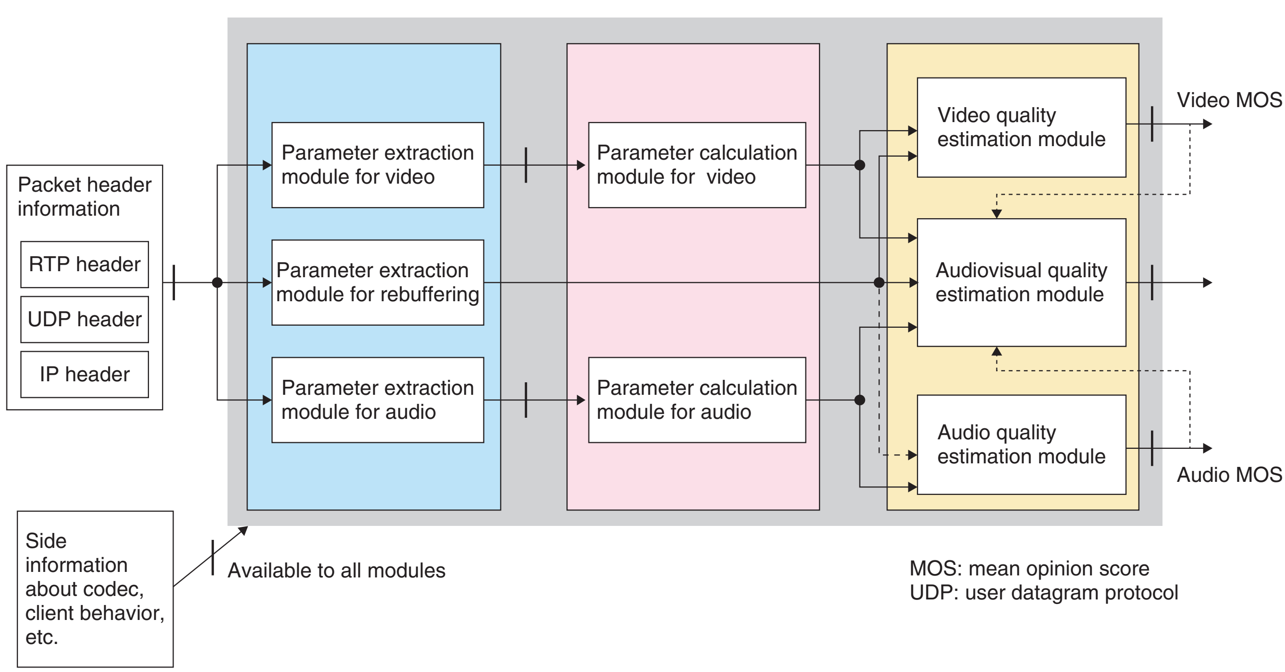

Fig. 2. Block diagram of P.1201.1 model [5].

bit from RTP headers and the rebuffering starting time and length from the client terminal.

2) Parameter-calculation module

This module calculates audio- and video-related quality parameters (e.g., coding bit rate and packet loss) using parameters extracted by the parameter-extraction module.

3) Quality-estimation module

This module estimates audio, video, and audiovisual quality using quality parameters calculated by the parameter-calculation module. Additionally, each module has a sub-module, as shown in Fig. 2.

This article summarizes the quality degradation factors processed by the quality-estimation modules. The details of the parameter extraction and calculation modules are not explained here, as they are beyond the scope of this article.

3.1 Audio quality estimation module

Audio quality is affected by the codec (coder-decoder) type, coding bit rate, packet loss, and rebuffering, so it is necessary to model the relationship between the following quality factors and the subjective audio quality.

- Effect of audio codec (i.e., AMR-NB (adaptive multi-rate narrowband), AMR-WB+ (extended adaptive multi-rate wideband), AAC-LC (advanced audio coding low complexity), HE-AACv1 (high-efficiency AAC, version 1), and HE-AACv2) on audio quality
- Effect of coding bit rate on audio quality
- Effect of lost audio frame length due to packet loss on audio quality
- Effect of rebuffering on audio quality (under study)

3.2 Video quality estimation module

As with audio quality, video quality is affected by the codec type, coding bit rate, packet loss, and rebuffering. Video quality is also affected by the number of bits per video frame type because it varies depending on the spatio-temporal information of the video content. Therefore, it is necessary to model the relationship between the following quality factors and the subjective video quality.

- Effect of video codec (i.e., MPEG-4 (Motion Picture Experts Group, version 4) and H.264) on video quality
- Effect of video resolution (i.e., QCIF, QVGA, and HVGA) on video quality
- Effect of coding bit rate, frame rate, and ratio of I-frame (intra-coded frame) bit count to the total bit count on video quality
- Effect of the number of packet-loss events, number of damaged video frames, and the video frame area damaged by the packet losses on video quality
- Effect of the number of rebuffering events, average rebuffering length, and average interval between rebuffering events on video quality

3.3 Audiovisual quality estimation module

Audiovisual quality is estimated based on audio- and video-related quality factors since audiovisual quality is affected by both audio and video quality.

Packet headers do not indicate the codec type and implementation or the video resolution, so coefficients for audio, video, and audiovisual quality estimation modules need to be optimized for these factors. ITU-T Recommendation P.1201.1 provides the optimized coefficients [6].

The rest of this section describes how the number of damaged video frames and the damaged video frame area are derived. When a video frame is lost, degradations of the video frame are propagated until the next I frame is received (**Fig. 3**). Therefore, determining the number of damaged video frames is an effective way of estimating quality. In addition, the damaged video frame area itself affects video quality. The damaged area is derived as follows: 1) the second packet of the “i+1”th video frame is lost, so the mod-

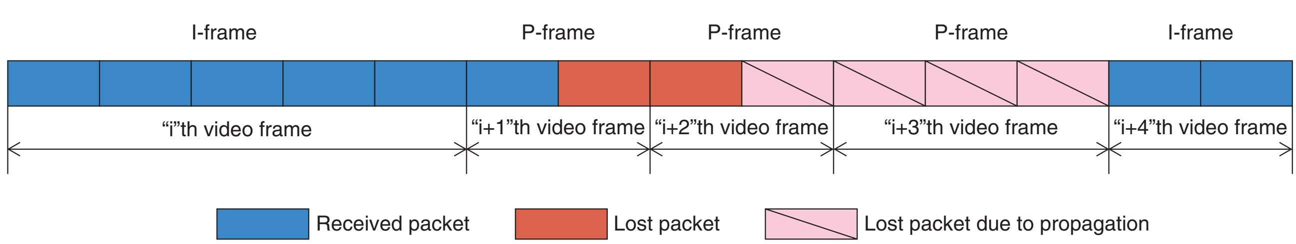

Fig. 3. Number of damaged video frames and damaged video frame area [5].

ule outputs 50% as the damaged video frame area; 2) the first packet of the "i+2"th video frame is lost, so the module outputs 100% as the damaged video frame area; 3) the "i+3"th video frame has not lost any packets, so the module outputs 100% as the damaged video frame area because the previous video frame was lost, and degradation propagation lasts until the next I-frame.

4. Performance of P.1201.1 model

ITU-T SG12 verified the validity of the P.1201.1 model by conducting numerous subjective tests using audio, video, and audiovisual sequences that were generated by varying the codec type, coding bit rate, packet-loss pattern, packet-loss concealment, and rebuffering pattern. The audio, video, and audiovisual quality estimation modules were verified by using the root mean square error (RMSE) and Pearson's correlation (PC). **Table 1** (see Appendix of ITU-T Recommendation P.1201 [8]) lists RMSE and PC values. Additionally, ITU-T verified that the P.1201.1 model reached a sufficient level of quality-estimation accuracy because the RMSE was small and PC was high, as listed in Table 1.

Table 1. Performance of P.1201.1 model [7].

	RMSE	PC
Audiovisual	0.470	0.852
Video	0.535	0.830
Audio	0.351	0.941

5. Application scenario

In-service quality monitoring is an important application of the P.1201.1 model. The P.1201.1 model can be applied as followed:

1) Customer complaints are often made because of quality degradation. Service providers can resolve them by monitoring the quality level. Even if the end-user does not notice any degradation, service providers can resolve problems due to quality degradation before customers complain about the quality.
2) Quality degradation locations can be detected quickly by gathering data on quality and the causes of quality degradation from many users.

6. Conclusion

This article introduced a quality estimation model we developed for IPTV services and the P.1201.1 model. The P.1201.1 model has reached a sufficient level of quality-estimation accuracy and can be applied to monitor the quality of mobile IPTV. As described previously, the P.1201.1 model cannot be used for transmission control protocol (TCP)-based progressive video streaming because it was developed for realtime IPTV services. Although there are different types of video streaming, quality factors such as coding and rebuffering in realtime video streaming are the same as those of TCP-based video streaming. Therefore, it may be possible to extend the model and apply it to TCP-based video streaming if the parameter-extraction module can process TCP headers. Although the application area of the P.1201.1 model is limited to lower resolution, it is desirable to extend the model to the higher resolution application area because the newer mobile terminals support higher resolution. ITU-T SG12 is planning to extend the P.1201.1 model because of these technological developments.

In addition to promoting the P.1201.1 model, to which NTT contributed, our group is involved in globally competitive research and development of a quality-monitoring technique for a new service.

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Playback State Estimation of Progressive Download-based Video Services

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Abstract

The recent proliferation of smartphones and tablet terminals has resulted in the rapid development of video applications. However, in wireless access lines such as those for LTE (Long Term Evolution) and 3G (third-generation) services, packet loss rates and variations in the delay are larger than those in fixed access lines. As a result, video services implemented in TCP (transmission control protocol), known as progressive download (PDL)-based video services, are especially pervasive on wireless access lines.

In this article, we introduce a method of playback state estimation of PDL-based video services. This method makes it possible to accurately estimate the playback state, which will improve the efficiency of monitoring the quality of video services in network and service operation processes.

1. Introduction

The use of smartphones and tablet terminals has increased substantially recently, and with this increase, more and more applications are now being provided via the Internet. In particular, video applications are remarkably pervasive; these are technically divided into three categories according to their delivery mechanism: progressive download (PDL)-based video, download type video, and real-time streaming. The first two are implemented in TCP (transmission control protocol) and the third in UDP (user datagram protocol). PDL-based video services account for about 40% of the volume of domestic Internet traffic. Representative PDL-based video services such as *YouTube* and *Hulu* are targeted to users of personal computers, smartphones, and tablet terminals that access the services over the Internet. In addition, some broadcasters also provide PDL-based video services that enable users to access their news and archived content. NTT Plala started *Hikari TV* (television) *dokodemo/mobile* in 2011, NTT WEST Corporation provides the network platform for *SKY PerfecTV! on Demand*, and NTT DOCOMO offers *BeeTV* and the *d-market* video store. The NTT Group believes that as a network and service provider, one of its most important tasks is improving the quality of these services.

The reasons for this are as follows. First, for network providers, PDL-based video services are considered to be the benchmark of the network. Low-quality PDL-based video services will directly lead to a decrease in customer satisfaction. Second, for a service provider, attractive content and reasonable prices are of course important, but so is the quality of the service itself. Nevertheless, there are no methods for either continuously monitoring the service quality or comparing the service quality with other competitive services. Thus, we have developed a method for addressing these issues for the network and service providers in the NTT Group.

2. Objectives of proposed method

The following two issues are considered to be the main objectives of the proposed method:

(1) Continuously monitoring the service quality;

(2) Comparing the service quality with other

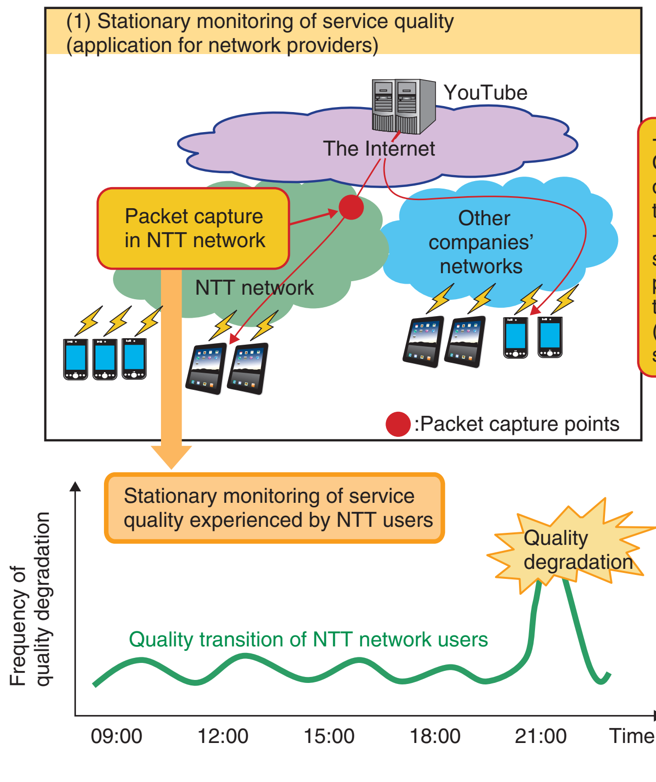


Fig. 1. Possible applications of proposed method.

competitive services.

The first item refers to efficiently enhancing service quality monitoring in the network operation processes, which will enable earlier detection and resolution of quality degradation incidents. The latter item implies that information on the actual service quality is provided, and this information is useful for making decisions on service strategies (**Fig. 1**).

3. Principal factor affecting the quality of PDL-based video services

In this section, we discuss the main factor affecting the quality of PDL-based video services (**Fig. 2**). In real-time streaming applications implemented in UDP, packet losses directly result in block noise of video images, or they halt the playback. When forward error correction (FEC)[*1] is applied, a packet loss that exceeds the redundancy of the FEC configuration leads to the block noise or to the halt of playback.

In PDL-based video, on the other hand, the lost packet will be retransmitted by the server thanks to the functionality of TCP. If the variation in the packet arrival interval caused by packet losses or jitter exceeds a certain level, the video playback will halt due to buffer starvation. Consequently, playback halt is the principal reason for quality degradation of PDL-based video services.

