

SONY

VIDEO COMMUNICATION SYSTEM-TECHNICAL DOCUMENTATION

Intelligent QoS

IPELA

PCS-HG90 Ver. 1.0

Introduction

On the current “best-effort” Internet, it is said that the average packet loss ratio (PLR) is about 2%. Packet loss causes video quality degradation, frame freeze, and audio glitches in visual communication sessions typified by videoconferencing. QoS (Quality of Service) features are crucial in overcoming these problems and realizing high-quality visual communication over the Internet.

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) features into existing PCS series SD visual communication systems, which contribute to high-quality SD visual communication.

However, the Sony PCS-HG90 visual communication system has to handle HD video transmission with a large amount of data flowing at about 2-8 Mb/s over heterogeneous and best-effort Internet. QoS features for HD visual communication need higher performance than 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 than SD video in the same network condition. It is, therefore, expected that 10^{-5} packet loss after recovery is necessary for HD visual communication. In an effort to solve these problems, Sony has innovated intelligent QoS features adaptive to HD visual communication.

Three major improvements have been made to the existing QoS features in PCS series SD visual communication systems, 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 with fine network measurement. 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 of the PCS-HG90 are described in the following section. Then we will look at the intelligent QoS allocation mechanism of the Integrated QoS Manager.

QoS Features

This section describes all QoS features of the PCS-HG90, which include terminal-supported technologies (Real-time ARQ™, Adaptive FEC, and ARC) and network-supported technologies such as Diffserv (Differentiated service).

Real-time ARQ™ (Automatic Repeat reQuest)

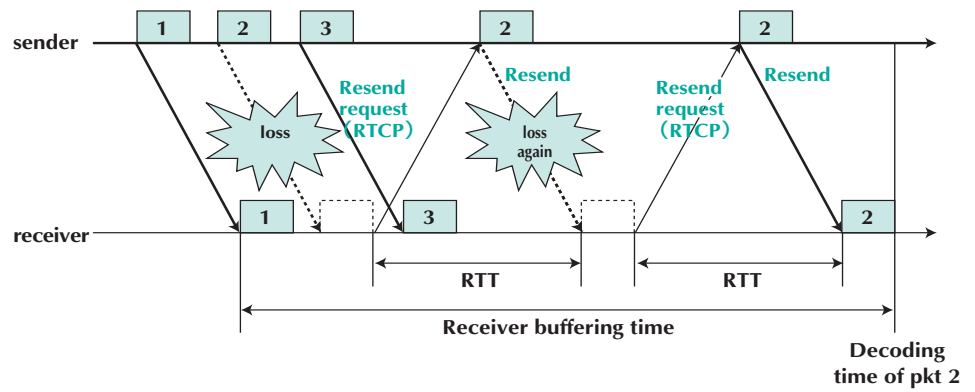
ARQ (Automatic Repeat reQuest) is a lost packet recovery technology where a sender retransmits the lost packets detected and requested by a receiver. Real-time ARQ™ is enhanced ARQ for real-time communication that takes into account both the network condition and interactive communication delays.

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 playing 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 ready for playing 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 playing 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 playing 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 sender and receiver can communicate retransmission packets several times till playing 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 playing time). From the other viewpoint, Real-time ARQ™ can minimize the buffering delay to playing time at the receiver for the measured RTT value, while satisfying the desired recovery performance related with PLR.

Fig.1: Real-time ARQ™ retransmission diagram



Adaptive FEC (Forward Error Correction)

