IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re patent of Melnyk et al.

U.S. Patent 7,987,285

Filed: July 9, 2008

Issued: July 26, 2011

Title: ADAPTIVE BITRATE MANAGEMENT FOR STREAMING MEDIA OVER PACKET NETWORKS

REQUEST FOR *EX PARTE* REEXAMINATION

Attorney Docket No.: OPT285

Customer No.: 165774

REQUEST FOR *EX PARTE* REEXAMINATION OF U.S. PATENT 7,987,285

Mail Stop "*Ex Parte* Reexam"
Attn: Central Reexamination Unit
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Dear Commissioner:

Pursuant to the provisions of 35 U.S.C. §§ 301-307, Unified Patents, LLC ("Requester") hereby requests an *ex parte* reexamination of claims 1, 6, 9-11, 14, and 15 (the "Challenged Claims") U.S. Patent 7,987,285 ("the '285 Patent," EX1001), which issued on July 26, 2011 to Miguel A. Melnyk *et al.* from U.S. Patent Application No. 12/170,347 filed on July 9, 2008. The '285 Patent claims priority to provisional application No. 60/948,917, filed July 10, 2007. The '285 Patent is currently assigned to OptiMorphix, Inc. ("OptiMorphix" or "Patent Owner"). The assignment to OptiMorphix is recorded in the U.S. Patent and Trademark Office ("USPTO") at reel/frame 064020/0183.

Requester submits that this Request presents prior art references and analysis that are non-cumulative of the prior art that was before the Examiner during the original prosecution of the '285 Patent and that claims 1, 6, 9-11, 14, and 15 of the '285 Patent are unpatentable over these references. Requester therefore requests that an order for reexamination and an Office Action rejecting claims 1, 6, 9-11, 14, and 15 be issued.

Ex Parte Patent Reexamination Filing Requirements

Pursuant to 37 C.F.R. § 1.510(b)(l), statements pointing out at least one substantial new question of patentability ("SNQ") based on material, non-cumulative reference patents and printed publications for the Challenged Claims of the '285 Patent are provided in Section I of this Request.

Pursuant to 37 C.F.R. § 1.510(b)(2), reexamination of the Challenged Claims of the '285 Patent is requested, and a detailed explanation of the pertinence and manner of applying the cited references to the Challenged Claims is provided in Section II and Exhibits AA-1, AA-2, and BB provided with this Request.

Pursuant to 37 C.F.R. § 1.510(b)(3), copies of every patent or printed publication relied upon or referred to in the statement pointing out each substantial new question of patentability or in the detailed explanation of the pertinence and manner of applying the cited references are provided as Exhibits 1001-1012 of this Request.

Pursuant to 37 C.F.R. §1.510(b)(4), a copy of the '285 Patent is provided as Exhibit 1001 of this Request, along with a copy of any disclaimer, certificate of correction, and reexamination certificate issued corresponding to the patent.

Pursuant to 37 C.F.R. § 1.510(b)(5), the attached Certificate of Service indicates that a copy of this Request, in its entirety, has been served on Patent Owner at the following address of record for Patent Owner, in accordance with 37 C.F.R. § 1.33(c):

109619-Citrix Systems, Inc./Finnegan
901 New York Avenue NW
Washington, DC 20001-4413
United States

Pursuant to 37 C.F.R. § 1.510(b)(6), Requester hereby certifies that the statutory estoppel provisions of 35 U.S.C. § 315(e)(l) and 35 U.S.C. § 325(e)(l) do not prohibit Requester from filing this *ex parte* patent reexamination request.

Also submitted herewith is the fee set forth in 37 C.F.R. § 1.20(c).

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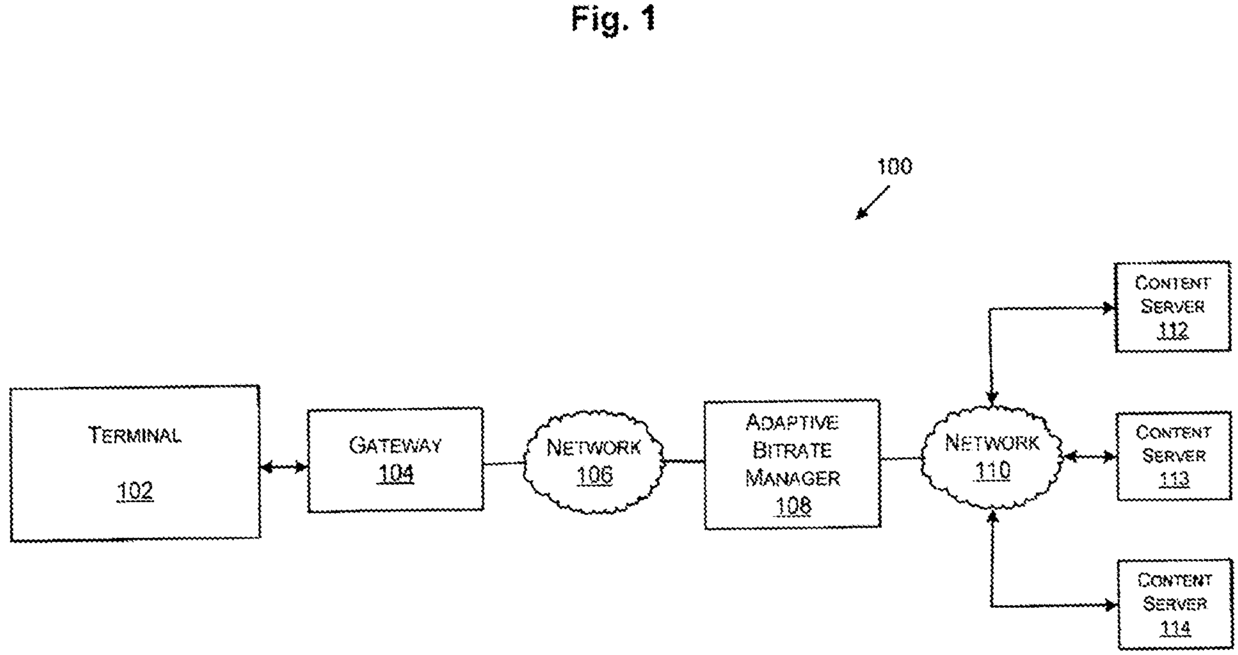
I. SUBSTANTIAL NEW QUESTIONS OF PATENTABILITY

Prior to describing the substantial new questions of patentability presented in this Request, provided below is an overview of the ’285 Patent and its prosecution, a discussion of claim construction, and a summary of the prior art in the present Request. Copies of the ’285 Patent and its file history are provided as Exhibits 1001 and 1002, respectively.

A. U.S. Patent 7,987,285

1. Summary

The ’285 Patent is generally directed to “adaptive bitrate management for streaming media over packet networks.” EX1001, Title. In general, the ’285 Patent includes two different sets of claims, both of which are included within the Challenged Claims. First, claims 9, 10, and 15 (which are challenged in the Request in SNQs 1-3) generally relate to a method for receiving an optimal session bitrate; allocating that bitrate between audio and video data, where one of audio or video data is privileged over the other; and encoding and transmitting the data. *See* EX1001, claim 9. Second, claims 1, 6, 11, and 14 (which are challenged in the Request in SNQ 4) also relate to transmitting media—but instead of privileging audio or video bitrate like in claim 9, these claims include many additional limitations related to “estimating one or more network conditions” and “determining stability criterion,” for example, by “comparing a media time in transit and a round trip time estimate” and “comparing a bitrate received with a current bitrate.” EX1001, claim 1. In this claim set, the optimal bit rate is explicitly “based at least in part on the media-network-stability criterion.” *Id.* Aspects of both claim sets will be summarized in the following paragraphs. The ’285 Patent’s system is shown in Figure 1:



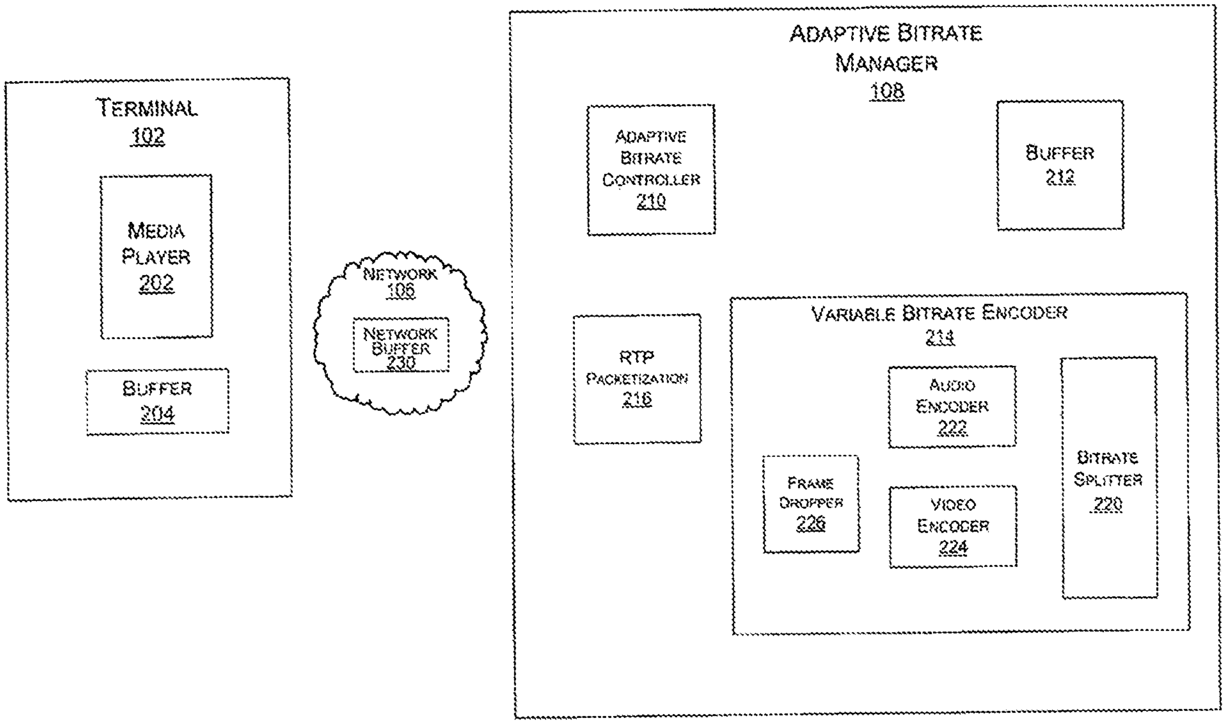
EX1001, FIG. 1.

The '285 Patent acknowledges that "[r]ate control is essential for media streaming over packet networks" and that 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." EX1001, 1:14-20. The '285 Patent identifies several problems (e.g., packet loss, reduction in bandwidth, incomplete network information, and low bitrates) as being challenges to overcome in this area. EX1001, 1:32-2:15.

Regarding the claimed optimal session bitrate, the '285 Patent describes exemplary ways of computing it; for example, the '285 Patent describes that 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," using well known network state estimators like media time in transit (MTT) or round trip time estimate (RTTE), which are explicitly claimed in claim 1 and its related claims. EX1001, 4:22-29. In other words, the '285 Patent's system and method analyze the state of the connection between a transmitting server (i.e., content servers 112-114) and a receiving terminal (i.e., terminal 102) to obtain a bitrate suitable for streaming media. *See* EX1001, 2:48-3:44.

Relevant particularly to the first claim set (claims 9, 10, and 15), Figure 2 of the '285 Patent depicts a more detailed view of adaptive bitrate manager 108:

Fig. 2

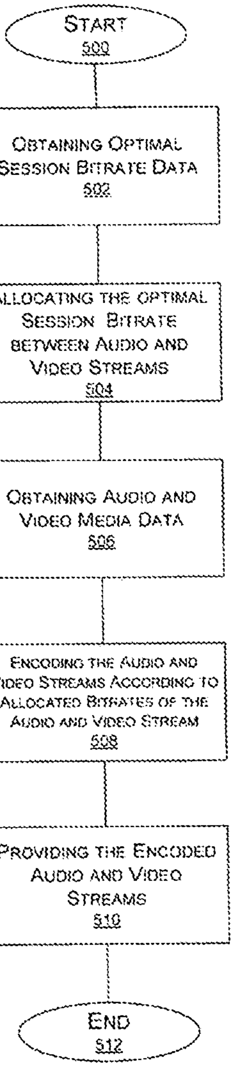


EX1001, FIG. 2.

For example, relevant to claim 9's limitations about splitting the optimal session bitrate between audio and video, the '285 Patent explains that variable bit rate encoder 214 "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." EX1001, 4:50-58. The encoder may also include a bitrate splitter 220, which "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." EX1001, 4:59-5:7. Notably, 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." *Id.* The '285 Patent provides an example where "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." *Id.*; *see also* EX1001, 8:3-42 (further describing the allocation of an optimal bitrate between audio and video).

Next, once an audio and video bitrate are determined, "it is the responsibility of each encoder to deliver maximum quality in the corresponding media track." EX1001, 5:16-18. Once encoded, the audio and video data are transmitted through the network. *See* EX1001, 5:38-58. Claim 9 and its related claims are also generally depicted and described by the '285 Patent in relation to Figure 5:

Fig. 5



EX1001, FIG. 5; *see also id.*, 10:52-11:26 (describing the operation of the method in Figure 5).

Relevant to claim 1 and its related claims, the ’285 Patent describes that its system may receive a “receiver report,” which “can provide feedback on the quality of service being provided by RTP packetization.” EX1001, 6:1-16. In an example, the receiver report includes information such as “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.” EX1001, 6:16-27. After receiving this report, the adaptive bitrate controller “can estimate the state of the network for determining whether to update the session bitrate for the next period.” EX1001, 6:28-40. As noted in claim 1, the adaptive bitrate controller can make network estimations (e.g., media time in transit, bitrate received, a round trip time estimate, and packet loss) to determine network stability and an optimal bitrate. EX1001, 6:38-67. These stability criteria are then used to determine the stability of the network and adjust the session bitrate accordingly. EX1001, 7:1-20; *see also id.*, FIG. 4.

These aspects of the ’285 Patent (e.g., transmitting audio and video content at an optimal bitrate, and determining network conditions), however, were all well known prior to the time of the ’285 Patent, as established and explained in connection with the prior art references in this Request—all of which were never presented during the ’285 Patent’s prosecution. *See* EX1002.

2. Prosecution History

U.S. Patent Application No. 12/170,347 filed on July 9, 2008 (claiming priority to provisional application No. 60/948,917, filed July 10, 2007), and matured into the ’285 Patent. *See* EX1001, EX1002. The ’285 Patent issued on July 26, 2011. The application received a notice of missing parts (*see* EX1002, 56-57), which was responded to, followed by a non-final Office Action (*see id.*, 135-153; rejecting the claims over the Chou and Birch references). The Applicant’s response to this Office Action did not include amendments, but rather argued the cited references had several deficiencies rendering them inapplicable to the claims (*see id.*, 164-182). Following this, the Examiner issued another non-final Office Action (*see id.*, 186-212) using the same references as the previous Office Action. The Applicant responded (*see id.*, 257-276), providing only clerical amendments to the claims (*see id.*, 261-262), and otherwise continuing to argue the Birch and Chou references were not applicable to the claims. A terminal disclaimer was filed by the Applicant (*see id.*, 306-308). Following this, a Notice of Allowance issued (*see id.*, 314-331) following a call from the Applicant, and included several substantial Examiner’s Amendments to the claims, while also cancelling several pending claims (*see id.*, 323-329). The stated reasons for allowance (*see id.*, 329-330) included most limitations from amended claim 1.

None of the references used in this Request were cited in any IDS or otherwise used at all during prosecution of the '285 Patent itself. *See* EX1002.

However, Requester notes that, while it was never cited at all in the prosecution of the '285 Patent, U.S. Patent Publication 2005/0021830 to Eduardo Urzaiz et al. ("Urzaiz," EX1005 – used in SNQ3) was cited by the Office in Office Actions of at least two child patent applications to the '285 Patent. For example, Urzaiz was used in Office Actions of U.S. Patent Application Nos. 13/557,086 and 14/077,139. Nonetheless, the single ground using Urzaiz (SNQ3) should not be discretionarily denied (e.g., under 35 U.S.C. § 325(d)). First, Urzaiz was never cited or applied in the context of the claims of the '285 Patent during its prosecution whatsoever. The child applications (where Urzaiz was applied) have different claims and different claim scope as compared to the Challenged Claims of the '285 Patent in this Request. Second, regardless of any arguments as to Urzaiz itself made in the child applications, this Request presents two new prior art references in SNQ3 combined together with Urzaiz, and these two new references were also not before the Examiner related to the '285 Patent's prosecution or the child applications at all. Accordingly, as it relates to SNQ3, the same or substantially the same arguments have not been previously presented to the Office with respect to the '285 Patent at all, nor have they been subject to any final decisions. *See* 35 U.S.C. § 325(d). Therefore, discretionary denial of SNQ3 would be inappropriate.

3. Challenged Claims

For the sake of reference, the claims for which reexamination is requested (1, 6, 9-11, 14, and 15) are reproduced below. Claims 1, 6, 9, 11, 14, and 15 are independent claims, while claim 10 depends on claim 9.

Claim 1

[1P] A method comprising:

[1a] receiving a receiver report from a terminal;

[1b] estimating one or more network conditions of a media network using the receiver report;

[1c] determining an optimal session bitrate using the estimated one or more network conditions, wherein determining the optimal session bitrate further comprises:

[1d] determining stability criterion using the estimated one or more network conditions, wherein determining stability criterion includes at least one of:

[1e] comparing a media time in transit and a round trip time estimate; and

[1f] comparing a bitrate received with a current bitrate session; and

[1g] determining the stability of the media network; and

[1h] providing the optimal session bitrate based at least in part on the media-network-stability determination; and

[1i] providing media data to the terminal according to the optimal session bitrate.

Claim 6

[6P] A method comprising:

[6a] receiving a receiver report from a terminal;

[6b] estimating one or more network conditions of a media network using the receiver report;

[6c] determining stability criterion, wherein determining stability criterion comprises at least one of:

[6d] comparing a media time in transit and a round trip time estimate; and

[6e] comparing a bitrate received with a current bitrate session; and

[6f] determining the stability of the media network using the determined stability criterion;

[6g] controlling a session bitrate based at least in part on the media-network-stability determination; and

[6h] providing the session bitrate to an encoder for transmitting media data according to the provided session bitrate.

Claim 9

[9P] A method comprising:

[9a] receiving an optimal session bitrate;

[9b] allocating the optimal session bitrate between audio and video media to produce an optimal audio bitrate and an optimal video bitrate,

[9c] 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;

[9d] encoding audio and video media data according to the optimal audio bitrate and the optimal video bitrate; and

[9e] providing the encoded audio and video data for transmittal to a terminal.

Claim 10

[10P] The method of claim 9, further comprising

[10a] dropping frames of the encoded video data.

Claim 11

[11P] A system comprising:

[11a] a terminal, having a media player, configured to provide a receiver report; and

[11b] an adaptive bitrate manager configured to:

[11c] receive the receiver report,

[11d] estimate one or more network conditions using the receiver report,

[11e] determine stability criterion using the estimated one or more network conditions, wherein determine stability criterion includes at least one of:

[11f] comparing a media time in transit and a round trip estimate, and

[11g] comparing a bitrate received with a current bitrate session, and

[11h] determine the stability of the media network,

[11i] determine an optimal session bitrate based at least in part on the media-network-stability determination, and

[11j] provide media data to the terminal according to the optimal session bitrate.

Claim 14

[14P] 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:

[14a] receiving the receiver report from a terminal;

[14b] estimating one or more network conditions of a media network using the receiver report;

[14c] determining stability criterion, wherein determining stability criterion comprises at least one of:

[14d] comparing a media time in transit and a round trip time estimate; and

[14e] comparing a bitrate received with a current bitrate session; and

[14f] determining the stability of the media network using the determined stability criterion;

[14g] controlling a session bitrate based at least in part on the media-network-stability determination; and

[14h] providing the session bitrate to an encoder for transmitting media data according to the provided session bitrate.

Claim 15

[15P] 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:

[15a] receiving the optimal session bitrate;

[15b] allocating the optimal session bitrate between audio and video media to produce an optimal audio bitrate and an optimal video bitrate,

[15c] 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;

[15d] encoding audio and video media data according to the optimal audio bitrate and the optimal video bitrate; and

[15e] providing the encoded audio and video data for transmittal to a terminal.

B. Claim Construction

The claims must be given their "broadest reasonable interpretation consistent with the specification." *See* MPEP § 2258; *In re Yamamoto*, 740 F.2d 1569, 222 USPQ 934 (Fed. Cir. 1984). This Request presents its claim analysis in a manner that is consistent with the broadest reasonable interpretation consistent with the specification.