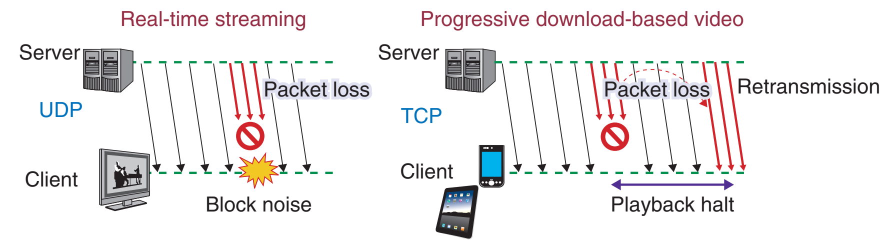
4. Proposed estimation method

4.1 Technical details

The explanation in the previous section indicates that it is most important to accurately estimate the playback state for PDL-based video applications. The proposed method makes it possible to estimate the playback state, especially the total halt duration and the number of playback halt events, by using packet capture data. While existing methods such as YouTube API (application programming interface) [1] are applicable only to specific services, the proposed method is basically applicable to arbitrary PDL-based

*1 FEC: When data are lost due to bit errors or packet losses, the FEC mechanism enables detection and correction at the client side by using the redundant data included in packets.

	Examples of quality degradation events	Representative services
Real-time streaming	- Video quality degradation - Block noise - Frame skip	Hikari TV/BS retransmission, etc.
Progressive download-based video	- Longer initial buffering time - Playback halt due to rebuffering	Hikari TV dokodemo/mobile, Hulu, BeeTV



BS: broadcast satellite

Fig. 2. Factors affecting service quality.

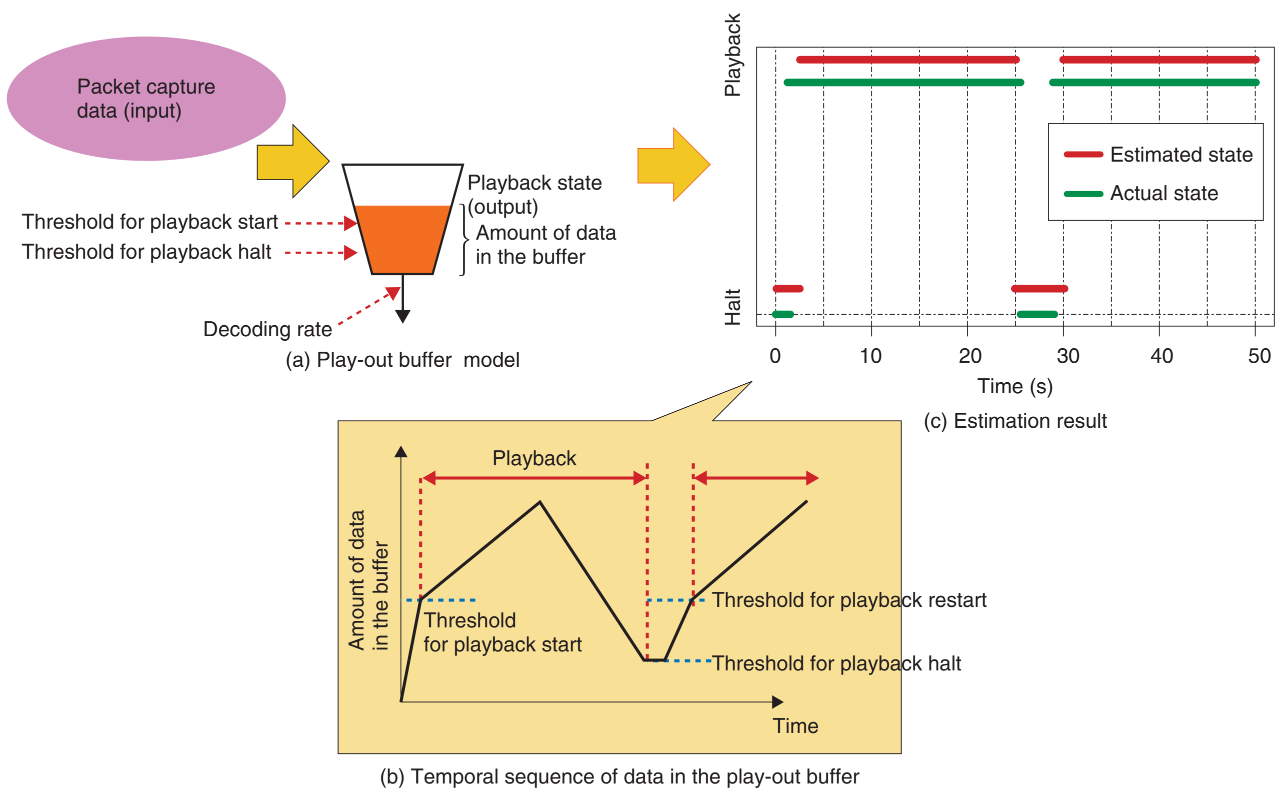


Fig. 3. Input and output of proposed method.

video services. Furthermore, it does not require any additional implementations to either applications or terminals, and is achieved with only passive monitoring. These are the main advantages of the proposed method.

We discuss here the technical details of the proposed method [2]–[4], which is depicted in **Fig. 3**. Before the estimation process is carried out, we first estimate parameters of the play-out buffer (Fig. 3(a)) such as the decoding rate, the thresholds of the playback start, and the halt and restart. This process is executed in advance of the estimation process. These parameters vary because of the different operating systems in client terminals, applications, and video resolution levels, so we have to estimate them through careful observation of the playback state and the corresponding packet capture data.

In the estimation process, we first extract the time sequence of the downloaded data volume in each session by using the TCP header in the packet capture data. Next, we calculate the amount of temporal data in the play-out buffer (Fig. 3(b)). The playback state at each measurement time, which is the final output of this method, is determined by the amount of data in the buffer at each measurement time and the thresholds estimated beforehand (Fig. 3(c)). In some cases, additional analysis of the payload information can enhance the accuracy of the playback state estimation.

Table 1. List of verified services.

Verified services
YouTube
Hikari TVdokodemo/mobile
SKY PerfecTV! on demand
d-market (BeeTV)
Hulu

4.2 Verification

We verified the effectiveness of the proposed method by applying it to the PDL-based video services listed in **Table 1**. An example of the playback state estimation is shown in **Fig. 4**. We evaluated the accuracy of the estimation by calculating the matching rate*2, which was about 92% in this case.

5. Future work

We plan to further enhance the accuracy of estimation by considering the variable decoding rate if the adaptive video quality transitions take place. We will also investigate the detection of the pause state initiated by users and the estimation of the pause state duration [4]. We will continue our research and development of methods to estimate the quality of various application services.

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*2 Matching rate: In this article, we define the matching rate as the total time when the estimated and actual playback states are consistently divided by the total time of the video playback.

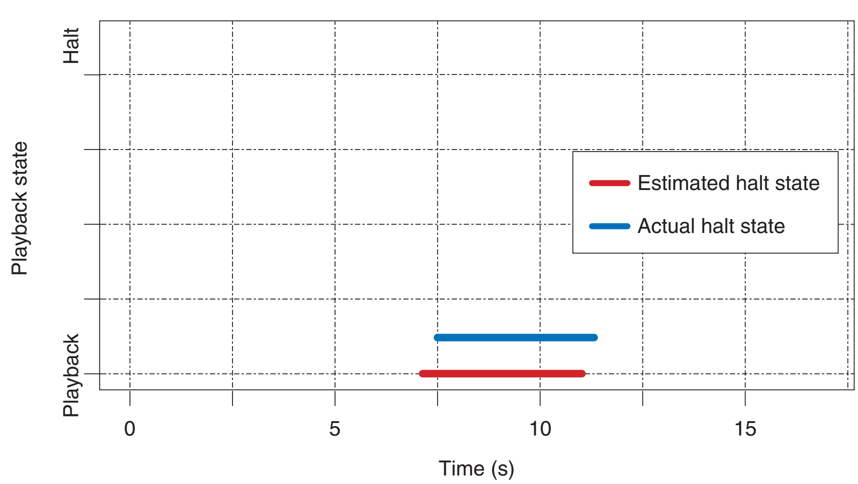

Fig. 4. Exampl of estimation result.

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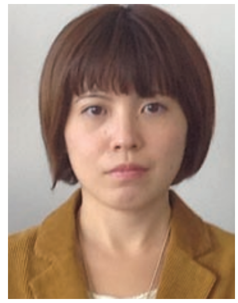

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Performance Estimation Techniques for Browser-based Applications

Hiroshi Yamamoto, Sorami Nakamura, Hirotada Honda, and Akira Takahashi

Abstract

Corporate application services using cloud computing are coming into wide use. These services include SaaS (Software as a Service), a delivery model for cloud-hosted software applications, and are provided via networks. Therefore, the state of the network and of the user terminal determines whether the performance assumed in the application development is achieved. This article describes methods we developed for estimating the waiting time experienced by the user and for determining whether or not a decrease in performance was caused by the user terminal. These methods make it possible to visualize application performance and to provide support when performance declines in browser-based applications, which are the main type of corporate application services.

1. Introduction

Traditionally, applications installed on terminals have been the mainstream, and applications with a web browser user interface started coming into wide use around the year 2000. Then web browsers gained the capability to serve as an application execution platform, which then led to the expanding use of technologies that enhance application interactivity, such as Ajax* or Flash in around 2007 (**Fig. 1**).

In conventional applications, web-browser-downloaded content such as HTML (hypertext markup language) and image files, is rendered and displayed in response to a user operation (hereafter referred to as *static content*). By contrast, in applications using technologies such as Ajax, the web browser first downloads executable code such as JavaScript and then executes the code in response to a user operation (hereafter referred to as *dynamic content*). With dynamic content, most of the processing is performed in the terminal, so application performance is less susceptible to network or server performance. However, it is strongly influenced by the terminal processing performance.

* Ajax (Asynchronous JavaScript+XML) enables dynamic updating of part of an otherwise static web page.

2. Performance indicators and problem with existing monitoring technology

2.1 Application performance metrics

Various indicators can be used to measure browser-based application performance. We used an indicator that correlates to the user experience and is based on the timing from the start of a user operation until the time the results are displayed. We call this the *experienced wait time*. The layer model of browser-based application performance indicators is shown in **Fig. 2**. The experienced wait time corresponds to a key quality indicator (KQI), and KQI consists of various key performance indicators (KPIs) such as the data transfer time and the terminal processing time.

We describe here a specific example of the relationship between these indicators. A schematic representation of signals to be exchanged between the terminal and the server in browser-based applications is shown in **Fig. 3**. The left and right illustrations are respective examples for static and dynamic content. With static content, a hypertext transfer protocol (HTTP) signal is issued synchronously with user operation and data reception, so *HTTP response time*, a KPI, and *experienced wait time,* a KQI, are generally consistent. In the case of dynamic content, the HTTP signal is issued asynchronously with user

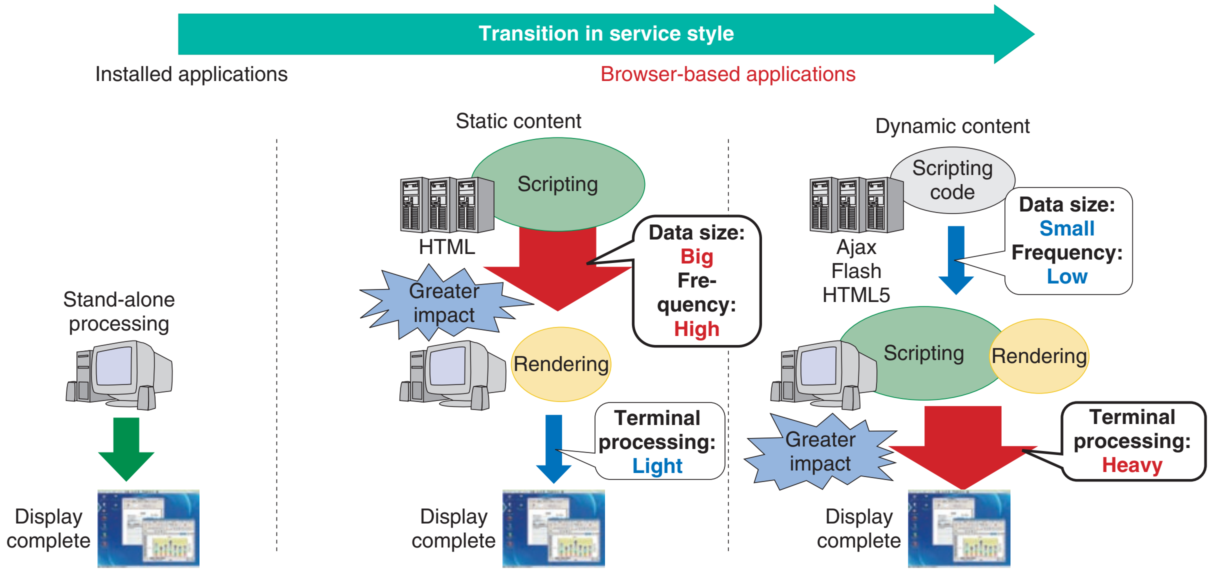


Fig. 1. Transition in style of application services.

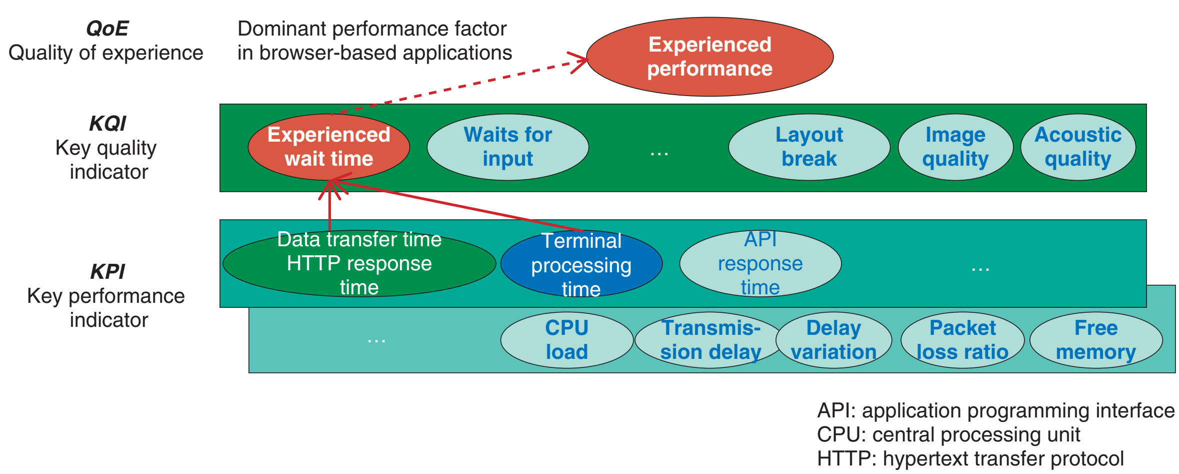


Fig. 2. Layer model of application performance.

operation and the processing performed only in the user terminal such as scripting, so *HTTP response time* and *experienced wait time* do not correlate in most cases [1].

