Peer-to-peer-based, High-quality Live Video Delivery System for Business-to-business Applications

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Abstract

As broadband services become widespread, various types of Internet broadcasting are emerging, such as the broadcasting of music concerts, market information, stock market news, and radio programs, especially in the business-to-business area. To meet such demands, we aim to deliver TV-like high-quality and economical video over the best-effort broadband network by applying the peer-to-peer delivery scheme. The system we have developed, which is the first of its kind in the industry, uses set-top boxes to connect TV sets to NTT's B-FLET's network, allowing easy delivery of live programs to shops or terminals on the street.

1. High-quality delivery of live programs for B2B applications

Today, most live video delivery services over the Internet assume that ADSL (asymmetric digital subscriber line) or CATV (cable TV) lines are being used, the application field is business-to-consumer (B2C) services, and encoded video content is being delivered to personal computers at a rate of 300 kbit/s to 1 Mbit/s. However, there is a growing business-to-business (B2B) demand for the delivery of higher-quality video streams like TV (around 6 Mbit/s) to flat-panel displays and TV sets installed in restaurants, convenience stores, fast food shops, coffee shops, and terminals on the street.

B2B video delivery has typically been provided using either leased video transmission lines, which guarantee transmission quality, or satellite delivery services, which have scalable broadcasting capability. However leased lines are expensive, so their use has been limited to special applications, while satellite delivery is more expensive than terrestrial lines unless the service covers more than 1000 sites, so it cannot be applied economically to applications with a small to medium level of coverage.

The system introduced here applies the peer-to-peer (P2P) delivery scheme to video delivery so that the delivery fee can be kept constant irrespective of the size of the service coverage. This article describes the topology control used to guarantee reliability and the quality control used to reduce video quality degradation due to packet loss in the broadband network.

2. Applications of P2P delivery

As shown in Fig. 1, the P2P scheme allows information to be shared or published through direct communication between terminals. Specifically, the end terminal sends an inquiry to a discovery mechanism within the network to find end terminals that have the requested file and then receives the file directly from one of the discovered terminals, without a server being involved. Through the repeated operation of such one-to-one information delivery to successive terminals, the same information can be progressively relayed to multiple terminals. One advantage of the P2P scheme over the client-server scheme is that it is far less expensive because there is no need to set up a data center.

Figure 2 shows examples of B2B video delivery business applications and the relative advantages of

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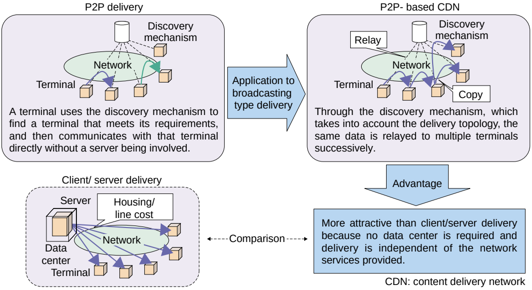

Fig. 1. Application of P2P delivery scheme to content delivery.

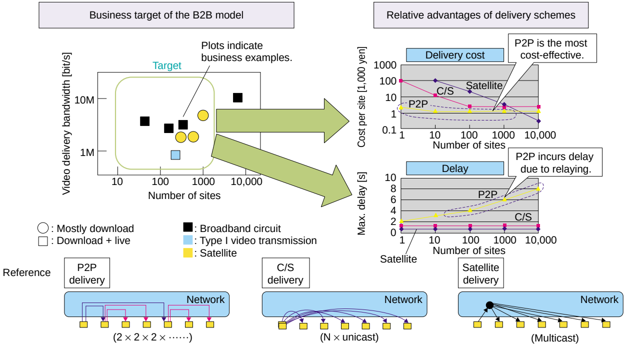

Fig. 2. Examples of B2B video delivery and the relative advantages of various delivery schemes.

various delivery schemes. A typical business application is delivering video to TV sets in shops. Demand is growing for high-quality live video delivery to 10–1000 shops. Compared with conventional client-server systems and satellite systems, P2P systems are less expensive but suffer from larger delay because the video content is successively relayed from one terminal to another. If this delay could be reduced to a tolerable level, the P2P system would be the most attractive solution.

When a video stream is relayed several times by terminals, as in a P2P system, the video quality will degrade due to network delay and data loss. To reduce the number of relays, terminals are usually connected in a tree structure, as shown in Fig. 3(a). A major problem with this topology is that it is vulnerable to

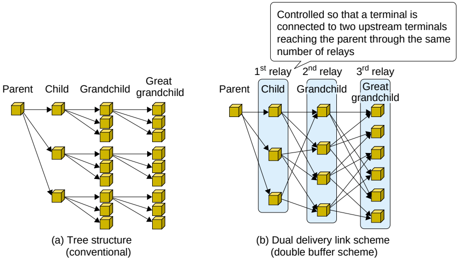

Fig. 3. P2P delivery topology.

both network and terminal failures, including power cuts. For example, if one upstream terminal fails, none of the terminals downstream can receive any video streams. In a B2B service, which charges a fee for video delivery, it is essential to provide mechanisms that ensure continued delivery even if an upstream terminal fails.