Requester submits that no terms of the '285 Patent warrant construction beyond their ordinary and customary meaning, or alternatively, only require construction to the extent necessary to determine whether the prior art teaches the claims. *Nidec Motor Corp. v. Zhongshan Broad Ocean Motor Co.*, 868 F.3d 1013, 1017 (Fed. Cir. 2017). In the claim charts, Requester provides comments on certain claim terms to explain how those terms are used in the specification of the '285 Patent; however, these comments are merely informative as to what the claim terms encompass under the broadest reasonable interpretation, and Requester does not propose any express constructions.

C. Listing of Prior Art Patents and Printed Publications

The following six references present several substantial new questions of patentability to the '285 Patent (having an earliest claimed priority date of July 10, 2007):

- **EX1004 ("van Beek"): U.S. Patent Publication 2005/0071876** to Petrus J. L. van Beek was filed on September 30, 2003 and published on March 31, 2005. Accordingly, van Beek qualifies as prior art to the '285 Patent under at least 35 U.S.C. §§ 102(a) and (b) (pre-AIA).
- **EX1005 ("Urzaiz"): U.S. Patent Publication 2005/0021830** to Eduardo Urzaiz et al. was filed on September 13, 2002 and published on January 27, 2005. Accordingly, Urzaiz qualifies as prior art to the '285 Patent under at least 35 U.S.C. §§ 102(a) and (b) (pre-AIA).
- **EX1006 ("Gupta"): U.S. Patent 7,734,800** to Anoop Gupta et al. was filed on August 25, 2003, published on February 26, 2004, and issued on June 8, 2010. Accordingly, Gupta qualifies as prior art to the '285 Patent under at least 35 U.S.C. §§ 102(a) and (b) (pre-AIA).

- **EX1007 ("Pogrebinsky"): U.S. Patent 7,142,506** to Vladimir Pogrebinsky was filed on February 2, 1999 and issued on November 28, 2006. Accordingly, Pogrebinsky qualifies as prior art to the '285 Patent under at least 35 U.S.C. §§ 102(a) and (e) (pre-AIA).
- **EX1011 ("Yano"): U.S. Patent Publication 2003/0037158** to Koichi Yano et al. was filed on August 20, 1998 and published on February 20, 2003. Accordingly, Yano qualifies as prior art to the '285 Patent at least under 35 U.S.C. §§ 102(a) and (b) (pre-AIA).
- **EX1012 ("Ogawa") U.S. Patent Publication 2006/0218264** to Akimichi Ogawa et al. was filed on March 22, 2006 and published on September 28, 2006. Accordingly, Ogawa qualifies as prior art to the '285 Patent at least under 35 U.S.C. §§ 102(a) and (e) (pre-AIA).

D. Level of Skill in the Art

A person of ordinary skill in the art ("POSITA") as of the earliest claimed priority date for the '285 Patent (i.e., July 10, 2007) would have had (i) a Bachelor's degree (or higher degree) in electrical engineering, and/or computer engineering, or equivalent and (ii) at least two years of experience working in the fields of signal processing, media (i.e., image, video, audio) processing, compression, and/or media transmission. Additional industry experience or technical training may offset less formal education, while advanced degrees or additional formal education may offset lesser levels of professional experience. EX1003, ¶¶ 28-30.[1]

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[1] Requester submits the declaration of Dr. Lina J. Karam (EX1003), an expert in the field of the '285 Patent. EX1003, ¶¶ 10, 30. Dr. Karam was a POSITA of the '285 Patent as of its earliest claimed priority date. EX1003, ¶¶ 28-30.

E. The Prior Art References, Arguments, and Evidence Present a Substantial New Question of Patentability

As shown below, Requester submits that the prior art references cited herein raise a new "substantial question of patentability" because "the teaching of the (prior art) patents and printed publications is such that a reasonable examiner would consider the teaching to be important in deciding whether or not the claim is patentable." *See* MPEP 2242.

The references discussed below when considered as an ordered combination, teach each limitation of the Challenged Claims, including the ideas of transmitting audio and video content at an optimal bitrate, and determining current network conditions that influence transmission bitrates, as recited in claims 1, 6, 9-11, 14, and 15. Further, the "same question of patentability as to the claim has not been decided by the Office in an earlier concluded examination or review of the patent" at least because none of the combinations of art referenced in this request was before the Office during prosecution of the '285 Patent or the subject of a final written decision in a prior post-grant proceeding challenging the claims of the '285 Patent, any of which that are known to Requester have been listed below in Section III, *infra*.

1. Overview of the SNQs

This Request presents several substantial new questions of patentability (SNQs) for resolution. These SNQs are referred to using a reference number for convenience herein (e.g., SNQ1, SNQ2, etc.). The SNQs presented in this Request are the following:

- **SNQ1:** Claims 9, 10, and 15 would have been anticipated under pre-AIA 35 U.S.C. §102 by van Beek.
- **SNQ2:** Claims 9, 10, and 15 would have been obvious under pre-AIA 35 U.S.C. §103 by van Beek.
- **SNQ3:** Claims 9, 10, and 15 would have been obvious under pre-AIA 35 U.S.C. §103 by Urzaiz, Gupta, and Pogrebinsky.
- **SNQ4:** Claims 1, 6, 11, and 14 would have been obvious under pre-AIA 35 U.S.C. §103 by Yano and Ogawa.

The following sections provide a brief description of how each reference and/or combination of references presents a substantial new question of patentability that should be considered, and are explained in full detail in **Exhibits AA-1, AA-2,** and **BB**.

2. SNQs 1-2: van Beek Anticipates (SNQ1) or Renders Obvious (SNQ2) Claims 9, 10, and 15 and Presents a Substantial New Question of Patentability

a) Overview of van Beek; van Beek Presents a Substantial New Question of Patentability

van Beek (EX1004, U.S. Patent Publication 2005/0071876) discloses a "transmission systems [*sic*] suitable for video" that provides "transmission of multiple data streams in a network that may have limited bandwidth." van Beek, Abstract, [0041]. The system is shown in Figure 1:

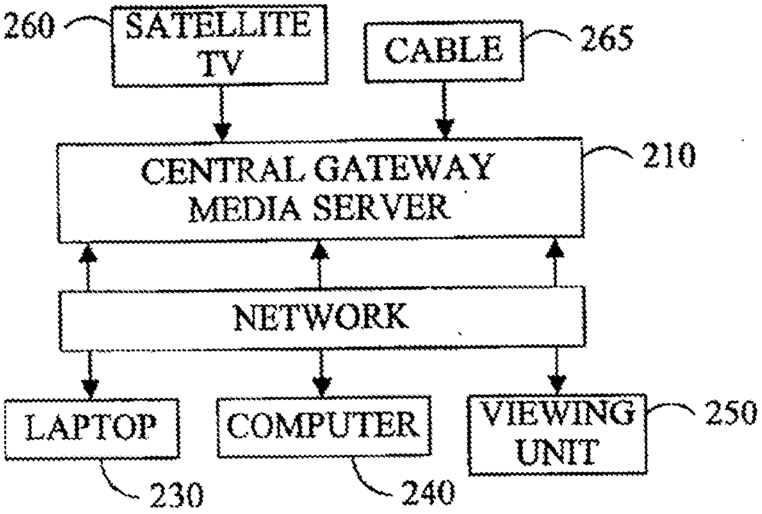


FIG. 1

EX1004, FIG. 1.

van Beek recognized that "[d]ifferent devices interconnected to the network have different resources and different usage paradigms. For example, different devices may have different microprocessors, different memory requirements, different display characteristics, different connection bandwidth capabilities, and different battery resources. . . . This results in unpredictable and dynamically varying network maximum throughput." van Beek, [0051]. van Beek acknowledged that a "more optimal approach to rate adaptation of multiple streams is to apply joint bit allocation/rate control." van Beek, [0124].

Like the '285 Patent, van Beek is also squarely directed at optimizing the transmission of audio and video data, disclosing that the improvements in its disclosure are directed towards "optimizing the quality of the AV data continuously, in real-time" and "adapting to the unpredictable and dynamically changing conditions of the network." van Beek, [0062]-[0066]. Similarly, van Beek describes that its "goal is to find the best set of output rates . . . that maximizes the overall quality of all output streams or, equivalently, minimizes an overall distortion criterion D, while the aggregate rate of all streams is within the capacity of the channel." van Beek, [0136].

To accomplish these goals, van Beek describes that its system estimates maximum bandwidth and throughput (*see* van Beek, [0221], [0225]) and adjusts the bitrate for various streams accordingly (*see* van Beek, [0124]-[0136]). And like the '285 Patent, van Beek acknowledges that for different (i.e., audio/video streams), its system "may attempt to allocate an equal amount of available bits to each stream," "attempt to allocate the available bits such that the quality of each stream is approximately equal," or "allow users to assign different priorities to different streams." van Beek, [0107]-[0108]. Specifically, van Beek noted that "the audio and video streams of an audiovisual stream may be separated and treated differently during their transmission," and like the '285 Patent (*see* EX1001, 4:59-5:7), "the audio part of an audiovisual stream may be assigned a higher priority than the video part." van Beek, [0121].

van Beek is analogous art to the '285 patent. It is in the same field of endeavor as the '285 Patent because both relate to streaming media content at an adaptive (or optimal) bitrate over a network connection. *See* EX1001, Title ("Adaptive Bitrate Management for Streaming Media Over Packet Networks"), 2:31-47 ("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 includes . . . the ability to implement joint session bitrate management for audio, video and/or other streams simultaneously."); *see* van Beek, [0062]-[0066] ("Accordingly a system that includes dynamic rate adaption is suitable to accommodate distribution of high quality audio/video streams over networks that suffer from significant dynamic variations in performance."), Abstract ("An adaptive bandwidth system on the gateway media server 210 determines the network bandwidth characteristics and adjusts the bandwidth for the output data streams in accordance with the bandwidth characteristics."); *see also* van Beek, [0086]-[0090], [0107]-[0108]; Karam Decl.

(EX1003), ¶ 44.

Furthermore, van Beek is reasonably pertinent to at least one problem that would have concerned the inventor of the '285 Patent. The '285 Patent purported to solve problems related to "rate control" and the "challenge in delivering bandwidth-intensive content like multimedia over capacity-limited, shared links," i.e., "quickly respond[ing] to changes in network conditions by adjusting the bitrate and the media encoding scheme to optimize the viewing and listening experience of the user." EX1001, 1:14-31. Likewise, van Beek seeks to solve problems in the prior art by providing an "adaptive bandwidth system [that] determines the network bandwidth characteristics and adjusts the bandwidth for the output data streams in accordance with the bandwidth characteristics." van Beek, [0041]. van Beek further provides a "system may robustly stream audio/visual data over (wireless) networks by: (1) optimizing the quality of the AV data continuously, in real-time; and (2) adapting to the unpredictable and dynamically changing conditions of the network." van Beek, [0062]-[0066]; Karam Decl. (EX1003), ¶ 45.

The summary and discussion above, together with **Exhibit AA-1**, supports a finding that van Beek discloses or renders obvious claims 9, 10, and 15, and a reasonable Examiner would consider the teachings of van Beek in determining whether claims 9, 10, and 15 are patentable at least because the reference was never considered during the prosecution of the '285 Patent. Accordingly, van Beek raises a substantial new question of patentability as to claims 9, 10, and 15 of the '285 Patent that should be resolved through reexamination.

3. SNQ3: Urzaiz in view of Gupta and Pogrebinsky Renders Claims 9, 10, and 15 Obvious and Presents a Substantial New Question of Patentability

a) Overview of Urzaiz; Urzaiz Presents a Substantial New Question of Patentability

Urzaiz (EX1005, U.S. Patent Publication 2005/0021830) discloses a "data transmission method . . . in which one or more data streams are transmitted at respective transmission rates." Urzaiz, Abstract. Urzaiz discloses that "data to be streamed is multi-media data such as, for example, audio and video data." Urzaiz, [0004]. Figure 3 of Urzaiz depicts a server transmitting audio and video data at separate bitrates to a client:

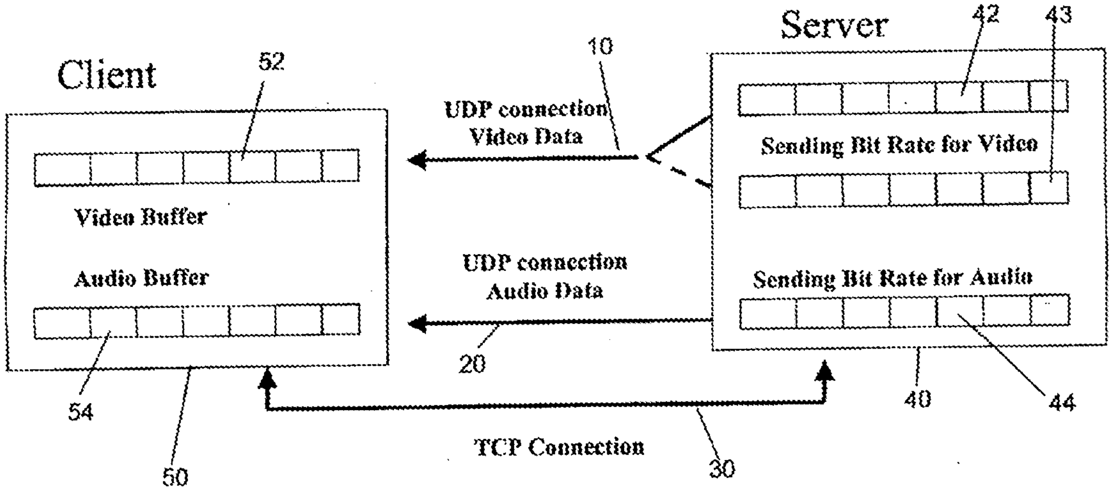


Fig.3

Urzaiz, FIG. 3.

Like the '285 Patent's "adaptive bitrate controller 210" (which, per the specification, is used to compute the optimal session bitrate), Urzaiz describes that "at step 2 [of Figure 11] sending rate calculator 46 calculates the total bandwidth available for all of the individual data streams which are to be transmitted from the server computer 40." Urzaiz, [0125]. Urzaiz explains that this "value total_rate represents the upper limit on transmission rate which the individual transmission rates of each separate data stream when summed together should not be greater." *Id.*

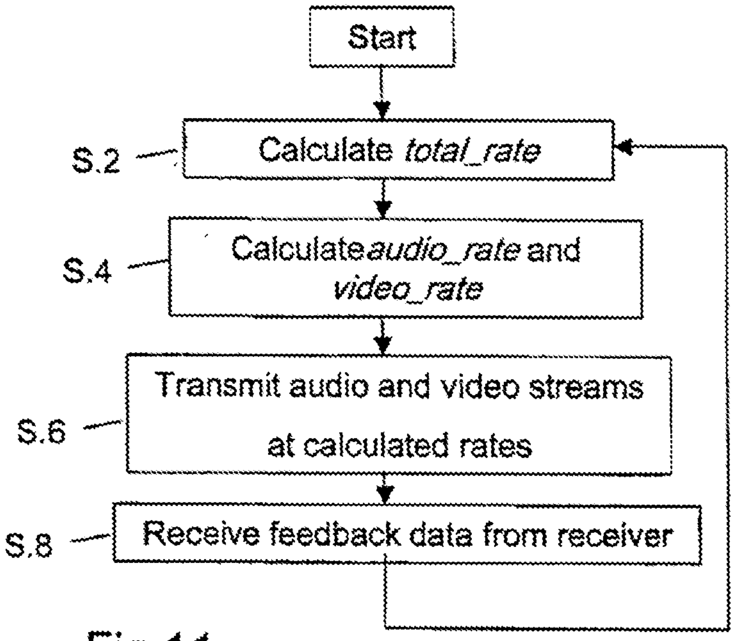


Fig.11

Urzaiz, FIG. 11.

Also like the ’285 Patent, Urzaiz describes that its server is “capable of calculating the maximum transmission rate available for the stream dependent upon the present conditions on the network, thereby optimising the transmission rate at which the stream is transmitted.” Urzaiz, [0035]. Urzaiz explains that in its system, “it becomes possible to control the respective audio sending rates and video sending rates to trade bit rate from one stream to the other depending upon the respective audio and video decode rates in the receiver.” Urzaiz, [0132]. For example, as described with respect to Figure 11, “[f]ollowing the calculation of the available total transmission rate, at step S4 the sending rate calculator 46 in the server calculates the individual transmission rates for each data stream, being in the second embodiment the transmission rate of the audio UDP stream (audio_rate) and the transmission rate of the video UDP stream (video_rate).” Urzaiz, [0127].

Urzaiz is analogous art to the ’285 patent. It is in the same field of endeavor as the ’285 Patent because both relate to streaming media content at an adaptive (or optimal) bitrate over a network connection. *See* EX1001, Title (“Adaptive Bitrate Management for Streaming Media Over Packet Networks”), 2:31-47 (“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 includes . . . the ability to implement joint session bitrate management for audio, video and/or other streams simultaneously."); *see* Urzaiz, [0125] (describing use of a "sending rate calculator" that "calculates the total bandwidth available for all of the individual data streams which are to be transmitted from the server computer"), [0127] (calculating transmission rates for audio and video streams), [0035] ("optimising the transmission rate at which the stream is transmitted"); Karam Decl. (EX1003), ¶ 53.

Furthermore, Urzaiz is reasonably pertinent to at least one problem that would have concerned the inventor of the '285 Patent. The '285 Patent purported to solve problems related to "rate control" and the "challenge in delivering bandwidth-intensive content like multimedia over capacity-limited, shared links," i.e., "quickly respond[ing] to changes in network conditions by adjusting the bitrate and the media encoding scheme to optimize the viewing and listening experience of the user." EX1001, 1:14-31. Likewise, Urzaiz seeks to solve problems in the prior art by providing a "sending rate calculator [for] calculat[ing] the total bandwidth available for all of the individual data streams which are to be transmitted from the server computer." Urzaiz, [0125]. Urzaiz seeks to optimize transmission rates "dependent upon the present conditions on the network," just like the '285 Patent. Urzaiz, [0035]; Karam Decl. (EX1003), ¶ 54.

The summary and discussion above, together with **Exhibit AA-2**, supports a finding that Urzaiz discloses or renders obvious claims 9, 10, and 15, and a reasonable Examiner would consider the teachings of Urzaiz in determining whether claims 9, 10, and 15 are patentable at least because the reference was never considered during the prosecution of the '285 Patent. Accordingly, Urzaiz (together with Gupta and Pogrebinsky, as explained in **Exhibit AA-2**) raises a substantial new question of patentability as to claims 9, 10, and 15 of the '285 Patent that should be resolved through reexamination.

b) Overview of Gupta; Gupta Presents and Substantial New Question of Patentability

Gupta (EX1006, U.S. Patent 7,734,800) discloses a composite media stream containing an audio stream and a video stream. Gupta, Abstract, 3:34-54, 7:43-54. While Gupta shares many aspects in common with the '285 Patent, it is primarily relied upon for its disclosure of assigning the audio stream and video stream different priorities (i.e., with respect to the quality and bandwidth priority assigned to a given stream) when streaming media.

For example, in its system, Gupta describes that "[e]ach stream is assigned a priority. Audio will generally have a high priority. The high-priority streams are given priority when allocating bandwidth. Thus, in the example above, the audio stream is streamed to the client at its full quality, while the video stream is reduced in quality to fit within the remaining bandwidth." Gupta, 12:60-13:12; *see also id.*, 11:37-57 (the system "selects a combination of individual media streams that provides the best quality while requiring no more than the available bandwidth"). Thus, much like the example in the '285 Patent (seeking to optimize audio quality—*see* EX1001, 4:59-5:7), Gupta seeks to optimize audio quality while still remaining within the available bandwidth of the connection.

Gupta is analogous art to the '285 patent. It is in the same field of endeavor as the '285 Patent because both relate to streaming media content at an adaptive (or optimal) bitrate over a network connection. *See* EX1001, Title ("Adaptive Bitrate Management for Streaming Media Over Packet Networks"), 2:31-47 ("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 includes . . . the ability to implement joint session bitrate management for audio, video and/or other streams simultaneously."); *see* Gupta, 13:4-23 (describing a "method of bandwidth utilization" that prioritizes an audio stream while reducing video stream quality "to fit within the remaining bandwidth"), 7:43-54 ("The composite media stream has a plurality of individual media streams as described above. For purposes of discussion, it is assumed in this example that the composite media stream has an audio stream and a video stream."); Karam Decl. (EX1003), ¶ 57.