The percentage of processing time, in which the terminal is occupied, out of the total experienced wait time for some sample application operations is shown in **Fig. 4**. These results show that although there are differences between the operations, the terminal processing time is a key factor in the experienced wait time, and the experienced wait time varies greatly in different types of terminals even when the operation is the same (e.g., Fig. 4, *docoiku* old terminal and other terminal). These evaluations show that the terminal has become a key performance factor. Furthermore, existing indicators such as data transfer time (HTTP response time) and server response time (API response time) are not taken into consideration in the terminal processing factor, so it is not possible to grasp the experienced wait time from those indicators.

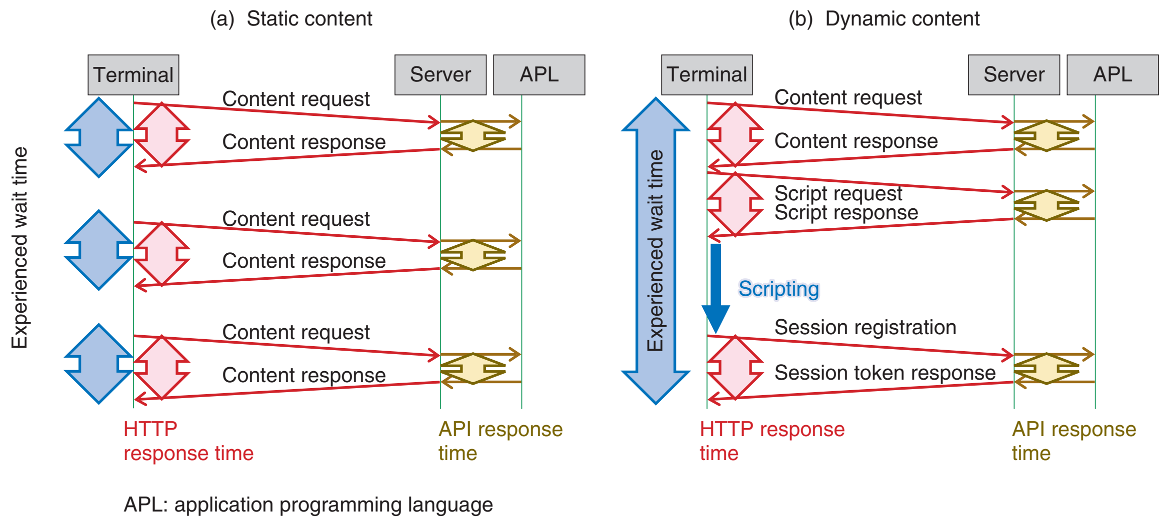
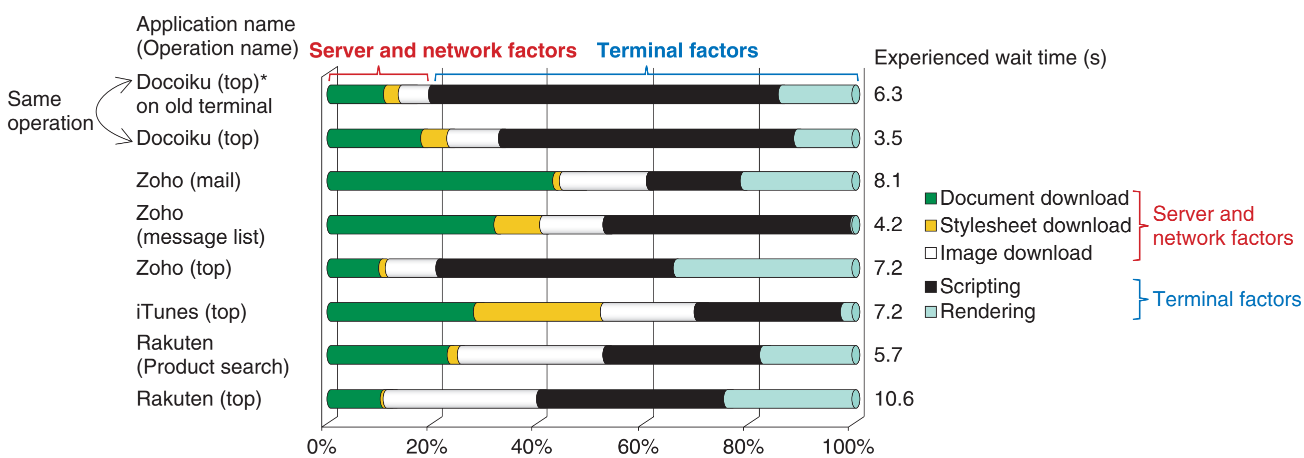


Fig. 3. Correlation of performance indicators.



*(top): Here, "top" means "top user page of docoiku application". This graph shows the experienced wait time for several user operations. For example "Docoiku (top) on old terminal" indicates a user operation to open the "Docoiku top page".

Fig. 4. Percentages of operation processing time out of total experienced wait time for different applications.

2.2 Performance monitoring problem

Most existing performance monitoring products measure the HTTP response time and the API response time mentioned above. These performance indicators are suitable for evaluating network performance and server performance, but not for evaluating experienced wait time. This is because the aforementioned terminal processing time is the key factor in the experienced wait time, and it is not taken into account in those performance indicators.

To address this problem, we developed a method to estimate the experienced wait time. This method is intended to close the gap that traditional performance indicators have in measuring the experienced wait time. We also developed a method to isolate the primary cause of deterioration. Our methods were developed for browser-based applications with dynamic content in order to (1) estimate the wait time experienced by the user and (2) determine whether or not a decrease in performance was caused by the terminal.

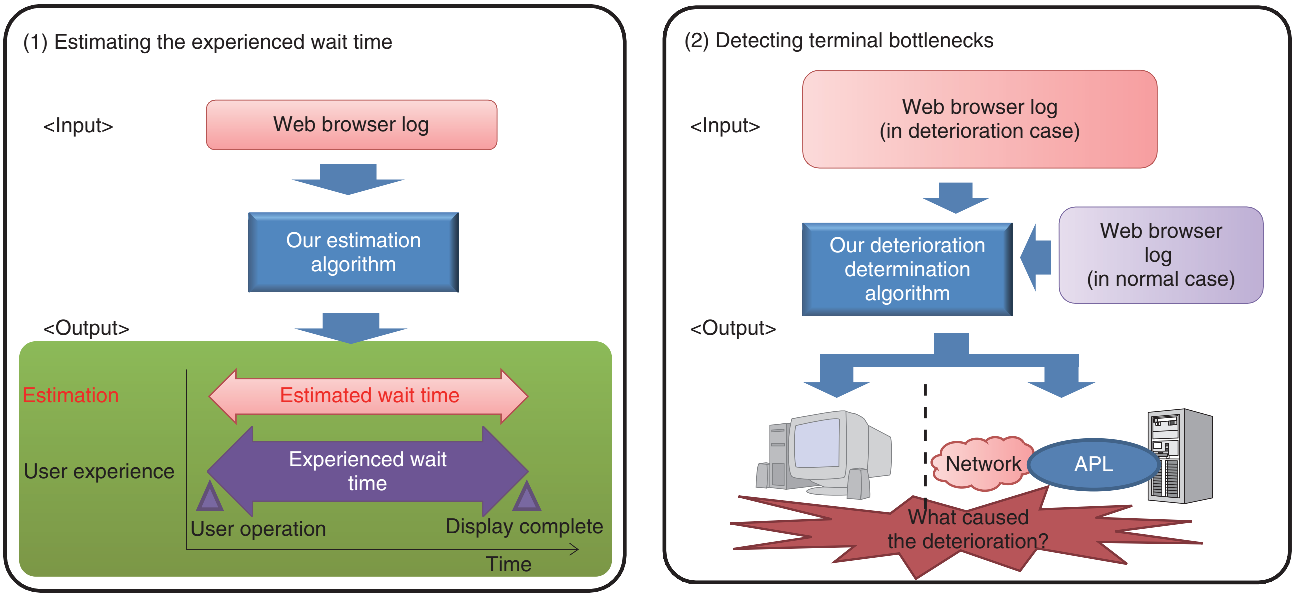


Fig. 5. Overview of estimation methods.

Table 1. Evaluated applications.

Category	Application name	Source	Accuracy (*)
Office	Microsoft Office Web Apps Google Docs Zoho Lotus Live	Microsoft Google Zoho IBM	Good Good Good Good
CRM	Salesforce.com KDDI Business Outlook	Salesforce KDDI	Good Good
WebOS	StartForce	StartForce	Good
Mail	Bizmail (Ajax edition)	NTT Communications	Good
Static content applications	Cybozu Bizmail (HTML edition)	Cybozu NTT Communications	Good Good

CRM: customer relationship management

(*) More than 80%: Good
Less than 80%: Unsatisfactory

3. Introduction of our methods

An overview of our methods is shown in **Fig. 5**. These methods are algorithms that take as input the web browser's processing log data for networking, scripting, and rendering tasks; the output is the user experienced wait time. This information makes it possible to determine whether or not a decrease in performance was caused by the terminal for an operation of any application. More specifically, our algorithms calculate a feature amount from the web browser's networking, scripting, and rendering logs, and output the estimated results by comparing the conditions of the expected user wait time feature amount pattern and the quality deterioration caused by the terminal feature amount pattern, which are prepared in advance.

The applications that were evaluated are listed in **Table 1**. As indicated, our methods can be applied to widely used applications such as *Salesforce* and *Microsoft Office Web Apps*. To use our estimation methods, it is necessary to install the browser plug-in for the terminal targeted for estimation. This installation requires the user's permission, so we applied the methods first to corporate applications.

4. Summary and future work

With the development of in-browser processing technology such as Ajax, the terminal processing time has become a key factor of the wait time that users experience. This has caused a gap between traditional performance indicators such as HTTP response time and API response time and the experienced wait time. We developed two methods for browser-based applications with dynamic content in order to deal with this problem. One method is used to estimate the wait time experienced by the user and the other to determine whether or not a decrease in performance was caused by the terminal.

In the future, we plan to develop an estimation method that does not require a browser plug-in in order to extend our methods to mass users.

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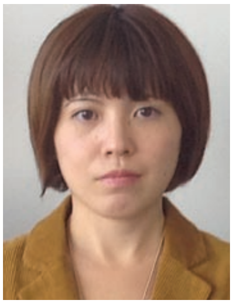

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QoE Assessment Methodologies for 3D Video Services

Kazuhisa Yamagishi, Taichi Kawano, and Kimiko Kawashima

Abstract

Technological advances in three-dimensional (3D) video enable users to watch 3D video content. However, watching such content may cause visual discomfort and fatigue. This article gives an outline of the relationship between 3D video degradation factors and quality of experience (QoE) and introduces QoE assessment methodologies and standardization activities.

1. Introduction

Users can now easily enjoy watching three-dimensional (3D) content on terminals such as 3D televisions (TVs), personal computers, and smartphones due to the recent advances made in these terminals. Video quality has also been improved by the introduction of high definition (HD). In addition, by introducing depth perception to video, 3D video gives a new experience to users. However, some users complain of visual discomfort and fatigue from watching 3D video. To provide high-quality 3D video content, it is important to design and manage services based on the quality of experience (QoE), and to do this, a 3D video QoE assessment methodology is essential.

2. 3D video QoE

This section describes QoE for 3D video services. The International Telecommunication Union, Telecommunication standardization section (ITU-T) Recommendation BT.2021 defines 3D video QoE in terms of visual quality, depth perception, and visual discomfort [1], [2]. As described in section 1, fatigue [3] from 3D video content is also an important factor. In this article, we define QoE in terms of visual quality, depth perception, discomfort, and fatigue. These QoE factors are affected by the 3D video processing chain, as shown in **Fig. 1**.

There are differences between the processing chain and the human vision system (HVS) in 3D video acquisition and display. For example, the position and angle of a camera do not match those of the human eyes. In addition, when a user views 3D video content, they see an image formed from two video images viewed separately by the left and right eye through stereoscopic glasses in the rendering phase of the processing chain. As a result, puppet theater[*1], cardboard[*2], and spatio-temporal asynchronous effects between the left and right views occur. Crosstalk[*3] due to the stereoscopic glasses also occurs.

3D video is downsized and/or encoded in order to reduce the network bandwidth and the amount of storage needed. To use the existing infrastructure for codec and transmission, the spatial resolution of the left and right views, which are arranged in a side-by-side frame-compatible format, is usually down-converted by half in the horizontal direction to maintain the spatial resolution of a full high-definition (HD) 2D video sequence. The video is encoded by MPEG-2 (Motion Picture Experts Group-2) or H.264/AVC (advanced video coding) and is transmitted to a user terminal such as a set-top box. Finally, the side-by-

*1 The puppet-theater effect makes a 3-D image look unnaturally small compared with the target image; people appear to be miniaturized puppets.

*2 The cardboard effect makes 3-D images look layered, i.e., consisting of flat objects against a flat background, though an observer can grasp the situation in front of and behind the shooting target.

*3 Crosstalk appears because of imperfect view separation when a small proportion of one eye's image is also perceptible by the other eye.

