SONY

VIDEO COMMUNICATION SYSTEM-TECHNICAL DOCUMENTATION

QoS

PCS-1/1P	Ver2.00 or later
PCS-11/11P	Ver2.00 or later
PCS-G50/G50P	All
PCS-G70/G70P	All
PCS-TL30	All
PCS-TL50	All

IPELA

Introduction

Currently, an estimated 5% of data packets sent over the Internet are lost. In a videoconferencing session, packet loss causes video frame freeze, error propagation, and audio glitches. For this reason, QoS (Quality of Service) and packet-loss robustness features are very important for videoconferences conducted over "best-effort" IP networks.

The main objective of QoS features used in conventional videoconferencing endpoints is to "conceal" the error. For example, certain technology is used to shuffle image data for transmission so that packet loss does not destroy wide areas of the video image. Other technology is used to increase "intra" macroblocks that do not require previous frame information. This prevents the packet loss error from propagating over a long period, but results in a lower frame rate as the increased amount of intra macroblocks requires more bits to carry the video frames.

Sony has developed new QoS features using a different approach: "recovering" the error rather than "concealing" it, and thereby maintaining the quality of real-time communication. With the QoS features of the Sony PCS terminals (PCS-1/1P/11/11P/G50/G50P/G70/G70P /TL30/TL50), users need not be concerned about bandwidth and network quality. Using the following two functions, the PCS terminals can automatically adjust their bandwidth, buffering size, and algorithm to maintain high-quality conferencing:

- Real-time ARQ (Automatic Repeat reQuest)
- ARC (Adaptive Rate Control)

Furthermore, the PCS-G50/G50P/G70/G70P terminals support FEC (Forward Error Correction) – a function that directly recovers the lost data by using parity packets attached to the data.

About this whitepaper

This whitepaper is split into three sections. The first describes the above-mentioned functions in more detail. The second outlines how users can select the most appropriate function for the condition of their network. The final section introduces additional functions integrated on the PCS terminals for achieving high-quality video and audio service over an IP network, such as network-level QoS, IP Precedence, Type of Service, and Differentiated Services.

Real-time ARQ (Automatic Repeat reQuest)

Real-time ARQ is a mechanism that automatically resends lost packets and reorders received packets, allowing a PCS terminal to recover almost completely from a packet loss rate of 10%, while maintaining the minimum latency adaptively determined according to the network environment.

Figure 1.1 shows how Real-time ARQ works. If an RTP packet of a video/audio bit stream is lost, this is detected on the receiver side and a "resend request" is sent to the transmission side using an RTCP (RTP Control Protocol) packet. On the transmission side, the transmitted packet is held, in preparation for resending according to the resend request.

By using an RTCP packet, the Round-Trip Time (RTT), or network latency, can be measured. If the RTT is large, it would be a waste of network traffic to send the "resend request" knowing that the resent packet would not arrive in time for decoding. Sony's Real-time ARQ is capable of determining whether or not the resending packet would arrive in time, and adaptively selects the optimum algorithm according to the RTT and packet loss rate.

