Priority Promotion Scheme (PPS)—An Autonomous and Distributed Admission Control for End-to-end Quality of Service for Interactive Multimedia Services

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Abstract

Quality of service (QoS) control, especially admission control, is essential for satisfactory end-to-end session-type communications such as VoIP (voice over IP) or video chatting over an IP (Internet protocol) network. Here, we introduce the priority promotion scheme as an autonomous and distributed admission control technique suitable for large and diversified networks. The appropriate combination of the underlying principle of the Internet with the carrier's needs makes end-to-end QoS, which used to look hard to achieve, more practical and deployable soon.

1. Need for admission control technology

In the conventional form of telecommunication where the telephone service is provided over a circuit-switched network, the line from the caller to the receiver is reserved at each switch at the time the caller dials. Thus, the call is established only if the entire circuit can be reserved; if the circuit cannot be reserved, the request to establish communication is rejected. Each circuit has a fixed bandwidth of 64 kbit/s, and no mutual bandwidth interference occurs. This special mechanism of circuit switching guarantees stable communication quality after call establishment and has provided customers with highly satisfactory network service.

However, consider the provision of interactive multimedia communication services such as VoIP, videophone, or video conferencing provided over an IP network. Quality degradation at the packet level due to low transport speed or router processing delay is already being eliminated, and the VoIP service is coming into its own. Looking at it at the session level, the decision as to whether or not communication can be set up is made by SIP (session initiation protocol) proxy machines, which are different from routers. In that case, the admission decision is made without regard for the router and transport path usage situation, so communication quality is not strictly guaranteed. In the worst case, the bandwidth of the transport path is insufficient, and intermittent data loss occurs. That can lead to disruption of the video stream and the breaking up of the voice stream, which reduces user satisfaction. It can also affect communication that has already been established, so the degradation of quality during communication, which does not happen in conventional circuit-switched networks, may make it difficult to apply usage-based charging. Considering extension to collaboration-type communication services, in which image communication is important, session-by-session admission control technology is required.

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2. Application of admission control technology to a large-scale network

Various methods referred to as admission control have been proposed as a means of checking for surplus bandwidth on the transport path and deciding whether or not communication can be established prior to the establishment of a new session [1]. All these methods are claimed to be usable in a large-scale network of several million subscribers, but several questions remain, such as whether or not they are applicable to diverse networks that include user networks and corporate networks, whether they can be implemented economically, and how high the barriers to introduction are.

One key to the success of the Internet is that minimal control is implemented in the network itself: most functions are implemented in the end systems. That approach allows a low-cost network configuration. Learning from this lesson, we turned our attention to a method of implementing admission control by distributing over the terminals the processing that has been centralized in the nodes of the conventional network. Based on the fundamental idea proposed in [2], we have extended measurement-based admission control to a comprehensive network system to achieve end-to-end QoS (quality of service).

3. Introduction to priority promotion scheme

The method for implementing autonomous and distributed quality control is called the priority promotion scheme (PPS) [3]. PPS deals with important issues that pose problems for a practical service. It begins with the remaining bandwidth estimation by measurement: to confirm that the transport path has sufficient bandwidth remaining to establish a new session, the terminal itself sends trial packets between the source and destination terminals at the bandwidth needed for the new session. The destination terminal measures the quality degradation when the trial packets are received and estimates the available bandwidth on the basis of that information to decide whether or not to admit the session (**Fig. 1**). In PPS, we initially selected the packet loss rate as a criterion, because it is easy to handle and can show the remaining bandwidth.

To ensure that trial packets do not degrade the quality of existing communication sessions and to clearly reveal degradation in the trial packets caused by insufficient bandwidth, a variant of differentiated services (DiffServ) priority control is used within the network, with the trial packets assigned a low priority and the actual media packets assigned a high priority for transmission. If the reception quality of the trial packets is good, the source judges that the new session can be admitted and the priority is then raised (promoted) from low to high. The priority promotion scheme gets its name from this feature. If the reception quality is not good, the session will be terminated or suspended and given another chance later. The use of the trial affects how the service appears to the end-user, as described later.

The overall configuration diagram with relevant protocols for PPS is illustrated in **Fig. 2**. We assume SIP for the session control provided by the SIP proxy. The packets are transmitted by realtime transport protocol (RTP). The packets for testing the reception quality use realtime transport control protocol (RTCP). These protocols are all standardized ones. The transport system (routers, layer-2 switches, etc.) performs DiffServ priority control, and the actual media packets are transmitted with higher priority than the trial packets. Considering that the interactive multimedia service provided by PPS should share the same path with other non-realtime services, the upper limit of the available bandwidth is regulated for PPS. This characteristic is described later. When PPS is implemented, this transfer operation, which is called measurable forwarding per-hop behavior (MF-PHB), is assumed to be implemented at each node in the transport system.

