EXHIBIT 1001

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Melnyk et al.

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(54) **ADAPTIVE BITRATE MANAGEMENT FOR STREAMING MEDIA OVER PACKET NETWORKS**

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(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 107 days.

This patent is subject to a terminal disclaimer.

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Related U.S. Application Data

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(51) **Int. Cl.**
G06F 15/16 (2006.01)

(52) **U.S. Cl.** **709/233**; 709/230; 709/232; 709/238; 370/230; 370/241; 375/240

(58) **Field of Classification Search** 709/230 237; 370/506
See application file for complete search history.

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(57) **ABSTRACT**

A method including receiving a receiver report from a terminal; estimating one or more network conditions of a media network based at least in part on the receiver report; determining an optimal session bitrate based on the estimated one or more network conditions; and providing media data to the terminal based on the optimal session bitrate.

16 Claims, 5 Drawing Sheets

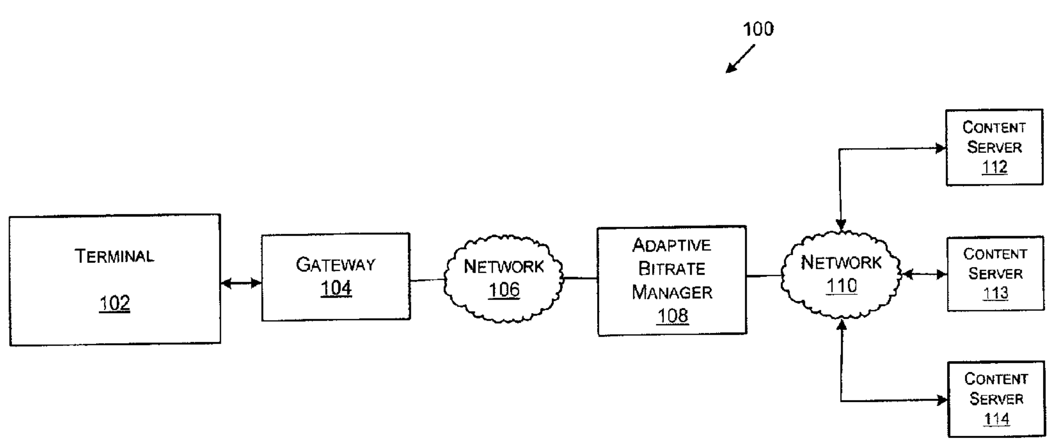

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Fig. 1

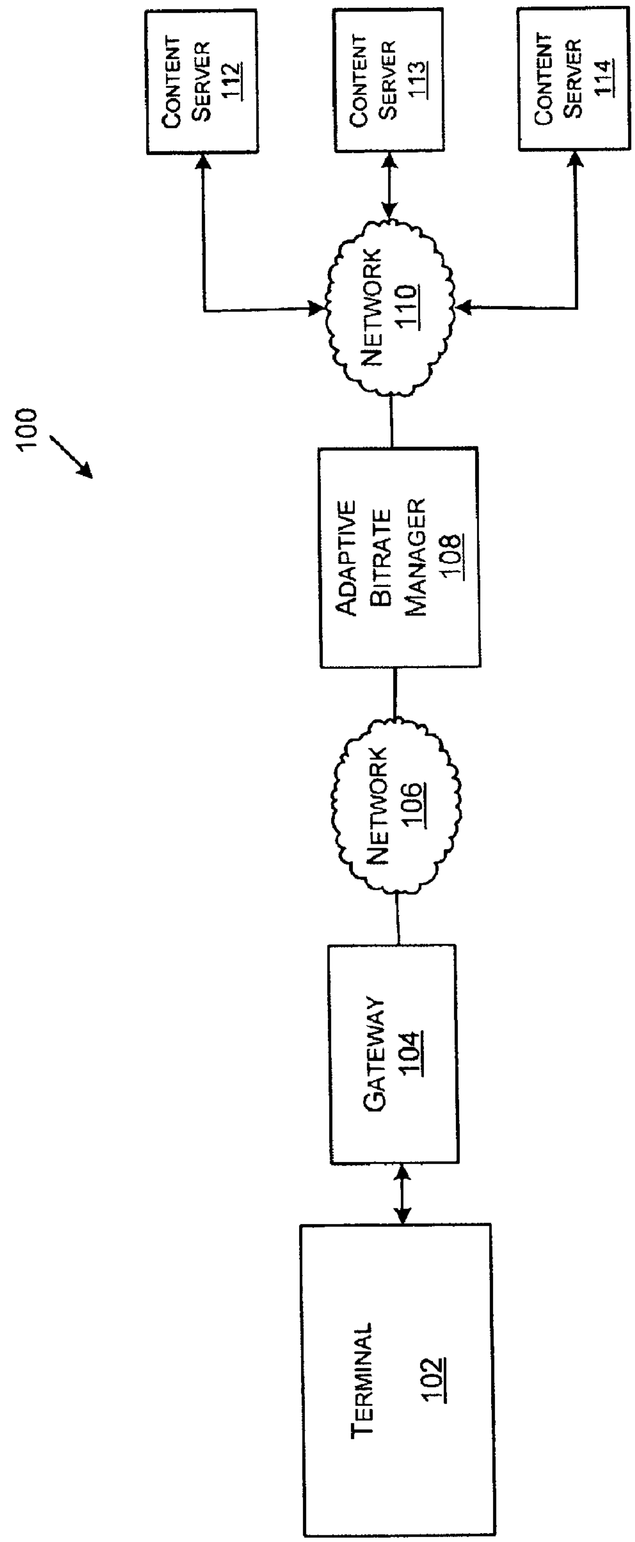

100
CONTENT SERVER 112
CONTENT SERVER 113
CONTENT SERVER 114
NETWORK 110
ADAPTIVE BITRATE MANAGER 108
NETWORK 106
GATEWAY 104
TERMINAL 102

