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APPARATUS FOR TRANSMITTING/RECEIVING BITSTREAM IN NETWORK AND METHOD THEREOF

Technical Field

The present invention relates to an apparatus for transmitting/receiving a bitstream in a network and a method thereof, and more particularly, to an apparatus for transmitting/receiving a bitstream, by which a video bitstream is stably transferred in a network including a wireless network and an Internet network, and a method thereof.

Background Art

In packet networks such as Internet, typically, two logical ports exist when a channel is set to achieve the communication between two spots. That is, a real time protocol (RTP) packet, which has been packetized by a request for comments (RFC) protocol via the Internet network, is transmitted via the hierarchical structure of a real time protocol/user datagram protocol/Internet protocol (RTP/UDP/IP) or a real time protocol/transmission control protocol/Internet protocol (RTP/TCP/IP).

The RTP/TCP/IP operates in the acknowledge mode and can transmit data stably, so that it usually transmits control information. The RTP/UDP/IP operates in the unacknowledged mode, and transmits video data which usually must be processed in real time.

Referring to FIG. 1, in a transmitting terminal, a video bit stream sequentially undergoes an application layer (video source codec), an RTP layer, an UDP/IP and TCP/IP layer, a radio link layer protocol (RLP) layer, a layer 2 (L2) and a layer 1 (L1). In each layer, header information is added to the video bit stream, and the video bit stream having header information is transmitted to a network. Here, the TCP transmits control information, and the UDP transmits an RTP packet. In a receiving terminal, the video bit stream undergoes an UDP/IP and TCP/IP layer, an RTP layer and an application layer (video source codec), and is decoded into video

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data.

FIG. 2 is a view illustrating an embodiment of a typical communication method of transceiving data between a server and a client. First, a server 200 reads a video bit stream produced by a video source codec to divide the read video bit stream into a high significant bitstream 210 and a low significant bitstream 220 before an RTP packet is produced. Next, the server 200 transmits the high significant bitstream 210 to a client 230 via an RTP/TCP/IP, as indicated by arrow ①, and receives an acknowledgment representing that transmission of the high significant bitstream 210 has been completed without error, as indicated by arrow ②. Then, the server 200 transmits the low significant bitstream 220 to the client 230 via an RTP/UDP/IP regardless of error as indicated by arrow ③. The client 230 reconstructs the high significant bitstream 210 and the low significant bitstream 220 back into the original video bitstream syntax.

As described above, under a conventional Internet environment, a bitstream produced by video source coding such as MPEG-4 or H.263 is transmitted to the UDP/IP layer. The UDP/IP layer cannot ascertain the completion of reception of all of the transmitted data since it operates in an unacknowledged mode. If a bitstream packet is transmitted under a communication environment in which an Internet network communicates with a wireless network, the packet data may have bit error under the wireless environment. That is, the bitstream packet may have a loss on the Internet while undergoing the Internet environment and the wireless environment, and bit error may be generated even though the bitstream packet has been transmitted without loss. Here, the bitstream packet includes a packet header and a payload header. If a bit error is included in these headers, the receiving terminal cannot perform suitable decoding.

When a video bitstream is transmitted in real time, the transmitting terminal classifies the video bitstream on the basis of the significance, and transmits significant portions first and then less significant portions. At this time, the receiving terminal must delay data until significant portions are

received first, so that real time processing of data is difficult. That is, if significant portions are consecutively transmitted in an Internet network and a wireless network, the network must be continuously stable. Also, if a bitstream is classified according to significance during video source coding, it cannot be reconstructed into a bitstream packet which confirms to a current RFC protocol. Furthermore, the server 200 and the client 230 must always perform a pre-process for classifying a bitstream according to the significance to transmit and receive the bitstream, and a post-process for reconstructing the received upper and lower significant bitstreams into the original bitstream. Also, the process for classifying a bitstream on the basis of the significance before packetizing can only be performed at a video codec level which has already known the syntax of the video bitstream.

Therefore, data communication in an Internet network combined with a wireless environment causes a packet loss and a bit error as described above, thus deteriorating the quality of an image.

Detailed Disclosure of the Invention

An objective of the present invention is to provide a method of transmitting a bitstream, by which a video bitstream is stably transmitted by the effective use of two logic channels when communication is established by the two logic channels during transmission of the video bitstream.

Another objective of the present invention is to provide an apparatus for transceiving a bitstream, by which a video bitstream is stably transceived by the effective use of two logic channels when communication is established by the two logic channels during transmission of the video bitstream.

To achieve the first objective, there is provided a method of transmitting a bitstream in a communication network, according to an embodiment of the present invention, the method including: (a) coding a source into a bitstream using a predetermined type of coding; (b) adding the header of each communication protocol layer to a payload while

transmitting the bitstream coded in the step (a) to each communication protocol layer; and (c) transmitting the header separately from the bitstream received from the step (b).