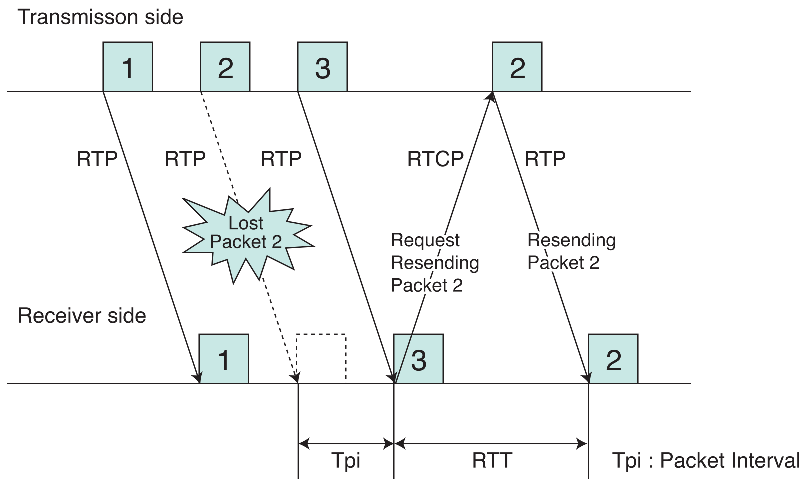


Figure 1.1 Real-time ARQ resending diagram

Minimum latency with optimum packet buffering

In general, the resending of packets requires a buffer at the receiving end for the reordering of packets, which increases the system latency. Sony PCS terminals have a variable-length reordering buffer for rearranging a packet in order of the sequence number of RTP headers. The size of the reordering buffer is optimized according to the measured network RTT and PLR (Packet Loss Rate). In an environment without packet loss, the reordering buffer is minimal and hence the communication latency is not affected.

Adaptive algorithm switching

In addition to the buffer size, the resending algorithm itself is adaptively switched according to the RTT and PLR. For example, if the network latency (RTT) is very large, packet resending is not used, but audio redundant transmission is used instead so that smooth conversation can be maintained.

MCU and presentation data

Real-time ARQ works in combination with the internal MCU function. When the MCU function is used, each link between the terminals and the PCS terminal with the MCU function uses ARQ to recover from packet losses. Additionally, when the PCS-DSB1 (Data Solution Box) is used, presentation data can also be recovered from packet losses.

Adaptive Rate Control (ARC)

Adaptive Rate Control is a mechanism used to slow down the video bit rate if network congestion occurs. Unlike conventional methods, the PCS terminal adopts TCP-Friendly Rate Control (TFRC) so it can adjust to other TCP traffic (e.g., FTP), which has a rate control function itself. TFRC is designed for unicast flows operating in the Internet environment and competing with TCP traffic. The TCP throughput equation in IETF RFC3448 is a function of packet loss rate and RTT should be suitable for use in TFRC. Figure 1.2 is an example of the calculated target video bit rate.

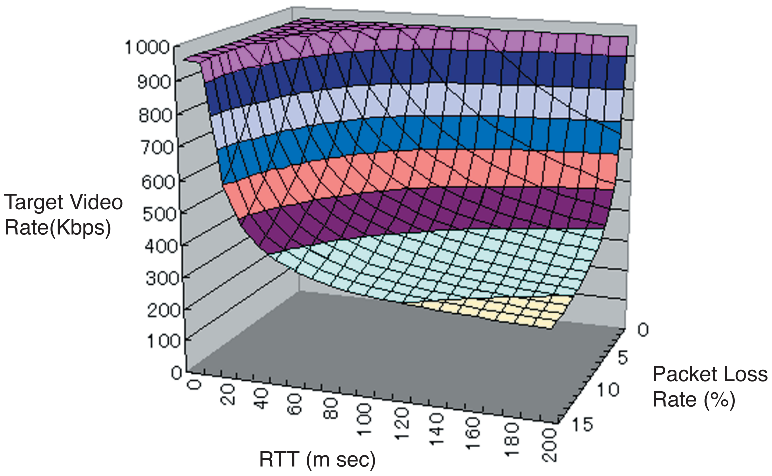


Figure 1.2 Adaptive Rate Control (for example, max video rate is 960 kbps)

Forward Error Correction (FEC) for Real-time Communications (For PCS-G50/G50P/G70/G70P)

FEC is another mechanism for recovering packet loss on the Internet.
The idea of FEC is to transmit parity packets to the receiver so any lost packets can be reconstructed. The Sony PCS system uses the Reed-Solomon (RS) code, which is perfectly suited for recovering erasure errors such as packet loss. Across the packets, RS (n, k) encodes k information symbols (each symbol per packet) into n symbols so as to construct the n-k parity packets. Here, the symbol size of all RS codes is set to eight for the convenience of accessing information in bytes. Figure 1.3 shows an example of the recovery procedure.
The first four packets – numbered 1 to 4 – are the data packets, and the latter two packets – F1 and F2 – are the parity packets, forming one RS block. By using FEC, a receiver can recover any lost pattern where the number of lost packets is no more than two.
For instance, if packet 2 is lost, the receiver can recover it by using the F1 packet. Compared with Real-time ARQ, the performance of FEC is not affected by the amount of RTT; this is one of the advantages of FEC that allows it to be used in a long RTT environment. By constructing a short packet matrix of RS blocks and adapting an original, fast-calculating RS algorithm, FEC decoding can be performed quickly so that PCS terminals can use FEC for real-time communication applications.
Since FEC always sends the parity packets, the overhead of the transmission rate is larger than that of Real-time ARQ.