Furthermore, Gupta is reasonably pertinent to at least one problem that would have concerned the inventor of the '285 Patent. The '285 Patent purported to solve problems related to "rate control" and the "challenge in delivering bandwidth-intensive content like multimedia over capacity-limited, shared links," i.e., "quickly respond[ing] to changes in network conditions by adjusting the bitrate and the media encoding scheme to optimize the viewing and listening experience of the user." EX1001, 1:14-31. Likewise, Gupta seeks to solve problems in the prior art by providing a system that "selects a combination of individual media streams that provides the best quality while requiring no more than the available bandwidth." Gupta, 11:37-57. Gupta seeks to optimize audio quality, just like a problem identified by the '285 Patent. Gupta, 12:60-13:12; Karam Decl. (EX1003), ¶ 58.

The summary and discussion above, together with **Exhibit AA-2**, supports a finding that Gupta discloses or renders obvious claims 9, 10, and 15, and a reasonable Examiner would consider the teachings of Gupta in determining whether claims 9, 10, and 15 are patentable at least because the reference was never considered during the prosecution of the '285 Patent. Accordingly, Gupta (together with Urzaiz and Pogrebinsky, as explained in **Exhibit AA-2**) raises a substantial new question of patentability as to claims 9, 10, and 15 of the '285 Patent that should be resolved through reexamination.

c) Overview of Pogrebinsky; Pogrebinsky Presents a Substantial New Question of Patentability

Pogrebinsky (EX1007, U.S. Patent 7,142,506) discloses a system "for adjusting of bit rate transmission in a communication network . . . in accordance with the network state detected" in the context of a "multimedia call." Pogrebinsky, Abstract. Pogrebinsky, like the '285 Patent, is concerned with the transmission of audio and video data packets over a communication network, such as the Internet. Pogrebinsky, 1:6-11.

Pogrebinsky describes the use of an "allocator" that is coupled to an "audio bit rate control device" and a "video bit rate control device." Pogrebinsky, 4:9-44, FIG. 2. Pogrebinsky's system includes a "network monitor 22, that monitors the network 1, for its condition, in real time, by receiving the network state, and at least one of the media bit rate controllers." Pogrebinsky, 4:9-44. Pogrebinsky also includes an "audio bit rate control device 19 and video bit rate control device 20 [with] hardware and software for adjusting the bit rate transmission to the available bandwidth of network 1." *Id.*

Thus, while Pogrebinsky also shares many aspects in common with the '285 Patent, like Gupta, it is also relied upon primarily for its disclosure prioritizing an audio channel when adjusting bit rate transmission during a multimedia call. For example, Pogrebinsky discloses that its "allocator 21 will know the total bit rate available, such that it can allocate bit rate between the audio and video bit rate controllers 19, 20, at step 242. In making the allocation, priority will always be given to the audio channel, such that the minimum bit rate for the audio is in accordance with the bit rates of the table FIG. 4." Pogrebinsky, 8:44-64.

Pogrebinsky is analogous art to the '285 patent. It is in the same field of endeavor as the '285 Patent because both relate to streaming media content at an adaptive (or optimal) bitrate over a network connection. *See* EX1001, Title ("Adaptive Bitrate Management for Streaming Media

Over Packet Networks"), 2:31-47 ("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 includes . . . the ability to implement joint session bitrate management for audio, video and/or other streams simultaneously."); *see* Pogrebinsky, Abstract ("[A]n apparatus and methods for adjusting of bit rate transmission in a communication network by monitoring the state of the network, detecting the state of the network and transmitting a multimedia call over the network in accordance with the network state detected."); Karam Decl. (EX1003), ¶ 62.

Furthermore, Pogrebinsky is reasonably pertinent to at least one problem that would have concerned the inventor of the '285 Patent. The '285 Patent purported to solve problems related to "rate control" and the "challenge in delivering bandwidth-intensive content like multimedia over capacity-limited, shared links," i.e., "quickly respond[ing] to changes in network conditions by adjusting the bitrate and the media encoding scheme to optimize the viewing and listening experience of the user." EX1001, 1:14-31. Likewise, Pogrebinsky seeks to solve problems in the prior art by providing an "apparatus and methods for adjusting of bit rate transmission in a communication network by monitoring the state of the network." Pogrebinsky, Abstract; *see also id.*, 2:13-18 (describing "bit rate adjustment according to the network available band width and state (condition), for improving received media quality at the receiver"); Karam Decl. (EX1003), ¶ 63.

The summary and discussion above, together with **Exhibit AA-2**, supports a finding that Pogrebinsky discloses or renders obvious claims 9, 10, and 15, and a reasonable Examiner would consider the teachings of Pogrebinsky in determining whether claims 9, 10, and 15 are patentable at least because the reference was never considered during the prosecution of the '285 Patent. Accordingly, Pogrebinsky (together with Urzaiz and Gupta, as explained in **Exhibit AA-2**) raises a substantial new question of patentability as to claims 9, 10, and 15 of the '285 Patent that should be resolved through reexamination.

d) Motivation to Combine Urzaiz, Gupta, and Pogrebinsky

A POSITA would have combined the teachings of Gupta and Pogrebinsky into Urzaiz's system—such as Gupta's teaching that each stream is assigned a priority, and Pogrebinsky's teaching that priority will always be given to the audio channel—thereby more explicitly disclosing *privileging either the audio media or the video media over the other* in limitation [9c]. Karam Decl. (EX1003), ¶ 64.

All three references are within the same field and highly relevant to one another. Urzaiz discloses a data transmission system where an audio and video stream can be transmitted and respective transmission rates. Urzaiz, Abstract, FIG. 3, [0004]. Similarly, Gupta discloses a composite media stream containing an audio and video stream. Gupta, Abstract, 3:34-54, 7:43-54. And finally, Pogrebinsky discloses a system for transmitting audio and video data packets over a communication network. Pogrebinsky, Abstract, 1:6-11. This similarity is further evidenced by the fact that all three references discuss addressing a similar problem—i.e., adjusting the bitrate of the transmitted media data in response to current network conditions and within the current bandwidth constraints of the network connection. *See* Urzaiz, [0125], [0132]; Gupta, 2:31-47, 13:4-23; Pogrebinsky, 4:9-44, FIG. 2. Accordingly, a POSITA would have readily identified all three references as relating to one another—all concerned with transmitting audio and video streams over a network, factoring in network conditions—and that the teachings of the references would be applicable to one another. Karam Decl. (EX1003), ¶ 65.

A POSITA would have been motivated to incorporate Gupta and Pogrebinsky's general teachings regarding how to prioritize a data stream into Urzaiz's data transmission system, and this would have merely been a matter of combining prior art elements according to known methods to yield predictable results. Specifically, a POSITA would have combined Urzaiz's disclosure of a data transmission system that is capable of transmitting audio and video streams at certain bitrates together with Gupta's teaching that each stream is assigned a priority when allocating bandwidth, and Pogrebinsky's teaching that it would be advantageous to prioritize an audio channel. For example, Urzaiz already provides general disclosure related to encoding and transmitting audio and video data at separate bitrates (i.e., an "audio_rate" and a "video_rate") that are within the total bitrate and bandwidth of a given network connection (i.e., a "total_rate"). Urzaiz, [0124]-[0128], [0004]-[0005], FIGs. 3, 11. Urzaiz also discloses that it is "possible to control the respective audio sending rates and video sending rates to trade bit rate from one stream to the other." Urzaiz, [0132], claim 36. Gupta and Pogrebinsky simply disclose specific situations, and reasons why to prioritize one of a video stream or an audio stream over the other, and, therefore, provide additional implementation details that would be applicable to Urzaiz's system already capable of controlling the bitrates of the streams. Gupta, 12:60-13:12; Pogrebinsky, 8:44-64;

Karam Decl. (EX1003), ¶ 66.

Further, a POSITA would have had a reasonable expectation of success in making the combination, as the proposed combination merely relates to the implementation details of how to prioritize the allocation of bitrates among audio and video data streams, and therefore, would have at most required minor modifications in software that would have yielded predictable results. Karam Decl. (EX1003), ¶ 67.

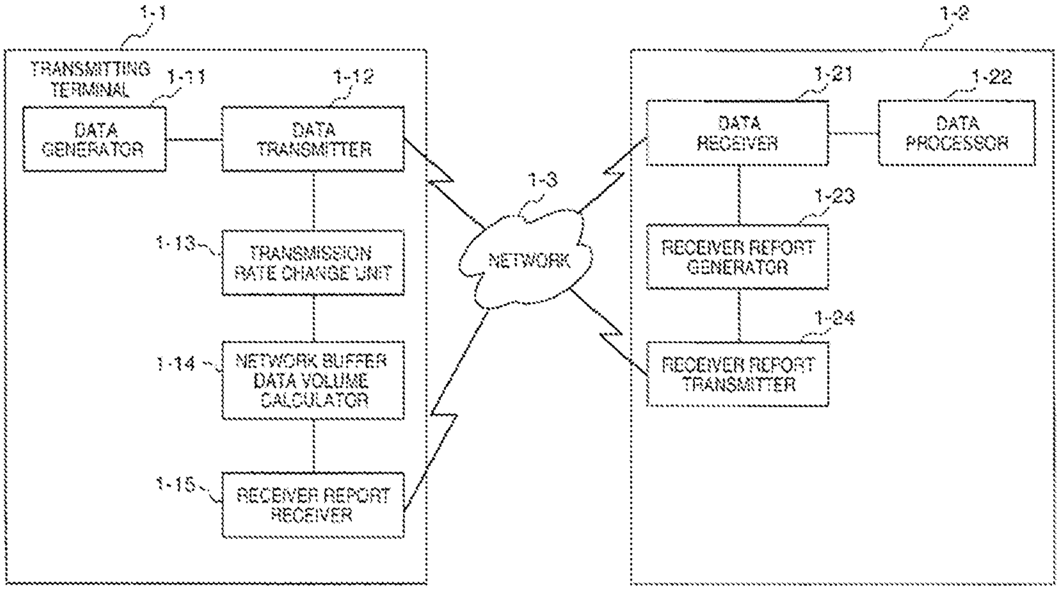
Separately, as it relates to limitation [10a] (*dropping frames of the encoded video data*), a POSITA would have been motivated to combine Gupta's teachings of dropping frames of a video stream to reduce the bitrate and bandwidth being consumed across a network connection with Urzaiz's system for transmitting a video stream. As discussed in the preceding paragraphs, Urzaiz and Gupta are highly similar to one another and a POSITA would have recognized the benefits of combining their teachings. As to limitation [10a], Urzaiz already identified situations where a lower video bitrate (and accordingly, lower bandwidth consumption) may be needed to prevent a video buffer at the receiver from emptying. Urzaiz, [0142]. In view of this, a POSITA would have been motivated to incorporate Gupta's teaching that "[o]ne easy way to reduce bandwidth is to simply drop lower-level dependent frames from the video stream." Gupta, 13:13-23. Thus, Gupta provides a specific solution and implementation details to a general disclosure already present in Urzaiz. And like in the earlier example, a POSITA would have had a reasonable expectation of success in making this combination, as it merely relates to implementation details of how to decrease bandwidth consumption by lowering the bitrate of a video stream by dropping frames, and therefore, would have at most required minor modifications in software that would have yielded predictable results. Karam Decl. (EX1003), ¶ 68.

4. SNQ4: Yano in view of Ogawa Renders Claims 1, 6, 11, and 14 Obvious and Presents a Substantial New Question of Patentability

a) Overview of Yano; Yano Presents a Substantial New Question of Patentability

Yano (EX1011, U.S. Patent Publication 2003/0037158) discloses an invention that "can make data communications at an optimal transfer rate on the basis of the unarrived data volume on a network between two end terminals." Yano, Abstract. This is depicted in Yano's Figure 1:

FIG. 1

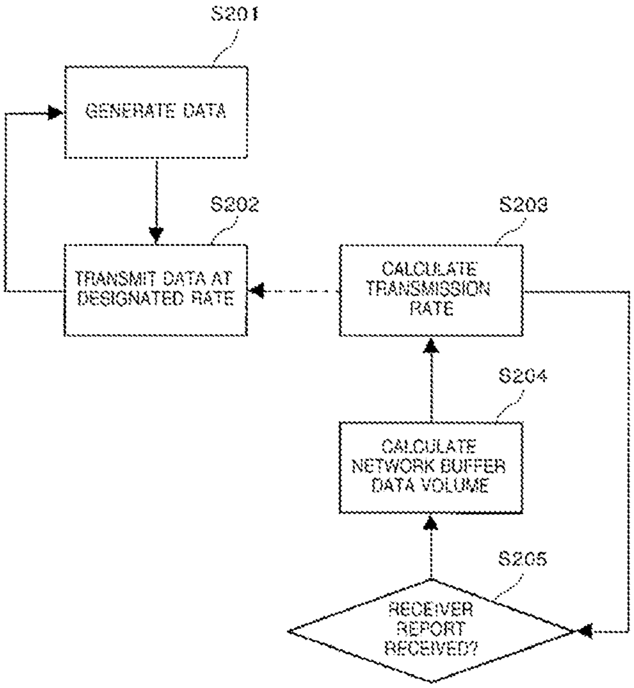


Yano, FIG. 1. For example, Figure 1 of Yano "shows the connection relationship and structure of the respective devices when data transmitted by a transmitting terminal 1-1 is received by a receiving terminal 1-2 via a network 1-3." Yano, [0031].

To accomplish its goal of an optimal transfer rate, "the receiving terminal 1-2 sends back a receiver report. A receiver report receiver 1-15 receives the receiver report, and sends the report contents to a network buffer data volume calculator 1-14." Yano, [0034]. Yano further explains that "[u]pon reception of the receiver report, the transmitting terminal calculates the volume of data which has been output from the transmitting terminal onto the network but has not reached the receiving terminal (step S204)." Yano, [0041]. Further, "the receiving terminal periodically transmits receiver reports to the transmitting terminal while transmitting/receiving data between the transmitting and receiving terminals." Yano, [0051].

Yano notes that the "transmitting terminal determines the transmission rate on the basis of a receiver report sent from the receiving terminal." Yano, [0081]. This is also depicted in Yano's Figure 2:

FIG. 2



Yano, FIG. 2.

Yano discloses various methods of calculating transfer rate using various network conditions. As one example, "transmitting terminal may calculate: (Ts2−Ts1)−(Tr2−Tr1) where Ts1 is the transfer time of data of a given sequence number, Tr1 is the reception time of that data at the receiving side, Tr2 is the transfer time of that receiver report, and Ts2 is the reception time of the receiver report including the sequence number." Yano, [0042]-[0044]. Yano also discloses calculating "data round-trip times" to estimate current network conditions. Yano, [0088]-[0094]; *see also id.*, FIG. 11. In other examples, Yano states that the "transmission rate is determined based on the reception rate." Yano, [0069]-[0071].

Yano is analogous art to the '285 patent. It is in the same field of endeavor as the '285 Patent because both relate to streaming media content at an adaptive (or optimal) bitrate over a network connection while considering network conditions. *See* EX1001, Title ("Adaptive Bitrate Management for Streaming Media Over Packet Networks"), 2:31-47 ("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 includes . . . the ability to implement joint session bitrate management for audio, video and/or other streams simultaneously."); *see* Yano, Abstract (disclosing a system for performing "data communications at an optimal transfer rate on the basis of the unarrived data volume on a network between two end terminals"); Karam Decl. (EX1003), ¶ 82.

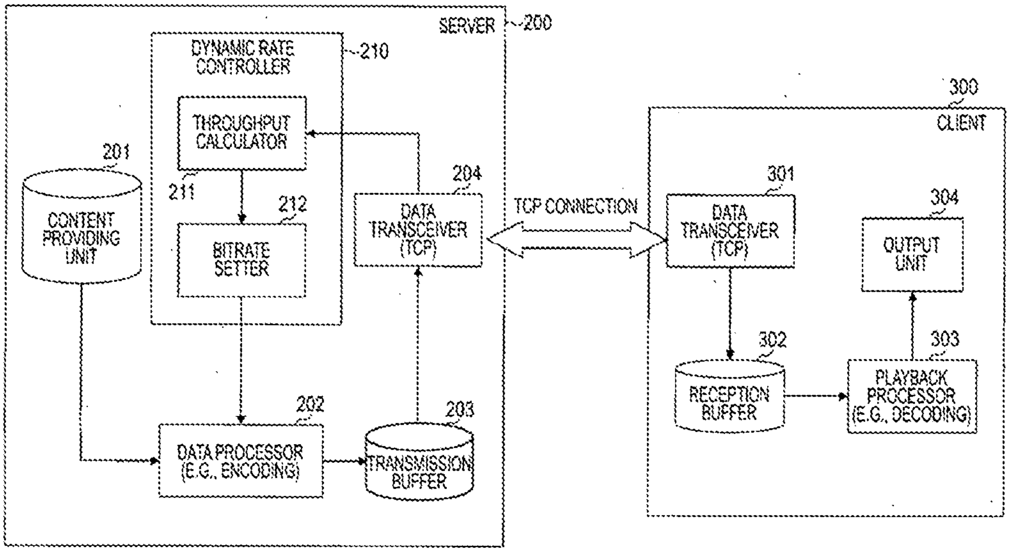
Furthermore, Yano is reasonably pertinent to at least one problem that would have concerned the inventor of the '285 Patent. The '285 Patent purported to solve problems related to "rate control" and the "challenge in delivering bandwidth-intensive content like multimedia over capacity-limited, shared links," i.e., "quickly respond[ing] to changes in network conditions by adjusting the bitrate and the media encoding scheme to optimize the viewing and listening experience of the user." EX1001, 1:14-31. Likewise, Yano seeks to solve problems in the prior art by providing a system that can receive a receiver report (*see* Yano, [0034], [0041], [0051]), where the receiver report includes information related to current network conditions (i.e., round trip times, current bitrate, etc.) and then uses that information to achieve the optimal transfer rate. *See* Yano, [0035], [0041]-[0044], [0088]-[0094], [0069]-[0071], FIGs. 2, 11; Karam Decl. (EX1003), ¶ 83.

The summary and discussion above, together with **Exhibit BB**, supports a finding that Yano discloses or renders obvious claims 1, 6, 11, and 14, and a reasonable Examiner would consider the teachings of Yano in determining whether claims 1, 6, 11, and 14 are patentable at least because the reference was never considered during the prosecution of the '285 Patent. Accordingly, Yano (together with Ogawa, as explained in **Exhibit BB**) raises a substantial new question of patentability as to claims 1, 6, 11, and 14 of the '285 Patent that should be resolved through reexamination.

b) Overview of Ogawa; Ogawa Presents a Substantial New Question of Patentability

Ogawa (EX1012, U.S. Patent Publication 2006/0218264) discloses "a data communication system, and a communication processing method that allow data to be transmitted in an optimal transmission mode in transmission and reception of streaming data." Ogawa, [0015]. "More specifically, it is desirable to provide a communication processing apparatus, a data communication system, and a communication processing method with which a server predicts an optimal value of bitrate of data transmitted in consideration of factors such as congestion on a communication path or disturbance on a communication link and with which the bitrate is dynamically controlled on the basis of the predicted value so that data streaming is carried out in an optimal data transmission mode." Ogawa, [0016]. This is depicted and described in Figure 2 of Ogawa:

FIG. 2



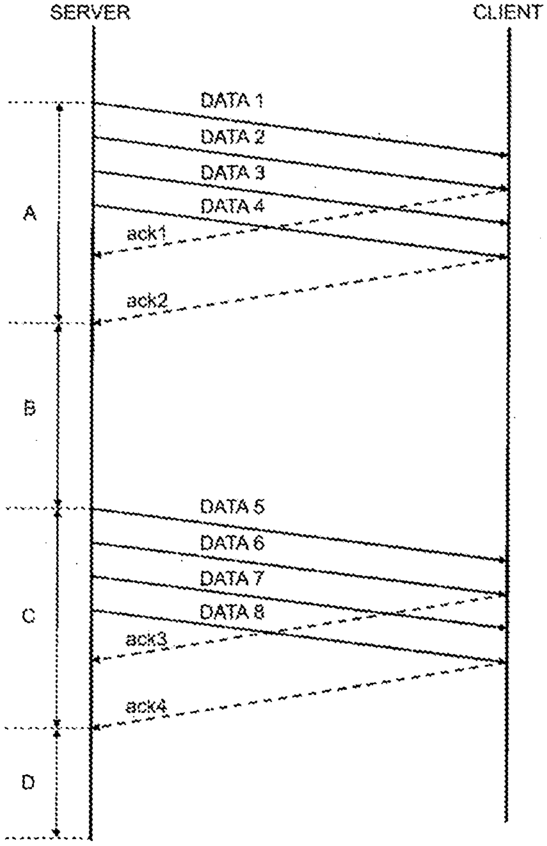
Ogawa, FIG. 2.