To achieve the first objective, there is provided a method of transmitting a bitstream in a communication network, according to another embodiment of the present invention, the method including: (a) coding a source into a bitstream using a predetermined type of coding; and (b) adding the header of each communication protocol layer to a payload while transmitting the bitstream coded in the step (a) to each communication protocol layer, and separately transmitting the payload and the header.

To achieve the second objective, there is provided an apparatus for transmitting a bitstream in a communication network, according to an embodiment of the present invention, the apparatus including: an encoder for encoding source data into a bitstream using a predetermined type of coding; a protocol processing unit for adding the header of each communication protocol layer to a payload while transmitting the bitstream encoded by the encoder to each communication protocol layer; and a packet processing unit for transmitting the bitstream processed by the protocol processing unit in an unacknowledged mode protocol and transmitting the header information in an acknowledged mode protocol.

To achieve the second objective, there is provided an apparatus for relaying and receiving a bitstream in a communication network, according to another embodiment of the present invention, the apparatus including: an extractor for separately extracting payloads and header information which corresponds to the header of each layer while transmitting a bitstream received in a separate transmission protocol in the communication network to each layer; an error determination processing unit for determining whether the header information extracted by the extractor has error, and, if it is determined that the header information has error, requesting re-transmission; a bitstream re-organizing unit for re-organizing a bitstream using the header information extracted by the

extractor, if it is determined that the header information has no error; and a decoder for decoding a bitstream re-organized by the bitstream re-organizing unit.

Brief Description of the Drawings

FIG. 1 is a view illustrating a method of transmitting a video bit stream under an H.323 environment;

FIG. 2 is a view illustrating a method of transceiving a file between a server and a client in a communication network;

FIG. 3 is a bock diagram of a video stream transmitting apparatus according to the present invention;

FIG. 4 is a block diagram of an apparatus for relaying and receiving a video stream, according to the present invention; and

FIG. 5 is a view illustrating a method of transmitting a video bitstream in a situation where a wireless network communicates with an Internet network.

Best mode for carrying out the Invention

MPEG-4 or H.263, which are widely used as a video coding method, include various types of standardized headers. In case that data is transmitted in a real time protocol (RTP) while undergoing each layer of an Internet protocol or a wireless protocol, the transmission format is used. Accordingly, when this video coding method is used, the header of each layer is added to a payload header in each layer. Thus, only when the header of each layer and the payload header are safe from a bit error, a receiving terminal can perform suitable decoding.

Referring to FIG. 3, a video codec unit 310 encodes data to a bitstream using an application program such as H.323. A protocol processing unit 320 transfers the bit stream encoded by the video codec unit 310 to each layer of a communication protocol, and simultaneously adds the header of each protocol to a payload. A packet processing unit

330 packetizes a bitstream processed by the protocol processing unit 320 and transmits the bitstream packet in a user datagram protocol (UDP), which is an unacknowledged mode transmission protocol, and transmits only header information in a transmission control protocol (TCP), which is an acknowledged mode transmission protocol. In another embodiment, the packet processing unit 330 transmits a payload, among a bitstream processed by the protocol processing unit 320, in an unacknowledged mode transmission protocol, and transmits only the added header information in an acknowledged mode transmission protocol.

FIG. 4 is a block diagram of an apparatus for relaying and receiving a video stream, according to the present invention. Referring to FIG. 4, a packet extractor 410 transfers to each layer a bitstream packet received in an unacknowledged or acknowledged mode transmission protocol, while separately extracting a payload and the header of each layer from the bitstream packet. An error determination processing unit 412 determines existence or non-existence of an error in the header information extracted by the packet extractor 410. If it is determined that an error exists in the header information, the error determination processing unit 412 requests re-transmission. On the other hand, if it is determined that there are no errors in the header, a bit stream re-organizing unit 420 re-organizes a video bitstream using the header of each layer extracted by the packet extractor 410. A video codec unit 430 decodes the bitstream re-organized by the bitstream re-organizing unit 420.

FIG. 5 is a view illustrating a method of transmitting a video bitstream in a situation where a wireless network is interlocked with an Internet network. Referring to FIG. 5, reference numeral 510 indicates a wireless terminal on a transmitting side including several layers, reference numeral 560 indicates a base station including several layers, reference numeral 570 indicates an inter working function (IWF) including several layers, and reference numeral 580 indicates an Internet terminal on a receiving side including several layers.

First, the wireless terminal 510 includes a video source codec layer 512, which corresponds to an application layer, at the top, and sequentially includes an RTP layer 514, an TCP/IP layer 516, a radio link layer protocol (RLP) layer 522, a MAC layer 524, a layer 1 (L1) 526. Here, a multimedia codec other than the video source codec can be used as the application layer. A video stream forms a packet made up for a header and a payload while passing through each layer.