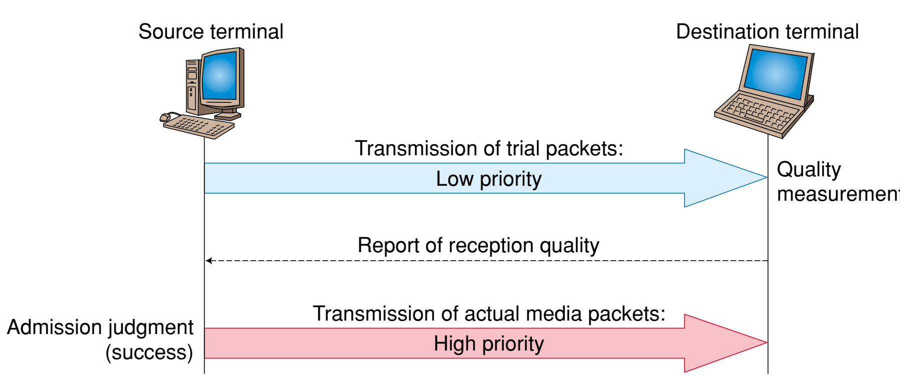

Fig. 1. PPS terminal behavior.

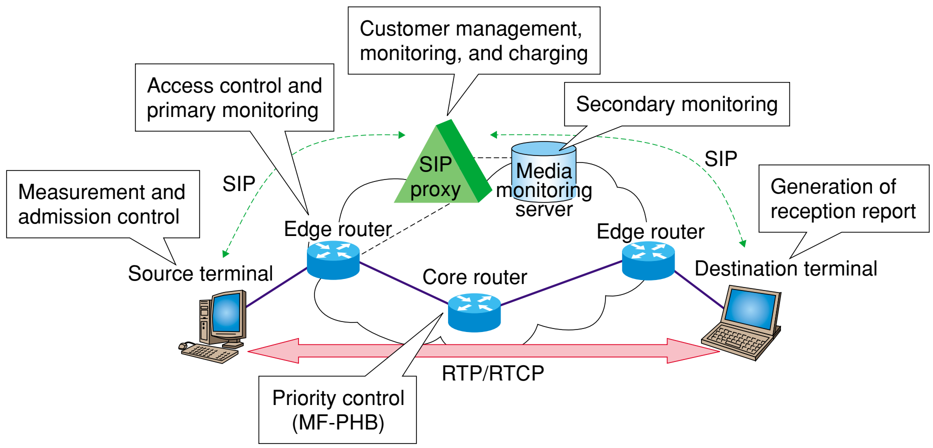

Fig. 2. Overall PPS configuration diagram.

4. Monitoring to identify invalid operation

In general, in packet network admission control, the admission decision is based on the initial communication bandwidth report, so if more packets than the reported value flow into the network because of incorrect or malicious terminal operations, then the actual packet volume will deviate from the initially assumed estimate, making it impossible to satisfy the desired communication quality. Therefore, to maintain quality within the network, packets other than the admitted media packets and any packets that exceed the admitted bandwidth must be stopped at the edge routers or some such location. This is a kind of access control and primary monitoring. In Fig. 2, it is shown performed at the edge router.

In PPS, because the admission decision is made at the terminal, there is concern that a false report of successful reception might be issued as a result of incorrect or malicious terminal operations, even though the quality of the trial packets was not acceptable. If any equipment in the network is careless about this, the quality in the network cannot be maintained. This is a fatal drawback for terminal-oriented techniques.

To prevent such a situation, we devised a method that involves selective monitoring of terminal operation by a media monitoring server (MMS). When a media stream is selected for monitoring, the SIP proxy reports the IP address of the MMS instead of the IP address of the original destination in the communication sent in response to the SIP request of the terminal being monitored. The result is that the originating terminal begins sending the media stream to the MMS, which allows the MMS to monitor for legitimate operation and relay the media stream. This concept achieves economical deployment of PPS in a real commercial network, while following the terminal-oriented Internet concept.

5. Coping with variable-bandwidth media streams

With PPS, in order to make the admission decision, the terminal estimates the available transport path bandwidth by sending trial packets. The available bandwidth is the transport path bandwidth minus the bandwidth being used by other terminals at the moment. Fluctuation in the bandwidth being used by other terminals may reduce the accuracy of the estimation and result in degradation of the quality of the admitted session. In video image communication for example, the amount of bandwidth needed depends strongly on movement in the video scene. Therefore, if the available bandwidth is estimated at a time when there is little movement in the video of the established sessions of the other terminals, the used bandwidth will be small and the available bandwidth will be overestimated. If the admission decision is made on that basis, it will be judged that the transmission can be received without degradation of quality, but actually the transport path bandwidth might be exceeded when the other terminals resume communication at the reported bandwidths, thus increasing the overall used bandwidth. This is another serious issue inherent to measurement-based admission control.