For example, "the dynamic rate controller 210 of the server 200 shown in FIG. 2 dynamically controls the transmission bitrate so that the streaming data will be transmitted at an optimal bitrate. The process will be described later more specifically. When the transmission of all the streaming data is finished, in step S14, the connection between the server and the client is closed, and the process is then exited." Ogawa, [0102].

Ogawa describes monitoring various network conditions to provide optimal data transfer. In one example, much like the '285 Patent, Ogawa's "bitrate setter 212 compares the current transmission-data bitrate with the maximum throughput calculated by the throughput calculator 211." Ogawa, [0145]. Ogawa also explains that "communication-bandwidth information includes, for example . . . a round trip time, a received signal strength indicator, and a transmission rate of communications between the access point and the client." Ogawa, [0027]; *see also id.*, [0037] (disclosing "the bitrate setter is configured to set a transmission bitrate on the basis of the communication-bandwidth information"), [0039]-[0040] (disclosing a "packet-interval measurer").

Ogawa also discloses that, with reference to Figure 4, "maximum throughput is calculated using measured values associated with 'effective data transmission and reception periods', i.e., the periods of transmission of successive data and reception of reception acknowledgments." Ogawa, [0147].

FIG. 4



Ogawa, FIG. 4.

Ogawa is analogous art to the '285 patent. It is in the same field of endeavor as the '285 Patent because both relate to streaming media content at an adaptive (or optimal) bitrate over a network connection while considering network conditions. *See* EX1001, Title ("Adaptive Bitrate Management for Streaming Media Over Packet Networks"), 2:31-47 ("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 includes . . . the ability to implement joint session bitrate management for audio, video and/or other streams simultaneously."); *see* Ogawa, [0015] ("a data communication system, and a communication processing method that allow data to be transmitted in an optimal transmission mode in transmission and reception of streaming data"), [0016] ("More specifically, it is desirable to provide a communication processing apparatus, a data communication system, and a communication processing method with which a server predicts an optimal value of bitrate of data transmitted in consideration of factors such as congestion on a communication path or disturbance on a communication link and with which the bitrate is dynamically controlled on the basis of the predicted value so that data streaming is carried out in an optimal data transmission mode."); Karam Decl. (EX1003), ¶ 88.

Furthermore, Ogawa is reasonably pertinent to at least one problem that would have concerned the inventor of the '285 Patent. The '285 Patent purported to solve problems related to "rate control" and the "challenge in delivering bandwidth-intensive content like multimedia over capacity-limited, shared links," i.e., "quickly respond[ing] to changes in network conditions by adjusting the bitrate and the media encoding scheme to optimize the viewing and listening experience of the user." EX1001, 1:14-31. Likewise, Ogawa seeks to solve problems in the prior art by providing a system that "dynamically controls the transmission bitrate so that the streaming data will be transmitted at an optimal bitrate." Ogawa, [0102]. Ogawa adjusts this bitrate in response to network conditions—e.g., its "server predicts an optimal value of bitrate of data transmitted in consideration of factors such as congestion on a communication path or disturbance on a communication link." Ogawa, [0016]; Karam Decl. (EX1003), ¶ 89.

The summary and discussion above, together with **Exhibit BB**, supports a finding that Ogawa discloses or renders obvious claims 1, 6, 11, and 14, and a reasonable Examiner would consider the teachings of Ogawa in determining whether claims 1, 6, 11, and 14 are patentable at least because the reference was never considered during the prosecution of the '285 Patent. Accordingly, Ogawa (together with Yano, as explained in **Exhibit BB**) raises a substantial new question of patentability as to claims 1, 6, 11, and 14 of the '285 Patent that should be resolved through reexamination.

c) Motivation to Combine Yano and Ogawa

A POSITA would have combined the teachings of Ogawa into Yano's data transmission system—such as Ogawa's teachings related to data transmission at an "optimal transmission mode" and "optimal value of bitrate" through monitoring various network conditions (such as current bitrate, and round trip times)—thereby more explicitly disclosing limitations in claim 1 (and related claims 4, 11, and 14) related to transmitting at optimal bitrates while monitoring network conditions. Karam Decl. (EX1003), ¶ 90.

Both Yano and Ogawa are within the same field and highly relevant to one another. Yano discloses a system for transmitting data "at an optimal transfer rate" between two terminals. Yano, Abstract, FIG. 1. To accomplish this, Yano's system receives a receiver report, which contains information relevant to the connection between the transmitting and receiving terminals. Yano, [0034], [0051]. Yano discloses monitoring current network conditions, e.g., data round-trip times. Yano, [0088]-[0094]. Similarly, Ogawa discloses a data communication system that allows data to be transmitted "in an optimal transmission mode." Ogawa, [0015]-[0016]. Ogawa includes a "dynamic rate controller 210" (like Yano's transmission rate change unit 1-13), which includes a "throughput calculator 211" and "bitrate setter 212" for setting the optimal bitrate. Ogawa, [0102], FIG. 2. Ogawa monitors various network conditions (e.g., "communication-bandwidth information"), which can include "a round trip time, a received signal strength indicator, and a transmission rate of communications between the access point and the client." Ogawa, [0027]. Accordingly, a POSITA would have readily identified both references as relating to one another and directed towards solving problems related to transferring media data at optimal bitrates based on current network conditions. Given their similarity, a POSITA would have recognized their teachings as being applicable to one another. Karam Decl. (EX1003), ¶ 91.

A POSITA would have been motivated to incorporate Ogawa's general teachings of monitoring specific "communication-bandwidth information" about a network connection into Yano's data transmission system that already considers monitoring network conditions in a receiver report, and this would have merely been a matter of combining prior art elements according to known methods to yield predictable results. Specifically, a POSITA would have combined Yano's disclosure of a data transmission system, together with Ogawa's teaching that it would be advantageous to monitor data reception periods, round trip times, and current transmission rates to transfer data at an optimal transfer rate. For example, Yano already provides general disclosure of data transmission system between two terminals, including a receiver report that contains information about the current network conditions. Yano, [0031], [0041], [0051], [0081], FIGs. 1-2. Ogawa simply provides supplemental disclosure and implementation details as to what network conditions would be advantageous to monitor in considering how to dynamically adjust the streaming transfer rate in an optimal manner, e.g., by monitoring factors such as a round trip time, signal strength, and current transmission rate. Ogawa, [0027], [0039], [0145], [0147], [0186], [0190], FIG. 4. Doing so would have been obvious to a POSITA, because it merely would have provided further implementation details to network conditions already considered by Yano itself, or supplemented the network conditions to be monitored by Yano's system to provide a more robust data transmission system. Karam Decl. (EX1003), ¶ 92.

Further, a POSITA would have had a reasonable expectation of success in making the combination, as the proposed combination merely relates to the implementation details of how to monitor specific network conditions to calculate an optimal transmission rate, and therefore, would have at most required minor modifications in software that would have yielded predictable results. Karam Decl. (EX1003), ¶ 93.

II. DETAILED APPLICATION OF THE PRIOR ART TO EVERY CLAIM FOR WHICH REEXAMINATION IS REQUESTED

For the reasons set forth in the various grounds below, claims 1, 6, 9-11, 14, and 15 are disclosed or rendered obvious and should be rejected as unpatentable. Requester presents four grounds of rejection as follows, as outlined in **Exhibits AA-1, AA-2,** and **BB**:

SNQ	Reference(s)	Basis	Claims
1	van Beek	§ 102	9, 10, 15
2	van Beek	§ 103	9, 10, 15
3	Urzaiz, Gupta, and Pogrebinsky	§ 103	9, 10, 15
4	Yano and Ogawa	§ 103	1, 6, 11, 14

In **Exhibits AA-1, AA-2,** and **BB** (appended to this Request, and also filed as Exhibits AA-1, AA-2, and BB), Requester provides detailed mapping and explanation as to how the references in each of the SNQs alone, or together as a combination, disclose each and every element of the challenged claims.

This Request, **Exhibits AA-1, AA-2,** and **BB**, and Exhibit 1003 from Requester's expert, Dr. Lina J. Karam, demonstrate that the Challenged Claims of the '285 Patent are unpatentable as obvious in view of the prior art references. Applicants did not identify any evidence of secondary considerations during prosecution. Further, the clear teachings in the prior art cannot be overcome by any supposed "secondary considerations." *Graham v. John Deere Co. of Kansas City*, 383 U.S. 1, 36 (1966).

III. DISCLOSURE OF CONCURRENT LITIGATION, REEXAMINATION, AND RELATED PROCEEDINGS

Based on information available to Requester, the ’285 Patent is or has been at issue in the following District Court litigations, as listed below:

- *OptiMorphix, Inc. v. Microsoft Corporation*, 5-23-cv-00150 (EDTX) Dec. 20, 2023
- *OptiMorphix, Inc. v. Cisco Systems, Inc.*, 5-23-cv-00126 (EDTX) Nov. 02, 2023
- *OptiMorphix, Inc. v. Amazon.com, Inc. et al.*, 5-23-cv-00123 (EDTX) Oct. 23, 2023
- *OptiMorphix, Inc. v. Alphabet, Inc. et al.*, 1-23-cv-01065 (DDE) Sep. 27, 2023

As of the filing date of this Request, and to the best knowledge of Requester, the ’285 Patent has not been involved in any post-grant proceedings.

IV. CONCLUSION

Reexamination and cancellation of claims 1, 6, 9-11, 14, and 15 of the ’285 Patent is respectfully requested. The Commissioner is hereby authorized to charge Deposit Account 50-6990 under Order No. OPT285 the *Ex Parte* Reexamination fee of $12,600 under 37 C.F.R. § 1.20(c). Requester believes no other fee is due with this submission, however the Commissioner is hereby authorized to charge any fee deficiency or credit any over-payment to Deposit Account 50-6990.

Please direct all correspondence in this matter to the undersigned.

Dated: May 24, 2024

Respectfully submitted,

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APPENDICES

EXHIBIT AA-1

Claim Chart

Comparing Claims 9, 10, and 15 of the ’285 Patent

to van Beek

I. GROUNDS OF UNPATENTABILITY

Ground	Claims	Statutes	Prior Art
1	9, 10, 15	35 U.S.C. § 102	van Beek
2	9, 10, 15	35 U.S.C. § 103	van Beek

A. Prior Art Relied Upon

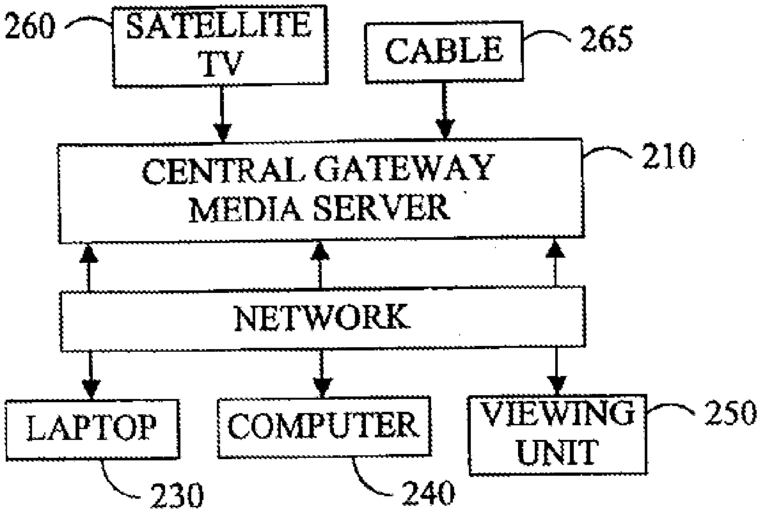
The ’285 Patent was filed on July 9, 2008 and claims priority to provisional application 60/948,917, filed July 10, 2007. Accordingly, the earliest possible priority date for the ’285 Patent is **July 10, 2007**.

Prior Art
EX1004 (“van Beek”): U.S. Patent Publication 2005/0071876 to Petrus J. L. van Beek was filed on September 30, 2003 and published on March 31, 2005. Accordingly, van Beek qualifies as prior art to the ’285 Patent under at least 35 U.S.C. §§ 102(a) and (b) (pre-AIA).

B. Claim Charts

The claim charts below first include an argument portion explaining why van Beek discloses and renders obvious the pertinent limitation, followed by additional supplementary citations to van Beek. Unless otherwise noted, all emphasis has been added by Requester (for example, claim language is denoted by italics, while the corresponding disclosure in the prior art is indicated in bold).

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Grounds 1 and 2)	
[9P] A method comprising:	To the extent the preamble is limiting, van Beek discloses *a method* (e.g., van Beek's adaptive bandwidth transmission system for transmitting multiple data streams, such as video and audio streams, in a network). * * * Requester provides the following disclosures from van Beek relevant to this claim limitation: A **transmission systems suitable** for video. van Beek, Abstract. FIG. 1 illustrate[s] a system for **transmission of multiple data streams in a network that may have limited bandwidth**. The system includes a **central gateway media server 210** and a **plurality of client receiver units 230, 240, 250**. The central gateway media server may be any device that can transmit multiple data streams. The input data streams may be stored on the media server or arrive from an external source, such as a satellite television transmission 260, a digital video disc player, a video cassette recorder, or a cable head end 265, and are **transmitted to the client receiver units 230, 240, 250 in a compressed format**. **The data streams can include display data, graphics data, digital data, analog data, multimedia data, audio data and the like**. An adaptive bandwidth system on the gateway media server 210 determines the network bandwidth characteristics and adjusts the bandwidth for the output data streams in accordance with the bandwidth characteristics. van Beek, [0041].

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Grounds 1 and 2)	
	 FIG. 1 van Beek, FIG. 1.
[9a] receiving an optimal session bitrate;	van Beek discloses, or at least renders obvious,[1] *receiving an optimal session bitrate* (e.g., van Beek's adaptive bandwidth system that receives information about a network connection and optimizes the aggregate channel bitrate), as claimed. Although the '285 Patent does not explicitly define the claim term *optimal session bitrate*, it does describe and claim exemplary ways of computing it; for example, the '285 Patent describes that 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," using well known network state estimators like media time in transit (MTT) or round trip time estimate (RTTE). EX1001, 4:22-29. In other words, the system analyzes the state of the connection between the transmitting

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[1] Throughout this claim chart, Requester's general discussion of van Beek and its disclosures apply to Ground 1 (i.e., anticipation under 35 U.S.C. § 102). Where Requester instead intends to instead to argue specifically under Ground 2 (i.e., obviousness under 35 U.S.C. § 103), those paragraphs and arguments are clearly delineated.

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Grounds 1 and 2)	
	terminal and the receiving terminal to obtain a bitrate suitable for streaming media. Accordingly, a POSITA would have understood that, in the context of transmitting (or streaming) video and audio media from a server to a client over a network connection, an optimal session bitrate would broadly refer to the maximum (or similarly, total) bitrate capable of being supported by the connection between the terminal (or server) transmitting the video and audio media to the receiving terminal (or client). Karam Decl. (EX1003), ¶ 47. Requester further notes that other claims of the '285 Patent (e.g., claim 1) claim a method for determining an optimal session bitrate (e.g., claim 1 recites "determining an optimal session bitrate using the estimated one or more network conditions" and includes claim limitations directed to network stability criteria, such as MTT, RTT, and current bitrate); but claim 9 is not so limited. All claim 9 requires is that the method being performed includes **receiving** an optimal session bitrate. Accordingly, this limitation of claim 9 is disclosed by any reference that describes a method for receiving the maximum (or total) bitrate that can be supported on a network connection for streaming audio and video media between terminals (e.g., a server and a client). Just like the '285 Patent's "adaptive bitrate controller 210" (which in the specification example is used to compute the optimal session bitrate), van Beek describes that an "adaptive bandwidth system on the gateway media server 210 **determines the network bandwidth characteristics and adjusts the bandwidth for the output data streams** in accordance with the bandwidth characteristics." van Beek, [0041]. van Beek also recognized that "[d]ifferent devices interconnected to the network have different resources and different usage paradigms. For example, different devices may have different microprocessors, different memory requirements, different display characteristics, **different connection bandwidth capabilities**, and different battery resources. . . . This results in unpredictable and **dynamically varying network maximum throughput**." van Beek, [0051]. These disclosures demonstrate that van Beek seeks to determine and adjust bandwidth for transmitting media based on varying network conditions in the same way that the '285 Patent discloses doing so for an optimal session bitrate.

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Grounds 1 and 2)	
	van Beek is also squarely directed to **optimizing** the transmission of audio and video media (i.e., an *optimal* session bitrate), because the stated improvements of its disclosure are directed towards "**optimizing** the quality of the **AV data** continuously, in real-time" and "**adapting** to the unpredictable and dynamically changing conditions of the network." van Beek, [0062]-[0066]. Similarly, van Beek describes that its "goal is to find the **best set of output rates** . . . that **maximizes** the overall quality of all output streams or, equivalently, minimizes an overall distortion criterion D, **while the aggregate rate of all streams is within the capacity of the channel**." van Beek, [0136]. Thus, while van Beek seeks to maximize quality of the AV data, it does so in a manner that does not exceed the capacity of the connection. van Beek further discloses "receiving an optimal session bitrate" through its network monitor module. *See* van Beek [0068], [0088]. van Beek explicitly describes that its coding/transcoding module receives a desired optimal output bitrate (i.e., ***receiving** an optimal session bitrate*) from the network monitor module: The **coding/transcoding module is provided with a desired output bit rate** (or other similar information) and **uses a rate control mechanism to achieve this bit rate**. The value of the desired output bit rate is part of information about the transmission channel provided to the extender by a network monitor module. The network monitor monitors the network and estimates the bandwidth available to the video stream in real time. The information from the network monitor is used to ensure that the video stream sent from the extender to a receiver has a bit rate that is matched in some fashion to the available bandwidth (e.g., channel rate). With a fixed video bit rate normally the quality varies on a frame by frame basis. **To achieve the optimal output bit rate**, the coder/transcoder may increase the level of compression applied to the video data, thereby decreasing visual quality slowly. In the case of a transcoder, this may be referred to as transrating. Note that the resulting decrease in visual quality by modifying the bit stream is minimal in comparison to the loss in visual quality that would be incurred if a

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Grounds 1 and 2)	
	video stream is transmitted at bit rates that can not be supported by the network. The loss of video data incurred by a bit rate that cannot be supported by the network may lead to Severe errors in video frames, such as dropped frames, followed by error propagation (due to the nature of video coding algorithms such as MPEG). The feedback obtained from the network monitor ensures that the output bit rate is toward an optimum level so that any loss in quality incurred by transrating is minimal. van Beek, [0068]. Each input stream is encoded or transcoded separately, although their bit rates are controlled by the (trans)coder manager. The (trans)coder manager handles competing requests for bandwidth dynamically. **The (trans)coding manager allocates bit rates to multiple video streams in such a way that the aggregate of the bit rates of the output video streams matches the desired aggregate channel bit rate. The desired aggregate bit rate, again, is obtained from a network monitor module**, ensuring that the aggregate rate of multiple video streams does not exceed available bandwidth. Each coder/transcoder again uses some form of rate control to achieve the allocated bit rate for its stream. van Beek [0088]; *see also id.*, FIGs. 2, 3, 6. van Beek also explicitly describes that bandwidth is controlled by adjusting *bitrates* so that the aggregate bitrate matches the desired aggregate channel bit rate, i.e., the desired optimal bit rate that is received from the network monitor module, e.g.: Each input stream is encoded or transcoded separately, although their bit rates are controlled by the (trans)coder manager. The (trans)coder manager handles competing requests for bandwidth dynamically. The (trans)coding manager allocates **bit rates** to multiple video streams in such a way that the **aggregate of the bit rates of the output video streams matches the desired aggregate channel bit**

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Grounds 1 and 2)	
	rate. The desired aggregate bit rate, again, is obtained from a network monitor module, ensuring that the aggregate rate of multiple video streams does not exceed available bandwidth. Each coder/transcoder again uses some form of rate control to achieve the **allocated bit rate for its stream**. van Beek, [0088]. Although the above disclosure only refers to video streams, van Beek also considers doing the same for audio streams. *See* van Beek, [0100]-[0102], [0121], [0055]. For the reasons set forth above, van Beek discloses *receiving an optimal session bit rate*, as claimed. (**Ground 2**) To the extent van Beek is not found to disclose this limitation, a POSITA would have nonetheless found this limitation obvious, both in view of the disclosures above, and additional disclosures cited below. * * * Requester provides the following disclosures from van Beek relevant to this claim limitation: To reduce such limitations one may (1) improve network technology to make networks more suitable to audio/visual data and/or (2) **one may modify the audio/visual data to make the audio/visual data more suitable to such transmission networks**. Therefore, a system may robustly stream audio/visual data over (wireless) networks by: (1) **optimizing the quality of the AV data continuously, in real-time**; and (2) **adapting** to the unpredictable and dynamically changing conditions of the network. Accordingly a **system that includes dynamic rate adaption is suitable to accommodate distribution of high quality audio/video streams over networks that suffer from significant dynamic variations in performance**. These