Fig. 2

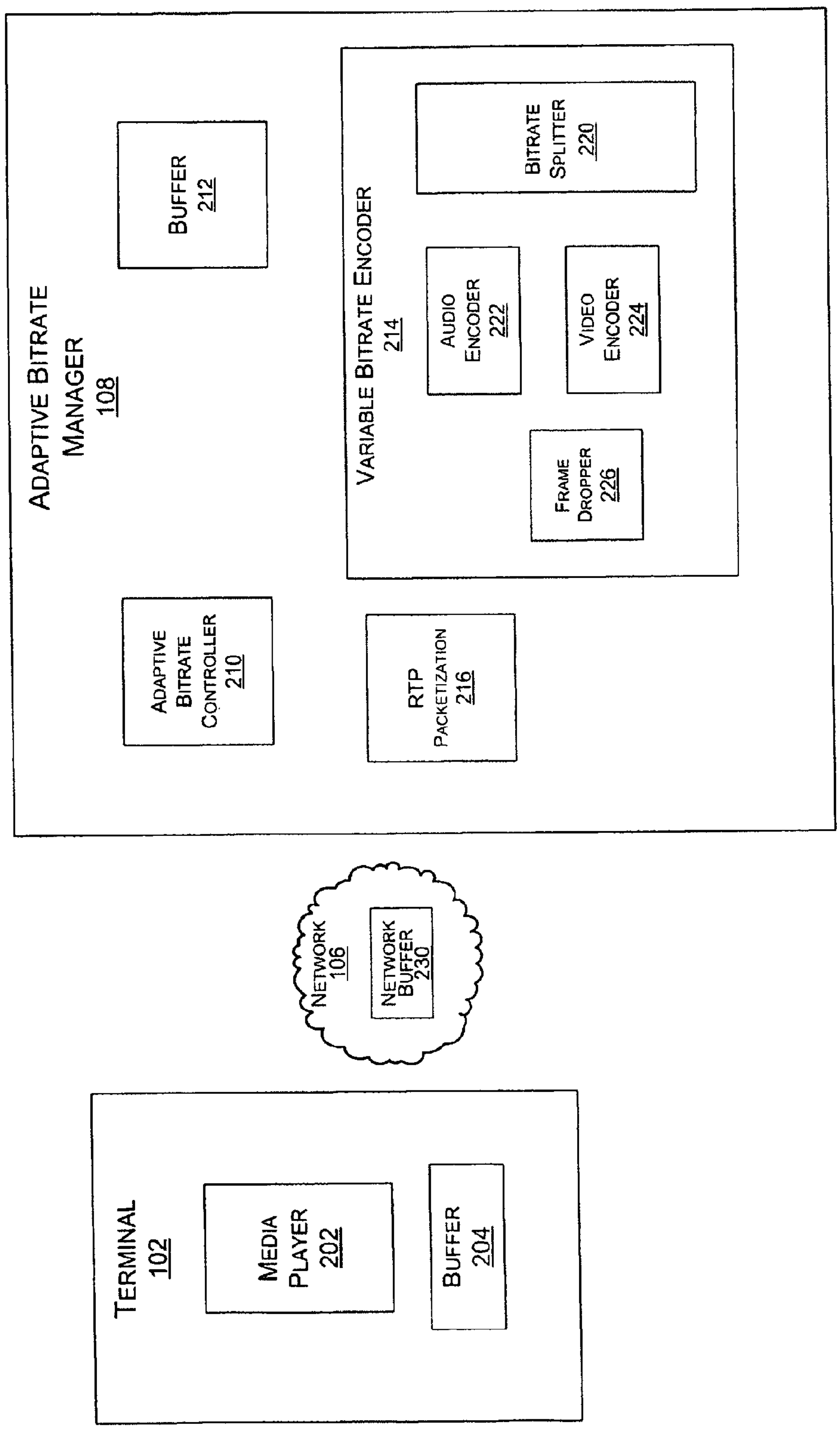

ADAPTIVE BITRATE MANAGER 108
BUFFER 212
VARIABLE BITRATE ENCODER 214
BITRATE SPLITTER 220
AUDIO ENCODER 222
VIDEO ENCODER 224
FRAME DROPPER 226
ADAPTIVE BITRATE CONTROLLER 210
RTP PACKETIZATION 216
NETWORK 106
NETWORK BUFFER 230
TERMINAL 102
MEDIA PLAYER 202
BUFFER 204

Fig. 3

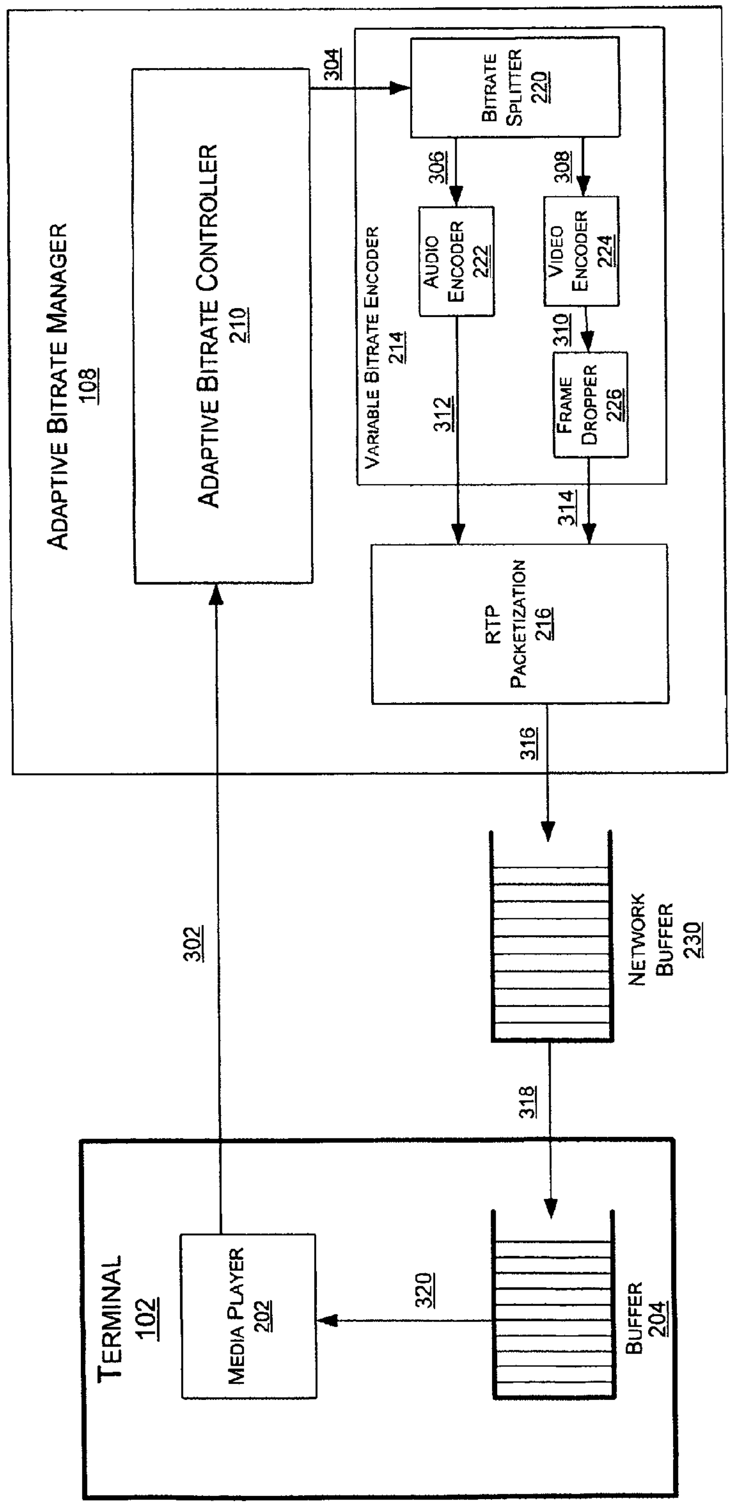

Adaptive Bitrate Manager 108
Adaptive Bitrate Controller 210
304
Variable Bitrate Encoder 214
Bitrate Splitter 220
306
308
Audio Encoder 222
Video Encoder 224
310
Frame Dropper 226
312
314
RTP Packetization 216
316
302
Network Buffer 230
318
Terminal 102
Media Player 202
320
Buffer 204