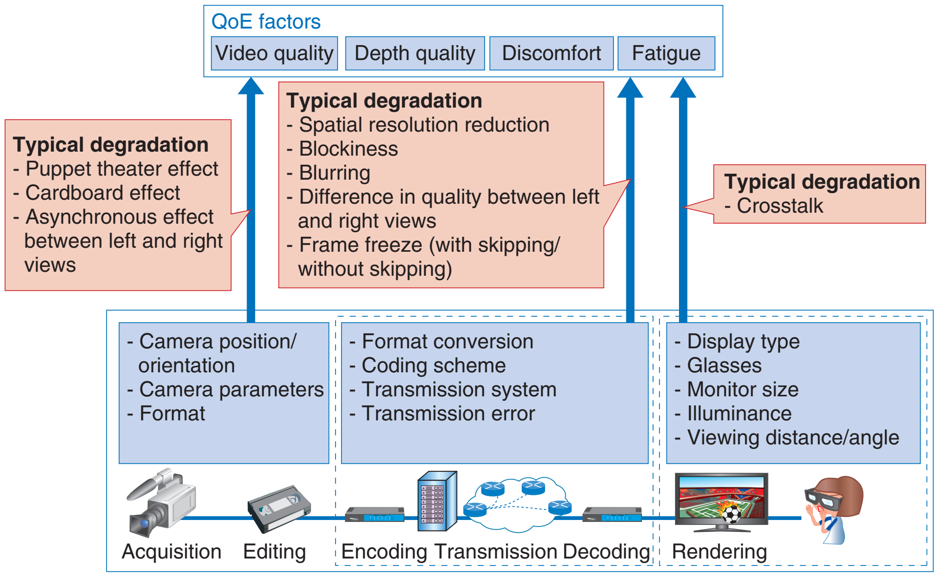

Fig. 1. 3D video processing chain.

side format video is decoded and up-converted to two full HD video signals for the left and right views. Thus, users perceive degradations in quality due to the reduced spatial resolution in addition to coding artifacts such as block noise. To prevent the degradation due to the reduced spatial resolution, the use of two full HD video signals for left and right views, which is called the frame-sequential format, is ideal. In this case, an H.264/MVC (multiview video coding) is often used, which involves an inter-view prediction technique that encodes the right-view video using both videos for left and right views in order to reduce the bit rate for the right-view video. With this system, service providers often encode the right-view video at a much lower bit rate than the left on the basis of binocular suppression. The two 2D video signals for the left and right views have full HD resolution, but they have an asymmetric quality in addition to coding artifacts such as block noise.

Encoded 3D video is packetized and transmitted over a network such as an IP (Internet protocol) or terrestrial network. Packet loss sometimes occurs in networks. Block noise occurs if there is no packet-loss concealment (PLC) technique applied in the user terminal. In contrast, if a PLC technique is applied in the user terminal, freezing artifacts will be introduced when the PLC scheme of the receiver replaces the erroneous frames (either due to packet loss or error propagation) with the previous error-free frame until a decoded picture without errors has been received. This type of artifact is also called *freezing with skipping*. The rebuffering artifacts come from rebuffering events at the receiver, which could be the result of a stream arriving late. This type of artifact is also called *freezing without skipping*.

Degradation perceptions may change due to the display size, room illuminance, viewing distance, and angle.

Therefore, as mentioned previously, because 3D video QoE is affected by many factors, methodologies are needed to assess 3D video QoE.

3. 3D video subjective assessment methodology

Subjective assessment, in which users subjectively evaluate 3D video QoE, is a fundamental quality assessment technique. ITU-R Recommendation BT.2021 was standardized for the 3D video quality subjective assessment method. As described in section 2, it is important to develop methodologies that assess depth perception, discomfort, and fatigue since 3D video QoE is affected by these factors, in addition to visual quality. The Video Quality Experts Group (VQEG) is currently investigating subjective assessment methodologies concerning depth perception and discomfort. However, it is often difficult to evaluate discomfort and fatigue using questionnaire-based subjective assessment because the levels of these indicators are sometimes low in questionnaires. Therefore, it is important to supplement information obtained from questionnaires with biological information such as heart rate, breathing rate, pupil changes, and eye-blink responses when assessing discomfort and fatigue. Our group has investigated the relationship between fatigue and biological information. The relationship between fatigue and the difference in video quality between left and right views is shown in **Fig. 2**. In this figure, the lower the number on the fatigue axis, the greater the amount of fatigue; i.e., a value of 1 represents high fatigue, whereas 5 represents low fatigue. The relationship between the eye-blink rate and the difference in video quality is shown in **Fig. 3**. As shown in Fig. 2, fatigue score increases as the difference in video quality increases. As shown in Fig. 3, the eye-blink rate increases as the difference in video quality increases. These results suggest that fatigue can be evaluated using the eye-blink rate.

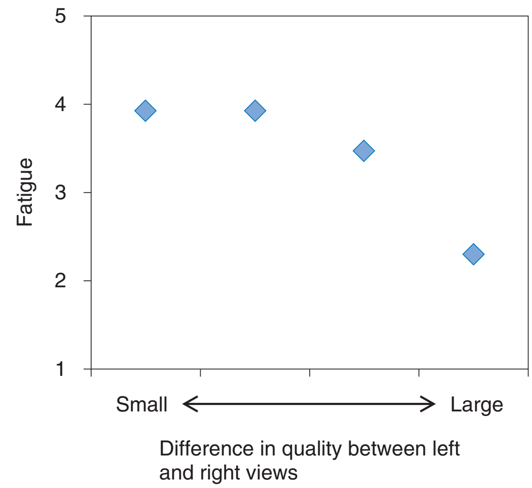

Fig. 2. Difference in quality between left and right views vs. fatigue.

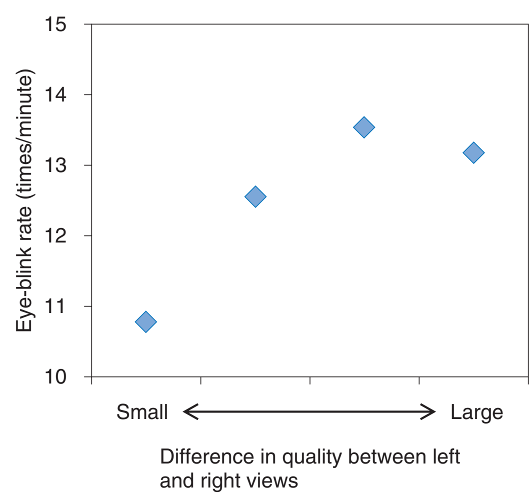

Fig. 3. Difference in quality between left and right views vs. eye-blink rate.

4. 3D video quality objective estimation methodology

Developing an objective quality estimation model that can be used to estimate QoE using information such as 3D video signals is essential for monitoring QoE.

It is important to take into account the degradation factors described in section 2 in order to develop such a model. In principle, block noise, blurring, and freezing due to encoding and transmission also occur in 3D video services. Therefore, the 2D video quality objective estimation model can be applied to 3D video quality estimation. However, since quality degradation factors such as the difference in video quality between left and right views, the asynchronous effect between left and right views, and crosstalk do not occur in 2D video services, these factors need to be taken into account in 3D video quality estimation.

Our group has been developing an objective quality estimation model that takes 2D video quality for left and right views, which is derived from a 2D video quality objective estimation model, as input. Video quality is denoted as a mean opinion score (MOS), where 2D video quality for the left view is denoted as MOS-L, 2D video quality for the right view is denoted as MOS-R, and the difference in video quality for left and right views is denoted as

dMOS-LR (= ABS(MOS-L – MOS-R)).

We compared the performance of our model with that of a conventional model used to calculate the average 2D video quality for left and right views. **Table 1** lists the performance values of our model and the conventional model, i.e., the root mean square errors in the range of

$0 \leq$ dMOS-LR ≤ 1 and $1 <$ dMOS-LR.

The results show that our model can estimate 3D video quality in the range of

$1 <$ dMOS-LR,

more accurately than the conventional model can.

VQEG is also investigating a 3D video quality objective estimation model and discussing a test plan that will be used to verify the validity of such future models.

Table 1. Quality estimation accuracy.

	Proposed model	Conventional model
0 ≤ dMOS-LR ≤ 1	0.22	0.24
1 < dMOS-LR	0.23	0.35

5. Conclusion

The introduction of 3D video has enabled service providers to provide a new visual experience, e.g., depth perception, to users. However, some users have complained of visual discomfort and fatigue from watching 3D video. Therefore, it is important to clarify factors that affect QoE and to develop a model to estimate QoE in order to provide high-QoE 3D video. Our group has been investigating subjective assessment methodologies for 3D video quality and fatigue as well as an objective quality estimation model. We plan to propose our model to VQEG in the future. We will also develop subjective and objective quality assessment methods for QoE other than that for 3D video quality using biological information. We will promote the practical use of these methods, which will contribute to providing a safe and pleasant 3D video streaming service.

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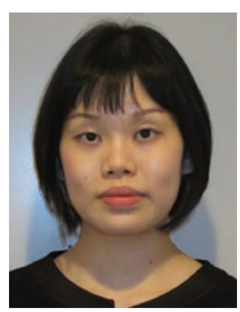

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Historical Overview of Semiconductor Device Reliability for Telecommunication Networks—

Field Data, Prediction Model of Device Failure Rate, and Wear-out Failure Analyses at NTT

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Abstract

The 1960s brought about the advent of the intense development of semiconductor devices for industrial and commercial applications. Since then, NTT has taken the initiative in introducing semiconductor devices into telecommunication networks to realize highly reliable and miniaturized equipment with high performance.

This article reviews the failure physics approach we have been using to develop highly reliable semiconductor devices. It also summarizes our findings regarding device failures in field use and discusses a failure rate prediction model based on the findings.

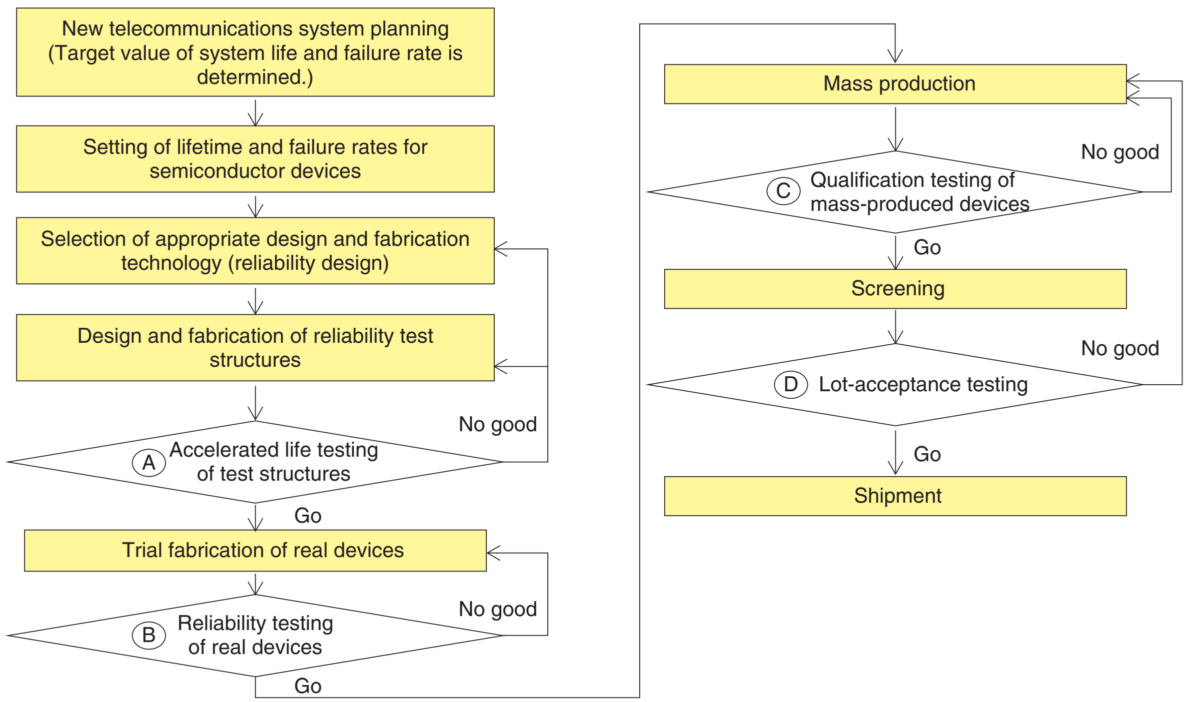
1. Introduction

From the 1960s to the 1980s, NTT's use of semiconductor devices was the primary factor in the successful development of highly reliable and miniaturized high-performance telecommunications equipment. In the 1960s, the semiconductor industry was in its infancy. Therefore, it was not really known how reliable semiconductor devices would be, and experience in using them in telecommunications equipment was lacking. At that time, NTT was the first company in the world to introduce transistors into telecommunications equipment [1]–[7].

In the early 1960s, NTT introduced a failure physics approach (also known as *the physics of failure*), described in the next section, in order to develop highly reliable semiconductor devices for telecommunications equipment. The target values for device reliability required in network system design were a lifetime longer than 25 years and a failure rate lower than 0.2–150 FIT (failure in time, which is defined as the number of failures per billion device hours (10^{-9}/h)), depending on the type of device (from diodes to LSIs (large-scale integrated circuits)) and the kind of equipment [4]. For semiconductor devices that had been developed at that time, accelerated life tests (longer than 10,000 hours) and long-term operation tests at room temperature using a large number of devices were conducted to confirm that those targets had been achieved. In addition, field failure tests were carried out with actual working equipment to confirm the high reliability of semiconductor devices and of passive and mechanical components as well.

On the basis of the results of those field failure tests, we developed a failure rate prediction model for electronic components including semiconductor devices to use in the reliability design of the next stage of our telecommunications equipment [8], [9]. The objectives of this failure rate prediction model were the same as those of the well-known MIL-HDBK-217 (Military Handbook) model for reliability prediction of electronic equipment [10], and the procedures were similar.

The development of reliability assurance standards



Criteria for each testing point:

A	B	C	D
- Failure mode extraction using test structures - Whether the estimated failure rate is lower than the target value or not	- Whether degradation modes are a matter or not - Whether lifetime and failure rate of wear-out failures are lower than the target values or not	- Whether lifetime and failure rate are lower than the target values for mass-produced devices or not	- Whether the produced lot is normal or not

Fig. 1. Procedure for developing highly reliable semiconductor devices for telecommunications equipment.

such as MIL-STD (Military Standard) -19500 [11] and MIL-STD-38510 [12] started in the late 1950s, and NTT applied them to semiconductor devices installed in NTT equipment to ensure their high reliability [8]. Over the years, NTT also developed and periodically revised its own assurance standards to reflect advances in device technology. These standards were finally withdrawn in 1987 because the reliability of semiconductor devices had been improved to the level where such standards were no longer necessary.