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Grounds 1 and 2)	
	variations may be caused by varying of distance of the receiving device from the transmitter, from interference, or other factors. The following discussion includes single-stream dynamic rate adaptation, followed by multi-stream dynamic rate adaptation, and then various other embodiments. van Beek, [0062]-[0066]. FIG. 1 illustrate[s] a system for **transmission of multiple data streams in a network that may have limited bandwidth**. The system includes a central gateway media server 210 and a plurality of client receiver units 230, 240, 250. The central gateway media server may be any device that can transmit multiple data streams. The input data streams may be stored on the media server or arrive from an external source, such as a satellite television transmission 260, a digital video disc player, a video cassette recorder, or a cable head end 265, and are transmitted to the client receiver units 230, 240, 250 in a compressed format. **The data streams can include display data, graphics data, digital data, analog data, multimedia data, audio data and the like. An adaptive bandwidth system on the gateway media server 210 determines the network bandwidth characteristics and adjusts the bandwidth for the output data streams in accordance with the bandwidth characteristics**. van Beek, [0041]. Bit Allocation in Joint Coding of Multiple Streams **A more optimal approach to rate adaptation of multiple streams is to apply joint bit allocation/rate control**. This approach applies to the case where the input streams to the multi-stream extender system are analog, as well as the case where the input streams are already in compressed digital form. Let the following parameters be defined: N_L denote the number of streams P_n denote a weight or priority assigned to stream n, with $p_n \geqq 0$

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Grounds 1 and 2)	
	a_n denote a minimum output rate for stream n, with $a_n \geqq 0$ b_n denote a maximum output rate for stream n, with $b_n \geqq a_n$ $D_n(r)$ denote the distortion of output stream n as a function of its output rate r (i.e. the distortion of the output with respect to the input of the encoder or transcoder) R_C denote the available bandwidth of the channel or maximum network maximum throughput R_n denotes the bit rate of input stream n R'_n denotes the bit rate of output stream n Note that R_n, R'_n and R_C may be time-varying in general; hence, these are functions of time t. The problem of the multi-stream extender can be formulated generically as follows: **The goal is to find the set of output rates R'_n, n=1, . . . , N_L, that maximizes the overall quality of all output streams or, equivalently, minimizes an overall distortion criterion D, while the aggregate rate of all streams is within the capacity of the channel**. van Beek, [0124]-[0136]. Another bit allocation approach in **joint coding of multiple streams i**n a LAN environment, such as those based on IEEE 802.11, is suitable for those networks that have multi-rate support. In this case an access point in the gateway may be communicating at different data link rates with different client devices. **For this, and other reasons, the maximum data throughput from the gateway to one device may be different from the maximum throughput from the gateway to another device, while transmission to each device contributes to the overall utilization of a single, shared, channel**. van Beek, [0173]. It is the **maximum throughput or bandwidth T that is estimated, in order to provide the transmitter with the right information to adapt the audio/video stream bandwidth (if necessary)**. The maximum throughput is

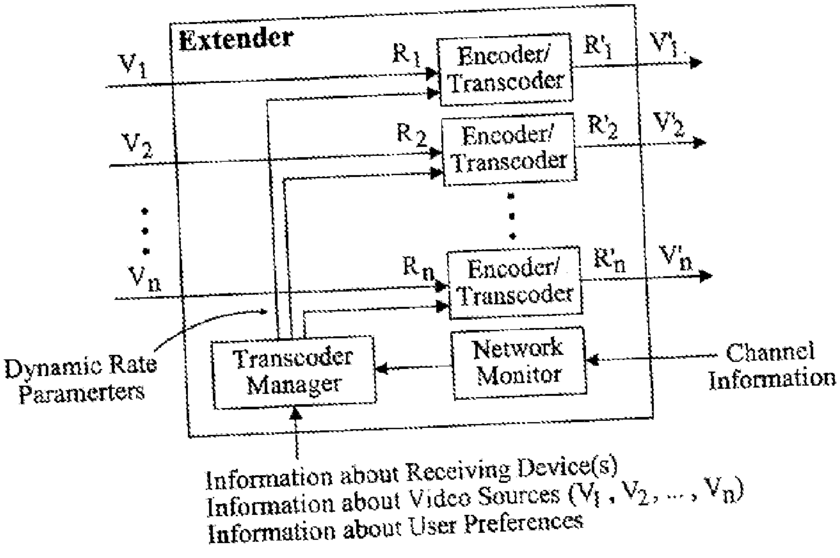
Claims	Relevant Disclosures in the Prior Art
Claim 9 (Grounds 1 and 2)	
	achieved, albeit temporarily, during transmission of a packet burst. Therefore, **the maximum throughput is estimated by computing the ratio of the number of bits transmitted during a burst, and the time duration of that burst**. van Beek, [0221]. The bandwidth measurements may be done on an ongoing basis, that is, more than just once. **Every burst of data packets during the streaming of audio/video data may be used to estimate bandwidth available during transmission of that burst. Such measurements performed at the receiver are sent back to the sender**. van Beek, [0225]. *See also* van Beek, FIGs. 28A, 28B, 29.
[9b] allocating the optimal session bitrate between audio and video media to produce an optimal audio bitrate and an optimal video bitrate,	van Beek discloses, or at least renders obvious, *allocating the optimal session bitrate between audio and video media to produce an optimal audio bitrate and an optimal video bitrate* (e.g., van Beek's disclosure that audio and video streams can be separated and treated differently), as claimed. First, van Beek describes *allocating* the optimal session bitrate between audio and video media by describing that the "(trans)coding manager **allocates bit rates** to multiple video streams in such a way that the **aggregate of the bit rates of the output video streams matches the desired aggregate channel bit rate**," where the aggregate channel bit rate is an optimal session bitrate, as described above in limitation [9a]. van Beek, [0088]. van Beek also notes that each "coder/transcoder again uses some form of rate control to achieve the **allocated bit rate** for its stream," and that "the bit rate of the multiple streams should be **controlled by some form of bit allocation and rate control** in order to satisfy such constraints." van Beek, [0089]-[0090]. Once again, while this specific section only refers to video, van Beek notes as to its disclosure generally "that while the system **may refer to audio/video, the concepts are likewise used for video alone and/or audio alone**." van Beek, [0055].

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Grounds 1 and 2)	
	van Beek further describes *allocating* in that its transcoder may employ several strategies for prioritizing a given video or audio stream, e.g., it “may attempt to **allocate** an equal amount of available bits to each stream,” “attempt to **allocate** the available bits such that the quality of each stream is approximately equal,” or “allow users to **assign different priorities** to different streams.” van Beek, [0107]-[0108]. Next, van Beek discloses allocating the optimal session bitrate *between* audio and video media by describing that the “bit rates of **individual audio/video streams on the network are subject to various constraints**. Firstly, the **aggregate rates of individual streams may be smaller than or equal to the overall channel capacity or network bandwidth** from sender to receiver.” van Beek, [0100]-[0102]. This makes clear that the individual bitrates for individual audio or video streams are allocated from van Beek’s “overall channel capacity or network bandwidth” (i.e., the claimed optimal session bitrate). van Beek is further explicit that audio and video streams can be treated separately: The relative weight of streams may also be set based on their modality. In particular, **the audio and video streams of an audiovisual stream may be separated and treated differently during their transmission.** For example, the audio part of an audiovisual stream may be assigned a higher priority than the video part. This case is motivated by the fact that when viewing a TV program, in many cases, loss of audio information is deemed more severe by users than loss of video information from the TV signal. This may be the case, for instance, when the viewer is watching a sports program, where a commentator provides crucial information. As another example, it may be that users do not wish to degrade the quality of audio streams containing hi-quality music. Also, the audio quality could vary among different speakers or be sent to different speakers. van Beek, [0121]. Finally, as described above, this allocating process in van Beek produces an *optimal* audio bitrate and an *optimal* video bitrate because the entire purpose of van Beek is to optimize the quality of

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Grounds 1 and 2)	
	an AV stream, e.g., "**optimizing** the quality of the AV data continuously, in real-time" and "adapting to the unpredictable and dynamically changing conditions of the network." van Beek, [0062]-[0066]. Similarly, van Beek describes that its "goal is to find the **best set of output rates** . . . that **maximizes the overall quality of all output streams** or, equivalently, **minimizes an overall distortion** criterion D, **while the aggregate rate of all streams is within the capacity of the channe**l." van Beek, [0136]. For the reasons set forth above, van Beek discloses *allocating the optimal session bitrate between audio and video media to produce an optimal audio bitrate and an optimal video bitrate*, as claimed. (**Ground 2**) To the extent van Beek is not found to disclose this limitation, a POSITA would have nonetheless found this limitation obvious, both in view of the disclosures above, and additional disclosures cited below. * * * Requester provides the following disclosures from van Beek relevant to this claim limitation: Multi-Stream Dynamic Rate Adaptation The basic extender for a single AV stream described above will encode an analog input stream or adapt the bit rate of an input digital bit stream to the available bandwidth without being concerned about the cause of the bandwidth limitations, or about other, competing streams, if any. In the following, the system may include a different extender system that processes multiple video streams, where the extender system assumes the responsibility of controlling or adjusting the bit rate of multiple streams in the case of competing traffic. The multi-stream extender, depicted in FIG. 6, employs a "(trans)coder manager" on top of multiple video encoders/transcoders. As shown in FIG. 6, the system operates on n video streams, where each source may be either analog (e.g. composite) or digital (e.g. MPEG-2 compressed bitstreams). Here, Vn denotes input stream n, while V'n denotes output stream n. Rn denotes the bit rate of input stream n (this exists only if input stream n is in already compressed

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Grounds 1 and 2)	
	digital form; it is not used if the input is analog), while R'n denotes the bit rate of output stream n. **Each input stream is encoded or transcoded separately, although their bit rates are controlled by the (trans)coder manager. The (trans)coder manager handles competing requests for bandwidth dynamically. The (trans)coding manager allocates bit rates to multiple video streams in such a way that the aggregate of the bit rates of the output video streams matches the desired aggregate channel bit rate. The desired aggregate bit rate, again, is obtained from a network monitor module, ensuring that the aggregate rate of multiple video streams does not exceed available bandwidth. Each coder/transcoder again uses some form of rate control to achieve the allocated bit rate for its stream.** In this case, the system may include multiple receivers (not shown in the diagram). Each receiver in this system has similar functionality as the receiver for the single-stream case. As in the single-stream case, **the bit rate of the multiple streams should be controlled by some form of bit allocation and rate control in order to satisfy such constraints**. However, in the case of a multi-stream system, a more general and flexible framework is useful for dynamic bit rate adaptation. van Beek, [0086]-[0090]. Various network technologies may be used for the gateway reception and transmission, such as for example, IEEE 802.11, Ethernet, and power-line networks (e.g., HomePlug Powerline Alliance). While such networks are suitable for data transmission, they do not tend to be especially suitable for audio/video content because of the stringent requirements imposed by the nature of audio/video data transmission. Moreover, the network capabilities, and in particular the data maximum throughput offered, are inherently unpredictable and may dynamically change due to varying conditions described above. The data throughput may be defined in terms of the amount of actual (application) payload bits (per second) being transmitted from the sender to the receiver successfully.

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Grounds 1 and 2)	
	It is noted that while the system may refer to audio/video, the concepts are likewise used for video alone and/or audio alone. van Beek, [0055] (like the '285 Patent (*see* EX1001, 4:66-5:7), van Beek teaches prioritizing one type of data over the other). Stream Prioritizing or Weighting The (trans)coder manager discussed above may employ several strategies. **It may attempt to allocate an equal amount of available bits to each stream; however, in this case the quality of streams may vary strongly from one stream to the other, as well as in time. It may also attempt to allocate the available bits such that the quality of each stream is approximately equal; in this case, streams with highly active content will be allocated more bits than streams with less active content.** Another approach is to allow users to assign different priorities to different streams, such that the quality of different streams is allowed to vary, based on the preferences of the user(s). This approach is generally equivalent to weighting the individual distortion of each stream when the (trans)coder manager minimizes the overall distortion. **The priority or weight of an audio/video stream may be obtained in a variety of manners, but is generally related to the preferences of the users of the client devices.** Note that the weights (priorities) discussed here are different from the type of weights or coefficients seen often in literature that correspond to the encoding complexity of a macro block, video frame, group of frames, or video sequence (related to the amount of motion or texture variations in the video), which may be used to achieve a uniform quality among such parts of the video. Here, weights will purposely result in a non-uniform quality distribution across several audio/video streams, where one (or more) such audio/video stream is considered more important than others. Various cases, for example, may include the following, and combinations of the following. van Beek, [0107]-[0108].

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Grounds 1 and 2)	
	In the following subsections, first is listed the type of constraints that the bit rate of the multiple streams in this system are subject to. Then, the notion of stream prioritizing is described, which is used to incorporate certain heterogeneous characteristics of the network as discussed above. Then, various techniques are described to achieve multi-stream (or joint) dynamic rate adaptation. Bit Rate Constraints for Multiple Streams **The bit rates of individual audio/video streams on the network are subject to various constraints**. Firstly, **the aggregate rates of individual streams may be smaller than or equal to the overall channel capacity or network bandwidth from sender to receiver**. This bandwidth may vary dynamically, due to increases or decreases in the number of streams, due to congestion in the network, due to interference, etc. van Beek, [0100]-[0102].  FIG. 6 van Beek, FIG. 6.

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Grounds 1 and 2)	
[9c] 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;	van Beek discloses, or at least renders obvious, *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* (e.g., van Beek's disclosure of treating audio and video streams differently, as well as prioritizing one over another), as claimed. As mentioned above in limitation [9b], van Beek explicitly describes that its allocating is *based at least in part on privileging either the audio media or the video media over the other*, e.g.: > The relative weight of streams may also be set based on their modality. In particular, **the audio and video streams of an audiovisual stream may be separated and treated differently during their transmission.** For example, **the audio part of an audiovisual stream may be assigned a higher priority than the video part.** This case is motivated by the fact that when viewing a TV program, in many cases, **loss of audio information is deemed more severe by users than loss of video information from the TV signal**. This may be the case, for instance, when the viewer is watching a sports program, where a commentator provides crucial information. **As another example, it may be that users do not wish to degrade the quality of audio streams containing hi-quality music**. Also, the audio quality could vary among different speakers or be sent to different speakers. van Beek, [0121]. Accordingly, van Beek's separating and assigning a "higher priority" to either an audio or video part of an audiovisual stream discloses the claimed *privileging either the audio media or the video media over the other*. For the reasons set forth above, van Beek discloses *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*, as claimed. (**Ground 2**) To the extent van Beek is not found to disclose this limitation, a POSITA would have nonetheless found this limitation

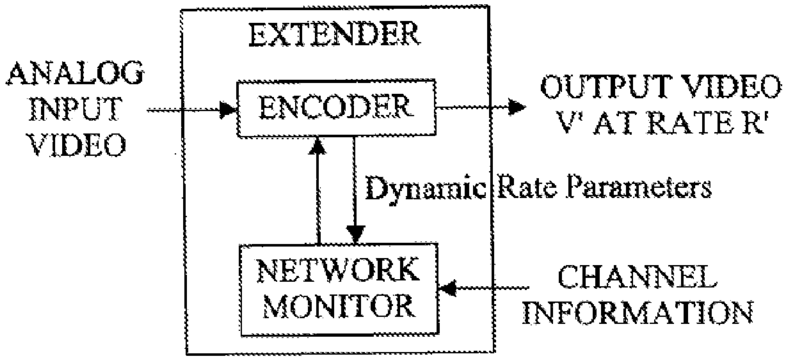
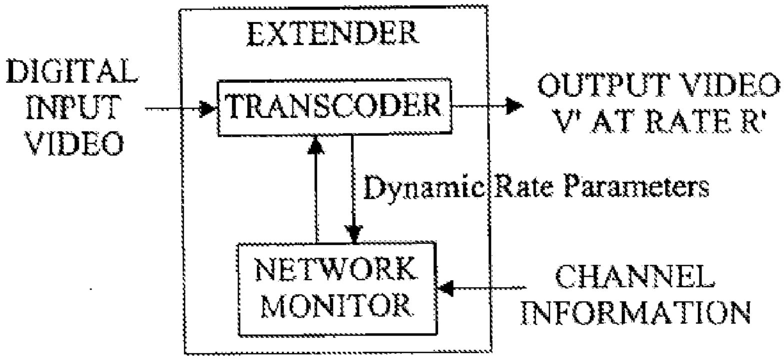
<table>
<tr><th>Claims</th><th>Relevant Disclosures in the Prior Art</th></tr>
<tr><td colspan="2">Claim 9 (Grounds 1 and 2)</td></tr>
<tr><td></td><td>
<p>obvious, both in view of the disclosures above, and additional disclosures cited below.</p>
<p>* * *</p>
<p>Requester provides the following disclosures from van Beek relevant to this claim limitation:</p>
<blockquote>
<p>Stream Prioritizing or Weighting</p>
<p>The (trans)coder manager discussed above may employ several strategies. It may attempt to allocate an equal amount of available bits to each stream; however, in this case the quality of streams may vary strongly from one stream to the other, as well as in time. It may also attempt to allocate the available bits such that the quality of each stream is approximately equal; in this case, streams with highly active content will be allocated more bits than streams with less active content. Another approach is to allow users to assign different priorities to different streams, such that the quality of different streams is allowed to vary, based on the preferences of the user(s). This approach is generally equivalent to weighting the individual distortion of each stream when the (trans)coder manager minimizes the overall distortion.</p>
<p>The priority or weight of an audio/video stream may be obtained in a variety of manners, but is generally related to the preferences of the users of the client devices. Note that the weights (priorities) discussed here are different from the type of weights or coefficients seen often in literature that correspond to the encoding complexity of a macro block, video frame, group of frames, or video sequence (related to the amount of motion or texture variations in the video), which may be used to achieve a uniform quality among such parts of the video. Here, weights will purposely result in a non-uniform quality distribution across several audio/video streams, where one (or more) such audio/video stream is considered more important than others. Various cases, for example, may include the following, and combinations of the following.</p>
</blockquote>
<p>van Beek, [0107]-[0108].</p>
</td></tr>
</table>

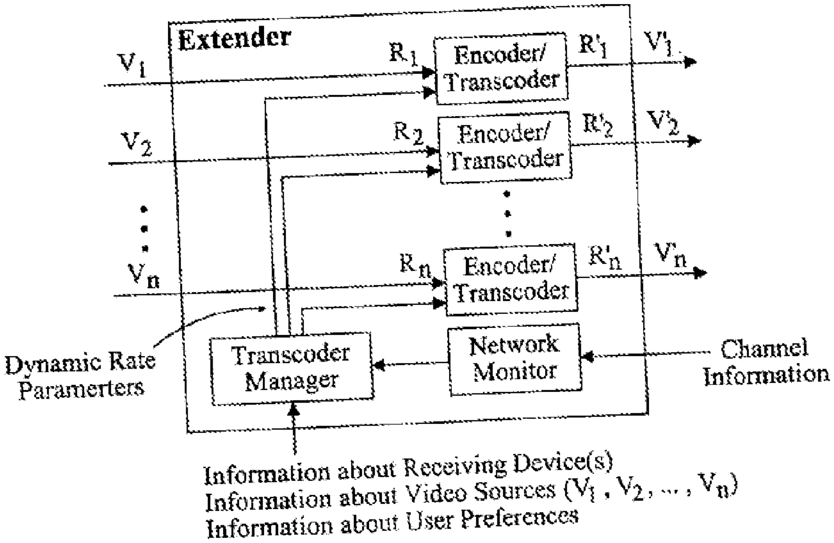
Claims	Relevant Disclosures in the Prior Art
Claim 9 (Grounds 1 and 2)	
	The relative weight of streams may also be set based on their modality. In particular, the audio and video streams of an audiovisual stream may be separated and treated differently during their transmission. **For example, the audio part of an audiovisual stream may be assigned a higher priority than the video part.** This case is motivated by the fact that when viewing a TV program, in many cases, loss of audio information is deemed more severe by users than loss of video information from the TV signal. This may be the case, for instance, when the viewer is watching a sports program, where a commentator provides crucial information. As another example, it may be that users do not wish to degrade the quality of audio streams containing hi-quality music. Also, the audio quality could vary among different speakers or be sent to different speakers. van Beek, [0121]. Various network technologies may be used for the gateway reception and transmission, such as for example, IEEE 802.11, Ethernet, and power-line networks (e.g., HomePlug Powerline Alliance). While such networks are suitable for data transmission, they do not tend to be especially suitable for audio/video content because of the stringent requirements imposed by the nature of audio/video data transmission. Moreover, the network capabilities, and in particular the data maximum throughput offered, are inherently unpredictable and may dynamically change due to varying conditions described above. The data throughput may be defined in terms of the amount of actual (application) payload bits (per second) being transmitted from the sender to the receiver successfully. **It is noted that while the system may refer to audio/video, the concepts are likewise used for video alone and/or audio alone**. van Beek, [0055].
[9d] encoding audio and video media data according to the optimal audio bitrate and the	van Beek discloses, or at least renders obvious, *encoding audio and video media data according to the optimal audio bitrate and the optimal video bitrate* (e.g., van Beek's disclosure of encoders and transcoders for video and audio streams), as claimed.