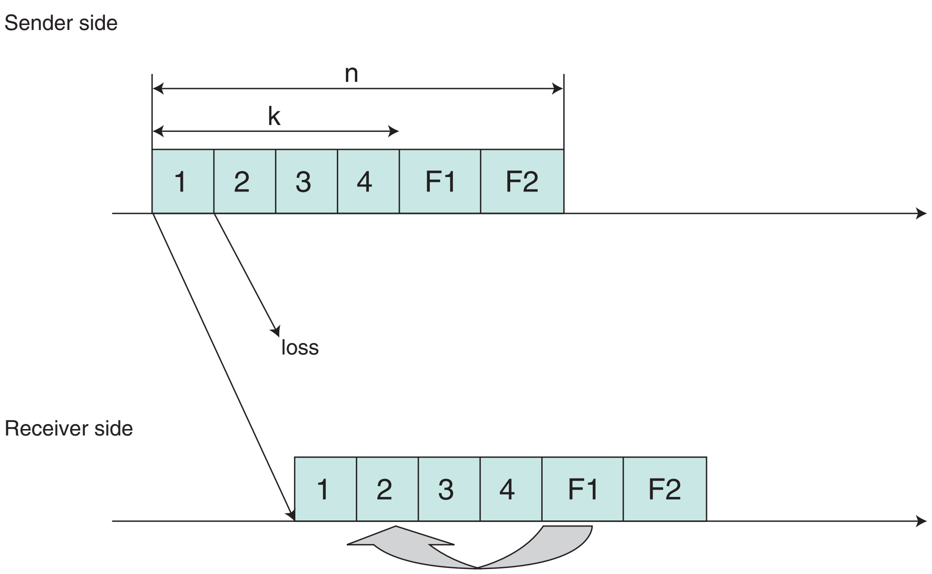


Figure 1.3 FEC recovery diagram

Selection of QoS mode

Selection method of the QoS mode differs depending on the type of terminal.

PCS-G50/G50P/G70/G70P

As discussed in the previous sections, the three modes are selected according to the setup of the PCS terminal's QoS setting. The following four modes are also supported, and these can be selected by the user:

Mode	Description
ARQ	Only ARQ is activated
FEC	Only FEC is activated
FEC&ARQ	Both ARQ and FEC are activated
Hybrid	ARQ/FEC/ARQ&FEC mode are selected according to the value of RTT

ARQ mode

ARQ for video data has two states: Standby and Active.
ARQ for audio data has three states: Standby, Active, and Audio double transmission. In audio double transmission mode, audio data is transmitted over the network twice with a large gap between each send – this prevents any increase in the delay caused by having to retransmit lost packets. When ARQ is set to on, ARC is automatically set to on as well.

FEC mode (for video only)

The FEC mode will only work if both the sender side and the receiver side have selected to turn it on. ARC is automatically turned on when FEC mode is selected. The ratio of the number of original packets to the number of parity packets is fixed.

FEC&ARQ mode

In this mixed mode, any packet loss that cannot be recovered by using FEC is recovered using ARQ. This mode becomes active when both FEC and ARQ are set to on.
Because the number of original packets and the number of parity packets in a single FEC block are fixed in FEC, it is easy to identify whether the packet loss can be recovered or not, simply by looking at the number of packets in the block. Where the received original packet is S_ORG, the original packet is ORG, and the received parity packet is S-PAR, the packet loss cannot be recovered if the following equation becomes true:

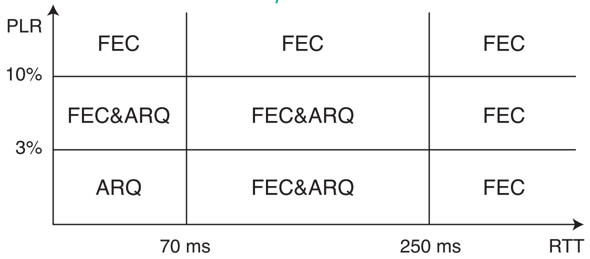
$$S\_PAR - (ORG - S\_ORG) < 0$$

In this case, where the packet loss cannot be recovered, a resend request will be issued.