2. Development of highly reliable devices based on the failure physics approach

The failure physics approach for developing highly reliable devices is as follows; first, various failure modes are investigated in accelerated life tests. Next, associated failure mechanisms are clarified, and device reliability is improved by optimizing device structures and fabrication processes. Finally, a reliability assurance procedure is carried out; this procedure includes conducting wafer acceptance tests, screening tests of encapsulated devices for eliminating early failures, and lot-acceptance tests.

The procedure we followed for developing highly reliable semiconductor devices is shown in **Fig. 1**, and examples of devices used in NTT equipment are summarized in **Table 1** [13], [14]. The relationship between failure rate and working time is known as the bathtub curve, which depicts the early failure period, random failure period, and wear-out failure period. Early failures are caused by latent defects induced during fabrication, and the failure rate decreases with time ($\beta < 1$, where β is the shape parameter in a Weibull distribution). Random failures are caused by

Table 1. Main devices used in various telecommunication systems, failure modes, and countermeasures for higher reliability.

System (System name/ year installed)	Main devices (type or process used)	Main failure modes	Countermeasures for higher reliability
Coaxial cable transmission system (CP-12MTr/1967)	Bipolar transistors (direct wire bonding on metallization of E-B junction)	- Wire bond breakage due to Au-Al compound formation between Au wire and Al metallization pad on junction - E-B junction short due to Au penetration of Au wire into junction	- Control thermal process during wire bonding and control bond strength - Change Au wire to Al wire
Electronic exchange system (D10/1972)	Logic ICs (bipolar CSL type)	- Open failure of Al metallization due to corrosion	- Improve passivation film
Electronic exchange system (D10-HCP/1977)	4K DRAM (n-channel MOS process)	- Increase in leakage current - Decrease in holding time	- Control contamination - Optimize screening conditions
Submarine coaxial cable system (CS-36M/1977)	Bipolar transistors (Au/Pt/Ti electrode, BeO substrate for high thermal conductivity)	- Metal penetration into junction - h_{FE} degradation	- Improve electrode metal structure from Al to Ti/Pt/Au electrode - Improve passivation film from single layer of SiO_2 to SiO_2/SiN - Use individual device assurance method
Digital electronic exchange system (D70/1982)	High-voltage bipolar LSIs (dielectric isolation, pn junction isolation, plastic encapsulation)	- Open failure of Al metallization due to corrosion - Degradation of breakdown voltage for isolation	- Adopt SiN passivation film - Adopt EPIC dielectric isolation structure combined with field plate
	64K DRAM	- Soft error	- Use chip coating with resin - Adopt ECC technique
Optical fiber transmission system (F-400M/1983)	InGaAsP/InP BH-LDs (Fabry-Perot type, buried-heterostructure)	- Increase in thermal resistance in die bonding to substrate - Short failure due to Sn whisker growth around die bonding - Degradation of optical power output	- Improve metallized layer and adopt silicon heat sink - Use Au-rich AuSn die bonding - Use individual device assurance method
ISDN network (ISDN/1988)	Submicrometer custom LSIs (CMOS, BiCMOS)	- Open failure due to electromigration and stress migration of Al metallization - Vth degradation due to hot-carrier effect - TDDB of gate oxide	- Use multilayer metallization structure (Al-Si-Cu/Ti/TiN/Ti, etc.) - Control process of H_2O content in interlayer dielectric film - Use high-voltage and low-temperature screening

BeO: beryllium oxide
BH: buried heterostructure
CSL: controlled saturation logic
Cu: copper
E-B: emitter-base
ECC: error-correcting code
EPIC: epitaxial passivated integrated circuit
H_2O: hydrogen dioxide (water)
InGaAsP: indium gallium arsenide phosphide
ISDN: integrated services digital network
Pt: platinum
$Si0_2$: silicon dioxide
SiN: silicon nitride
Sn: tin
TDDB: time-dependent dielectric breakdown (gate oxide breakdown)

small latent defects that occur after relatively long-term operation at a constant failure rate ($\beta = 1$). Wear-out failures occur after long-term operation due to wear-out and fatigue, and the failure rate tends to

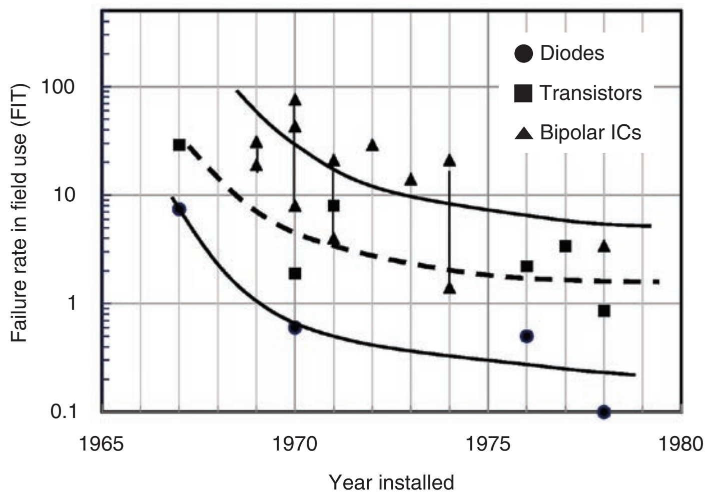


Fig. 2. Failure rate in field use vs. year installed in NTT equipment for semiconductor devices.

increase rapidly during this period (β > 1).

Countermeasures consisting of fabrication process control and proper screening were effective in reducing the failure rate of early failures. For random failures, long-term tests conducted under actual working conditions confirmed that failure rates were lower than the target value. Wear-out failures, which should never occur during the guaranteed lifetime of a device, were prominent for miniaturized LSIs and optical devices. These failures are classified into degradation-related failures and catastrophic ones. Examples of degradation failures are a decrease in the optical output power of laser diodes (LDs) or changes in the current gain (h_{FE}) of bipolar transistors during operation. In a circuit affected by such degradation, the extent of the degradation has to be accurately estimated in order to analyze its effect on circuit operation in the intended working conditions. Examples of catastrophic failures are the lifting of Au (gold) wire bonding and stressmigration in Al (aluminium) interconnections in plastic-encapsulated LSIs. Wear-out failure analyses that we performed for plastic encapsulated LSIs and LDs are described in more detail in section 4.

3. Failure rate in field use and reliability prediction model

3.1 Failure rate data in field use

From the 1960s to the 1980s, NTT collected a large amount of data on field failures of devices in NTT telecommunications equipment. The devices were typically operated in an air-conditioned room with a maximum junction temperature of 55ºC (ambient temperature 40ºC) and relative humidity of 20–70% [5], [15]–[25].

The relationships between the failure rate of devices (in failures/10^9 hours = FIT) in field use and the installation year for diodes, bipolar transistors, and bipolar integrated circuits (ICs) are shown in **Fig. 2**. These values were selected from field data of more than 1×10^{10} device•hours for diodes, 4×10^9 device•hours for bipolar transistors, and 5×10^8 device•hours for bipolar ICs, which corresponded to one to three years of operation. The lower limits of the failure rate estimation are 0.01 FIT for diodes, 0.25 FIT for transistors, and 2 FIT for bipolar ICs. These lower limits are one order lower than the observed failure rate as described below.

As shown in Fig. 2, the failure rate decreased with time and approached a certain constant value depending on the device group. These curves are similar to the so-called learning curve, and the constant value corresponds to the failure rate in the random failure period for mature devices. The failure rate seemed to become higher as the process complexity increased. The failure rate in the random failure period was estimated to be about 0.2 FIT for diodes, about 2 FIT for transistors, and about 5 FIT for bipolar ICs. The failure rate observed in field use from the 1960s to the 1980s for more precisely divided device groups in the random failure period is plotted in **Fig. 3**. The figure shows that somewhat high failure rates were observed for metal-oxide semiconductor (MOS) ICs, which

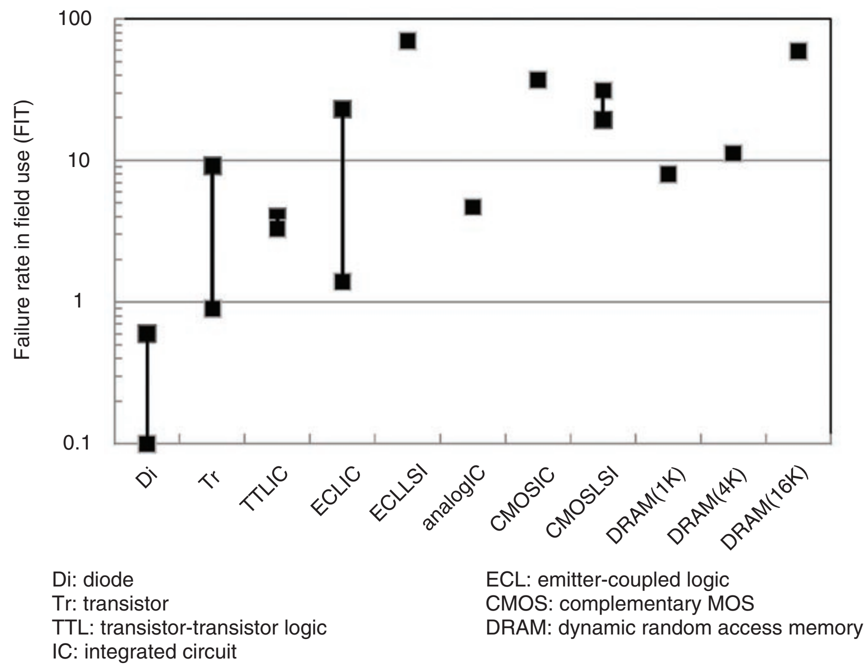


Fig. 3. Failure rate in field use in random failure period for various semiconductor devices.

had just been developed and had not yet matured in those days, though the failure rates were less than 100 FIT. We also examined the failure rate for LSIs developed in the 1990s by conducting accelerated life tests with a large number of LSIs for about 3000 hours. We found that the failure rates of memory LSIs such as 64-K–1-M DRAM (dynamic random access memory) were below 100 FIT, and those of CMOS (complementary MOS) logic LSIs (about 150-K gates) fabricated with 0.25–0.5 μm technology were below 200 FIT. The failure rates of these LSIs were expected to decrease in the accelerated life tests and approach a constant value below 100 FIT if the tests were continued beyond 3000 hours, because no failures were observed within the first 3000 hours.

3.2 Failure rate prediction models

3.2.1 NTT model

A failure rate prediction model was developed on the basis of the collected field data and used in the next stage of equipment design [8], [9], [26], [27]. The NTT failure rate prediction model (called the NTT model hereafter) was sought to determine the constant failure rates in the random failure period and was similar to the well-known MIL-HDBK-217 model. It was established for semiconductor devices and passive and mechanical components used for NTT equipment. The failure rate of ICs is expressed as

$$\lambda = \lambda_b \cdot \pi_Q \cdot (\pi_E + \pi_T \cdot \pi_V), \qquad (1)$$

where λ is the predicted failure rate, λ_b is the basic failure rate (FIT), π_Q is the quality factor, π_E is the environmental factor, π_T is the temperature factor, and π_V is the power supply voltage factor.

The basic failure rate λ_b was defined for each device type, integration level, and technology. For example, the λ_b of bipolar digital logic ICs and MOS DRAM is listed in **Table 2**.

Quality factor π_Q was given a value from 1 to 4, depending on the quality grade defined by NTT; it was 1 for highly reliable ICs (namely, quality-controlled devices used in NTT equipment). Environmental factor π_E was given a value from 0.3 to 1.8, depending on the operating environment such as an air-conditioned room, vehicle, or mobile telephone; it was 0.3 for an air-conditioned room. Power supply voltage factor π_V was expressed as $\pi_V = 0.35 \times \exp(0.21V_{DD})$ for CMOS ICs and 1 for other devices.

Temperature factor π_T is given as

$$\pi_T = \exp\left(\frac{0.3}{k}\left(\frac{1}{339} - \frac{1}{273+T_j}\right)\right) + \exp\left(\frac{0.7}{k}\left(\frac{1}{356} - \frac{1}{273+T_j}\right)\right), \qquad (2)$$

Table 2. Example of basic failure rate of ICs.

Category	λ_b (FIT)					
Bipolar digital logic ICs	5.0	for	$Q \leqq 100$	$0.50Q^{0.50}$	for	$Q \geqq 100$
MOS DRAM	$3.81B^{0.25}$	for	$B \leqq 16$ K	$0.337B^{0.50}$	for	$B \geqq 16$ K

Q: number of active transistors
B: bits; K = 1024

where k is Boltzmann's constant and T_j is junction temperature at the operating condition. This formula was derived by considering the contribution to the failure of two activation energies (Ea) of 0.3 and 0.7 eV. Here, various failure mechanisms showed different activation energies, and the major failure mechanisms for MOS ICs were gate oxide breakdown (TDDB) with Ea=0.3 eV and electromigration or stressmigration of metallization with Ea=0.7 eV. Activation energies for these major failure mechanisms were therefore introduced to define the temperature factor in the NTT model.

Unlike the MIL-HDBK-217 model, the NTT model considered the effect of packaging only for hermetic or non-hermetic (plastic-encapsulated) sealing and did not take the effect of package type or size into account.

3.2.2 Comparison of NTT model with other failure rate prediction models

The final revised version of the NTT model was completed in 1981 and published in 1982 [8]. In those days, comparable prediction models for electronic components were MIL-HDBK-217 (version C (1979) and version D (1982)), and Bellcore TR-332 [28]. Later, other failure rate prediction models were developed such as PRISM [29] and IEC/TR (International Electrotechnical Commission/Technical Report) 62380 [30]. These models consider not only individual components but also the total system operation, and they differ from device-oriented models such as the NTT model and MIL-HDBK-217. Quantitative comparisons of the predicted trend of failure rates among these various models were needed but had not been carried out [31], [32]. We therefore performed comparisons between six device-oriented models of the NTT model (1981), three versions of the MIL-HDBK-217 model (C(1979), E(1986), F(1991)), and two versions of the Bellcore model (TR-332 issue (1997), SR-332 issue 2 (2006)). (Note that TR-332 was changed to SR-332 in 2001.) Operating conditions were assumed to be $T_j = 55$ºC (ambient temperature of 40ºC) and an air-conditioned room, which corresponded to "Ground, Benign" in MIL and also to the NTT electronic switching system environment. The quality grade was selected as the highest reliable level at NTT, i.e., $\pi_Q = 1$, which was equivalent to MIL-class B. The power supply voltage factor was a standard voltage, and $\pi_V = 1$. Under these conditions, comparisons of failure rate were performed for NMOS (n-channel MOS) DRAM and bipolar digital logic ICs as a function of the scale of integration. The results are shown in **Figs. 4** and **5**.