Fig. 4

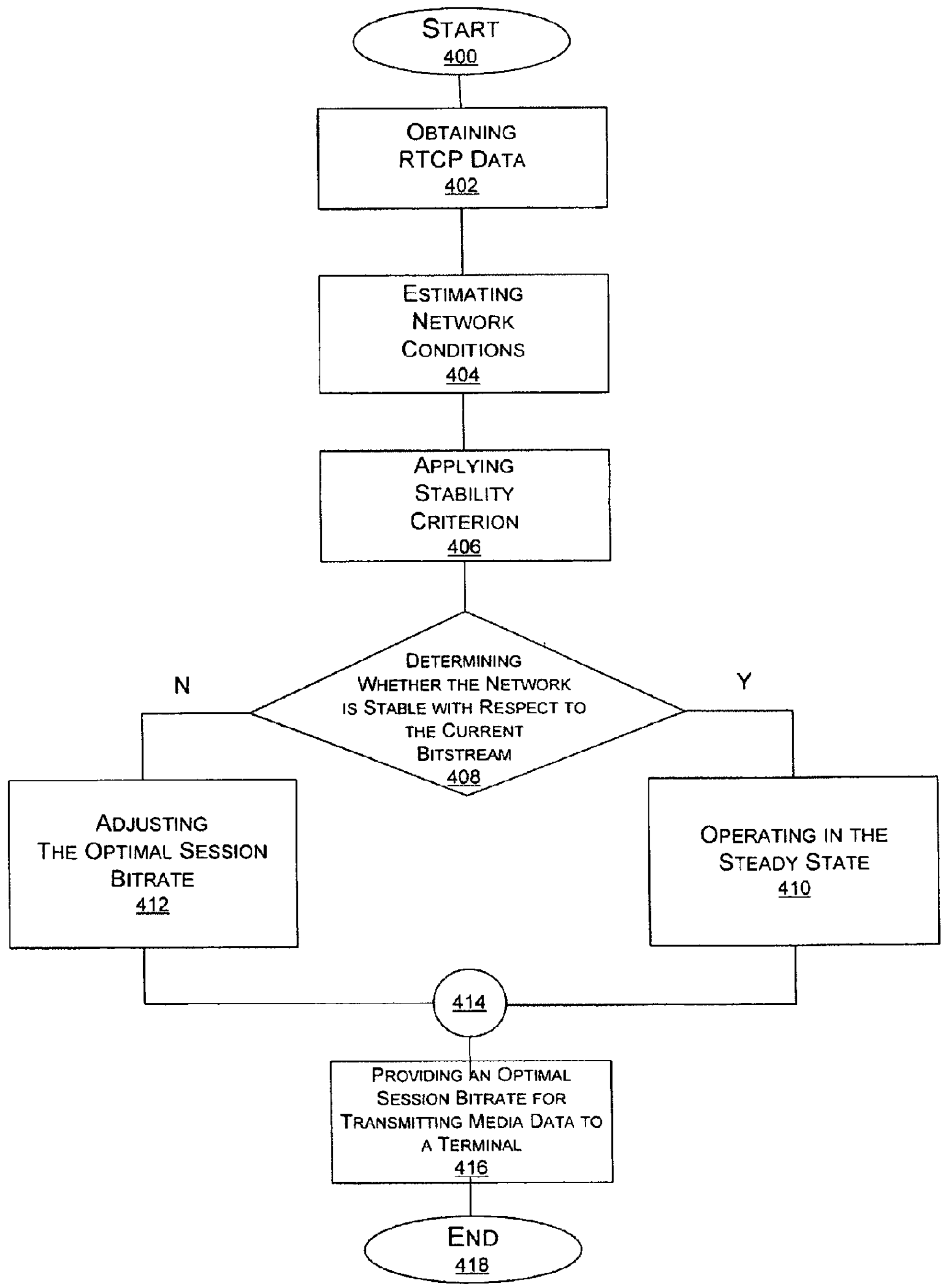
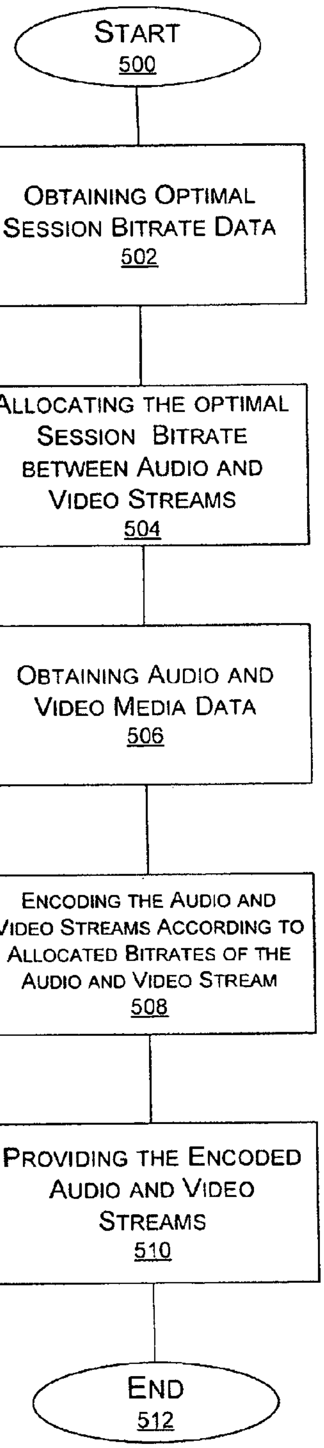

START
400
OBTAINING RTCP DATA
402
ESTIMATING NETWORK CONDITIONS
404
APPLYING STABILITY CRITERION
406
DETERMINING WHETHER THE NETWORK IS STABLE WITH RESPECT TO THE CURRENT BITSTREAM
408
N
Y
ADJUSTING THE OPTIMAL SESSION BITRATE
412
OPERATING IN THE STEADY STATE
410
414
PROVIDING AN OPTIMAL SESSION BITRATE FOR TRANSMITTING MEDIA DATA TO A TERMINAL
416
END
418

Fig. 5


START
500
OBTAINING OPTIMAL SESSION BITRATE DATA
502
ALLOCATING THE OPTIMAL SESSION BITRATE BETWEEN AUDIO AND VIDEO STREAMS
504
OBTAINING AUDIO AND VIDEO MEDIA DATA
506
ENCODING THE AUDIO AND VIDEO STREAMS ACCORDING TO ALLOCATED BITRATES OF THE AUDIO AND VIDEO STREAM
508
PROVIDING THE ENCODED AUDIO AND VIDEO STREAMS
510
END
512

ADAPTIVE BITRATE MANAGEMENT FOR STREAMING MEDIA OVER PACKET NETWORKS

CROSS REFERENCE TO RELATED PATENTS

This application claims the benefit of U.S. Provisional Application No. 60/948,917, filed Jul. 10, 2007, "Adaptive Bitrate Management for Streaming Media over Packet Networks," which is incorporated herein by reference.