We therefore devised an extension to the queuing scheme at the router (MF-PHB). It transfers previously established sessions but limits the bandwidth allotted to the trial packets. This method can completely avoid quality degradation in established sessions, even if media streams have variable bandwidth requirements.

6. Measurable Forwarding Per-Hop Behavior (MF-PHB)

Before explaining our extension to MF-PHB, let us review the basic operation of MF-PHB. To distinguish between them, we use the term ‘static mode’ to refer to the ordinary method and ‘elastic mode’ to refer to the extended method. The bandwidth usage states of these two modes of MF-PHB are shown in **Fig. 3**. The vertical axis in the figure represents the used bandwidth and the horizontal axis represents time.

In the static mode, to reserve a constant bandwidth for non-PPS services (e.g., best-effort data communication), the bandwidth is kept below an upper limit, BWmf. Then, at the same time, BWmf is reserved for PPS use so that the bandwidth available to PPS services is not reduced by the use of the other services. Neither the trial packets nor the actual media stream used by PPS can use more bandwidth than BWmf (Fig. 3(a)).

The elastic mode, however, features separate upper bandwidth limits for the trial and actual media streams. For a total flow of trial and actual media streams of up to BWm, either type of media stream can be sent without packet loss. If BWm is exceeded, however, some trial packets are discarded, while actual media packets are transferred. In this way, new sessions are admitted as long as BWm is not exceeded, but are rejected if it is exceeded. Thus, transmission without loss of actual packets is possible, even if the bandwidth used by the actual media stream changes temporarily and is transmitted at a bandwidth higher than BWm. If an upper limit is not set for the bandwidth available to the actual media stream, then the bandwidth may become unavailable for use by other services. Therefore, the maximum bandwidth BWh is set (Fig. 3(b)).

When the elastic mode is applied, PPS can be used in usage scenarios such as the switching operation at the time of line failure and hand-over in a mobile environment. There are significant benefits when we apply PPS to a commercial network.

7. Examples of services using PPS

Having explained the technical aspects of PPS end-to-end admission control, we now introduce some examples of services that can be implemented with PPS. PPS provides the user with a means of confirming that communication can be accomplished without

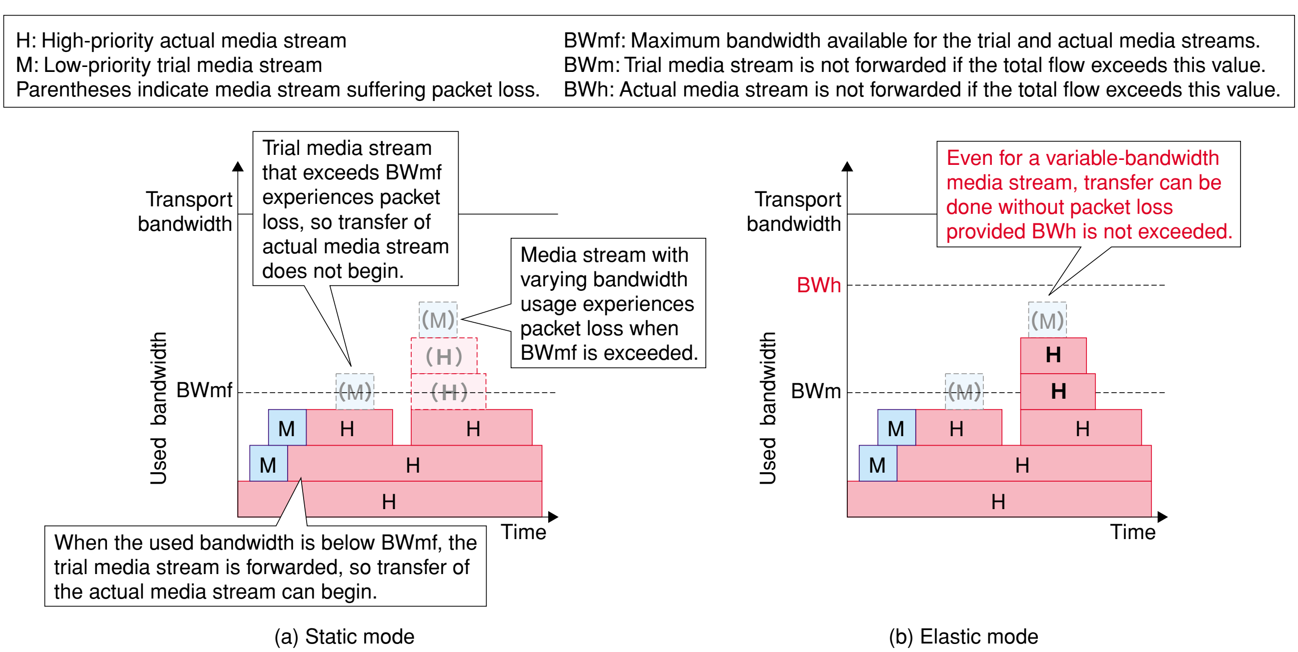

Fig. 3. MF-PHB router output.