For DRAM, the failure rate prediction curve obtained with the NTT model lies between MIL-HDBK-217 C(1979) and 217 E(1986), and the relationship between failure rate and DRAM size seems to shift in parallel with subsequent versions, i.e., the year of publication. For digital logic ICs, the prediction curves tend to converge at around 200 FIT with increases in the number of gates, regardless of the model. The features of the relationship between failure rate and scale for DRAM are somewhat different from those for bipolar logic ICs. This difference can be explained as follows; the process technology for DRAM had been revised with the very large scale production more quickly than it had for bipolar logic ICs. Consequently, the yield and reliability were improved more quickly compared with bipolar digital ICs. The trend in the predicted results indicated that the failure rate of LSIs would converge to a constant value regardless of the scale of integration, and that value was estimated to be 100–200 FIT even for the maximum scale of integration for a given year. On the whole, the relationship between failure rate and scale of integration can be assumed to shift in parallel with the fabrication year due to advances in technology. Although these prediction models involving the NTT model are old, they can still be used as references for predicting failure rates of recently developed devices.

4. Examples of analyses for wear-out failure modes

Wear-out failure is divided into catastrophic failure

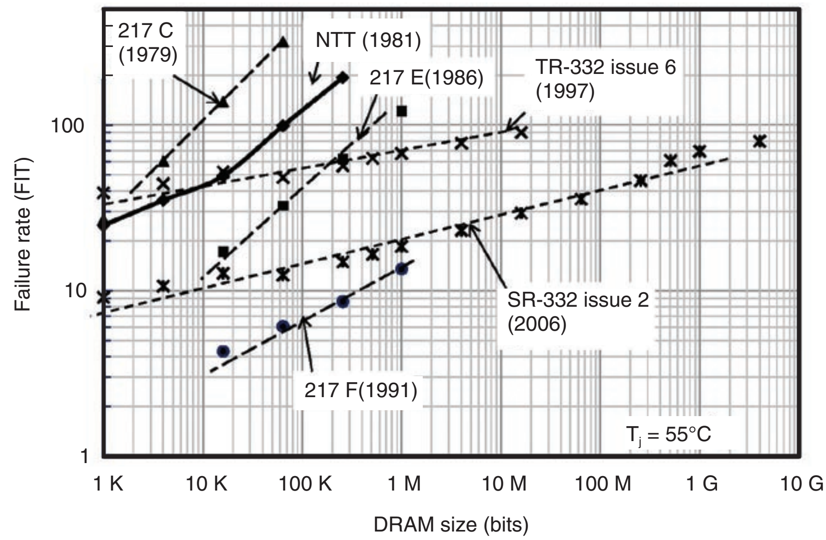


Fig. 4. Predicted failure rate vs. size of DRAM for various failure rate prediction models.

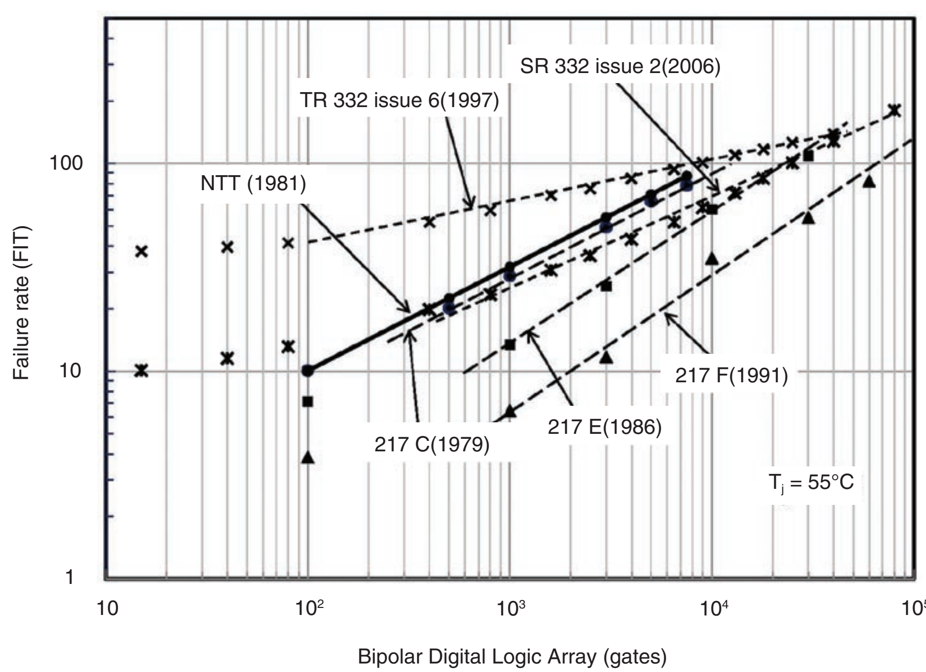
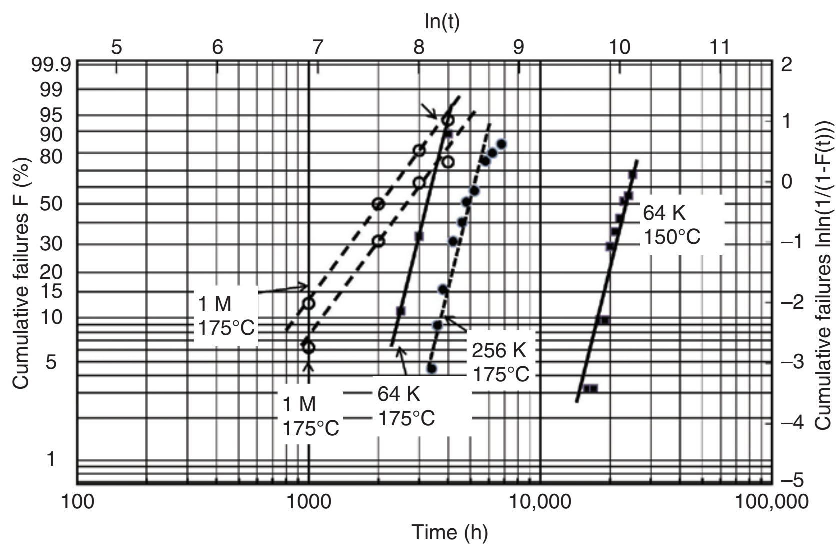


Fig. 5. Predicted failure rate vs. number of gates in bipolar logic IC gate arrays for various models.

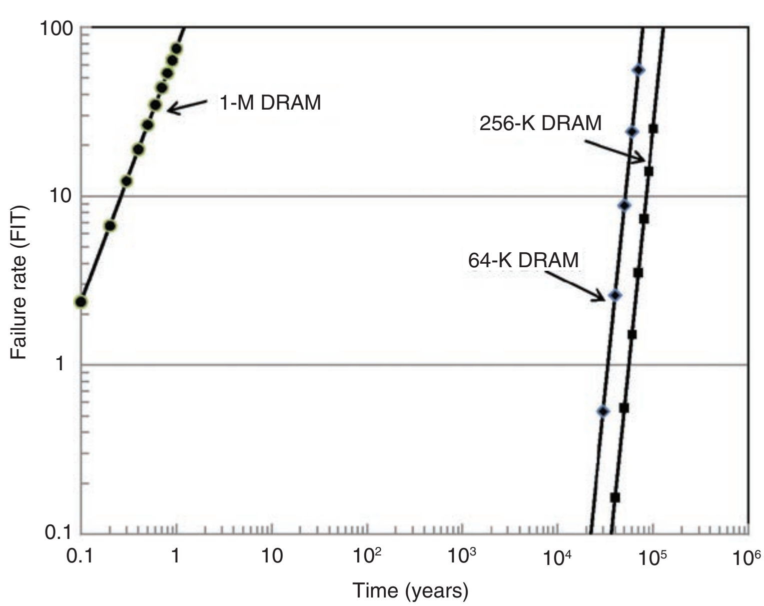
and degradation failure modes, as mentioned previously. The time to reach wear-out failures has tended to decrease with the rapid progress in the miniaturization of device dimensions used in LSIs since the latter half of the 1980s. Reliability analyses of plastic-encapsulated LSIs and LDs are described below as respective examples of catastrophic and degradation failures.

4.1 Reliability analysis for catastrophic failures

A reliability analysis was carried out for plastic-encapsulated MOS DRAMs, in which typical failure modes were open failures caused by the lifting of Au wire-bonds and stressmigration of Al(Si) metallization [13]. Weibull plots of failures observed in high-temperature operating tests for 64-K, 256-K, and 1-M DRAMs are shown in **Fig. 6**. Failure analyses clarified that the failures involved the lifting of Au wire-bonds for the 64-K and 256-K DRAM and stressmigration of Al(Si) metallization for the 1-M DRAM. Au wire-bond failures were induced by thermal expansion mismatch among the molding compounds (10–20 ppm/°C), Au wire (14 ppm/°C), and Si (3 ppm/°C), and also by decreases in the thermal expansion coefficient of molding compounds during high-temperature aging. Stressmigration failure was



(a) Weibull failure distribution



(b) Failure rate vs. time in use

Fig. 6. (a) Failures observed for plastic encapsulated DRAM in high-temperature operation tests and (b) estimated failure rate vs. time under normal operating temperature (T_j = 55°C).

Table 3. Weibull parameters determined from high-temperature operation tests for NMOS DRAM.

Device	η (scale parameter) at 175°C (h)	β (shape parameter)	Activation energy (eV)	Failure mode (open failure)
64-K DRAM	3400	6.5	1.25	Lifting of Au wire-bonds
256-K DRAM	5200	6.5	1.25	Lifting of Au wire-bonds
1-M DRAM	2300	2.5	0.55	Stressmigration of Al(Si) interconnection

caused by tensile stress due to the thermal expansion difference between the Al interconnection and the intermediate layer.

Weibull parameters determined from the results of the accelerated life tests shown in Fig. 6(a) are listed in **Table 3**. The values of the shape parameter β were 6.5 for 64-K and 256-K DRAM and 2.5 for 1-M DRAM, which indicates the wear-out failures. The activation energy of the Au wire-bond open failure was estimated to be 1.25 eV for 64-K DRAM, and was assumed to be the same for the 256-K DRAM. The activation energy of the Al(Si) stressmigration failure observed in the 1-M DRAM in the 150ºC and 175ºC tests was estimated to be 0.15 eV. However, the failure rate for stressmigration was known to have a plateau around 180ºC and to show complicated temperature dependence [33]. Therefore, 0.55 eV was adopted as the activation energy for failure rate estimation, in which the energy was derived from testing below 150ºC using other test structures [34].

The relationship between estimated failure rate and operating time at the operating temperature of 55ºC derived from the above parameters is shown in Fig. 6(b) in a Weibull distribution. Here, the failure rate in the Weibull distribution was obtained by

$$\lambda(t) = \left(\frac{\beta}{\eta}\right)\left(\frac{t}{\eta}\right)^{\beta-1} \tag{3}$$

The target value of the failure rate for catastrophic wear-out failures was set at 10 FIT after 25 years. The failure rates of 64-K and 256-K DRAM were far below the target value, while that of the 1-M DRAM reached the target value after only one year, as shown in Fig. 6(b), which indicates an obvious need for improvement. The1-M DRAM was subsequently improved by using Al(Si, Cu) interconnections instead of Al(Si) interconnections, and the lifetime of the improved 1-M DRAM was confirmed to be 100 times longer than the former one.

Further, room-temperature operating tests were carried out over 17,400 hours with 320 samples, and the results confirmed that the rates of random failures of 64-K DRAM were below 165 FIT. For the 1-M DRAM, 150ºC operating tests using 110 samples for a duration of 10,000 hours confirmed that the rates of random failures were below 62 FIT. The analyses of wear-out failures and random failures confirmed that the total failure rates satisfied the target values.

4.2 Reliability analysis for degradation failures

Degradation of electrical characteristics is a major concern for semiconductor devices such as LDs, whose optical output power degrades with time. Degradation analyses, therefore, are needed in addition to random failure examination and are typically obtained by conducting accelerated life tests. It is imperative to confirm that the amount of degradation during the guaranteed duration of use is within the designated value by analyzing the degradation characteristics. A reliability analysis was done to determine the characteristic degradation of LDs used in the FS-400M submarine optical telecommunications system.

Samples were Fabry-Perot buried type InGaAsP-LDs with a 1.3 μm wavelength. In actual applications, an automatic power control (APC) circuit was used to maintain a constant optical output power; i.e., degradation of the constant output power of LDs was maintained by the increase in driving current in the APC circuit. The failure criterion of the LD degradation was normally defined by a 50% increase in driving current. The accelerated life tests were conducted in conditions of 50ºC and 70ºC temperatures with an output power of 5 mW, and also at 10ºC and 5 mW, which was the actual operating condition. In total, 1000 samples were tested, with 100 samples used for the 10ºC test, 800 samples for the 50ºC test, and 100 samples for the 70ºC test. The rate of random failures was also estimated by testing the same 1000 samples [35].