BACKGROUND INFORMATION

Rate control is essential for media streaming over packet networks. The challenge in delivering bandwidth-intensive content like multimedia over capacity-limited, shared links is to quickly respond to changes in network conditions by adjusting the bitrate and the media encoding scheme to optimize the viewing and listening experience of the user. In particular, when transferring a fixed bitrate over a connection that cannot provide the necessary throughput, several undesirable effects arise. For example, a network buffer may overflow resulting in packet loss causing garbled video or audio playback, or a media player buffer may underflow resulting in playback stall. Standard bodies have recommended protocols to address these issues. Internet Engineering Task Force (IETF), in RFC 3550, specifies RTCP as the fundamental building block to implement bit rate/packet rate control in streaming media. Several extensions to RTCP, suited for high capacity networks, follow this original recommendation.

Even with these recommended protocols, delivering a multimedia session over wireless networks can be particularly challenging, due in part to the following:

- Sudden Adjustment of nominal transmission rate. Due to interference, fading, etc, 3+G networks negotiate physical layer parameters on the fly. Nominal transmission bitrates can change by a factor of 10;
- Packet Loss: caused by either link transmission errors or by network congestion;
- Reduction of Effective bandwidth: The wireless link is a shared resource at Layer 2, with MAC (Media Access Control) mechanism and scheduling. This means that an increased load presented by other wireless terminals in the same sector can reduce the effective bandwidth or capacity that a terminal will see; and
- Limited Capacity: Available capacity is typically a fraction to that obtained in traditional wireline internet access technologies, where currently capacity is not an issue. Fixed internet media sessions can typically offer to the network loads between 250 and 400 kbps. Despite the fact that current 3G cellular networks can sustain throughputs of 500 kbps and above, the total bitrate budget for a wireless multimedia session is typically kept under 150 kbps to ensure scalability.

For wireless mobile devices, providing a good experience in streaming media sessions is particularly difficult, due to

- Infrequent and incomplete network state information. The typical wireless media player support RTCP receiver report as defined in RFC 3550, and the report generation frequency is fixed. As a result, the network state information obtained at the sender end is limited and sporadic. In its Packet Streaming Service specification, 3GPP recommends several extensions to the basic IETF RTCP Receiver Report (i.e. RTCP Extended Reports, or XR). Unfortunately, very few handsets implement these enhancements;

- Different media streams are handled separately. Despite the fact that they are both transmitted over the same network link, audio and video streams are handled separately by RTCP. Both RTCP reports provide state information about the same network, therefore a joint analysis; and
- Low bitrates available: The bitrate budget for a wireless multimedia session is generally very low (under 150 kbps). The adjustment of audio and video bitrates can have large perceptual impact on the session, and the total available network bitrate, even for 3G networks, can fall well below acceptable quantities. With these issues, to wireless networks and wireless mobile devices it has been difficult to set up a consistent streaming media session.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. **1** is a block diagram of an exemplary system.

FIG. **2** is a block diagram illustrating an embodiment of the exemplary system of FIG. **1**.

FIG. **3** is a functional diagram illustrating an exemplary communication flow in the exemplary system of FIG. **2**.

FIG. **4** is a flowchart representing an exemplary method for processing an RTCP packet.

FIG. **5** is a flowchart representing an exemplary method for processing optimal session bitrate data.

DETAILED DESCRIPTION OF DRAWINGS

Reference will now be made in detail to the exemplary embodiments consistent with the invention, the examples of which are illustrated in the accompanying drawings. Wherever possible, the same reference numbers will be used throughout the drawings to refer to the same or like parts.

Adjusting the bitrate of streaming media sessions according to instantaneous network capacity can be a critical function required to deliver streaming media over wireless packet networks. Adaptive bitrate management is a comprehensive framework and method that enables the delivery of self-adjusting streaming sessions to media players, for example, such as standard 3GPP-compliant media players. Adaptive bitrate management includes, among other things, an adaptive bitrate controller and a variable bitrate encoder, both of which allow the adaptive bitrate management the ability to implement joint session bitrate management for audio, video and/or other streams simultaneously.

FIG. **1** is a block diagram of an exemplary system. Exemplary system **100** can be any type of system that transmits data packets over a network. For example, the exemplary system can include a mobile terminal accessing streaming media data from content servers through the Internet. The exemplary system can include, among other things, a terminal **102**, a gateway **104**, one or more networks **106**, **110**, an adaptive bitrate manager **108**, and one or more content servers **112-114**.

Terminal **102** is a hardware component including software applications that allow terminal **102** to communicate and receive packets corresponding to streaming media. Terminal **102** provides a display and one or more software applications, such as a media player, for displaying streaming media to a user of terminal **102**. Further, terminal **102** has the capability of requesting and receiving data packets, such as data packets of streaming media, from the Internet. For example, terminal **102** can send request data to content servers **112-114** for a particular file or object data of a web page by its URL, and the content server of the web page can query the object data in a database and send the corresponding response data to terminal **102**. In some embodiments, response data may be routed through adaptive bitrate manager **108**.

While terminal **102** can be a wired terminal, some embodiments of the invention may prefer using a mobile terminal because mobile terminals are more likely to be in networks that would benefit more from an adaptive bitrate manager. The network connection tends to be less stable as compared to wired network connection due to, for example, the changing position of the mobile terminal where data rate transmissions between the mobile terminal and the network can fluctuate, in some cases quite dramatically.

Gateway **104** is a device that converts formatted data provided in one type of network to a particular format required for another type of network. Gateway **106**, for example, may be a server, a router, a firewall server, a host, or a proxy server. Gateway **104** has the ability to transform the signals received from terminal **102** into a signal that network **106** can understand and vice versa. Gateway **104** may be capable of processing audio, video, and T.120 transmissions alone or in any combination, and is capable of full duplex media translations.

Networks **106** and **110** can include any combination of wide area networks (WANs), local area networks (LANs), or wireless networks suitable for packet-type communications, such as Internet communications Further, networks **106** and **110** can include buffers for storing packets prior to transmitting them to their intended destination.

Adaptive bitrate manager **108** is a server that provides communication between gateway **104** and content servers **112-114**. Adaptive bitrate manager **108** can optimize performance by adjusting a streaming media bitrate according to the connection, i.e., media network, between adaptive bitrate manager **108** and terminal **102**. Adaptive bitrate manager **108** can include optimization techniques, further described below.

Content servers **112-114** are servers that receive the request data from terminal **102**, process the request data accordingly, and return the response data back to terminal **102** through, in some embodiments, adaptive bitrate manager **108**. For example, content servers **112-114** can be a web server, an enterprise server, or any other type of server. Content servers **112-114** can be a computer or a computer program responsible for accepting requests (e.g., HTTP, RTSP, or other protocols that can initiate a media session) from terminal **102** and serving terminal **102** with streaming media.

FIG. **2** is a block diagram illustrating an embodiment of the exemplary system of FIG. **1**. Terminal **102** may include, among other things, a media player **202** and a buffer **204**. Adaptive bitrate manager **108** can include, among other things, an adaptive bitrate controller **210**, a buffer **212**, a variable bitrate encoder **214**, and a Real-time Transport Protocol (RTP) packetization **216**.

Media player **202** is computer software for playing multimedia files (such as streaming media) including video and/or audio media files. Such popular examples of media player **202** can include Microsoft Windows Media Player, Apple Quicktime Player, and RealOne Player. In some embodiments, media player **202** decompresses the streaming video or audio using a codec and plays it back on a display of terminal **102**. Media player **202** can be used as a stand alone application or embedded in a web page to create a video application interacting with HTML content. Further, media player **202** can communicate with adaptive bitrate manager **108** by sending RTCP receiver reports.