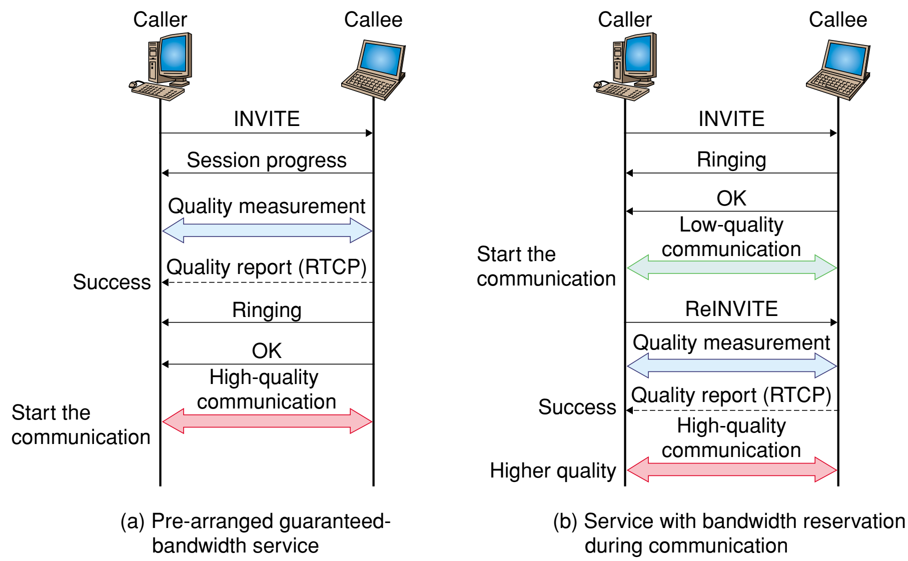

Fig. 4. Service examples.

quality degradation by actually conducting a trial transmission. The trial packets can be used in various services, depending on what stage of communication the trial packets are transmitted.

(1) Pre-arranged guaranteed-bandwidth service (**Fig. 4(a)**)

This is the same as the current telephone service. When the user on the originating side (the caller) lifts the receiver and dials, trial packets are transmitted at the bandwidth corresponding to the desired video image size and sound quality. Likewise, the terminal on the receiving side (the callee) also begins transmitting trial packets. If it is found that there is no quality degradation in either direction, transmission of the actual media stream begins and the receiving terminal rings. The conversation begins when the receiving user picks up the receiver. If quality degradation is found for the trial packets traveling in either direction, the caller is informed that the line is busy and transmission of the trial packets is stopped. Depending on the estimated available bandwidth, communication with a smaller image or lower audio quality may be possible.

(2) Service with bandwidth reservation during communication (**Fig. 4(b)**)

When the caller dials, the callee's terminal rings immediately. If the caller answers, transmission by best-effort or other non-PPS priority begins. Then, if the users feel that the quality is poor and choose high-quality PPS communication, even if they have to pay for it, a PPS admission trial is executed and the users can switch to the high-priority communication if the bandwidth is available. This service can also be used as a last resort to begin communication with best-effort or other low priority when high-quality communication cannot be established with a pre-arranged guaranteed bandwidth service, allowing the users to increase the quality later.

8. Current achievements and future potential of PPS

Concerning the observation time (a key PPS parameter), we learned that an interval of about one second is sufficient to reveal that the margin for the packet loss rate is 1% or less for a practical network configuration and communication in which the bandwidth does not change (e.g., VoIP without compression). For communication in which the bandwidth does change, we are clarifying the quality that can be achieved according to the degree of change and the degree of multiplexing. We have verified that most commercial routers can perform MF-PHB, which is newly introduced here, if the queuing configuration is manually optimized, something that is not usually done. The terminal observation function makes use of the RTCP function, which has already been established for VoIP. Regarding the selected monitoring by the MMS, we have shown that there are no technical difficulties in terminal-behavior monitoring by laboratory trials. All the results to date indicate that there are no technical obstacles and that PPS is a technology that has very high capabilities with respect to the functions and achievable quality required for the terminal and the network [4]. PPS employs packet discarding (the simplest behavior for routers) to judge quality and operates at the terminal where the service is finally received. Therefore, PPS should find widespread use as a general admission control technology for the large-scale and complex packet-based networks of the future.

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