4.2.1 Degradation mode of electrical characteristics

The degradation mode was a decrease in optical output power, and the degradation corresponded to an increase in the driving current in the APC circuit, which could be easily monitored. The typical degradation characteristics in the 70ºC test are shown in **Fig. 7**. The driving current increased with time as $\Delta I_d/I_o(t) \propto t^n$, where ΔI_d is the increase in driving current, I_o is the initial driving current, t is the test time, and n is a constant. The value of n varied with the samples in the range of 0.4–1.0. The lifetime, which is defined as the time to reach a 50% increase in driving current, was estimated by extrapolating the relation shown in Fig. 7 for each sample. A log-normal plot of cumulative failures versus lifetime for the 70ºC operating test is shown in **Fig. 8(a)**. This log-normal distribution was approximated by the median life of $\mu = 1.4 \times 10^5$ hours and the standard deviation of $\sigma = 0.99$, except for the long lifetime portion. Although the activation energy was estimated to be about 0.8 eV from the data at 50ºC and 70ºC, the lowest activation energy of 0.35 eV used in semiconductor devices was applied in order to obtain a more severe prediction in the actual lifetime estimation. The failure rate is given by the following equation for the log-normal distribution [36]:

$$\lambda(t) = \frac{\sqrt{2}\cdot\exp\left\{-\frac{1}{2\sigma^2}\cdot\left(1n(\frac{t}{\mu})^2\right)\right\}}{\sqrt{\pi}\cdot t\cdot\sigma\cdot\mathrm{erfc}\left(\frac{1}{\sqrt{2}\sigma}\cdot 1n(\frac{t}{\mu})\right)}. \tag{4}$$

The estimation of failure rate versus time (years) at 10ºC in the actual undersea environment is shown in **Fig. 8(b)**. From this plot, a failure rate of 220 FIT after 25 years was estimated, while 0.001 FIT after 25 years was estimated when 0.8-eV activation energy was applied.

4.2.2 Random failure mode

Long-term reliability testing for 19,000 hours at maximum was carried out for 1000 samples as described above. Except for optical output degradation, no catastrophic failures were observed in these tests. A failure rate lower than 13 FIT (CL=60%) was estimated from these tests of 7.3×10^7 device•hours at 10ºC, where 0.35-eV activation energy was used. Thus, the total rate of wear-out failure and random failure was estimated to be below 233 FIT, which was below the target value of 300 FIT at 10ºC operation and 25 years duration.

4.2.3 Screening

The operating test results showed a relatively good correlation between the increase in driving current at 1000 hours and the lifetime estimated using the procedure described above with a correlation coefficient of -0.62. This indicates that a failure rate one order of

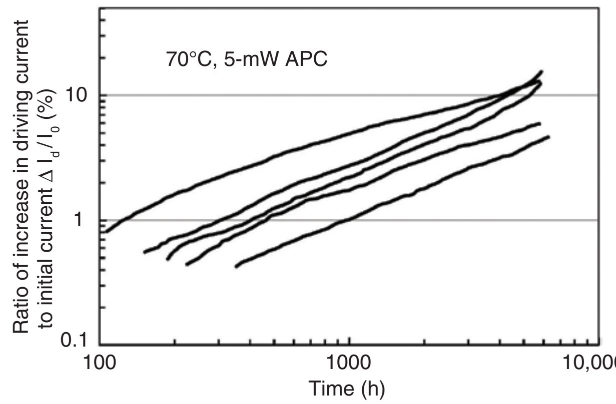
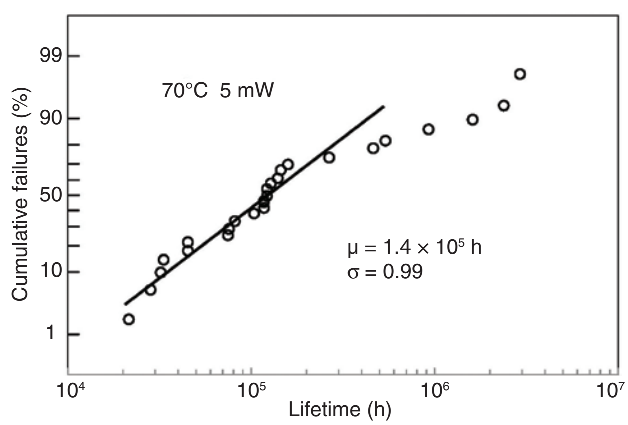
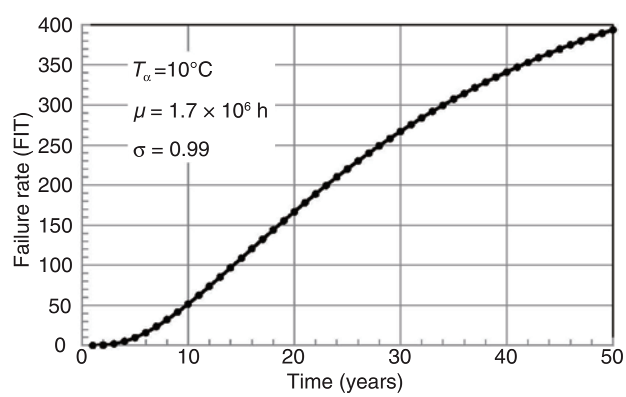


Fig. 7. Driving current degradation of LDs used in FS-400M system during high-temperature operating test.



(a) Log-normal failure distribution



(b) Failure rate vs. time (years)

Fig. 8. (a) Lifetime distribution of LDs derived from Fig. 7 and (b) failure rate prediction at 10°C (failure criterion is 50% increase in driving current).

magnitude lower than 300 FIT could be obtained by selecting LDs with a smaller increase in driving current after 1000 hours of aging. This procedure was implemented as the third step of screening, where the total screening consisted of three steps: 70ºC/150 mA for 100 h, 70ºC/8 mW with APC for 100 h, and 70ºC/5 mW with APC for 1000 h [35]. The first and second steps were used to avoid early failures and stabilize the electrical characteristics, and the third step was done to select a long-lifetime device. This screening method was called the *individual device assurance method* because the electrical characteristics of each LD at all stages of production and screening were acquired individually, and because it was expected that this method would make it possible to select LDs with better stability and longer lifetime. This individual device assurance method was also applied for the selection of bipolar transistors used in the submarine coaxial transmission system (CS-36M), in which transistors with high stability, i.e., degradation of h_{FE} of less than 3%, were required during the intended duration of service [37].

4.3 Future challenges

Recently, wear-out failure has become the most significant issue in the reliability of advanced LSIs because the process dimensions have been miniaturized to a size of about several tens of nanometers, and thin plastic encapsulation of high pin counts has been applied, which results in shorter lifetimes. For optical semiconductor devices, the time to a wear-out failure such as the degradation of optical output power has been the most important issue ever since the introduction of optical devices. For such devices, there is a need for more precise reliability analyses and accurate failure rate estimation for both catastrophic failure and degradation modes.

In line with the recent ITRS (International Technology Roadmap for Semiconductors) [38], it has been proposed that the failure rate for long-term reliability should be determined according to system scale. For example, 1000 FIT/chip is required for one chip/system, while 1 FIT/chip is needed for 1000 chips/system. However, 1 FIT/chip is not reasonable because it is very difficult or even impossible to ensure by reliability testing when considering the device•hours and testing cost. The future challenge in ensuring high reliability of advanced devices is to provide the most accurate reliability information based on the advanced failure physics approach to system designers, who will then be able to design reliable systems by introducing a redundant unit such as multiple devices used in parallel.

5. Summary

This article presented a review of the development of highly reliable semiconductor devices used in NTT telecommunications equipment over a period of several decades. The findings obtained and the future challenges are summarized as follows:

(1) Highly reliable semiconductor devices were developed through the failure physics approach and were assured through accelerated life testing and field failure examination to have a lifetime longer than 25 years and failure rates from 0.1 to150 FIT (depending on device type), which were adequate for application to NTT telecommunications equipment.

(2) Field failure data for semiconductor devices were collected for various types of operating equipment from the 1960s to 1980s. The data revealed that the failure rate decreased with time and tended to approach a constant value of less than about 100 FIT regardless of the device type or scale of integration.

(3) The failure rate prediction model, called the NTT model, was developed based on field failure data collected and used in equipment reliability design. Comparisons between the NTT model and other models used in those days, e.g., MIL-HDBK-217 and Bellcore TR-332, showed almost the same prediction results.

(4) Since wear-out failure modes vary with device type or technology, it is necessary to clarify failure modes and failure mechanisms for relevant devices by conducting accelerated life tests and failure analyses, and to predict accurate failure rates in actual-use conditions. The use of such accurate failure rate data will lead to the design of reliable equipment.

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Overview of MPLS-TP Standardization

Makoto Murakami and Yoshinori Koike

Abstract

The recent prevalence of IP (Internet protocol) based services has increased the demand for transport network technology that can efficiently accommodate both packet-based and circuit-based traffic while achieving almost the same operation and maintenance capabilities as traditional transport networks such as synchronous digital hierarchy or optical transport networks. Thus, ITU-T (International Telecommunication Union, Telecommunication Standardization Sector) has developed a packet transport technology called MPLS-TP (multiprotocol label switching-transport profile) in collaboration with IETF (Internet Engineering Task Force).

This article introduces the overview and history of MPLS-TP standardization.

1. Outline of MPLS-TP technology

Telecommunication carriers have employed transport technologies standardized by ITU-T (International Telecommunication Union, Telecommunication Standardization Sector) including synchronous digital hierarchy (SDH) or optical transport networks (OTNs) in their core networks that convey large capacity traffic over long distances. Recently, a packet-based technology that can accommodate packet-based data more efficiently than the current SDH or OTN has been demanded because packet-based traffic has become dominant in these networks as the number of Internet protocol (IP)-based services has increased. However, traditional packet network technologies have been problematic for telecommunication carriers to adopt in networks because of a lack of sufficient OAM (operation, administration, and maintenance) functions, fault localization, and fast switching in the event of a failure. Thus, a new packet-based transport network technology that has the same operation and maintenance capabilities as traditional transport network technology is required.

One of the significant features of packet transport network technology is that a path for packets between two network elements (NEs) is explicitly determined, and the connectivity of the path is periodically monitored to allow network operators to manage the path condition; this is called being *connection-oriented*. Other key functions should include protection switching, which promptly recovers a particular path when the route of the path has a fault; alarm transfer, which promptly transmits information about faults to other NEs; and traffic engineering, which enables operators to flexibly assign a bandwidth on a path.

MPLS (multiprotocol label switching) has been standardized in IETF to make it possible to explicitly determine the path route of IP packets by attaching a label to an IP packet and forwarding the packet by inspecting only the label instead of the IP address header itself, as shown in **Fig. 1**. MPLS, however, does not have sufficient maintenance operation functions as required in transport networks, and thus, some inherent features, including PHP (penultimate hop popping), ECMP (equal cost multi-path), and label merging could disrupt the management of a connection-oriented path between the two end points because of a lack of traceability. Moreover, MPLS technology makes it difficult to secure user data traffic in telecommunication carrier networks. This is because all the paths in MPLS are basically controlled by the control plane (part of the autonomous control) through a soft state procedure in which any fault that occurs during the exchange of control messages about a path causes a disconnection of the paths in the data plane even though they are not experiencing

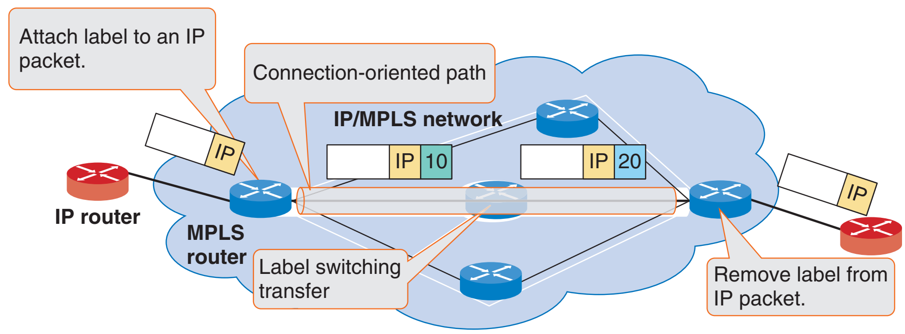

Fig. 1. Data forwarding in MPLS label switching.

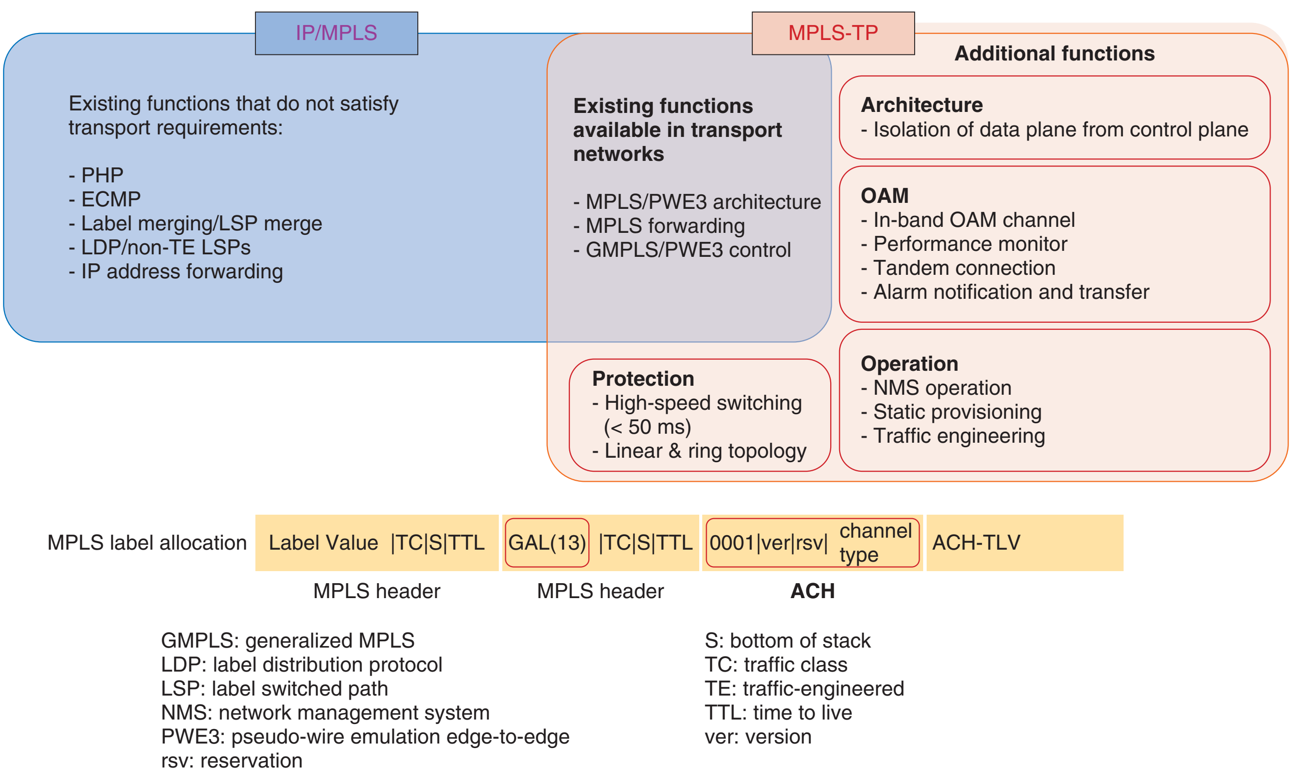

Fig. 2. Differences between IP/MPLS and MPLS-TP.

a failure.