Buffer **204** (also known as terminal buffer **204**) is a software program and/or a hardware device that temporarily stores multimedia packets before providing the multimedia packets to media player **202**. In some embodiments, buffer **204** receives the multimedia packets from adaptive bitrate manager **108** via network **106**. These packets can be configured based on the real-time transport protocol (RTP). In some embodiments, buffer **204** receives the multimedia packets from a device other than adaptive bitrate manager **108**. Once buffer **204** receives multimedia packets, it can provide the stored multimedia packets to media player **202**. While FIG. **2** illustrates that terminal buffer **204** and media player **202** are separate components, one of ordinary skill the art will appreciate that terminal buffer **204** can be a part of media player **202**. Further, while FIG. **2** illustrates only a single buffer, one of ordinary skill the art will appreciate that multiple buffers can exist, for example, one or more buffers for audio media packets and one or more buffers for video media packets.

Adaptive bitrate controller **210** of adaptive bitrate manager **108** is a software program and/or hardware device that preriodically receives RTCP receiver reports from terminal **102** and provides an optimal session bitrate to be used during the next period for encoding multimedia data to be sent to terminal **102**. In some embodiments, adaptive bitrate controller **210** includes a buffer for storing the current and previous RTCP receiver reports. To compute the optimal session bitrate, adaptive bitrate controller **210** uses one or more network state estimators for estimating the state of the streaming media network and computing the optimal session bitrate to be used in the next RTCP interval. For example, these network state estimators can estimate a media time in transit (MTT), a bitrate received at terminal **102**, a round trip time estimate (RTTE), and a packet loss count. Adaptive bitrate controller **210** can use the history and statistics of the estimator to implement different control algorithms to compute the optimal session bitrate. Further, adaptive bitrate controller **210** may update the optimal session bitrate by determining the stability of the streaming media network. This can be done by checking the newly computed estimators for compliance to one or more stability criterion. Using the estimations and the stability criterion, adaptive bitrate controller **210** can determine whether to adjust the outgoing bitrate or keep the current outgoing bitrate unchanged for the next period. After this determination, adaptive bitrate controller **210** provides the optimal session bitrate value to variable bitrate encoder **214**.

Buffer **212** of adaptive bitrate manager **108** is a software program and/or a hardware device that temporarily stores media data before providing the media data to variable bitrate encoder **214**. In some embodiments, buffer **212** receives the media data from one or more content servers **112-114** via network **110**. In some embodiments, buffer **212** receives the media data from a device other than content servers **112-114**.

Variable bitrate encoder **214** of adaptive bitrate manager **108** is a software program and/or hardware device that receives optimal session bitrate data from adaptive bitrate controller **210** and provides, to RTP packetization **216**, audio and/or video data that are encoded at a bitrate matching the optimal session bitrate provided by adaptive bitrate controller **210**. Variable bitrate encoder can include, among other things, a bitrate splitter **220**, an audio encoder **222**, a video encoder **224**, and, for some embodiments, a frame dropper **226**.

Bitrate splitter **220** is a software program and/or a hardware device that receives the optimal session bitrate data from adaptive bitrate controller **210** and allocates optimal bitrates to be used when encoding the audio and video media data during the next interval. The allocation is such that the summation of bitrates for all tracks, when combined, can be substantially equal to the optimal session bitrate specified by adaptive bitrate controller **210**. For example, this allocation could be based on a predetermined allocation, user preference, optimal performance data, privileging one type of data over the other, the amount of audio and video data to be provided, and/or any combination of the above. For example, bitrate splitter **220** may privilege audio quality in a way that if a reduced bitrate is specified, bitrate splitter **220** will reduce the video bitrate first and postpone reducing the audio bitrate as much as possible.

Audio encoder **222** and video encoder **224** are software programs and/or hardware devices that receive their respective bitrate allocation from bitrate splitter **220** and provide outgoing media data encoded to match the bitrate of their respective bitrate allocation for the next RTCP interval. Both audio encoder **222** and video encoder **224** receive their respective media data from buffer **212** and output this media data according to its respective bitrate allocation from bitrate splitter **220**. After the bitrate has been determined for both audio and video, it is the responsibility of each encoder to deliver maximum quality in the corresponding media track. For example, audio encoder **222** can generate variable bitrates by adjusting spectral quantization and cutoff frequency. Further, video encoder **224** can generate variable bitrates, for example, by adjusting Discrete Cosine Transform (DCT) coefficient quantization or by introducing frame dropping. This frame dropping can be executed, when needed, by frame dropper **226**.

Frame dropper **226** is a software program and/or a hardware device that can be triggered when the desired bitrate is less than a quality threshold. This threshold can be codec dependent, and represents the bitrate value below which the use of coarser quantization leads to intolerable artifacts in the image. Frame dropper **226** can dynamically determine a frame dropping rate based on the desired video bitrate and the bitrate being generated by video encoder **224**. To compensate inherent bitrate fluctuations in the video bitrate at the output of the encoder, frame dropper **226** can dynamically update the dropping rate by using a sliding window covering the byte size history of recently encoded frames.

RTP packetization **216** is a software program and/or a hardware device that receives the audio and video media data from audio encoder **222** and video encoder **224** and translates this data into a packet format. RTP defines a standardized packet format for delivering audio and video over the Internet. Besides carrying the audio and media data, these packets can include, among other things, a payload-type identifier for identifying the type of content, a packet sequence number, time stamping for allowing synchronization and jitter calculations, and delivery monitoring data. This type of data can later assist adaptive bitrate controller **210** in determining the quality of service provided by the network when adaptive bitrate controller **210** receives a corresponding RTCP receiver report from terminal **102**. Upon translating this data into a packet format, RTP packetization **216** transmits the data through network buffer **230** of network **106** to terminal buffer **204** of terminal **102**. In addition adaptive bitrate manager **108** saves the history of sent RTP packets in the audio and video tracks. This history data can include, among other things, the time that each packet is sent, the sequence number, and the size of each RTP packet.

FIG. **3** is a functional diagram illustrating an exemplary communication flow in the system of FIG. **2**. It is assumed for purposes of explaining this exemplary embodiment that terminal **102** has already received at least some of the media data of the requested media data package. Further, it is assumed that the media data package includes both audio and video media data. After receiving packets, media player **202** transmits (**302**) an RTCP receiver report to adaptive bitrate manager **108**.

RTCP is a protocol for providing quality control information for an RTP flow, such as the transmission provided by RTP packetization **216** of adaptive bitrate manager **108**. More specifically, RTCP partners with RTP packetization **216** of adaptive bitrate manager **108** in the delivery and packaging of multimedia data. In some embodiments, media player **202** periodically transmits the RTCP receiver report. RTCP receiver report can provide feedback on the quality of service being provided by RTP packetization **216**. While RTP/RTCP is used as an exemplary embodiment to explain the adaptive bitrate control method, one of ordinary skill could appreciate that this adaptive bitrate control method is applicable to any protocol that fulfills the functions of media transport with sequencing and timing information and media transport feedback with information about received packets (covering sequencing, timing, loss rate, etc.)