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Grounds 1 and 2)	
optimal video bitrate; and	First, the *optimal audio bitrate* and *optimal video bitrate* claim terms in this limitation are disclosed by van Beek for the reasons set forth above in limitations [9a] and [9b]; this limitation merely requires that the method perform the *encoding* of the *audio and video media data* according to those rates. As to the step of *encoding audio and video media data*, van Beek describes that its system includes "**a video encoder or transcoding module**." van Beek, [0067]. It further describes the use of "multiple video encoders/transcoders" and that "**[e]ach input stream is encoded or transcoded separately**," where "their bit rates are controlled by the (trans)coder manager." van Beek, [0087]-[0088]; *see also id.*, [0200] ("Each encoder and/or transcoder produces a corresponding output bitstream."). As mentioned above, the "(trans)coding manager allocates bit rates to multiple video streams in such a way that the aggregate of the bit rates of the output video streams matches the desired aggregate channel bit rate." van Beek, [0088]. van Beek further states that each "coder/transcoder again uses some form of rate control to achieve the allocated bit rate for its stream." *Id.* For the reasons discussed above in limitations [9a] and [9b], the encoding and transcoding that takes place in van Beek is done with the purpose of optimizing and maximizing the quality of both the audio and video portions of an AV stream. Finally, Requester again notes that while certain disclosures in van Beek may only refer to a video stream, van Beek explains as to its disclosure generally, "that while the system may refer to audio/video, the concepts are likewise used for video alone and/or audio alone." van Beek, [0055]. Furthermore, van Beek also notes that "the audio and video streams of an audiovisual stream may be separated and treated differently during their transmission." van Beek, [0121]. van Beek also provides the following disclosures: The data streams can include display data, graphics data, digital data, analog data, multimedia data, audio data and the like. van Beek, [0041]. There are many characteristics that the present inventors identified that may be considered for an

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Grounds 1 and 2)	
	audio/visual transmission system in order to achieve improved results over the technique described above. van Beek, [0049]; *see also* van Beek, [0062]-[0064], [0101], [0105], [0108], [0121], [0203], [0221], [0225], [0250]-[0251]. For the reasons set forth above, van Beek discloses *encoding audio and video media data according to the optimal audio bitrate and the optimal video bitrate*, as claimed. **(Ground 2)** To the extent van Beek is not found to disclose this limitation, a POSITA would have nonetheless found this limitation obvious, both in view of the disclosures above, and additional disclosures cited below. * * * Requester provides the following disclosures from van Beek relevant to this claim limitation: Multi-Stream Dynamic Rate Adaptation The basic extender for a single AV stream described above will encode an analog input stream or adapt the bit rate of an input digital bit stream to the available bandwidth without being concerned about the cause of the bandwidth limitations, or about other, competing streams, if any. In the following, the system may include a different extender system that processes multiple video streams, where the extender system assumes the responsibility of controlling or adjusting the bit rate of multiple streams in the case of competing traffic. The multi-stream extender, depicted in FIG. 6, employs a **"(trans)coder manager" on top of multiple video encoders/transcoders**. As shown in FIG. 6, **the system operates on n video streams**, where each source may be either analog (e.g. composite) or digital (e.g. MPEG-2 compressed bitstreams). Here, Vn denotes input stream n, while V′n denotes output stream n. Rn denotes the bit rate of input stream n (this exists only if input stream n is in already compressed digital form; it is not used if the input is analog), while R′n denotes the bit rate of output stream n.

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Grounds 1 and 2)	
	Each input stream is encoded or transcoded separately, although their bit rates are controlled by the (trans)coder manager. The (trans)coder manager handles competing requests for bandwidth dynamically. **The (trans)coding manager allocates bit rates to multiple video streams in such a way that the aggregate of the bit rates of the output video streams matches the desired aggregate channel bit rate. The desired aggregate bit rate, again, is obtained from a network monitor module, ensuring that the aggregate rate of multiple video streams does not exceed available bandwidth**. Each coder/transcoder again uses some form of rate control to achieve the allocated bit rate for its stream. In this case, the system may include multiple receivers (not shown in the diagram). Each receiver in this system has similar functionality as the receiver for the single-stream case. As in the single-stream case, the bit rate of the multiple streams should be controlled by some form of bit allocation and rate control in order to satisfy such constraints. However, in the case of a multi-stream system, a more general and flexible framework is useful for dynamic bit rate adaptation. . . . van Beek, [0086]-[0090]. Another embodiment is a multi-stream system, as illustrated in FIG. 9. This multi-stream system has multiple AV sources, where some sources may be in analog form, and other sources may be in digital form (e.g., MPEG-2 or MPEG-4 bit streams). **These AV sources are input to a processing module that contains zero or more encoders (analog inputs) as well as zero or more transcoders (digital inputs). Each encoder and/or transcoder produces a corresponding output bitstream.** The bit rate of these bit streams are dynamically adapted to the conditions of the channel, so as to optimize the overall quality of all streams. The system may also adapt these streams based on information about the capabilities of receiver devices. The system may also adapt streams based on information about the preferences of each user. **All encoded/transcoded bit streams are sent to a network access point, which transmits each bit stream to the corresponding receiver.** Each receiver contains an AV decoder that decodes the digitally compressed bit stream.

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Grounds 1 and 2)	
	van Beek, [0200].  FIG. 2  FIG. 3

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Grounds 1 and 2)	
	 FIG. 6 van Beek, FIGs. 2, 3, 6.
[9e] providing the encoded audio and video data for transmittal to a terminal.	van Beek discloses, or at least renders obvious, *providing the encoded audio and video data for transmittal to a terminal* (e.g., van Beek's disclosure that input media streams are processed by a server and "transmitted to the client receiver units"), as claimed. van Beek itself is titled as a "Wireless Video **Transmission** System," and throughout its disclosure regularly refers to transmitting data streams from a server to a client device (i.e., the claimed *terminal*). For example, van Beek discloses: FIG. 1 illustrate[s] a system for transmission of multiple data streams in a network that may have limited bandwidth. The system includes **a central gateway media server 210** and a plurality of client receiver units 230, 240, 250. The **central gateway media server may be any device that can transmit multiple data streams. The input data streams may be stored on the media server . . . and are transmitted to the client receiver units 230, 240, 250 in a compressed format**. The data streams can include **display data, graphics data**, digital data, analog data, multimedia data, **audio data** and the like.

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Grounds 1 and 2)	
	An adaptive bandwidth system on the gateway media server 210 determines the network bandwidth characteristics and adjusts the bandwidth for the output data streams in accordance with the bandwidth characteristics. van Beek, [0041]. This portion of van Beek makes clear that its data streams include, among other things, display data, graphics data, and audio data (i.e., the claimed *audio and video data*); that this data is in a compressed format (i.e., the audio and video data is *encoded*); and that it is sent from a central gateway media server to client receiver units (i.e., *providing . . . for transmittal to a terminal*). *See also* van Beek, [0055]. And as mentioned in the previous limitations, van Beek describes encoding or transcoding input streams separately (*see id.*, [0088]) and that audio and video streams may be separated and treated differently during transmission (*see id.*, [0121]). van Beek also describes that "[e]ach encoder and/or transcoder produces a corresponding output bitstream," and that "[a]ll **encoded/transcoded bit streams are sent to a network access point, which transmits each bit stream to the corresponding receiver**." van Beek, [0200]. For the reasons set forth above, van Beek discloses *providing the encoded audio and video data for transmittal to a terminal*, as claimed. **(Ground 2)** To the extent van Beek is not found to disclose this limitation, a POSITA would have nonetheless found this limitation obvious, both in view of the disclosures above, and additional disclosures cited below. * * * Requester provides the following disclosures from van Beek relevant to this claim limitation: Another embodiment is a multi-stream system, as illustrated in FIG. 9. This multi-stream system has multiple AV sources, where some sources may be in analog form, and other sources may be in digital form (e.g., MPEG-2 or MPEG-4 bit streams). These AV sources are input to a processing module that contains zero or more encoders (analog inputs) as well as zero

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Grounds 1 and 2)	
	or more transcoders (digital inputs). Each encoder and/or transcoder produces a corresponding output bitstream. The bit rate of these bit streams are dynamically adapted to the conditions of the channel, so as to optimize the overall quality of all streams. The system may also adapt these streams based on information about the capabilities of receiver devices. The system may also adapt streams based on information about the preferences of each user. **All encoded/transcoded bit streams are sent to a network access point, which transmits each bit stream to the corresponding receiver.** Each receiver contains an AV decoder that decodes the digitally compressed bit stream. van Beek, [0200]. FIG. 1 illustrate[s] a system for transmission of multiple data streams in a network that may have limited bandwidth. The system includes a central gateway media server 210 and a plurality of client receiver units 230, 240, 250. **The central gateway media server may be any device that can transmit multiple data streams**. The input data streams may be stored on the media server or arrive from an external source, such as a satellite television transmission 260, a digital video disc player, a video cassette recorder, or a cable head end 265, and are transmitted to the client receiver units 230, 240, 250 in a compressed format. **The data streams can include display data, graphics data, digital data, analog data, multimedia data, audio data and the like. An adaptive bandwidth system on the gateway media server 210 determines the network bandwidth characteristics and adjusts the bandwidth for the output data streams in accordance with the bandwidth characteristics**. van Beek, [0041]. Multi-Stream Dynamic Rate Adaptation The basic extender for a single AV stream described above will encode an analog input stream or adapt the bit rate of an input digital bit stream to the available bandwidth without being concerned about the cause of the bandwidth limitations, or about other, competing streams, if any. In the following, the system may include a different extender system that processes

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Grounds 1 and 2)	
	multiple video streams, where the extender system assumes the responsibility of controlling or adjusting the bit rate of multiple streams in the case of competing traffic. **The multi-stream extender, depicted in FIG. 6, employs a "(trans)coder manager" on top of multiple video encoders/transcoders**. As shown in FIG. 6, the system operates on n video streams, where each source may be either analog (e.g. composite) or digital (e.g. MPEG-2 compressed bitstreams). Here, Vn denotes input stream n, while V'n denotes output stream n. Rn denotes the bit rate of input stream n (this exists only if input stream n is in already compressed digital form; it is not used if the input is analog), while R'n denotes the bit rate of output stream n. **Each input stream is encoded or transcoded separately**, although their bit rates are controlled by the (trans)coder manager. The (trans)coder manager handles competing requests for bandwidth dynamically. The (trans)coding manager allocates bit rates to multiple video streams in such a way that the aggregate of the bit rates of the output video streams matches the desired aggregate channel bit rate. The desired aggregate bit rate, again, is obtained from a network monitor module, ensuring that the aggregate rate of multiple video streams does not exceed available bandwidth. Each coder/transcoder again uses some form of rate control to achieve the allocated bit rate for its stream. In this case, the system may include multiple receivers (not shown in the diagram). Each receiver in this system has similar functionality as the receiver for the single-stream case. As in the single-stream case, the bit rate of the multiple streams should be controlled by some form of bit allocation and rate control in order to satisfy such constraints. However, in the case of a multi-stream system, a more general and flexible framework is useful for dynamic bit rate adaptation. . . . van Beek, [0086]-[0090]. Various network technologies may be used **for the gateway reception and transmission,** such as for example, IEEE 802.11, Ethernet, and power-line networks (e.g., HomePlug

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Grounds 1 and 2)	
	Powerline Alliance). While such networks are suitable for data transmission, they do not tend to be especially suitable for audio/video content because of the stringent requirements imposed by the nature of audio/video data transmission. Moreover, the network capabilities, and in particular the data maximum throughput offered, are inherently unpredictable and may dynamically change due to varying conditions described above. **The data throughput may be defined in terms of the amount of actual (application) payload bits (per second) being transmitted from the sender to the receiver successfully. It is noted that while the system may refer to audio/video, the concepts are likewise used for video alone and/or audio alone.** van Beek, [0055].

Claims	Relevant Disclosures in the Prior Art
Claim 10 (Grounds 1 and 2)	
[10P] The method of claim 9, further comprising	To the extent the preamble is limiting, van Beek discloses, or at least renders obvious, the method of claim 9. *See* claim 9 above.
[10a] dropping frames of the encoded video data.	van Beek discloses, or at least renders obvious, *dropping frames of the encoded video data* (e.g., van Beek's disclosure of a transcoder changing frame rate in response to bandwidth usage), as claimed. The '285 Patent itself describes that "frame dropping can be executed, when needed, by frame dropper 226." EX1001, 8:36-27. Further, "[w]hen 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," and that "[f]rame dropper 226 can drop the frames accordingly to deliver the optimal session bitrate." EX1001, 8:43-57. van Beek discloses that, in a situation where its system "has information about the resources available to the client device consuming the video signal," it "may further **increase or decrease the output video quality in accordance with the device resources by adjusting bandwidth usage** accordingly." van Beek, [0070]. Several methods for adjusting bandwidth usage using van Beek's

<table>
<tr><th>Claims</th><th>Relevant Disclosures in the Prior Art</th></tr>
<tr><td colspan="2">Claim 10 (Grounds 1 and 2)</td></tr>
<tr><td></td><td>transcoder are disclosed: “a transcoder may for example: change bit rate, change frame rate, change spatial resolution, and change the compression format.” <i>Id</i>. Based on this disclosure, in a situation where a system needs to decrease the bandwidth being consumed across a network connection, one such solution provided by van Beek would be to have the transcoder decrease the frame rate—resulting in <i>dropping frames of the encoded video data</i>, as claimed. This is similar to the ’285 Patent’s disclosure that its system can dynamically determine a “frame dropping rate.”

van Beek also teaches that a reduced bandwidth can be achieved by “reducing the resolution of the target stream,” e.g.:

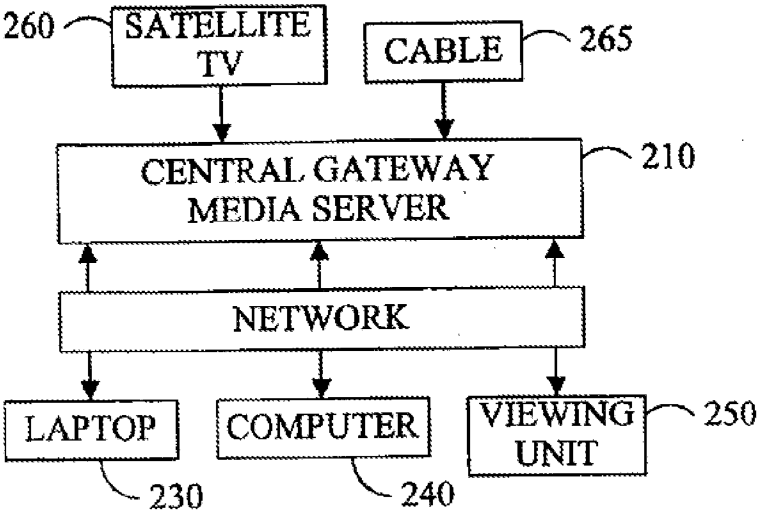
<blockquote>In one existing system, the start time of each unit of media for each stream is matched against the estimated transmission time for that unit. When any one actual transmission time exceeds its estimated transmission time by a predetermined threshold, the network is deemed to be close to saturation, or already saturated, and the system may select at least one stream as a target for lowering total bandwidth usage. Once the target stream associated with a client receiver unit is chosen, the target stream is modified to transmit less data, which may result in a lower data transmission rate. For example, a decrease in the data to be transmitted can be accomplished by a gradual escalation of the degree of data compression performed on the target stream, thereby reducing the resolution of the target stream. If escalation of the degree of data compression alone does not adequately reduce the data to be transmitted to prevent bandwidth saturation, the resolution of the target stream can also be reduced. For example, if the target stream is a video stream, the frame size could be scaled down, reducing the amount of data per frame, and thereby reducing the data transmission rate.</blockquote>
van Beek, [0042].

For the reasons set forth above, van Beek discloses <i>dropping frames of the encoded video data</i>, as claimed.</td></tr>
</table>

Claims	Relevant Disclosures in the Prior Art
Claim 10 (Grounds 1 and 2)	
	(**Ground** 2) To the extent this disclosure from van Beek alone does not disclose limitation [10a], such would have been obvious to a POSITA because the concept of dropping frames of encoded video data was well known at the time of the '285 Patent. Karam Decl., ¶ 48. A POSITA would have understood that the resolution of the target video stream includes a spatial resolution (i.e., size of frame) and a temporal resolution (i.e.., number of frames per second or frame rate). Karam Decl., ¶ 48. Thus, reducing the frame rate comprises "a change of frame rate" by "reducing [the temporal] resolution of the target stream" as taught by van Beek. *Id.* For example, U.S. Patent 7,734,800 to Gupta et al. (EX1006), states that "[o]ne easy way to reduce bandwidth is to simply drop lower-level dependent frames from the video stream." Gupta, 13:13-23. Thus, just like the frame rate regulation disclosed by van Beek, a POSITA would have been well aware that dropping frames of encoded video data was a common method for decreasing the bit rate of a stream. Karam Decl., ¶ 49. In view of this knowledge a POSITA would have possessed at the time, it would have been obvious that van Beek's system—seeking to reduce video bitrate by lowering frame rate—renders obvious *dropping frames of the encoded video data*. Karam Decl., ¶¶ 48-49. * * * Requester provides the following disclosures from van Beek relevant to this claim limitation: > If the system, including for example the extender, has information about the resources available to the client device consuming the video signal as previously described, **the extender may further increase or decrease the output video quality in accordance with the device resources by adjusting bandwidth usage accordingly**. For example, consider an MPEG-1 source stream at 4 Mbps with 640 by 480 spatial resolution at 30 fps. If it is being transmitted to a resource-limited device, e.g., a handheld with playback capability of 320 by 240 picture resolution at 15 fps, the transcoder may reduce the rate to 0.5 Mbps by simply subsampling the video without increasing the quantization levels. Otherwise, without subsampling, the transcoder may

Claims	Relevant Disclosures in the Prior Art
Claim 10 (Grounds 1 and 2)	
	have to increase the level of quantization. In addition, the information about the device resources also helps prevent wasting shared network resources. A transcoder may also convert the compression format of an incoming digital video stream, e.g., from MPEG-2 format to MPEG-4 format. **Therefore, a transcoder may for example**: change bit rate, **change frame rate**, change spatial resolution, and change the compression format. van Beek, [0070].

Claims	Relevant Disclosures in the Prior Art
Claim 15 (Grounds 1 and 2)	
[15P] 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:	To the extent the preamble is limiting, van Beek discloses, or at least renders obvious, *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* (e.g., van Beek's adaptive bandwidth transmission system for transmitting multiple data streams, such as video and audio streams, in a network), for the reasons discussed above in claim 9. *See* claim 9 above. In addition, van Beek makes clear that its disclosures are performed by a computer executing instructions, e.g.: FIG. 1 illustrate[s] a system for transmission of multiple data streams in a network that may have limited bandwidth. The system includes a **central gateway media server 210** and a plurality of **client receiver units 230, 240, 250**. The central gateway media server may be any device that can transmit multiple data streams. The input data streams may be stored on the media server or arrive from an external source, such as a satellite television transmission 260, a digital video disc player, a video cassette recorder, or a cable head end 265, and are transmitted to the client receiver units 230, 240, 250 in a compressed format. The data streams can include display data, graphics data, digital data, analog data, multimedia data, audio data and the like. An adaptive bandwidth system on the **gateway media server 210 determines the network**

Claims	Relevant Disclosures in the Prior Art
Claim 15 (Grounds 1 and 2)	
	bandwidth characteristics and adjusts the bandwidth for the output data streams in accordance with the bandwidth characteristics. van Beek, [0041].  FIG. 1 van Beek, FIG. 1 (e.g., depicting central gateway media server 210 and computer 240). A test setup was implemented using **software running on two Windows 2000 laptop PCs**, both equipped with IEEE 802.11b WLAN client cards. These WLAN cards on these laptops were configured to communicate in the 802.11 ad-hoc mode, and the IP protocol settings were configured to create a 2 laptop private network. **Software running on one PC acted as a server, sending packets over the network to the receiver using the UDP, IP and 802.11b protocols**. Note that UDP may be used instead of TCP, as UDP is more suitable for real-time traffic. It is noted that the system may use other protocols, such as for example, the Powerline Communication networks or other LANs. van Beek, [0226].