In contrast, MPLS-TP emphasizes operation and maintenance capabilities by improving the data plane control mechanism, introducing static and central operations by the management plane, and implementing various OAM functions and high-speed protection switching, while some MPLS features that could disrupt transport network operation are disabled, as shown in **Fig. 2**. A GAL (generic associated channel label) is newly defined in the MPLS header stack for identifying the OAM packet, and each OAM function is differentiated by an ACH (associated channel). MPLS-TP also enables the paths in the data plane to be controlled by the management plane and allows network operators to manually and intentionally manage all the paths. Moreover, when a control plane is introduced, MPLS-TP employs an ASON (automatically switched optical network), described in

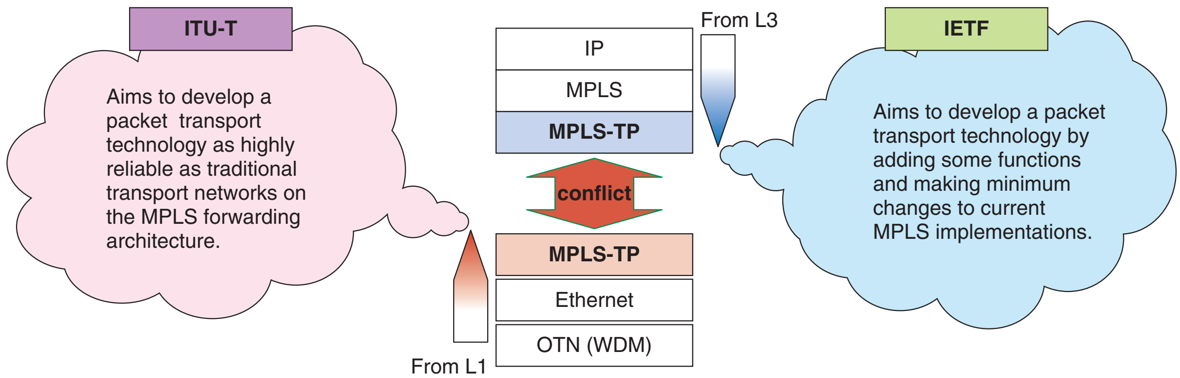

Fig. 3. Differences between ITU-T and IETF concepts.

ITU-T Recommendation G.8080, which isolates paths in the data plane from any failure in the control plane so that a fault in the control plane never affects the user data traffic. Thus, MPLS-TP can provide highly reliable network services, which is one of the most significant attributes required for transport networks.

As such, MPLS-TP was created as a merger of the transport network technology mainly standardized by ITU-T and MPLS technology mainly standardized by IETF. This merger has brought out some differences in the concepts of the two standardization organizations, as shown in **Fig. 3**. ITU-T's aim was to develop packet transport technology with the high reliability of traditional SDH or OTN transport networks by creating a new mechanism on the MPLS forwarding architecture irrespective of existing MPLS implementations. By contrast, IETF has persisted with the current MPLS implementations in their aim to develop an MPLS-TP technology with minimum changes. Therefore, the MPLS-TP specifications tended to deviate from the original concept and to be complicated due to the constraints from the existing MPLS implementations. Another point in the controversy is that IETF follows a process called *rough consensus*, in which there are no definite rules in the decision-making process in developing their standard RFCs (requests for comments), while ITU-T has clearly defined rules for approving Recommendations based on the unanimous agreement among member states as an affiliate in the United Nations. With these backgrounds, MPLS-TP standardization has created a major controversy between the organizations.

2. History of MPLS-TP standardization

2.1 2005 to 2010

ITU-T began standardizing T-MPLS (Transport-MPLS) in 2005. This standardization effort introduced OAM and protection functions to the MPLS data forwarding mechanism in order to satisfy packet transport network requirements and was based on ITU-T's experience in developing standards for transport network technologies. ITU-T approved the main T-MPLS recommendations in 2006 and 2007; these include G.8110.1 (Architecture of Transport MPLS (T-MPLS) layer network), G.8121 (Characteristics of Transport MPLS equipment functional blocks), G.8112 (Interfaces for the Transport MPLS (T-MPLS) hierarchy), and G.8131 (Linear protection switching for Transport MPLS (T-MPLS) networks.

However, IETF claimed that T-MPLS did not conform to the existing MPLS specifications when ITU-T had made progress on the LC (last call) of G.8114 (Operation & maintenance mechanism for T-MPLS layer networks) and G.8113 in January 2008. In anticipation of future collaboration with IETF on this topic, ITU-T stopped the LC of G.8114 and established a Joint Working Team (JWT) with IETF so they could cooperate on the packet transport network standardization in February 2008.

In December 2008, after T-MPLS was renamed as MPLS-TP, ITU-T and IETF agreed to publish some IETF RFCs based on T-MPLS Recommendations by June 2009, and they therefore developed several MPLS-TP RFCs, including those regarding the framework and requirements.

At the 74th IETF meeting in March 2009 ITU-T experts asserted their intention to employ the OAM tools defined in Y.1731 for MPLS-TP because they had already been proven to meet the requirements of the transport network and were sufficiently mature for deployment. At the 75th IETF meeting in July 2009, however, the IETF MPLS working group took a unilateral decision to give priority to backwards compatibility with existing IP/MPLS protocols and said that they would develop and support only one solution based on extensions of bidirectional forwarding detection (BFD) and label switched path (LSP)-ping, which they had been developing for IP/MPLS.

The ITU-T experts regarded the complex OAM tool sets developed for compatibility with the existing IP/MPLS implementations preferred by IETF as a deviation from the primary MPLS-TP concept. Therefore, at the SG (Study Group) 15 plenary in June 2010, ITU-T decided, with support from many member states, to develop an MPLS-TP OAM Recommendation, G.8113.1 and to proceed with the standardization of both types of OAM tools to satisfy the demands of ITU-T and IETF. ITU-T also requested IETF/IANA (Internet Assigned Numbers Authority) to assign a codepoint to differentiate these two OAM solutions. In August 2010, the ITU-T TSB (Telecommunication Standardization Bureau) director had a direct talk with the IETF chairman to settle the controversy, but they were unable to reach a resolution. The IETF chairman thus unilaterally declared that IETF would not accept any other OAM solutions than those developed by IETF.

2.2 2011

At the SG15 plenary in February 2011, the approval of two OAM Recommendations, G.8113.1 advocated by ITU-T and G.8113.2 by IETF, were discussed, but the progress of the meeting stalled due to an intense dispute. Finally, the ITU-T SG15 chairman decided to move G.8113.1 forward to the approval process based on the result of a vote among member states, although it was exceptional to take such action at an ITU-T SG meeting. Then, IETF issued a comment in a newsletter criticizing the decision made in ITU-T [1], and ITU-T also issued an article titled “THE FACTS” in the ITU-T newslog to reveal the history of MPLS-TP standardization [2]. Hence, the serious conflict between the two organizations came to light. Japanese telecom carriers and vendors had dedicated discussions in the TTC (Telecommunication Technology Committee) in Japan to decide the position of Japanese member states on the MPLS-TP standardization, because any final decisions must be made at the member state level in ITU-T.

In June 2011, a circular letter was issued by the ITU-T TSB asking member states to respond to an inquiry on proceeding with the traditional approval process for G.8113.1. With support from 33 member states including Japan and against opposition from 5 member states, it was decided that G.8113.1 would move forward for approval at the following SG15 plenary. Since a further dispute at the SG15 plenary was expected, Japan submitted a contribution to propose a compromise as a member state in a relatively neutral position after holding repeated discussions with the stakeholders including the ITU-T bureau, the SG15 chairman, and management members. However, G.8113.1 was finally disapproved at the SG15 plenary held in December 2011 due to opposition from 4 member states (the USA, Israel, the UK, and Finland) that was unable to be resolved despite the intense formal and informal discussions and negotiations conducted throughout the day and night. As a result, the SG15 Chairman declared a deadlock in the approval of Recommendation G.8113.1, and the decision was committed to the WTSA (World Telecommunication Standardization Assembly)-12 held in November 2012. The ITU-T TSB director issued a comment on the ITU-T newslog in which he gave a detailed account of the event and expressed his great appreciation to the member state of Japan for its efforts to settle the situation [3].

2.3 2012

NTT participated in further discussions on MPLS-TP standardization in CJK (China Japan Korea) meetings held with Chinese and Korean experts who shared the same opinion and submitted a common proposal document by the three countries to move G.8113.1 forward as well as a series of related MPLS-TP Recommendations at the APT (Asia-Pacific Telecommunity) meeting held in August 2012. The proposal was supported at the meeting and approved to be an APT common proposal to WTSA-12.

At the SG15 meeting in September 2012, a degree of progress was made, as some MPLS-TP Recommendations were approved or consented to. A proposal from Canada that G.8113.2 should also be approved in WTSA-12 was agreed, and the approval of both draft Recommendations became a matter for simultaneous discussion at WTSA-12.

At the beginning of the WTSA-12 meeting held in November 2012, these two draft Recommendations were simultaneously submitted for approval and were

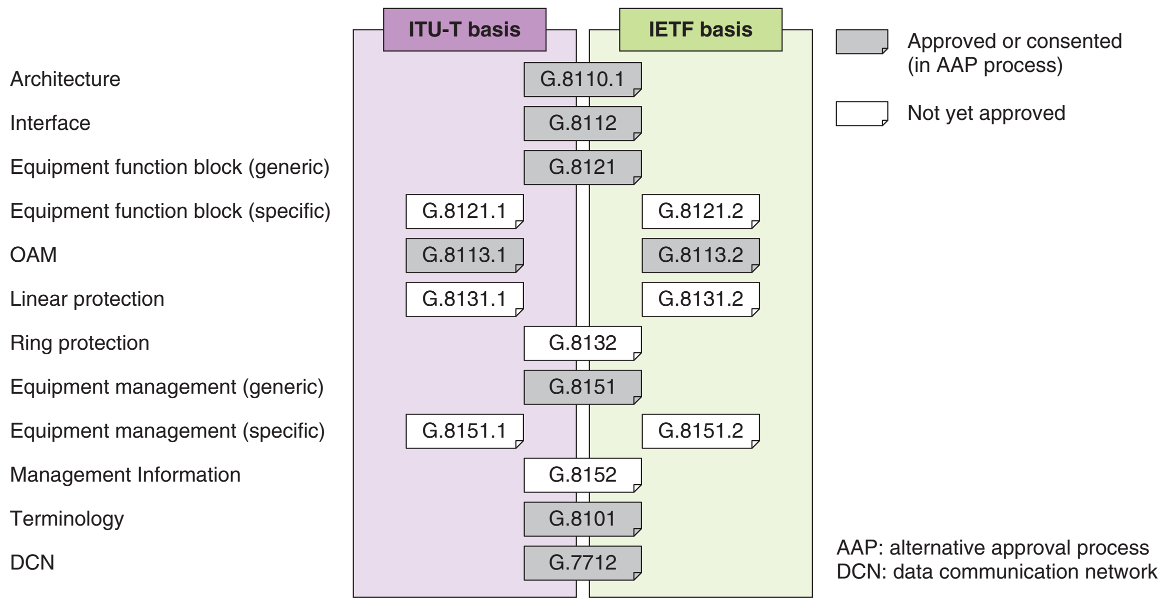

Fig. 4. Structure of MPLS-TP Recommendations and current status.

successfully approved with no opposition. Finally, IANA formally notified ITU-T that it would assign an ACH codepoint for identification of an OAM packet based on G.8113.1, and the two OAM solutions respectively preferred by ITU-T and IETF finally came to be international standards.

3. Future activities

The controversy of MPLS-TP standardization that continued for more than seven years since the standardization of T-MPLS was successfully resolved in part because of the work done to gain approval of the OAM Recommendations at WTSA-12. The status of international standardization of MPLS-TP is now as shown in **Fig. 4**. Other draft Recommendations including equipment functional blocks and protection schemes will be further discussed and approved. Additionally, some other significant issues related to MPLS-TP including a point-to-multipoint connection scheme and layer integration are expected to be standardized in the future.

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NTT Establishes NTT Innovation Institute, Inc. as its New R&D Center in North America

On April 1, 2013, NTT Innovation Institute, Inc. (NTT I^3) was established in North America as a new research and development (R&D) center in line with NTT's "Toward the Next Stage" business strategy, which was announced on November 8, 2012. NTT I^3 was established as a venue to speedily develop world-class security cloud technology and from which to supply the related services to the North American market, where the competition is the most intense.

At NTT I^3, the aim is to develop services by combining the technical capabilities of the Institute with know-how (IP: intellectual property) and to promote open innovation in cloud services, which are changing very rapidly and require swift deployment in the market.

Specifically, the service development in North America will be promoted through *security* and *cloud* research and development. Additionally, the successful deployment of "global cloud services", a growth driver of the NTT Group, will be strengthened.

In addition, the cloud services refined in North America can be deployed in Japan and in newly emerging markets as modularized IP.

At a kick-off meeting held at NTT I^3 on April 5, Hiromichi Shinohara, NTT Executive Vice President and Director of the Research and Development Planning Department, offered greetings saying, "We expect NTT I^3 to play a major role in accelerating NTT's global business."

The newly appointed Chief Operating Officer (COO) of NTT I^3, Eiji Kuwana (formerly, General Manager of NTT Secure Platform Laboratories), went on to say, "We must aggressively take up the challenge of making the most of new business opportunities."

These words underscore the resolve of the entire NTT I^3 staff to become a core base in strengthening the global R&D of the NTT Group.

Greetings from COO Eiji Kuwana