Further, in some embodiments, the receiver report can be a single report having both audio and video report data or it can be separated into multiple reports (such as in the RTCP case), for example, such as a receiver report for audio report data and another receiver report for video report data. The receiver report data can include, among other things, data regarding the sequence number of the most recently received RTP packet at terminal **102**, the timestamp of the last packet received by terminal **102** reported in the RTCP receiver report, the number of bits sent from this report, a round trip time, and a number of packets lost.

After receiving the receiver report, adaptive bitrate controller **210** can estimate the state of the network for determining whether to update the session bitrate for the next period. Adaptive bitrate controller **210** can save the newly received receiver report in a cumulative history and record the time at which the packet was received. To estimate the state of the network, adaptive bitrate controller **210** can combine data from the received RTCP receiver report, the previously received RTCP receiver reports stored by the adaptive bitrate manager **108**, and the history of sent RTP packets stored by adaptive bitrate manager **108**. Adaptive bitrate controller can estimate the following exemplary data by using network state estimators:

- Media Time in Transit (MTT), computed as the difference between the timestamp of the most recently sent RTP packet and the timestamp of the last RTP packet received by the player reported in RTCP receiver report;
- Bitrate received, computed as the bits received between the current and previously received RTCP receiver reports, divided by the time elapsed between these two receiver reports. The bits received between receiver reports are computed by cross referencing sequence numbers in the receiver report with the history of bytes sent stored at adaptive bitrate manager **108**;
- Round Trip Time Estimate (RTTE) can be obtained by averaging a number of the lower MTT values stored at the adaptive bitrate manager **108**. For example, RTTE could be calculated by averaging the lowest 3 MTT values out of all stored MTT values for that streaming media network. Further, adaptive bitrate manager **108** can calculate the RUE from data within an RTCP sender report. While these exemplary embodiments are illustrated, any method can be used to estimate a round trip time for the streaming media network; and
- Packet Loss count, captured directly from RTCP receiver report.

Adaptive bitrate controller **210** can use these estimates to implement several different control algorithms. For example, the Streaming Media stability criterion can be used to compute the session bitrate for the next RTCP interval.

Adaptive bitrate controller **210** uses the stability criterion to determine the stability of the streaming media network. While any number of algorithms can be used to determine the stability, one exemplary embodiment compares the estimated MTT with the RTTE. If the MTT and the RTTE remain close, adaptive bitrate controller **210** can determine that the streaming media network can properly support the current bitrate. Further, by comparing the bitrate received with the current bitrate session, adaptive bitrate controller **210** can determine that the network can cope with the load imposed by adaptive bitrate manager **108**.

Adaptive bitrate controller **210** uses the estimations and the stability criterion to implement control algorithms for discovering the network capacity and adjusting the session bitrate accordingly. Adaptive bitrate controller **210** can define the variations of the control algorithms to operate in two different modes: (1) acquisition mode and (2) normal mode. While two modes have been illustrated in this exemplary embodiment, one of ordinary skill in the art will appreciate that multiple modes of operation can be defined.

In the normal mode, adaptive bitrate controller **210** operates in the steady state condition, indicating that the network is either maintaining or incrementally increasing the effective capacity seen by the system. In some embodiments, while operating in normal mode, the control algorithms can increase the session bitrate while the MTT is not increasing and the bitrate received remains close to the current session bitrate.

Adaptive bitrate controller **210** generally triggers the acquisition mode when it detects high packet loss, a sudden increase in the MTT, and/or a value of the MTT higher than a threshold (MTT threshold), which can be a fixed value or can be obtained dynamically for an adaptive control mechanism. Once triggered, acquisition mode sets the optimal session bitrate to a value, such as the bitrate received or a fraction of the received bitrate. Because the bitrate received can be the best estimation of the actual bitrate that the network can support at that particular point in time, adaptive bitrate manager **108** should quickly return back to a stable condition. In some embodiments, the new session bitrate is simply set to be a fraction of the current session bitrate.

In this embodiment, while only terminal **102** is illustrated for communicating with adaptive bitrate manager **108**, one of ordinary skill in the art will appreciate that multiple terminals can communicate with adaptive bitrate manager **108**, where each of the terminals can be located in substantially different network environments. Such environments can vary significantly, as different underlying wireless technologies and fixed network topologies can be used. Therefore, for some embodiments, it may be desirable to discover characteristics of the network environment beforehand so that key parameters in the framework are adjusted automatically. For example, adaptive bitrate controller **210** could set the MTT threshold at the beginning of the multimedia session to a value correlated to the RTTE. In this way, the system can attempt to follow the general stability criterion provided by adaptive bitrate controller **210**. As indicated above, this stability criterion could be based on, independent of the network environment (a prior unknown), the comparison between the MTT and the RTTE, which is largely advantageous given that the actual network infrastructure type can rarely be determined a priori. In some embodiments, the optimal session bitrate can be updated by determining the difference between the MTT and the RTTE and adjusting the session bitrate according to the difference. For example, the larger the difference, the greater adjustment from the current session bitrate to an optimal session bitrate. In some embodiments, the MTT used for this determination can be based on the one or more historical values of MTT.

Using the control algorithms to compute a session bitrate update as described above, adaptive bitrate controller **210** determines an optimal session bitrate for transmitting media data to terminal **102**. Adaptive bitrate controller **210** provides (**304**) the optimal session bitrate data to bitrate splitter **220** of variable bitrate encoder **214**. Upon receiving the optimal session bitrate data, bitrate splitter **220** allocates the optimal session bitrate between the audio and video streams. For example, this allocation could be based on a predetermined allocation, a user preference optimal performance data, privileging one type of data over the other, the amount of audio and video data to be provided, and/or any combination of the above. For example, bitrate splitter **220** may privilege audio quality in a way that if a reduced bitrate is specified, bitrate splitter **220** reduces the video bitrate first and postpones reducing the audio bitrate as much as possible.

After splitting the optimal session bitrate into an optimal audio bitrate and an optimal video bitrate, bitrate splitter provides (**306**) the optimal audio bitrate to audio encoder **222** and provides (**308**) the optimal video bitrate to video encoder **224**. Upon receiving their respective bitrate, both audio encoder **222** and video encoder **224** receive their respective media data from buffer **212** and output their respective audio media data and video media data according to the respective bitrate allocation from bitrate splitter **220**. After the bitrate has been determined for both audio and video, it is the responsibility of each encoder to deliver maximum quality in the corresponding media track by maintaining the requested bitrate until the next RTCP interval. For example, audio encoder **222** can generate variable bitrates by adjusting quantization and cutoff frequency. Further, video encoder **224** can generate variable bitrates, for example, by adjusting Discrete Cosine Transform (DCT) coefficient quantization or by introducing frame dropping. This frame dropping can be executed, when needed, by frame dropper **226**. In some embodiments, the encoding parameters of the encoders are not modified until they receive optimal bitrate data from bitrate splitter **220**, which would be provided in a subsequent RTCP interval, because the encoders **222**, **224** are slave devices to bitrate splitter **220**.