Claims	Relevant Disclosures in the Prior Art
Claim 15 (Grounds 1 and 2)	
	(**Ground 2**) To the extent van Beek is not found to disclose this limitation, a POSITA would have nonetheless found this limitation obvious in view of the disclosures above.
[15a] receiving the optimal session bitrate;	van Beek discloses, or at least renders obvious, *receiving the optimal session bitrate* for the same reasons as discussed above with respect to element [9a] of claim 9 (reciting "receiving an optimal session bitrate"). *See* limitation [9a] above.
[15b] allocating the optimal session bitrate between audio and video media to produce an optimal audio bitrate and an optimal video bitrate,	van Beek discloses, or at least renders obvious, *allocating the optimal session bitrate between audio and video media to produce an optimal audio bitrate and an optimal video bitrate* for the same reasons as discussed above with respect to element [9b] of claim 9 (reciting an identical limitation to [15b]). *See* limitation [9b] above.
[15c] 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;	van Beek discloses, or at least renders obvious, *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* for the same reasons as discussed above with respect to element [9c] of claim 9 (reciting an identical limitation to [15c]). *See* limitation [9c] above.
[15d] encoding audio and video media data according to the optimal audio bitrate and the optimal video bitrate; and	van Beek discloses, or at least renders obvious, *encoding audio and video media data according to the optimal audio bitrate and the optimal video bitrate* for the same reasons as discussed above with respect to element [9d] of claim 9 (reciting an identical limitation to [15d]). *See* limitation [9d] above.

Claims	**Relevant Disclosures in the Prior Art**
Claim 15 (Grounds 1 and 2)	
[15e] providing the encoded audio and video data for transmittal to a terminal.	van Beek discloses, or at least renders obvious, *providing the encoded audio and video data for transmittal to a terminal* for the same reasons as discussed above with respect to element [9e] of claim 9 (reciting an identical limitation to [15e]). *See* limitation [9e] above.

EXHIBIT AA-2

Claim Chart

Comparing Claims 9, 10, and 15 of the ’285 Patent

to Urzaiz, Gupta, and Pogrebinsky

I. GROUNDS OF UNPATENTABILITY

Ground	Claims	Statutes	Prior Art
3	9, 10, 15	35 U.S.C. § 103	Urzaiz, Gupta, and Pogrebinsky

A. Prior Art Relied Upon

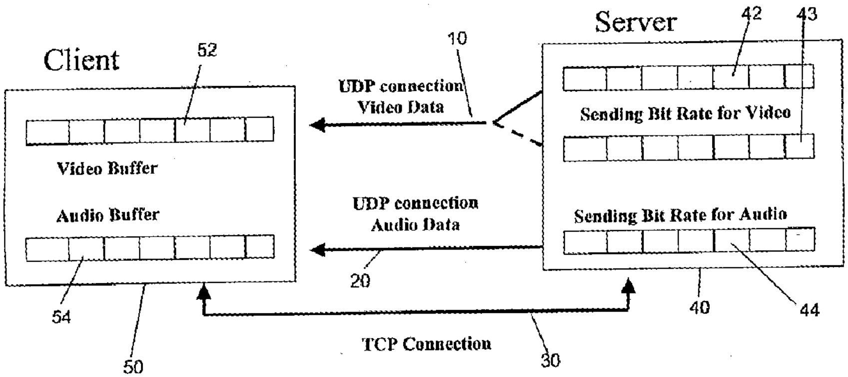
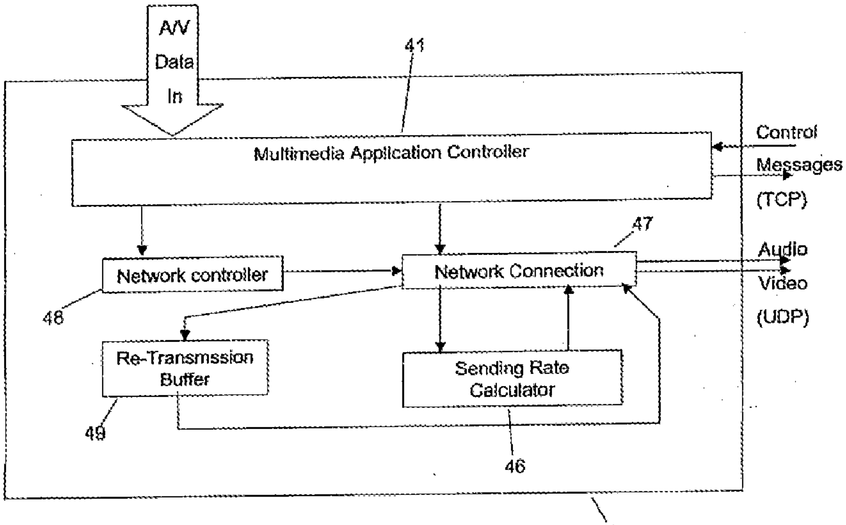
The ’285 Patent was filed on July 9, 2008 and claims priority to provisional application 60/948,917, filed July 10, 2007. Accordingly, the earliest possible priority date for the ’285 Patent is **July 10, 2007**.

Prior Art
EX1005 (“Urzaiz”) U.S. Patent Publication 2005/0021830 to Eduardo Urzaiz et al. was filed on September 13, 2002 and published on January 27, 2005. Accordingly, Urzaiz qualifies as prior art to the ’285 Patent under at least 35 U.S.C. §§ 102(a) and (b) (pre-AIA).
EX1006 (“Gupta”) U.S. Patent 7,734,800 to Anoop Gupta et al. was filed on August 25, 2003, published on February 26, 2004, and issued on June 8, 2010. Accordingly, Gupta qualifies as prior art to the ’285 Patent under at least 35 U.S.C. §§ 102(a) and (b) (pre-AIA).
EX1007 (“Pogrebinsky”) U.S. Patent 7,142,506 to Vladimir Pogrebinsky was filed on February 2, 1999 and issued on November 28, 2006. Accordingly, Pogrebinsky qualifies as prior art to the ’285 Patent under at least 35 U.S.C. §§ 102(a) and (e) (pre-AIA).

B. Claim Charts

The claim charts below first include an argument portion explaining why Urzaiz, Gupta, and Pogrebinsky disclose and render obvious the pertinent limitation, followed by additional supplementary citations to Urzaiz, Gupta, and Pogrebinsky. Unless otherwise noted, all emphasis has been added by Requester (for example, claim language is denoted by italics, while the corresponding disclosure in the prior art is indicated in bold).

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Ground 3)	
[9P] A method comprising:	To the extent the preamble is limiting, Urzaiz discloses *a method* (e.g., Urzaiz's data transmission method and system for transmitting audio and video data from a server to a client). * * * Requester provides the following disclosures from Urzaiz relevant to this claim limitation: A **data transmission method** and system **is disclosed in which one or more data streams are transmitted at respective transmission rates** which are controlled to prevent data buffers in the receiver from overflowing. In some embodiments feedback data concerning the state of each buffer in a receiving client is received at the transmitting server, and used to adapt the sending rates to achieve the effect. Information indicative of the data decode rates or the fill extent of each buffer is communicated to the server as the feedback data. In other embodiments the server makes an open-loop estimate of the remaining space in the buffer, and controls the transmission rate accordingly. A data receiving method and system adapted to receive the data streams is also disclosed. Urzaiz, Abstract. Commonly, the **data to be streamed is multi-media data such as, for example, audio and video data.** The audio and video data may be from a live audio visual broadcast such as a news or sports event, or may be sourced from, for example, a video-on-demand service which permits subscribers to watch television programmes and films of their choice as and when they choose. Whatever the source of the data, however, the respective audio and video feed data must first be suitably digitally encoded in order to compress the audio and video data signals to a size suitable for transmission over a network. Commonly, audio and video encoding is performed in accordance with one of the various MPEG standards. Following encoding of the **audio and video data, the encoded data is passed to a network server, where it is stored in separate audio and video buffers prior to transmission over the network to a client**. Urzaiz, [0004]-[0005].

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Ground 3)	
	 Fig.3 Urzaiz, FIG. 3.  Fig. 4 Urzaiz, FIG. 4.
[9a] receiving an optimal session bitrate;	Urzaiz discloses, or at least renders obvious, *receiving an optimal session bitrate* (e.g., Urzaiz's disclosure of a "sending rate calculator" that computes and receives feedback about a "total_rate"), as claimed. Although the '285 Patent does not explicitly define the claim term *optimal session bitrate*, it does describe and claim exemplary ways of computing it; for example, the '285 Patent describes that to "compute the optimal session bitrate, adaptive bitrate controller 210

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Claim 9 (Ground 3)	
	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," using well known network state estimators like media time in transit (MTT) or round trip time estimate (RTTE). EX1001, 4:22-26. In other words, the system analyzes the state of the connection between the transmitting terminal and the receiving terminal to obtain a bitrate suitable for streaming media. Accordingly, a POSITA would have understood that, in the context of transmitting (or streaming) video and audio media from a server to a client over a network connection, an optimal session bitrate would broadly refer to the maximum (or similarly, total) bitrate capable of being supported by the connection between the terminal (or server) transmitting the video and audio media to the receiving terminal (or client). Karam Decl. (EX1003), ¶ 70. Requester further notes that other claims of the '285 Patent (e.g., claim 1) claim a method for determining an optimal session bitrate (e.g., claim 1 recites "determining an optimal session bitrate using the estimated one or more network conditions" and includes claim limitations directed to network stability criteria, such as MTT, RTT, and current bitrate); but claim 9 is not so limited. All claim 9 requires is that the method being performed includes ***receiving** an optimal session bitrate*. Accordingly, this limitation of claim 9 is disclosed by any reference that describes a method for receiving the maximum (or total) bitrate that can be supported on a network connection for streaming audio and video media between terminals (e.g., a server and a client). Just like the '285 Patent's "adaptive bitrate controller 210" (which in the specification example is used to compute the optimal session bitrate), Urzaiz describes that "at step 2 [of Figure 11] sending rate calculator 46 calculates **the total bandwidth available for all of the individual data streams which are to be transmitted from the server computer 40**." Urzaiz, [0125]. Urzaiz explains that this "value **total_rate represents the upper limit on transmission rate which the individual transmission rates of each separate data stream** when summed together should not be greater." *Id.*

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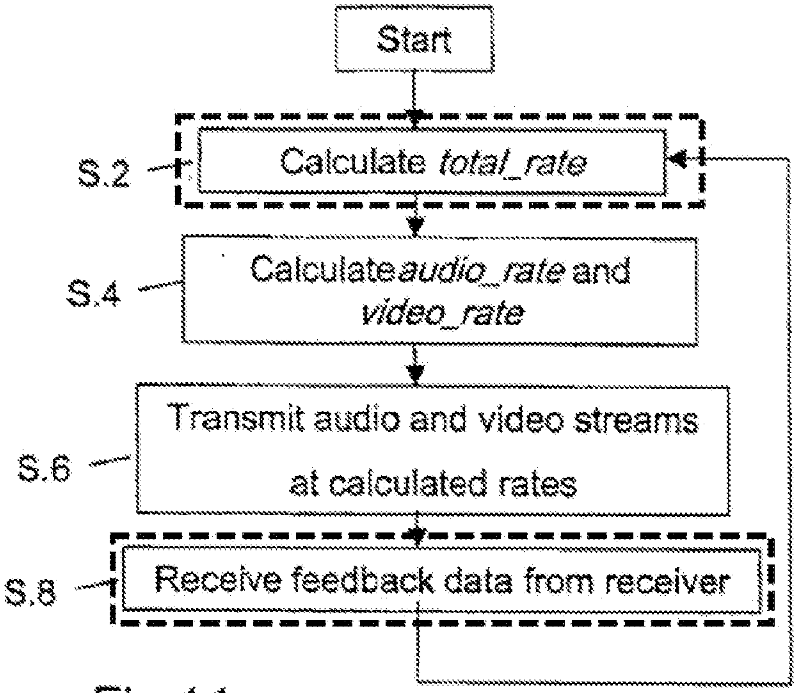
Claim 9 (Ground 3)	
	 Fig.11 Urzaiz, FIG. 11 (annotated). These disclosures demonstrate that Urzaiz seeks to determine and adjust bandwidth for transmitting audio and video data streams in the same way that the '285 Patent discloses doing so for an *optimal session bitrate*.

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Ground 3)	
	Urzaiz is also directed to determining an *optimal* bit rate because it describes that its server is "capable of calculating the **maximum transmission rate** available for the stream **dependent upon the present conditions on the network, thereby optimising the transmission rate at which the stream is transmitted**." Urzaiz, [0035].[1] By optimizing the transmission rate based on network conditions, Urzaiz calculates its "total_rate" in the same manner as the '285 Patent. Finally, Urzaiz describes ***receiving*** *an optimal session bitrate* by disclosing that at step "S8 of FIG. 11 **the server computer 40 <u>receives</u> feedback data from the client computer 50**, which in the preferred embodiment **is that data which is required to perform the total transmission rate and data stream transmission rate calculations of steps S2 and S4**." Urzaiz, [0135]. Accordingly, when Urzaiz *receives* the feedback data from the client computer (i.e., represented by the line in Figure 11 from S.8 to S.2), which is the information used to calculate Urzaiz's "total_rate," Urzaiz's system discloses *receiving an optimal session bitrate*. For the reasons set forth above, Urzaiz discloses *receiving an optimal session bit rate*, as claimed. * * * Requester provides the following disclosures from Urzaiz relevant to this claim limitation:

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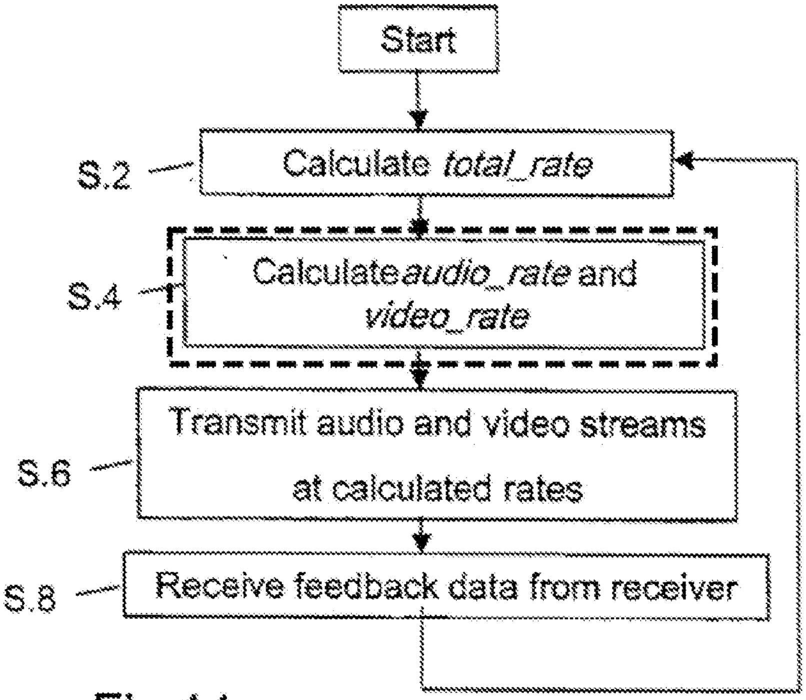
[1] Requester notes that, while Urzaiz is directed to three primary embodiments (i.e., the first embodiment being described in paragraphs 87-122, and the second embodiment being described in paragraphs 123-144, and a third embodiment in paragraphs 145-164), the majority of disclosure cited in these charts is from the first two embodiments—the only differences between the two being that the second embodiment is "particularly concerned with **sending more than one data stream to the same client**, and in particular with sending simultaneous real-time audio and video data in separate audio and video data streams" (Urzaiz, [0124]), while the first embodiment only describes one or more streams (Urzaiz, [0088]). These embodiments, however, are complementary, and Urzaiz is explicit in describing its second embodiment that, e.g., "[t]he same considerations **for the calculation of the transmission rate of each stream apply in the second embodiment as in the first embodiment**." Urzaiz, [0126]. Accordingly, while Requester cites primarily to the second embodiment of Urzaiz (because it is the most like the '285 Patent's sending audio and video data streams to the same client), citations to other portions of Urzaiz are made as well given their application to all of Urzaiz's embodiments.

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Ground 3)	
	> Preferably, the invention is further arranged to **receive feedback data from the or each receiver indicative of one or more of a round trip time (RTT), a loss rate value, and/or a receiving rate value at the receiver**, and furthermore to **calculate the total transmission rate as a function of one or more of the received values indicated by the feedback data**. The round trip time is a measure of the it takes for data to travel from a transmitter to the receiver and back to the transmitter, whereas the loss rate value is a measure of the amount of data transmitted to the receiver which is lost in the network. The receiving rate value is the number of bits received by the receiver in the round trip time. > By **providing feedback from the receiver to the server it is possible to provide the server with up to date information** indicative of, for example, congestion conditions on the network resulting in packet losses. **The server then becomes capable of calculating the maximum transmission rate available for the stream dependent upon the present conditions on the network, thereby optimising the transmission rate at which the stream is transmitted.** Urzaiz, [0034]-[0035]. > In the first embodiment, **the total transmission rate parameter max_rate is calculated using a transmission rate formula which has been derived so as to model the average throughput over time of a TCP connection**, and therefore total rate is calculated so as to provide a TCP-friendly transmission rate. Urzaiz, [0092]. > Equation 1 gives a value **bit_rate_stream which is an estimate of the average bandwidth that a single TCP connection would achieve** in the present network conditions. However, in the first embodiment we do not use this estimate directly as the total transmission rate for a stream, but rather this value bit_rate_stream is placed into equation 2 as set out below: > max_rate=min(bit_rate_stream,2*receiving_rate_stream) Eq. 2 Urzaiz, [0095].

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Ground 3)	
	The operation of a second embodiment of the present invention will now be described with reference to FIGS. 8 to 13. The second embodiment of the invention is particularly concerned with sending more than one data stream to the same client, and in particular with sending simultaneous real-time audio and video data in separate audio and video data streams. Furthermore, as with the first embodiment the second embodiment is also concerned with controlling the transmission rate of the stream in a closed-loop manner[.] FIG. 11 is a flow diagram of the steps performed by the server computer **40** in accordance with the second embodiment of the present invention. **Firstly, at step 2 the sending rate calculator 46 calculates the total bandwidth available for all of the individual data streams which are to be transmitted from the server computer 40. This value total_rate represents the upper limit on transmission rate which the individual transmission rates of each separate data stream when summed together should not be greater**. The value total_rate is calculated in accordance with the following principles. The **same considerations for the calculation of the transmission rate of each stream apply in the second embodiment as in the first embodiment**, and we therefore apply equations 1 and 2 as previously described in respect of the first embodiment separately to each stream to obtain a value max rate for each stream, representing the maximum individual transmission rate for each of the audio and video data streams. However, in the present embodiment we are concerned with the transmission of multiple streams, and hence the above calculations must be performed separately for each stream to be transmitted. **That is, both Equations 1 and 2 are applied in order to each stream (i.e. the audio and video streams in the second embodiment) and the value max rate found for each stream. The respective values thus found for each stream are then summed together to give the value total rate, being the total bandwidth available to all streams to provide for TCP-friendly performance**, and thereby taking into account possible network congestion. **Following the calculation of the available total transmission rate,** at step S4 the **sending rate calculator 46**

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Ground 3)	
	in the server calculates the individual transmission rates for each data stream, being in the second embodiment the transmission rate of the audio UDP stream (audio_rate) and the transmission rate of the video UDP stream (video_rate). The values of audio_rate and video_rate are calculated as follows. As mentioned previously with respect to FIG. 3, the audio data is transmitted in a UDP stream separately from the video data which is transmitted in another UDP stream, and there are therefore two separate UDP connections one for each stream. Although it could be thought that each stream is competing for the same network bandwidth, in reality this is not true because it is not possible to send video and audio data packets at the same instant. Therefore, in the case of two data streams being audio and video streams, the previously calculated total sending bit rate can be made the equivalent of the audio sending bit rate plus the video sending bit rate. Furthermore, as will be described later, in the second embodiment the server is receiving information from the client about the state of the video and audio buffers, and the decoding rate for the video and audio packets. It therefore becomes possible to control the sending rates of the audio and video data streams to control the filling rate of the buffers in the client. This is achieved as follows. Urzaiz, [0124]-[0128]. Thus, as will be apparent from the above, it becomes possible to control the respective audio sending rates and video sending rates to trade bit rate from one stream to the other depending upon the respective audio and video decode rates in the receiver. **Furthermore, it should be noted above that the parameter total_rate is the value calculated previously from the application of Equations 1 and 2 to give the total available bandwidth available for the transmission of all the data streams i.e. total_rate = total_rate_stream_1 + total_rate_stream_2 + . . . + total_rate_stream_n** wherein n is the number of data streams being transmitted simultaneously. Returning to FIG. 11, after the calculation of the audio and video sending rates for each stream, at step S6 the network connection 47 in the server transmits the audio and video streams as separate UDP data streams, at the calculated audio

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Ground 3)	
	and video sending rates. It should be noted that as the audio and video steams [*sic*] are continuously transmitted, the steps of FIG. 11, although depicted sequentially, are actually performed in parallel, such the transmission rates of the audio and video streams are in reality updated once new values for the audio and video transmission rates have been calculated. While the new calculations are being performed, however, these streams continue to be transmitted at the previously calculated rate. FIG. 13 shows a plot of the measured transmission rate of one data stream controlled in accordance with the embodiments of the present invention, when transmitting the same data as that transmitted by the TCP connection plotted in FIG. 2. From FIG. 13 it will be seen that after initial transient variations experienced at the opening of the session, the transmission rate of the stream steadies out, and continues with relatively little variance over time. Furthermore, when compared to the transmission rate experienced by the TCP connection shown in FIG. 2 it will be seen that an almost identical average throughput is achieved as TCP, but without the large changes in transmission rate which result from TCP's multiplicative decrease control algorithm. This property of providing a smooth transmission rate with respect to time renders the present invention particularly suitable for use in transmitting data which requires continuous streaming. At **Step S8 of FIG. 11 the server computer 40 receives feedback data from the client computer 50, which in the preferred embodiment is that data which is required to perform the total transmission rate and data stream transmission rate calculations of steps S2 and S4**. In particular for each stream the server receives data informing it of the round trip time presently being experienced at the client, the loss rate of packets at the client, the respective decoding rates of the audio and video buffers in the client, and the data receiving rate of each data stream at the client. **These quantitative values are transmitted back to the server via the TCP connection from the client.** Urzaiz, [0132]-[0135].