The video source codec layer 512 encodes a video source into a video bitstream using a video source coding method such as MPEG-4 or H.263 to form a payload header 532 and a video payload 534 as shown in (a). Here, the payload header 532 and the video payload 534 can be replaced by multimedia data. Then, the RTP layer 514 forms a packet by adding a video payload 545 filled with video data, a payload header 544, and an RTP header 543, and the UDP/IP or TCP/IP layer 516 adds an IP header 541 and an UDP or TCP header 542 to the formed packet, as shown in (b). The RTP layer 522 and the L2 layer 524 add an L2 header 551 and an RLP header 552 to the packet (b) as shown in (c).

Next, a video bitstream to which the header of each layer is added is transmitted to the base station 560 including identical layers, through an UDP or TCP. The video bitstream (c) including headers can be divided into a portion which is transmitted through the UDP, which is an UNACK transmission protocol, and a portion which is transmitted through the TCP, which is an ACK transmission protocol. As described above, the wireless terminal 510 can transmit a video bitstream using the following methods.

In a first embodiment of the first method, a video bitstream, to which header information is added, is transmitted in the UDP, and the header information is separately transmitted in the TCP. When a bitstream is transmitted only in the UDP, if header information included in the bitstream is damaged, it is difficult for a receiving side to process the bitstream. Hence, in order to prevent the packet loss, the wireless terminal 510 individually packetizes the header of each layer, that is, the L2 header 551,

the RLP header 552, the IP header 553, the UDP or TCP header 554, the RTP header 555 and the payload header 556, which are added to a video bitstream after the video bitstream passes through each layer. At the same time or when re-transmission is requested, the wireless terminal 510 stably transmits the packetized headers in a TCP. Here, the transmission in the TCP at the request of re-transmission is performed in units of IP packets or RLPs.

In a second embodiment, in order to solve delay that may occur in real time environment, a video bitstream to which header information is added is transmitted in a UDP, and the header information is separately packetized and transmitted in a UDP simultaneously or when re-transmission is requested.

In the second method according to an embodiment, a video bitstream is separated into a payload portion and a header portion, and these portions are separately packetized. The payload portion is transmitted in a UDP, and simultaneously, the header portion is separately transmitted in a TCP. In another embodiment, the payload portion is transmitted in a UDP, and simultaneously, the header portion is separately transmitted in a UDP. In still another embodiment, in order to reduce a transmission time, a bitstream packet except for a portion from which a bit error is removed by the TCP layer can be transmitted to a UDP layer.

In the third method, when a bitstream transmitted via the TCP layer is re-transmitted a small number of times, the channel of transmission is determined to be stable to some extent. Accordingly, a small bitstream transmitted via the UDP can be transmitted via the TCP.

The base station 560 relays the layers of a wireless protocol, that is, an RLP layer, an L2 and an L1, to the layers of an Internet protocol, an UDP layer, an IP layer and an L1 (or ATM), in order to tunnel a bitstream received from the wireless terminal 510. At this time, when re-transmission is requested, data transmitted in a TCP is re-transmitted in units of IP packets or RLPs.

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The IWF 570 relays a bitstream, which has passed through the layers of the base station 560, that is, a UDP, an IP, and an L1, to an UDP or TCP, an IP and an L1, in order to interface with the Internet terminal 580.

The Internet terminal 580, which is a final receiving side, decodes a bitstream received from the IWF 570 through an L1 576, an IP layer 572, a UDP or TCP layer 566, an RTP layer 564, and a video source codec layer 562. The Internet terminal 580 can properly decode a video bitstream which probably has a bit error, using an error resilient tool of video coding, referring to a payload and separately-received header information, when a packet received via the UDP layer has a bit error.

The present invention is not limited to the aforementioned embodiment, and it is apparent that modifications to this particular embodiment may be effected by those skilled in the art without departing from the spirit of the present invention. That is, the present invention can be used in a case where a bitstream is bidirectionally communicated in real time or streamed in a one-way system in a packet network such as an Internet. The present invention can also be applied to a case where an audio source codec other than a video source codec, or a source codec having the same function as the audio source codec has an error resilient tool with respect to a payload or a function which conforms to the error resilient tool, that is, in cases of an adaptive multirate (AMR) for an MPEG-4 audio mobile, an AMR for an UMTS, a speech codec, and the like.

Also, the above-described embodiment of the present invention can be written in a program that can be executed in computers, and can be realized in general-use digital computers which operate the program from a medium which is used in computers. The medium includes a magnetic storage medium (for example, a ROM, a floppy disc, a hard disc, and the like), an optical read-out medium (for example, a CD-ROM, a DVD and the like), and a storage medium such as a carrier wave (for example, transmission via the Internet).