3. Topology control to guarantee quality

To eliminate this drawback of P2P, we have developed a dual delivery link topology control scheme (called the double buffer scheme). As shown in Fig. 3(b), every terminal (except those directly connected to the delivery server) receives the same video data from two different upstream terminals, which receive data via the same number of relays from the delivery server. Even if one upstream link fails, the downstream terminal will continue to receive video data from the other upstream terminal without interruption: the effects of an upstream failure do not propagate downstream. In addition, when the system detects that a terminal cannot receive data from one of the two upstream terminals, it connects it to an

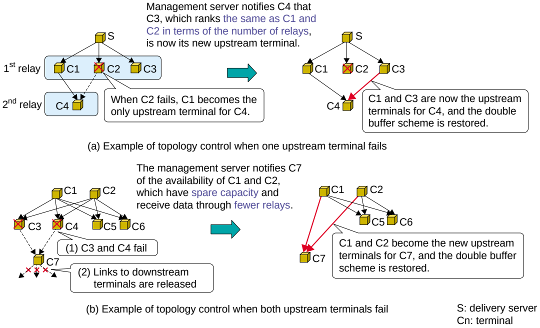

Fig. 4. Example of topology control after a terminal failure.

alternative upstream terminal to restore the double buffer scheme. Figure 4 outlines how topology control is executed when a failure occurs.

For the topology control, we applied the content routing device, called Smart Director, of the MDS-Dome [1] content delivery network (CDN), developed at NTT Information Sharing Platform Laboratories. By taking advantage of Smart Director's ability to quickly select a server based on precise server status and load information, we achieved high-performance double buffer topology control.

4. Quality control

The video quality of MPEG-2 streams, based on subjective evaluation, is known to degrade dramatically when the packet loss probability exceeds 0.1%. Packet loss can cause lost video frames, sound skips, and random noise. Therefore, a packet loss compensation method is essential if TV-quality video is to be delivered to terminals via broadband circuits whose transmission quality is not guaranteed.

There are two packet loss compensation techniques: forward error correction (FEC), which transmits redundant data in anticipation of problems, and packet re-transmission which responds to problems. In a real network, packet loss occurs randomly, so the packet loss probability is not constant. With FEC, if the packet loss probability is high, a lot of redundant data will be sent, which reduces transmission efficiency, even if the actual packet loss rate later turns out to be low. In contrast, packet re-transmission provides efficient transmission because the amount of re-transmitted data is proportional to the actual packet loss rate.

Since our system will be used in real networks and requires reliable transmission irrespective of the packet loss probability, we chose to use packet re-transmission. Specifically, a serial number assigned to each packet is monitored in real time, and if a skip in the sequence of serial numbers is detected, re-transmission is immediately requested. With packet re-transmission, there is a trade-off between reliability and delay time. To increase reliability, it is necessary to increase the waiting time for re-transmission. However, the current system delivers live video through multiple relays using P2P distribution, so it is necessary to minimize the delay caused by these relays to maintain the real time nature of the delivered video. Therefore, the waiting time for packet re-transmission is limited to one second, and packets are re-transmitted only once to reduce transmission delay. Any possible reduction in reliability is made up for by the dual delivery link topology. If the packet re-transmission in one link cannot recover the lost packets, the packets received on the other link can be used. If switching the incoming link does not recover the lost packets, quality degradation in the network is recognized, and the topology is reconfigured. Namely, the connection to the current upstream terminal is released and a connection to a different upstream terminal is set up.

To control the quality of video streams, we used the Live Stream Switching (LSS) system [1], developed at NTT Information Sharing Platform Laboratories, to copy and transmit a high quality video stream.

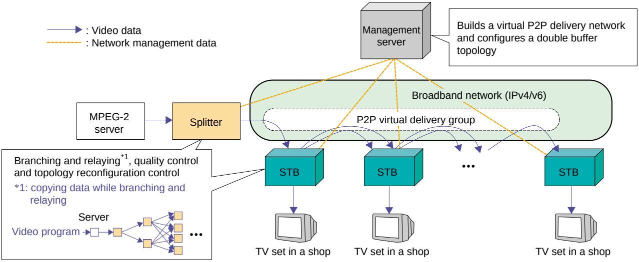

Fig. 5. System configuration.

5. System configuration

The developed system consists of a splitter, set-top boxes (STBs) (which are personal computers with terminal control software and MPEG-2 players built in), and a management server (Fig. 5). The splitter receives video packets from the MPEG-2 server, and sends them to an STB in response to a connection request. An STB is located at each site and participates in the multi-relay transmission of packets. It also has a video output port to send video signals to a TV set. The management server manages the configuration of the dual delivery link topology, monitors failures, and controls the STBs.

To use this system, a user must connect an STB, with a terminal ID and pre-assigned IP address, to a broadband network, such as NTT's local IP network. When power is supplied, the STB automatically asks the management server to send back delivery-related information. The management server searches for two upstream STBs and returns the IP addresses of the discovered STBs. Based on the returned information, the STB participates in the delivery chain of the network and starts receiving a video stream. In this way, the system can be configured and operated easily and at a low cost.

Once video reception has started, the topology and quality control techniques ensure that TV-quality video continues to be received even if a failure occurs in an upstream STB or if the network quality degrades. The system supports video stream encryption using IPsec (security architecture for the Internet protocol) and IPv6 networks, so it can provide high-quality live video delivery while meeting the requirements of individual customers.

6. Future studies

We are planning to hold a field trial using a real network (B-FLET's) to further evaluate the quality that can be achieved using this system.

Reference

[1] K. Yamada, T. Shiroshita, and S. Ushijima, "Large-capacity Content Delivery System for B-to-E and B-to-C: MDS-Dome/Megacast, LSS," NTT Technical Journal, Vol. 14, No. 4, pp. 46-49, 2002 (in Japanese).

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