SONY

VIDEO COMMUNICATION SYSTEM-TECHNICAL DOCUMENTATION

Intelligent QoS

PCS-HG90	Ver. 2.0 or later
PCS-XG55S	All
PCS-XG80S	All

IPELA™

Introduction

On the current "best-effort" Internet, it is said that the packet loss ratio (PLR) can reach up to several percent. Packet loss causes video quality degradation, frame freeze, and audio glitches in visual communication sessions over the Internet typified by videoconferencing. QoS (Quality of Service) features are crucial in overcoming these problems and realizing high-quality visual communication.
Several QoS features have been designed and implemented to visual communication systems. These features can be classified into two main technologies: loss recovery, which recovers lost packets between sender and receiver through packet retransmission and packet-level error correction codes, and error concealment, which conceals or alleviates the negative impact of lost packets on decoded images, by showing previous error-free images for example.
Sony has always focused on maintaining real-time communication through loss recovery technology, as this is a fundamental solution for realizing high-quality visual communication. Sony has already developed Real-time ARQ (Automatic Repeat reQuest), FEC (Forward Error Correction), and ARC (Adaptive Rate Control) functions into existing PCS series which contribute to high-quality SD visual communication.
However, the Sony HD visual communication system has to handle HD video transmission with a large amount of data flowing of up to 10 Mb/s*1 over heterogeneous and best-effort Internet. QoS features for HD visual communication need higher performance than that of SD visual communication. For example, HD video with a larger amount of packets needs more network bandwidth resource and suffers from more lost packets in one video frame which causes noticeable artifacts than SD video in the same network condition. It is, therefore, expected that less than 10^{-5} packet loss after recovery (specified by ITU-T Y.1541) is necessary for HD video. In an effort to solve these problems, Sony has innovated intelligent QoS features.
Three major improvements have been made to the existing QoS features in PCS series handling SD video. First, FEC is performed with a variable number of parity packets in large FEC blocks, which corrects a large number of lost packets in an FEC block and realizes effective bandwidth utilization. Second, ARC quickly calculates a stable transmission rate based on the accurate measurement of network condition. Finally, Integrated QoS Manager intelligently allocates the amount of ARQ, FEC, and video packets based on Round-Trip Time (RTT) delay, PLR, and available bandwidth measured by ARC. This manager optimizes QoS recovery performance and available bandwidth utilization for HD video.
All QoS features and Auto Bandwidth Detection of the PCS-HG90/XG55S/XG80S Sony visual communication system are described in the following section. Then we will look at the intelligent QoS allocation mechanism of the Integrated QoS Manager.

*1: Up to 4Mbp/s for PCS-XG55S

QoS Features

This section describes all QoS features of the PCS-HG90/XG55S/XG80S, which include end-to-end QoS supported between terminals (Real-time ARQ, Adaptive FEC, and ARC) and network-level QoS supported by the networks such as Diffserv (Differentiated services).

Real-time ARQ (Automatic Repeat reQuest)

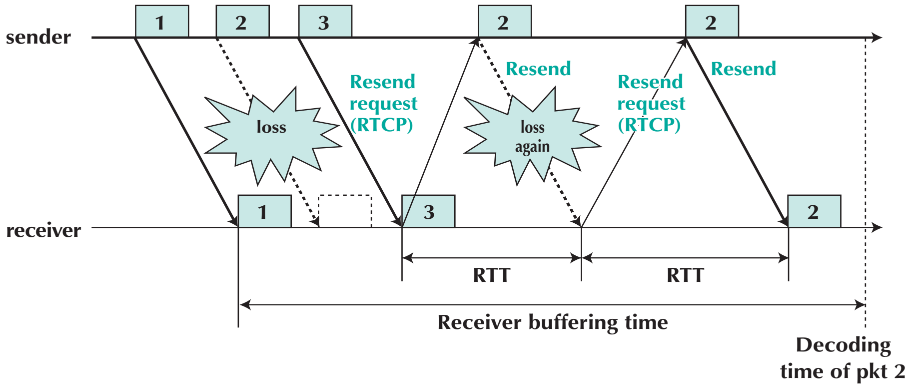
ARQ (Automatic Repeat reQuest) is a lost packet recovery technology where a sender retransmits the lost packets detected and requested by a receiver. Real-time ARQ is enhanced ARQ for real-time communication that takes into account both the network condition and interactive communication delays.