In some embodiments, where frame dropping is preferred, video encoder **224** can provide (**310**) the video media data to frame dropper **226** when the optimal session bitrate is less than a quality threshold. This threshold can be codec dependent, and represents the bitrate value below which the use of coarser quantization leads to intolerable artifacts in the image. When frame dropping is triggered, frame dropper **226** can dynamically determine a frame dropping rate based on the desired video bitrate and the bitrate being generated by video encoder **224**. To compensate inherent bitrate fluctuations in the video bitrate at the output of video encoder **224**, frame dropper **226** can dynamically update the dropping rate by using a sliding window covering the byte size history of recently encoded frames. Frame dropper **226** can drop the frames accordingly to deliver the optimal session bitrate. In addition, in some embodiments, video encoder **224** can utilize the network state estimator of adaptive bitrate controller **210** to encode video in a more resilient manner. In some embodiments, packet loss information can be used in conjunction with the MTT by video encoder **224** to determine if a Group of Picture (GOP) value should be reduced, increasing the number of I-Frames per second sent in the video stream. In some embodiment, if frame dropping is not needed, video encoder **224** can simply provide the video media data to RTP packetization **216**. Audio encoder **222** and, for this embodiment, frame dropper **226** provide (**312**, **314**) the audio media data and the video media data, respectively, to RTP packetization **216**.

Upon receiving the audio media data and the video media data, RTP packetization **216** translates this data into a packet format. RTP defines a standardized packet format for delivering audio and video over the Internet. Upon translating this data into a packet format, RTP packetization **216** transmits (**316**) the audio and video media packets to network buffer **230** of network **106**. While only one transmission is shown, one of ordinary skill in the art will appreciate that transmission **316** can include separate transmission for one or more audio media packets and another for one or more video media packets. Furthermore, one of ordinary skill in the art will appreciate that network **106** can include multiple networks, each having their own one or more buffers. Besides carrying the audio and media data, these packets can include, among other things, a payload-type identifier, a packet sequence number, a timestamp, and delivery monitoring data. This type of data can later assist adaptive bitrate controller **210** in determining the quality of service provided by the network when adaptive bitrate controller **210** receives the RTCP receiver report from terminal **102**. Moreover, adaptive bitrate manager **108** can also store a history of sent RTP packets so that it can later adjust the bitrate accordingly.

Upon receiving the packets, network buffer **230** of network **106** can store the packets until it is the packets turn to be provided to terminal **102**. While only buffer **230** is illustrated, one of ordinary skill in the art will appreciate that one or more separate buffers can exist for each of the audio media packets and the video media packets. When it is the packets turn, network buffer **230** transmits (**318**) the packets to terminal buffer **204**.

Upon receiving the packets, terminal buffer **204** of terminal **102** can store the packets until it is the packets turn to be provided to media player **202**. While only buffer **230** is illustrated, one of ordinary skill in the art will appreciate that one or more separate buffers can exist for each of the audio media packets and the video media packets. When it is the packets turn, buffer **204** provides (**320**) the packets to media player **202**. In turn, media player **202** can extract the relevant data out of packets and provide this data to adaptive bitrate manager **108** in a subsequent RTCP receiver report.

FIG. **4** is a flowchart representing an exemplary method for processing an RTCP packet. Referring to FIG. **4**, it will be readily appreciated by one of ordinary skill in the art that the illustrated procedure can be altered to delete steps or further include additional steps. It is assumed for this exemplary method that RTCP packet includes data concerning both audio and video media data. While both types exists, one of ordinary skill in the art will appreciate that RTCP data can include either audio or video data. After initial start step **400**, an adaptive bitrate manager obtains (**402**) RTCP data, which can include one or more RTCP receiver reports. This RTCP data can correlate to the quality and quantity of audio and video media packets received at a media player of a terminal. The RTCP data can include, among other things, a sequence number of a last packet received by the terminal, a timestamp corresponding to such packet, a number of bits sent, a round trip time, and number of packets lost during a transmission from the adaptive bitrate manager to the terminal. RTCP data can be obtained by receiving an RTCP report from the terminal and by cross-correlating the contents of the last received RTCP packet with the history of RTP packets stored at the adaptive bitrate manager.

After receiving RTCP data, the adaptive bitrate manager estimates (**404**) network conditions of a streaming media network. To estimate the state of the network, the adaptive bitrate manager can combine data from the received RTCP data from step **402** and previously received RTCP data stored by the adaptive bitrate manager. Adaptive bitrate controller can estimate an MITT, a bitrate received, an RTTE, and a packet loss. The adaptive bitrate manager can use these estimates to implement several different control algorithms.

After estimating the network conditions, the adaptive bitrate manager applies (**406**) stability criterion to determine the stability of the streaming media network. If needed, the stability criterion can assist in adjusting the bitrate for attempting to stabilize the streaming media network, e.g., such as avoiding buffer overflows in the network and underflows at the terminal. While any number of algorithms can be used to determine the stability criterion, one exemplary embodiment compares the estimated MTT with the estimated RTTE, both of which are estimated in step **404**. If the MTT and the RTTE remain close, the adaptive bitrate manager can use this comparison to determine that the streaming media network can properly support the current bitrate. Further, by comparing the bitrate received with the current bitrate session, the adaptive bitrate manager can determine that the streaming media network can cope with the load.

After establishing the stability criterion, the adaptive bitrate manager determines (**408**) whether the network is stable with respect to the current bitstream based on estimation step **404** and/or stability criterion establishment step **406**. If the network is stable, the adaptive bitrate manager operates (**410**) in a steady state condition by either maintaining or incrementally increasing the current bitrate. In some embodiments, the optimal session bitrate can be computed by determining the difference between the MTT and the RTTE and adjusting the session bitrate according to the difference. For example, if the current session bitrate is less than a set target session bitrate, the adaptive bitrate manager can incrementally increase the optimal session bitrate if the values of the MTT and the RTTE are comparable. Then, the adaptive bitrate manager provides (**416**) an optimal session bitrate for transmitting media data to a terminal. After providing step **416**, the method can proceed to end **418**.

If determining that the network is not stable, the adaptive bitrate manager adjusts (**412**) the bitrate so that adaptive bitrate manager can reach a stable condition. For example, in some embodiments, the adaptive bitrate manager can use the estimated bitrate received from step **404** because, in some embodiments, the bitrate received can be the best estimation of the actual bitrate that the network can support at that particular point in time. Then, the adaptive bitrate manager provides (**416**) the optimal session bitrate for transmitting media data to the terminal. After providing step **416**, the method can proceed to end **418**.

FIG. **5** is a flowchart representing an exemplary method for processing optimal session bitrate data. Referring to FIG. **5**, it will be readily appreciated by one of ordinary skill in the art that the illustrated procedure can be altered to delete steps or further include additional steps. It is assumed for this exemplary method that both audio and video media data exists. While both types exists, one of ordinary skill in the art will appreciate that either audio or video data can exist. After initial start step **500**, an adaptive bitrate manager obtains (**502**) optimal session bitrate data for transmitting media data to a terminal.

Upon receiving the optimal session bitrate data, the adaptive bitrate manager allocates (**504**) the optimal session bitrate between audio and video streams to produce an optimal audio bitrate and an optimal video bitrate. For example, this allocation could be based on a predetermined allocation, user preference, optimal performance data, privileging one type of data over the other, the amount of audio and video data to be provided, and/or any combination of the above. For example, the adaptive bitrate manager may privilege audio quality in a way that if a reduced bitrate is specified, the adaptive bitrate manager can reduce the video bitrate first and postpone reducing the audio bitrate as much as possible.