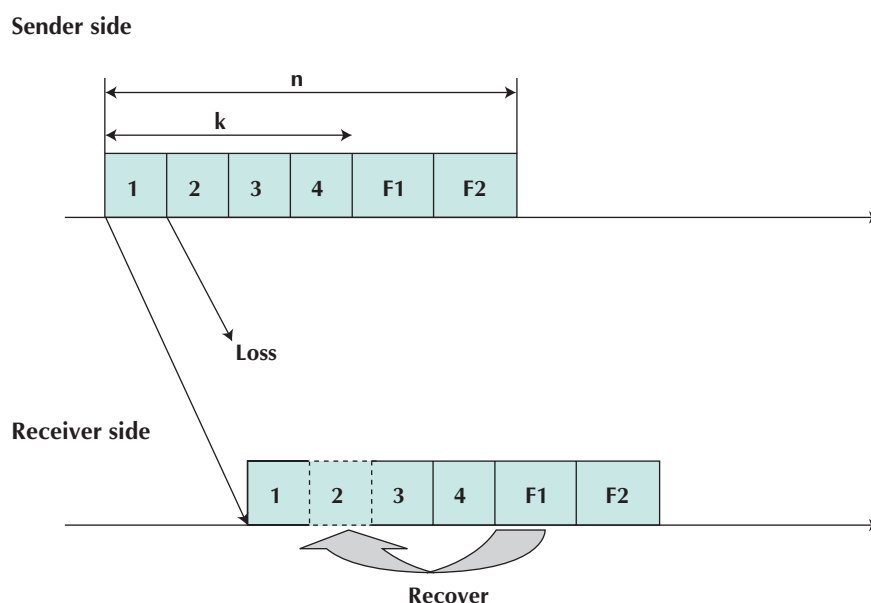
FEC (Forward Error Correction) is another loss recovery technology using parity data. The sender generates parity packets from original data packets and the receiver recovers the lost packets from the original packets and parity packets that have been already received.

The Sony PCS series including the PCS-HG90 applies 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 new Adaptive FEC feature of the Sony PCS-HG90 improves loss recovery performance of HD video by accommodating larger FEC block sizes than the PCS series SD visual communication systems, which can tolerate more lost packets. Moreover, FEC in the PCS-HG90 can adjust the ratio of parity packets in the FEC block according to network conditions such as RTT and PLR. It reduces parity packets in satisfying expected recovery performance. 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 SD visual communication systems adopt the TCP-friendly rate control (TFRC) algorithm, which controls the transmission rate by considering RTT and PLR in the Internet environment.

For the Sony PCS-HG90, ARC has to support HD video streams at a transmission rate of over 2 Mb/s, 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 feature on the PCS-HG90 improves the rate calculation function and network measurement function of TFRC.

Network level QoS

The PCS series including the PCS-HG90 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

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) features. As described in the previous section, all QoS features have their specific network conditions that enhance their loss recovery performance. As the ratio of recovery packets is increased, the loss recovery capability 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 in the measured network condition in order to maximize video quality.

Integrated QoS Manager located at the sender (IQM in Figure 4) performs this intelligent QoS allocation. It monitors the network condition (RTT, PLR, available bandwidth, etc.) through the ARC, ARQ, and FEC features. 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 5 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.

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.4: QoS module diagram

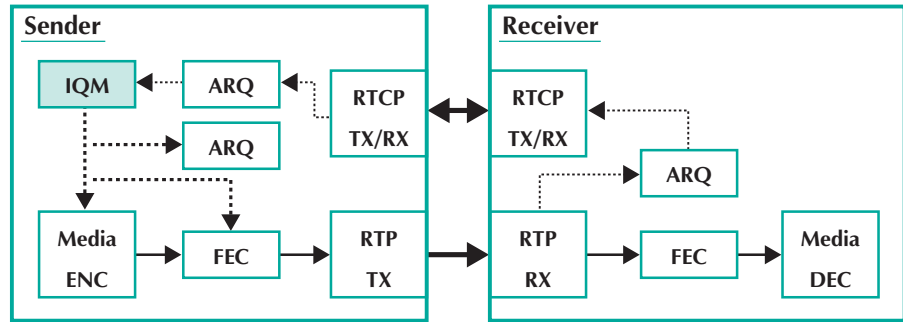
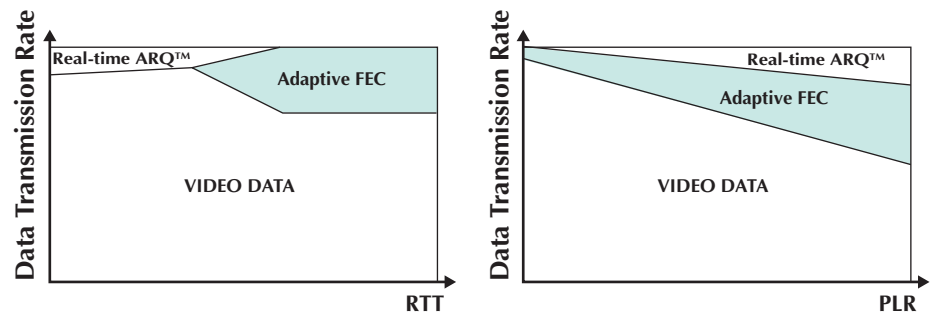


Fig.5: Example for rate allocation table



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