Figure 1 shows how Real-time ARQ works. All packets are given a sequence number and timestamp by RTP (Real-time Transport Protocol), which designates audio/video decoding timing. As a basic ARQ mechanism, the receiver detects the loss of RTP packets by monitoring sequence numbers that do not appear (e.g., packet No.2, in Figure 1) and sends a "retransmission request" with the lost sequence numbers in RTCP (RTP Control Protocol) format to the sender. The sender has a sending buffer to store the packets sent, from which the packet requested in the retransmission request is retransmitted. The receiver reorders all received packets, including the retransmission packet, in a receiving buffer and moves them to the decoder before the decoding time.

Real-time ARQ mechanism, in addition to ARQ, can optimize ARQ performance while maintaining real-time constraints for interactive communication. When RTT has passed after a retransmission request is sent, the receiver can receive the requested packet as shown in Figure 1. Whereas, the requested packet needs to arrive at the receiver before its decoding time. Considering these two conditions, Real-time ARQ measures RTT between the terminals, and judges whether the retransmission packet of the Nth retransmission request (N=1,2,...) would arrive by its decoding time or not. In this way, Real-time ARQ prevents wasteful retransmission and enhances ARQ's recovery performance.

In general, Real-time ARQ can realize good recovery performance on short RTT networks because the retransmission can be attempted several times until the decoding time. On long RTT networks, however, Real-time ARQ may not satisfy the desired performance or may not even work (for example, when RTT = 150ms and buffering delay is 100ms, no retransmission packet arrives by its decoding time). In other words, Real-time ARQ can shorten the buffering delay before the decoding time at the receiver up to the measured RTT value, while satisfying the desired recovery performance related with PLR.

Fig.1: Real-time ARQ retransmission diagram
(In case of Retransmission request N=2)



Adaptive FEC (Forward Error Correction)

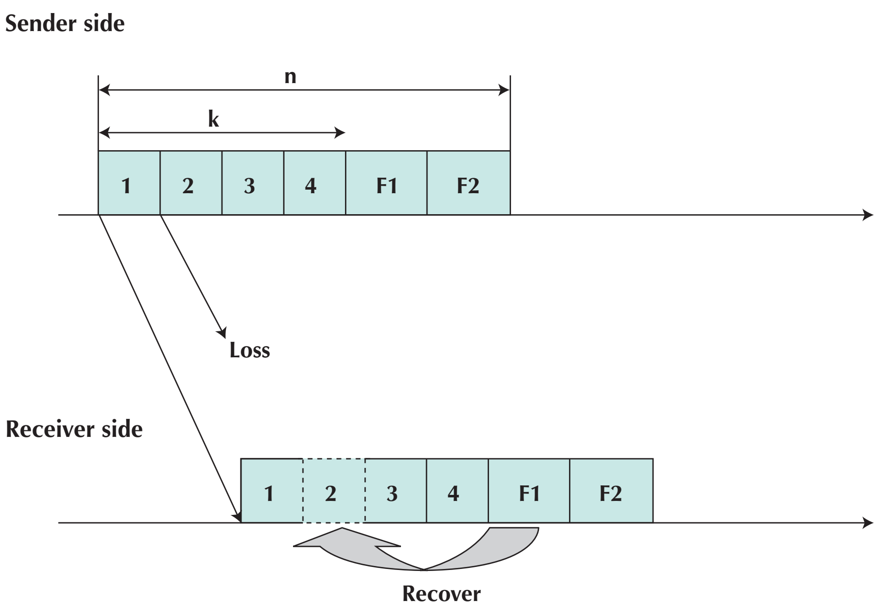
FEC (Forward Error Correction) is another loss recovery technology using parity data. The sender generates parity packets from original data packets and the receiver recovers the lost packets from the original packets and parity packets that have been already received.

