Recent QoS Standardization Activities in ITU-T SG12

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Abstract

To provide users with telecommunication services of satisfactory quality, it is important to develop methods for evaluating, designing, and managing the quality of service (QoS). ITU-T SG12 (International Telecommunication Union, Telecommunication Standardization Sector, Study Group 12) is responsible for standardization work on the end-to-end transmission performance of terminals and networks. This article introduces recent standardization activities for speech and video quality evaluation and management and for network performance parameters and objectives.

1. ITU-T SG12

SG12 is the lead study group in ITU-T (International Telecommunication Union, Telecommunication Standardization Sector) on network performance and quality of service (QoS). In January 2005, SG12 was restructured by incorporating some questions on network performance that had previously been studied in Study Group 13. Standardization work on speech and video quality evaluation is mainly being carried out in ITU. On the other hand, standardization of network performance parameters and objectives is being carried out by various standardization organizations, but their efforts are being coordinated with SG12's.

2. Speech and video quality evaluation and management

To provide better service quality to users, we should evaluate the quality before a service starts and manage the quality while services are being provided. Here, we describe recent standardization activities on speech and video quality evaluation and management methods.

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2.1 Quality evaluation method

The prime criterion for the quality of speech and video communication services is subjective quality, i.e., the user's perception of service quality. Subjective quality is evaluated in psychological experiments, which evaluate the user's perception of transmitted speech and video. The opinion test is widely used for subjective quality evaluation, and results derived from opinion tests are called the mean opinion score (MOS).

In subjective quality evaluation, it is necessary to prepare exclusive test facilities such as acoustically shielded chambers to ensure constant characteristics so that the quality evaluation test is conducted stably and reproducibly. Moreover, results are needed for many subjects to accurately represent various conditions of telecommunications equipment and systems. This makes subjective quality evaluation time-consuming and expensive. Therefore, a method that estimates subjective quality by measuring physical characteristics of transmitted speech and video signals is desirable. This is called objective quality evaluation.

SG12 standardized Recommendation P.862 (PESQ: Perceptual Evaluation Speech Quality) as an objective speech quality evaluation method based on measuring the physical characteristics of transmitted speech signals. At the last SG12 meeting in October 2005, the following two techniques concerning PESQ went to AAP (alternative approval process).

1. Wideband PESQ (P.862.2): Recommendation P.862 has not been able to evaluate wideband (7 kHz) speech quality because it was targeted at telephone-band (300 Hz to 3.4 kHz) speech signals. SG12 considered wideband extension of PESQ and standardized it as Recommendation P.862.2 “Wideband PESQ”.
2. Application guide for P.862 (P.862.3): PESQ quantifies a degradation by means of a comparison between the original speech signal and a degraded speech signal. When one measures quality using the PESQ algorithm applied to hardware devices, it is known that measurement noise introduced at the digital-to-analog (D/A) interface of the products influences the resultant PESQ score. This recommendation points out such practical problems in applying P.862, P.862.1, and P.862.2 and provides solutions to them for PESQ users.

PESQ and Wideband PESQ use electrical interfaces with the devices being tested and assume they have the normal electroacoustic characteristics of handsets. SG12 has been considering a new objective speech quality evaluation model using acoustic interfaces. One of the application scenarios of this technology is evaluation of hands-free applications with microphones and loudspeakers. Currently, the scope and requirements of this recommendation are being studied.

The objective speech quality evaluation mentioned above evaluates the “listening” quality. Besides this, the speech communication quality depends on conversational factors such as the delay time and talker echo. An overall conversational quality evaluation method estimates the combined effects of these factors on subjective conversational quality, which is called the “Opinion Model”.

SG12 has already standardized Recommendation G.107, the “E-model”, as one of the opinion models, which is useful as a transmission planning tool. Currently, a wideband (7-kHz) extension of the E-model is being studied. For example, application of the equipment impairment factor in the E-model to the wideband speech codec has been discussed. After this, SG12 needs to determine codec-specific values for ITU standard wideband speech codecs. Since the E-model has been widely used as a quality evaluation tool for IP telephony in Japan, the standardization work on its extension to include wideband speech quality evaluation should be closely monitored.

The E-model is an opinion model for speech communication services, but SG12 has started studying an opinion model for videophone applications. At the last SG12 meeting in October 2005, agreement was reached about the proposed framework of the recommendation which has a speech quality estimation function, video quality estimation function, delay and synchronization quality estimation function, and multimedia quality integration function.

2.2 Quality management

Since IP networks do not necessarily guarantee the transmission performance, it is important to monitor and manage the quality during service. SG12 has been studying a method that estimates subjective listening quality based on information obtained from realtime protocol (RTP) headers and RTP control protocol (RTCP) packets so as to manage the quality of IP telephony during service. This recommendation specifies the requirements for the performance of a model. Since the draft recommendation is almost stable, it will be submitted for SG12 consent at the next meeting.

3. Network performance

Studies of network performance metrics and objectives studied in SG13 as well as in FG-NGN (focus group on next generation networks) have been transferred to SG12. IP network performance is being discussed separately for connection setup/release quality parameters and IP packet transfer quality parameters. Here, we introduce recent standardization activities concerning IP packet transfer quality parameters.

3.1 Network performance objectives for IP-based services (Y.1541)

This recommendation defines objectives for IP network performance parameters. Network performance objectives are specified between a pair of UNIs (user network interfaces). Currently, six network QoS classes are defined. Each class specifies maximum values for IP packet transfer delay (IPTD), IP packet delay variation (IPDV), IP packet loss ratio (IPLR), and IP packet error ratio (IPER), which are defined in Recommendation Y.1540.

With services over IP networks diversifying, new network QoS classes have been proposed. For example, provisional classes suitable for PSTN/ISDN (public switched telephone network and integrated services digital network) emulation and simulation and high-speed transport protocol have been proposed. These classes feature more stringent loss/error requirements than traditional classes and include a new parameter: IP reordered ratio (IPRR).

3.2 Framework for achieving end-to-end performance objectives (G.fepo)

Draft recommendation G.fepo aims to specify a framework for achieving the end-to-end performance objectives specified in Recommendations Y.1540 and Y.1541.

This draft text describes two main approaches for achieving network performance between UNIs. These approaches have been extensively discussed.

1. Top-down approach
   This approach divides network performance objectives among network providers. Each provider plans, controls, and manages its performance objectives within its own networks to achieve the overall objectives.
2. Bottom-up approach
   This approach accumulates providers' offered performance objectives along a path between UNIs. If the path does not meet the requested objectives, either an alternative path or a different service class is negotiated.

An IP network is expected to have various network topology and quality control mechanisms that are different from those in the conventional telephone network. Comparison of specific methods and a study of the application area need to be considered.

The NGN aims to achieve high-quality speech and video communication, high-speed data transfer, and so on. Since recommendations and draft recommendation described here are the basis of frameworks for network performance planning and management in the NGN, it is expected that standardization work on the network performance will be conducted in cooperation with NGN discussions in other SGs.

4. Future of these activities

In the standardization of speech and video quality evaluation methods, efforts to handle wideband speech communications and multimedia applications have become active. A quality evaluation method that can evaluate the user's perception of services properly is essential to provide the customer with such new services with appropriate quality. On the other hand, in the work on network performance, interconnection between network providers will be an important issue. Study items such as how to measure network performance and how to achieve end-to-end network performance objectives in an interconnection network environment should be considered thoroughly. From these viewpoints, we need to monitor standardization activities in SG12 hereafter.

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