Industrial Applicability

According to the present invention as described above, header information or the like is stably transmitted separately from a payload when a wireless network and an Internet network are linked, so that it can correct and check a bit error of a packet which has been passed through each layer. Also, in contrast with an existing method of transmitting data according to significance, a packet to which the present invention is applied can be processed independently of a video syntax. Furthermore, under an application environment where communication is achieved using a UDP, a bitstream including a bit error can be appropriately decoded by separately-received header information using an error resilient tool.

What is claimed is:

1. A method of transmitting a bitstream in a communication network, the method comprising:

(a) coding a source into the bitstream using a predetermined type of coding;

(b) adding a header of each communication protocol layer to a payload of the bitstream while transmitting the bitstream coded in the step (a) to each communication protocol layer; and

(c) transmitting the header and the bitstream received from the step (b) and, therefore, comprising both the header and the payload, separately from each other.

2. A method of transmitting a bitstream in a communication network, the method comprising:

(a) coding a source into the bitstream using a predetermined type of coding;

(b) adding a header of each communication protocol layer to a payload of the bitstream while transmitting the bitstream coded in the step (a) to each communication protocol layer, and separately transmitting the payload and the header.

3 The method of claim 1, wherein a bitstream, to which headers have been added by undergoing each communication protocol layer, is transmitted in an unacknowledged mode protocol, and only the header information in the bitstream is separately transmitted in an acknowledged or unacknowledged mode protocol.

4. The method of claim 2, wherein a payload in a bitstream, which has passed through each communication protocol layer, is transmitted in an unacknowledged mode protocol, and the added header is separately transmitted in an acknowledged mode protocol

5. The method of claim 3 or 4, wherein, when the number of times of re-transmission of a bitstream transmitted to an acknowledged mode protocol is equal to or less than a predetermined number of times, the bitstream, which has been transmitted in an unacknowledged mode protocol, is transmitted in an acknowledged mode protocol.

6. The method of claim 3, wherein a bitstream, to which headers have been added by undergoing each communication protocol layer, is transmitted in an unacknowledged mode protocol, and simultaneously the header information in the bitstream is transmitted in an acknowledged mode protocol.

7. The method of claim 4, wherein a payload in a bitstream, which has passed through each communication protocol layer, is transmitted in an unacknowledged mode protocol, and simultaneously the header information in the bitstream is transmitted in an acknowledged mode protocol.

8. The method of claim 3, wherein a bitstream, to which headers have been added by undergoing each communication protocol layer, is transmitted in an unacknowledged mode protocol, and simultaneously the header information in the bitstream is also transmitted in the unacknowledged mode protocol.

9. The method of claim 3, wherein a bitstream, to which headers have been added by undergoing each communication protocol layer, is transmitted in an unacknowledged mode protocol, and, when a transmission failure occurs, the bitstream is re-transmitted in an acknowledged or unacknowledged mode protocol.

10. The method of any of claims 3 through 7, wherein the

acknowledged mode protocol is a transmission control protocol (TCP), and the unacknowledged mode protocol is a user datagram protocol (UDP).

11. The method of any of claims 3 through 9, wherein the acknowledged mode protocol re-transmits data of individual Internet protocol (IP) packets or individual radio link layer protocols (RLP).

12. The method of claim 1 or 2, wherein the headers are a payload header, a real time protocol (RTP) header, a user diagram protocol (UDP) or transmission control protocol (TCP) header, an Internet protocol (IP) header, a radio link layer protocol (RLP) header, and a layer 2 (L2) header which are added after a bitstream is passed through each layer

13. The method of claim 1 or 2, wherein the payload includes multimedia data.

14. An apparatus for transmitting a bitstream in a communication network, the apparatus comprising

an encoder for encoding source data into a bitstream using a predetermined type of coding;

a protocol processing unit for adding the header of each communication protocol layer to a payload while transmitting the bitstream encoded by the encoder to each communication protocol layer; and

a packet processing unit for transmitting the bitstream processed by the protocol processing unit in an unacknowledged mode protocol and transmitting the header information in an unacknowledged or acknowledged mode protocol.

15. An apparatus for relaying and receiving a bitstream in a communication network, the apparatus comprising:

an extractor for separately extracting payloads and header information, which corresponds to the header of each layer, while transmitting a bitstream received in a separate transmission protocol in the communication network to each layer;

an error determination processing unit for determining whether the header information extracted by the extractor has error, and requesting re-transmission if it is determined that the header information has error;

a bitstream re-organizing unit for re-organizing a bitstream using the header information extracted by the extractor; and

a decoder for decoding a bitstream re-organized by the bitstream re-organizing unit.