The PCS-HG90/XG55S/XG80S and the PCS-G series apply Reed-Solomon (RS) code for FEC codec, RS (*n, k*) means that the sender encodes *n-k* parity packets from *k* original data packets. A unit of these *n* packets, including original data packets and parity packets, is called an FEC block. Figure 2 shows an example of the FEC recovery procedure of RS (*6, 4*). The first four packets, numbered 1 to 4, are original data packets, and the latter two packets, F1 and F2, are the parity packets, which altogether form one FEC block. In this case, the receiver can recover no more than 2 (= number of parity packets) lost packets in any lost pattern. In Figure 2, the lost packet #2 can be recovered from the other 4 packets in the same FEC block (ex. #1, #3, #4, and parity packet F1).

In general, the recovery performance of FEC is not affected by RTT. Therefore, FEC is more effective and suitable than ARQ on long RTT networks. However, FEC needs a larger amount of packets for recovery than ARQ.

The Adaptive FEC function of the Sony PCS-HG90/XG55S/XG80S improves loss recovery performance of HD video by accommodating larger FEC block sizes than the PCS series handling SD video, which can tolerate more lost packets. Moreover, FEC in the PCS-HG90/XG55S/XG80S can adjust the ratio of parity packets in the FEC block according to network conditions such as RTT and PLR. It reduces parity packets and yet could satisfy the expected recovery performance, which improves image quality. This ratio is specified by the Integrated QoS Manager, which is described in the next section.

Fig.2: FEC recovery diagram



ARC (Adaptive Rate Control)

Adaptive Rate Control is a mechanism to calculate the optimum transmission rate of the video stream according to the network condition. In general, ARC increases the rate additively to seek the available network bandwidth and slows down the rate if network congestion occurs. The existing PCS series which handle SD video adopt the TCP-friendly rate control (TFRC) algorithm, which controls the transmission rate by considering RTT and PLR of the Internet environment.
For the PCS-HG90/XG55S/XG80S, ARC has to support HD video streams at a high transmission rate, even in trans-ocean communication with long RTT (around 100ms). Unfortunately, the existing TFRC mechanism seems to underestimate the transmission rate and its convergence is slow and unstable.
To solve this problem, the ARC function on the PCS-HG90/XG55S/XG80S improves the rate calculation function and network measurement function of TFRC.

Network level QoS

The PCS series including the PCS-XG80S can enter the values of IP Precedence, Type of Service, and Differentiated Services. The TOS (Type of Service) field in the IP header is used for either defining IP Precedence and Type of Service or DSCP (Differentiated Services Code Point) bits of Differentiated Services. The field can be used for both services if the network administrator so desires. The TOS field structures are defined as follows:

Fig.3: Network level QoS

● **IP Precedence and Type of Service**

0	1	2	3	4	5	6	7
Precedence			Delay	Throughput	Reliability	Minimum Cost	CU

CU: Currently Unused

● **Diffserv (Differentiated Services)**

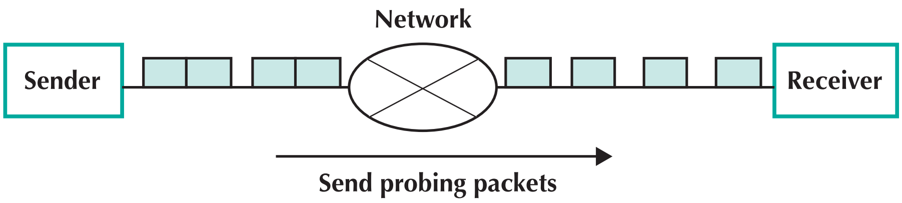
0	1	2	3	4	5	6	7
DS5	DS4	DS3	DS2	DS1	DS0	CU	CU

CU: Currently Unused

Auto Bandwidth Detection

Auto Bandwidth Detection is a technology that estimates the network bandwidth between terminals. This mechanism is useful for all QoS features.
Figure 4 shows how Auto Bandwidth Detection works. The basic mechanism is as follows.
A sender sends probing packets before starting visual communication. The interval of the packet extends when the packets passes the route of narrowband that is the bottleneck in the network. A receiver measures the packet reception interval and estimates the bandwidth in the route. By using this value for QoS parameter setting, you can start a communication with the optimized QoS.