Adaptive bitrate manager obtains (**506**) audio and video media data. In some embodiments, obtaining step **506** can occur prior to allocating step **504** or obtaining step **502**. After allocating step **504** and obtaining step **506**, the adaptive bitrate manager encodes (**508**) the audio and video media data according to their respective allocated bitrate specified at step **504**.

After encoding the audio and video streams according to the allocated bitrate, the adaptive bitrate manager provides (**510**) the encoded audio and video media data for transmitting to the terminal. In some embodiments, an RTP packetization receives the encoded audio and video media data and translates this data into a packet format. RTP defines a standardized packet format for delivering audio and video over the Internet. Upon translating this data into a packet format, the RTP packetization can then transmit the audio and video media packets to the terminal. After providing the encoded audio and video media data, the method can proceed to end **512**.

The methods disclosed herein may be implemented as a computer program product, i.e., a computer program tangibly embodied in an information carrier, e.g., in a machine readable storage device or in a propagated signal, for execution by, or to control the operation of, data processing apparatus, e.g., a programmable processor, a computer, or multiple computers. A computer program can be written in any form of programming language, including compiled or interpreted languages, and it can be deployed in any form, including as a stand alone program or as a module, component, subroutine, or other unit suitable for use in a computing environment. A computer program can be deployed to be executed on one computer or on multiple computers at one site or distributed across multiple sites and interconnected by a communication network.

In the preceding specification, the invention has been described with reference to specific exemplary embodiments. It will however, be evident that various modifications and changes may be made without departing from the broader spirit and scope of the invention as set forth in the claims that follow. The specification and drawings are accordingly to be regarded as illustrative rather than restrictive. Other embodiments of the invention may be apparent to those skilled in the art from consideration of the specification and practice of the invention disclosed herein.

What is claimed is:

1. A method comprising:

receiving a receiver report from a terminal;

estimating one or more network conditions of a media network using the receiver report;

determining an optimal session bitrate using the estimated one or more network conditions, wherein determining the optimal session bitrate further comprises:

determining stability criterion using the estimated one or more network conditions, wherein determining stability criterion includes at least one of:

comparing a media time in transit and a round trip time estimate; and

comparing a bitrate received with a current bitrate session; and

determining the stability of the media network; and

providing the optimal session bitrate based at least in part on the media-network-stability determination; and

providing media data to the terminal according to the optimal session bitrate.

2. The method of claim **1**, further comprising maintaining or incrementally increasing a current bitrate when the stability of the media network is considered normal.

3. The method of claim **1**, further comprising adjusting a current bitrate when the stability of the network is not normal.

4. The method of claim **1**, further comprising allocating the optimal session bitrate between audio media data and video media data to produce an optimal audio bitrate and an optimal video bitrate; and encoding audio and video media data according to the optimal audio bitrate and the optimal video bitrate.

5. The method of claim **4**, wherein providing media data to the terminal includes providing the encoded audio media data and the encoded video media data based on the optimal audio bitrate and the optimal video titrate.

6. A method comprising:

receiving a receiver report from a terminal;

estimating one or more network conditions of a media network using the receiver report;

determining stability criterion, wherein determining stability criterion comprises at least one of:

comparing a media time in transit and a round trip time estimate; and

comparing a bitrate received with a current bitrate session; and

determining the stability of the media network using the determined stability criterion;

controlling a session bitrate based at least in part on the media-network-stability determination; and

providing the session bitrate to an encoder for transmitting media data according to the provided session bitrate.

7. The method of claim **6**, wherein controlling the session bitrate includes maintaining or incrementally increasing a current bitrate when the stability of the media network is considered normal.

8. The method of claim **6**, wherein controlling the session bitrate includes adjusting a current bitrate when the stability of the network is considered not normal.

9. A method comprising:

receiving an optimal session bitrate;

allocating the optimal session bitrate between audio and video media to produce an optimal audio bitrate and an optimal video bitrate, wherein allocating the optimal session bitrate between audio and video media is based at least in part on privileging either the audio media or the video media over the other;

encoding audio and video media data according to the optimal audio bitrate and the optimal video bitrate; and

providing the encoded audio and video data for transmittal to a terminal.

10. The method of claim **9**, further comprising dropping frames of the encoded video data.

11. A system comprising:

a terminal, having a media player, configured to provide a receiver report; and

an adaptive bitrate manager configured to:

receive the receiver report,

estimate one or more network conditions using the receiver report, determine stability criterion using the estimated one or more network conditions, wherein determine stability criterion includes at least one of:

comparing a media time in transit and a round trip estimate, and

comparing a bitrate received with a current bitrate session, and

determine the stability of the media network,

determine an optimal session bitrate based at least in part on the media-network-stability determination, and

provide media data to the terminal according to the optimal session bitrate.

12. The system of claim **11**, wherein the adaptive titrate manager further comprises an adaptive titrate controller for receiving the receiver report and calculating an optimal session bitrate.

13. The system of claim **12**, wherein the adaptive bitrate manager further comprises an encoder for obtaining the optimal session bitrate, allocating the optimal session titrate between audio and video media to produce an optimal audio titrate and an optimal video titrate, encoding audio and video data according to the optimal audio titrate and the optimal video bitrate, and providing the encoded audio and video data for transmittal to the terminal.

14. A non-transitory computer readable storage medium storing instruction that, when executed by a computer, cause the computer to perform a method for processing a receiver report, the method comprising:

receiving the receiver report from a terminal;

estimating one or more network conditions of a media network using the receiver report;

determining stability criterion, wherein determining stability criterion comprises at least one of:

comparing a media time in transit and a round trip time estimate; and

comparing a bitrate received with a current bitrate session; and

determining the stability of the media network using the determined stability criterion;

controlling a session bitrate based at least in part on the media-network-stability determination; and

providing the session bitrate to an encoder for transmitting media data according to the provided session bitrate.

15. A non-transitory computer readable storage medium storing instruction that, when executed by a computer, cause the computer to perform a method for processing an optimal session bitrate, the method comprising:

receiving the optimal session bitrate;

allocating the optimal session bitrate between audio and video media to produce an optimal audio bitrate and an optimal video bitrate, wherein allocating the optimal session bitrate between audio and video media is based at least in part on privileging either the audio media or the video media over the other;

encoding audio and video media data according to the optimal audio bitrate and the optimal video bitrate; and

providing the encoded audio and video data for transmittal to a terminal.

16. A terminal comprising:

a buffer configured to receive media data packets transmitted by an adaptive bitrate manager over a media network; and

a media player configured to receive media data packets and provides a receiver report to the adaptive bitrate manager configured to:

receive the receiver report,

estimate one or more network conditions of the media network using the receiver report,

determine stability criterion using the estimated one or more network conditions, wherein determine stability criterion includes at least one of:

comparing a media time in transit and a round trip time estimate, and

comparing a bitrate received with a current bitrate session; and

determine the stability of the media network,

determine an optimal session bitrate based at least in part on the media-network-stability determination, and

provide media data to the buffer according to the optimal session bitrate.

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