Hybrid mode

When both the PCS terminal (the sender) and the remote party (the receiver) are set to Hybrid on, the Hybrid mode is activated. This automatically selects the most appropriate mode according to the network conditions, as shown in the following figure. When the remote terminal is set to Hybrid off, the mode selected for the local terminal will depend on the capability (FEC or ARQ) of both the remote and the local terminals.

Hybrid Mode



The threshold values of the above figure might be changed in future versions.

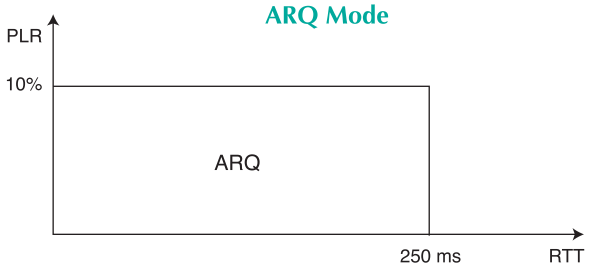
PCS-1/1P/11/11P/TL50/TL30

ARQ mode can be selected according to the QoS settings of the PCS terminals.

ARQ mode

ARQ for video data has two states: Standby and Active.
ARQ for audio data has three states: Standby, Active, and Audio double transmission.
In audio double transmission mode, audio data is transmitted over the network twice with a large gap between each send - this prevents any increase in the delay caused by having to retransmit lost packets. When ARQ is set to on, ARC is automatically set to on as well.

ARQ Mode



Network-level QoS

The PCS terminal 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 usage of the field for either service is up to the service administrator of the network.

IP Precedence and Type of Service

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

CU: Currently Unused

The value of IP Precedence and Type of Service can be defined in the setup menu.

IP Precedence bits The IP precedence bits indicate the priority with which a packet is handled, as follows:

1 1 1: Network control
1 1 0: Inter-network control
1 0 1: Critical
1 0 0: Flash override
0 1 1: Immediate
0 1 0: Priority
0 0 1: Routine

Delay bit Used when the time taken for a datagram to travel from the source PCS terminal to the destination PCS terminal or host (latency) is important. When the delay bit is set to on, the network provider is requested to select a route with the minimum delay.

Throughput bit Used when the volume of data transmitted in any period of time is important. When the throughput bit is set to on, the network provider is requested to select a route that produces the maximum throughput.

Reliability bit Used when it is important that the data arrives at the destination without requiring retransmission. When the reliability bit is set to on, the network provider is requested to select the route with the maximum reliability.

Minimum cost bit Used when minimizing the cost of data transmission is important. When the minimum cost bit it set to on, the network provider is requested to have datagrams routed via the lowest-cost route.

Unless the value of this field is set by the administrator, the default value of this field is always 0.

Differentiated Services

The Differentiated Services (Diffserve) architecture is based on a network model implemented over a complete Autonomous System (AS) or domain. As this domain is under administrative control, it is possible to take provisions to establish clear and consistent rules to manage traffic entering and flowing through the networks that conform to the domain. Diffserve defines a field in the IP header called the Differentiated Services Code Point (DSCP), which is a six-bit field as shown below.

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

CU: Currently Unused

The type of service field of the IP header is used to define the DSCP. Hosts in a network that supports DiffServe make each packet with a DSCP value.
Routers within the DiffServe network use these values to classify the traffic into distinct service classes according to the DSCP value. Thus, routers control packets not on a flow-by-flow basis, but by traffic classes based on DSCP marking. Since the routers are not required to maintain any elaborate state information to identify the flows, the routers can handle a large number of flows. PHB (Per Hop Behavior) is defined according to the traffic classes based on DSCP marking. For example, if the routers receive packets with a DSCP that equals 101110 - which means expedited forwarding (EF) – the routers are requested to forward the packets for low latency and low loss service. Whereas Assured Forwarding (AF), which defines three levels of drop precedence, is used to guarantee the minimum bandwidth.
The default value of this field when the field is used as Differentiated Services is 000000, meaning simply that the best effort service is provided.

SONY