Fig.4: Auto Bandwidth Detection Mechanism



Intelligent QoS Allocation

Intelligent QoS allocation is the network bandwidth allocation mechanism that controls the amount of original video packets and recovery packets generated by the ARQ (retransmission) and FEC (parity) functions. As described in the previous section, all QoS features have their specific network conditions that enhance their loss recovery performance. As the ratio of the recovery packets is increased, the loss recovery capability also improves. However, video quality will deteriorate due to the smaller ratio of original video packets. Therefore, this mechanism should seek a trade-off allocation point between recovery capability and video quality under the measured network conditions in order to maximize video quality.

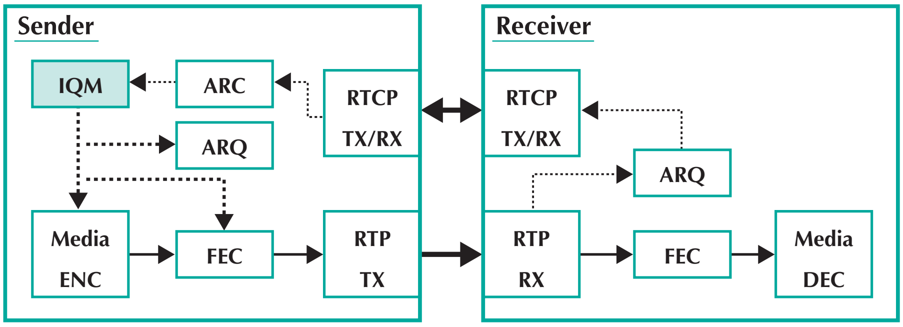
Integrated QoS Manager located at the sender (IQM in Figure 5) performs this intelligent QoS allocation. It monitors the network conditions (RTT, PLR, available bandwidth, etc.) through the ARC, ARQ, and FEC functions. The total transmission rate for both recovery packets and original video packets is then reported from ARC and is allocated to that for ARQ retransmission packets, FEC parity packets, and original video packets. The transmission rate for the retransmission packets and loss recovery capability of ARQ are estimated from the RTT and PLR. The transmission rate for parity packets of FEC is calculated from the evaluated PLR after ARQ recovery, RTT, and the total transmission rate. Then the transmission rate for original video packets is calculated from the transmission rate of the retransmission packets (ARQ) and parity packets (FEC).

Figure 6 illustrates an example of intelligent QoS allocation. On short RTT networks (i.e., communication within a country), a large amount of transmission rate is allocated to the ARQ function. Whereas on long RTT networks (i.e., communication between trans-ocean countries), a large amount of transmission rate is allocated to FEC parity data, since ARQ recovery does not work as effectively. As the PLR increases, more of the transmission rate is allocated to recovery packets and less is allocated to the original video packets.

Thus, intelligent control ensures above a certain quality level regardless of the RTT length and provides high QoE (Quality of Experience) to the users in the HD visual communications.

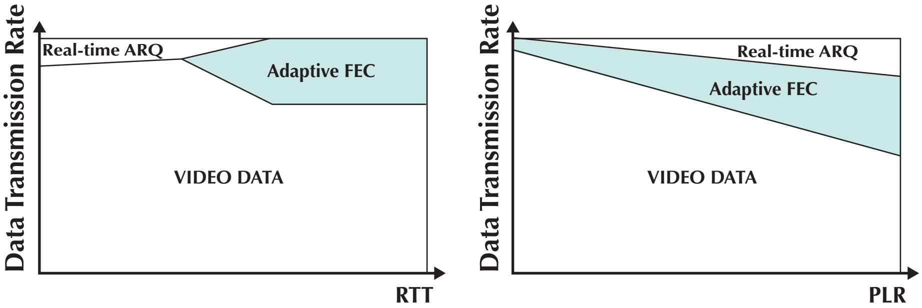
For audio transmission, audio packets are transmitted twice rather than FEC – this provides a redundant audio source to use should the original packet fail. This is possible because audio packet is much smaller in size than video.

Fig.5: QoS module diagram



TX: Transmission, RX: Reception

Fig.6: Example for rate allocation table



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