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Ground 3)	
[9b] allocating the optimal session bitrate between audio and video media to produce an optimal audio bitrate and an optimal video bitrate,	Urzaiz discloses, or at least renders obvious, *allocating the optimal session bitrate between audio and video media to produce an optimal audio bitrate and an optimal video bitrate* (e.g., Urzaiz's disclosure that describes calculating individual transmission rates for each data stream, including an "audio_rate" and a "video_rate"), as claimed. First, Urzaiz describes *allocating* the total_rate described in the previous limitation (i.e., the claimed optimal session bitrate) between audio and video in step S.4 of Figure 11, by describing that "the sending rate calculator 46 in the server **calculates the individual transmission rates for each data stream**, being in the second embodiment the **transmission rate of the audio UDP stream (audio_rate)** and the **transmission rate of the video UDP stream (video_rate)**." Urzaiz, [0127].  Fig.11 Urzaiz, FIG. 11 (annotated). Urzaiz explains that "in the case of **two data streams being audio and video streams, the previously calculated total sending bit rate can be made the equivalent of the audio sending bit rate plus the video sending bit rate**." Urzaiz, [0128]. Because the audio and video bitrates in Urzaiz when summed are equivalent to the total rate,

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Claim 9 (Ground 3)	
	Urzaiz has accordingly *allocated* the total_rate among the audio_rate and video_rate when it performs its calculation in step S.4. Finally, as described above, this process in step S.4 of Urzaiz produces an *optimal* audio bitrate and an *optimal* video bitrate because the entire purpose of Urzaiz is "**optimising the transmission rate** at which the stream is transmitted." Urzaiz, [0035]; *See also* Urzaiz, [0126] (teaching that Equation 2 is used to compute the optimal max_rate for each stream and that the optimal total_rate (*optimal session bitrate*), being the value of the total bandwidth available to all streams, is then determined by summing the respective computed max_rate values found for each stream), [0128] (teaching that once the optimal total_rate (*optimal session bitrate*) is computed, sending rate calculator calculates the individual transmission rates for each data stream, i.e., audio-rate for the audio stream and video_rate for the video stream (*allocating the optimal session bitrate between audio and video media to produce an optimal audio bitrate and an optimal video bitrate*)). For the reasons set forth above, Urzaiz discloses *allocating the optimal session bitrate between audio and video media to produce an optimal audio bitrate and an optimal video bitrate*, as claimed. * * * Requester provides the following disclosures from Urzaiz relevant to this claim limitation: Commonly, the **data to be streamed is multi-media data such as, for example, audio and video data.** The audio and video data may be from a live audio visual broadcast such as a news or sports event, or may be sourced from, for example, a video-on-demand service which permits subscribers to watch television programmes and films of their choice as and when they choose. Whatever the source of the data, however, the respective audio and video feed data must first be suitably digitally encoded in order to compress the audio and video data signals to a size suitable for transmission over a network. Commonly, audio and video encoding is performed in accordance with one of the various MPEG standards. Following **encoding of the audio and video data, the encoded data is passed to a network server, where it is**

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Ground 3)	
	stored in separate audio and video buffers prior to transmission over the network to a client. Urzaiz, [0004]-[0005]. The operation of a second embodiment of the present invention will now be described with reference to FIGS. **8** to **13**. The **second embodiment of the invention is particularly concerned with sending more than one data stream to the same client, and in particular with sending simultaneous real-time audio and video data in separate audio and video data streams**. Furthermore, as with the first embodiment the second embodiment is also concerned with controlling the transmission rate of the stream in a closed-loop manner[.] FIG. 11 is a flow diagram of the steps performed by the server computer **40** in accordance with the second embodiment of the present invention. Firstly, at step 2 the sending rate calculator 46 calculates the total bandwidth available for all of the individual data streams which are to be transmitted from the server computer 40. This value total_rate represents the upper limit on transmission rate which the individual transmission rates of each separate data stream when summed together should not be greater. The value total_rate is calculated in accordance with the following principles. The same considerations for the calculation of the transmission rate of each stream apply in the second embodiment as in the first embodiment, and we therefore apply equations 1 and 2 as previously described in respect of the first embodiment **separately to each stream to obtain a value max rate for each stream, representing the maximum individual transmission rate for each of the audio and video data streams. However, in the present embodiment we are concerned with the transmission of multiple streams, and hence the above calculations must be performed separately for each stream to be transmitted.** That is, both Equations 1 and 2 are applied in order to each stream (i.e. the audio and video streams in the second embodiment) and the value max rate found for each stream. The respective values thus found for each stream are then summed together to give the value total rate, being the total bandwidth available to all streams to provide for TCP-friendly performance, and thereby taking into account possible network congestion.

Claims	**Relevant Disclosures in the Prior Art**
Claim 9 (Ground 3)	
	Following the calculation of the available total transmission rate, **at step S4 the sending rate calculator 46 in the server calculates the individual transmission rates for each data stream, being in the second embodiment the transmission rate of the audio UDP stream (audio rate) and the transmission rate of the video UDP stream (video rate)**. The values of audio_rate and video_rate are calculated as follows. As mentioned previously with respect to FIG. 3, the audio data is transmitted in a UDP stream separately from the video data which is transmitted in another UDP stream, and there are therefore two separate UDP connections one for each stream. Although it could be thought that each stream is competing for the same network bandwidth, in reality this is not true because it is not possible to send video and audio data packets at the same instant. **Therefore, in the case of two data streams being audio and video streams, the previously calculated total sending bit rate can be made the equivalent of the audio sending bit rate plus the video sending bit rate**. Furthermore, as will be described later, in the second embodiment the server is receiving information from the client about the state of the video and audio buffers, and the decoding rate for the video and audio packets. It therefore becomes possible to control the sending rates of the audio and video data streams to control the filling rate of the buffers in the client. This is achieved as follows. Urzaiz, [0124]-[0128]. Thus, as will be apparent from the above, **it becomes possible to control the respective audio sending rates and video sending rates to trade bit rate from one stream to the other depending upon the respective audio and video decode rates in the receiver**. Furthermore, it should be noted above that the parameter total_rate is the value calculated previously from the application of Equations 1 and 2 to give the total available bandwidth available for the transmission of all the data streams i.e. **total_rate = total_rate_stream_1 + total_rate_stream_2 + . . . + total_rate_stream_n** wherein n is the number of data streams being transmitted simultaneously.

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Ground 3)	
	Returning to FIG. 11, after the calculation of the audio and video sending rates for each stream, at step S6 the network connection 47 in the server transmits the audio and video streams as separate UDP data streams, at the calculated audio and video sending rates. It should be noted that as the audio and video steams [*sic*] are continuously transmitted, the steps of FIG. 11, although depicted sequentially, are actually performed in parallel, such the transmission rates of the audio and video streams are in reality updated once new values for the audio and video transmission rates have been calculated. While the new calculations are being performed, however, these streams continue to be transmitted at the previously calculated rate. Urzaiz, [0132]-[0133].
[9c] 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;	Urzaiz in view of Gupta and Pogrebinsky, discloses, or at least renders obvious, *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* (e.g., Urzaiz's disclosure of controlling audio and video bitrates by trading bitrates "from one stream to the other"; supplemented by Gupta's disclosure that data streams can be "assigned a priority" where "high-priority streams are given priority when allocating bandwidth" and Pogrebinsky's disclosure that in certain situations when allocating between video and audio streams, "priority will always be given to the audio channel"), as claimed. Urzaiz describes that its allocating described above in limitation [9b] is *based at least in part on privileging either the audio media or the video media over the other*, e.g.: Thus, as will be apparent from the above, it becomes **possible to control the respective audio sending rates and video sending rates to trade bit rate from one stream to the other** depending upon the respective audio and video decode rates in the receiver. Furthermore, it should be noted above that the parameter total_rate is the value calculated previously from the application of Equations 1 and 2 to give the total available bandwidth available for the transmission of all the data streams i.e. total_rate = total_rate_stream_1 + total_rate_stream_2 + . . . +

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Ground 3)	
	total_rate_stream_n wherein n is the number of data streams being transmitted simultaneously. Urzaiz, [0132]. As described in the paragraph above, Urzaiz's system will control (i.e., *privileging*) the audio and video bit rates depending on the audio and video decode rates in the receiver. This is further described in other portions of Urzaiz's disclosure in the context of the audio and video buffers, e.g., "where the sending rate of each stream can be controlled by the invention so as to be substantially smooth, and so as to prevent the receiver buffers from overflowing." Urzaiz, [0033]. Indeed, Urzaiz recognized the flexibility of prioritizing particular data streams over the other was an important problem to be solved in improving problems in the art. For example, Urzaiz notes that in older TCP-based systems, problems arise when one transmission stream applies its own control algorithm "without any regard to the transmission rate of the other stream," and that this is a problem because "for most audio video sources there is commonly much more video data to be transmitted per unit time than audio data." Urzaiz, [0011]. Without considering both streams and the total bitrate the connection can handle, Urzaiz noted this "can cause problems with the data buffers in the receiver, in that where temporary large differences occur, the audio buffer, for example, might fill and overflow thereby losing data, whereas the corresponding video buffer may have emptied therefore preventing AV reproduction from taking place." Urzaiz, [0012]. Accordingly, to fix these problems, Urzaiz describes and claims, e.g., "controlling the respective data transmission rates of at least a subset of the respective data streams **to trade bit-rate between said streams**." Urzaiz, claim 36. When Urzaiz's system does so, it performs *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* because whichever stream is having its bitrate increased (i.e., among the audio_rate and video_rate) is being privileged over the other given the current conditions of the system. To the extent Urzaiz does not disclose this limitation, it would have been obvious nonetheless in view of the teachings of Gupta and Pogrebinsky. Karam Decl., ¶ 71.

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Ground 3)	
	Gupta, like Urzaiz and the '285 Patent, is directed to a composite media stream containing an audio stream and a video stream. Gupta, Abstract, 3:34-54, 7:43-54. Gupta provides an example as follows: Now, suppose that a client requests the multimedia content over a communications **channel having a bandwidth of 56 Kbps**, at a speed factor of 2.0. At this speed factor, the client consumes **audio data** at twice the normal rate, which in this case is **32 Kbps**. That leaves **24 Kbps of available bandwidth**. Accordingly, the server selects the **low bandwidth video stream** with the timeline modified by a factor of 2.0, and **combines it with the audio stream to form a composite media stream for streaming to the client**. The total required communications bandwidth is 52 Kbps, which is within the limits of the available bandwidth. Although the example give[n] with reference to FIG. 11 is relatively specific, this method of bandwidth utilization can be generalized to include other types of media streams. **Each stream is assigned a priority. Audio will generally have a high priority. The high-priority streams are given priority when allocating bandwidth.** Thus, in the example above, the audio stream is streamed to the client at its full quality, while the video stream is reduced in quality to fit within the remaining bandwidth. Gupta, 12:60-13:12. Gupta, accordingly, discloses the concept of each stream being assigned a "priority," where "high-priority streams are given priority when allocating bandwidth." *Id.* Gupta also provides a specific example where, if the video speed is set to a higher rate, an audio stream can be given higher priority to be streamed "at its full quality," while decreasing the bitrate assigned to the video stream. *Id.* As applied to Urzaiz, it would have been obvious to a POSITA to assign priorities to the different audio and video streams already present in Urzaiz's system by "**trading bit-rate between said streams**" as taught by Urzaiz. A POSITA would have been motivated to look for prior art teaching how to accomplish this bit-

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Ground 3)	
	rate trading and would have looked for Gupta as a suitable solution to accomplish the bit-rate trading in Urzaiz's system. Further, a POSITA would have had a reasonable expectation of success making such a combination because modifying the method of Urzaiz in view of Gupta would have involved applying a known technique to improve a similar method in a similar way to yield predictable results of prioritizing the audio stream over the video stream. Karam Decl., ¶ 71. Similar again to Urzaiz and the '285 Patent, Pogrebinsky discloses a system "for adjusting of bit rate transmission in a communication network . . . in accordance with the network state detected," but in the context of a "multimedia call." Pogrebinsky, Abstract. Pogrebinsky describes the use of an "allocator" that is coupled to an "audio bit rate control device" and a "video bit rate control device." Pogrebinsky, 4:9-44, FIG. 2. In an example, Pogrebinsky describes: > The continuation of this first algorithm is such that the audio and video bit rate controls 19, 20 respectively, are queried for the total bit rate between the audio and video channels, at step 240. During this step 240, at least one, and preferably both the audio and video channels, are sampled by the audio sampling device 2 and video sampling devise 23 respectively, in communication with their respective bit rate controllers 19, 20. The allocator 21 includes hardware and software that can detect the total bit rate by querying the bit rate in the audio and video bit rate controllers 19, 20 (as per the audio and video channels respectively) and combining the bit rates to find the total bit rate at step 240. Accordingly, the allocator [sic] 21 will know the quality of the audio transmission in accordance with the Table of FIG. 4. **Also, in accordance with the Table of FIG. 4, the allocator 21 will know the total bit rate available, such that it can allocate bit rate between the audio and video bit rate controllers 19, 20, at step 242. In making the allocation, priority will always be given to the audio channel. Such that the minimum bit rate for the audio is in accordance with the bit rates of the table FIG. 4**.

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Ground 3)	
	Pogrebinsky, 8:44-64. Accordingly, because Pogrebinsky is directed to a multimedia call, when allocating bitrate among audio and video, priority is always given to the audio channel (i.e., because the communication aspect of a multimedia call may be deemed more important than the video content). Once again, Pogrebinsky discloses and describes the claimed step of *privileging either audio media or the video media over the other*, and as applied to Urzaiz, provides a POSITA with a specific example where Urzaiz's audio and video streams would be prioritized over the other to maintain acceptable audio quality. As applied to Urzaiz, it would have been obvious to a POSITA to assign priorities to the different audio and video streams already present in Urzaiz's system by "**trading bit-rate between said streams**" as taught by Urzaiz. A POSITA would have been motivated to look for prior art teaching how to accomplish this bit-rate trading and would have looked for Pogrebinsky as a suitable solution to accomplish the bit-rate trading in Urzaiz's system. Further, a POSITA would have had a reasonable expectation of success making such a combination because modifying the method of Urzaiz in view of Pogrebinsky would have involved applying a known technique to improve a similar method in a similar way to yield predictable results of prioritizing the audio stream over the video stream. Karam Decl., ¶ 72. Furthermore, several other contemporaneous references and papers describe the concept of *privileging either the audio media or the video media over the other* and would have been known to a POSITA. For example, one paper investigated "the optimal trade-off between bits allocated to audio and to video under global bitrate constraints." *See* EX1009 (S. Winkler et al., Perceived Audiovisual Quality of Low-Bitrate Multimedia Content, IEEE Transactions on Multimedia, Vol. 8, No. 5 (October 2006)), at 973. This paper noted that in certain types of content, "relatively more bits should be allocated to the video," while in others, "a higher relative bitrate for the audio seems favorable." *See id.* at 977-978. Another paper recognized that in a system with separate audio and video streams, a controller can produce "an adaptive decision" to either "degrade audio, degrade video, upgrade audio, or upgrade video" depending on a selected option and network congestion. *See* EX1010 (M. A. Talaat et al., Content-Aware Adaptive Video Streaming System, International Conference on Information and Communication Technology, Cairo (December 2005)), at 267-268, 272. Finally, van Beek (EX1004, used in Grounds 1 and 2 of this request), explicitly

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Ground 3)	
	describes an example of prioritizing audio and video streams. *See* van Beek, [0121]. All of these contemporaneous references further confirm that a POSITA reading Urzaiz's discussion of trading and changing bitrates between audio and video streams would have understood that this can simply be accomplished by means or prioritizing, i.e., privileging the audio stream or video stream over the other as this was exceedingly well known in the prior art. Further, a POSITA would have had reasonable expectation of success in doing so. Therefore, Urzaiz in view of Gupta and Pogrebinsky discloses this limitation. Karam Decl., ¶ 71. Gupta and Pogrebinsky's teachings would have been combined with Urzaiz for the reasons discussed in the Request at Section I.E.3.d. Karam Decl., ¶¶ 64-68. For all of the reasons set forth above, Urzaiz in view of Gupta and Pogrebinsky, discloses *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*, as claimed. * * * Requester provides the following disclosures from Urzaiz relevant to this claim limitation: Thus, as will be apparent from the above, **it becomes possible to control the respective audio sending rates and video sending rates to trade bit rate from one stream to the other depending upon the respective audio and video decode rates in the receiver**. Furthermore, it should be noted above that the parameter total_rate is the value calculated previously from the application of Equations 1 and 2 to give the total available bandwidth available for the transmission of all the data streams i.e. **total_rate = total_rate_stream_1 + total_rate_stream_2 + . . . + total_rate_stream_n** wherein n is the number of data streams being transmitted simultaneously. Returning to FIG. 11, after the calculation of the audio and video sending rates for each stream, at step S6 the network connection 47 in the server transmits the audio and video streams as separate UDP data streams, at the calculated audio and video sending rates. It should be noted that as the audio

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Ground 3)	
	and video steams [*sic*] are continuously transmitted, the steps of FIG. 11, although depicted sequentially, are actually performed in parallel, such the transmission rates of the audio and video streams are in reality updated once new values for the audio and video transmission rates have been calculated. While the new calculations are being performed, however, these streams continue to be transmitted at the previously calculated rate. Urzaiz, [0132]-[0133]. A method of data transmission across a network, comprising the steps of: calculating a total transmission rate for the transmission of data using a transmission rate formula; transmitting data onto the network for transmission to a receiver in at least two separate data streams each at a respective data transmission bit rate; and **controlling the respective data transmission rates of at least a subset of the respective data streams to trade bit-rate between said streams**; wherein the sum of the respective transmission rates of each data stream is substantially equal to or less than the calculated total transmission rate. Urzaiz, claim 36. The problems associated with the frequent changes in data transmission rate using TCP for streaming data are further compounded when two or more data streams which contain related data, such as, for example, audio and video data, are to be transmitted simultaneously. In this case, when using TCP and taking the transmission of audio and video data in separate data streams as an example, because the audio stream is transmitted over a separate TCP connection from the video stream then each respective connection will apply its own transmission rate control algorithm without any regard to the transmission rate of the other stream. This has the resulting effect that over time the data throughput of the audio stream over the network becomes substantially the same as that of the video stream, whereas in reality for most audio visual sources there is commonly much more video data to be transmitted per unit time than audio data. This equality in transmission rate between the audio and video streams thus achieved by TCP

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Ground 3)	
	can have the effect at the receiver of affecting proper reproduction of the data, in that because the two types of data are not transmitted at respective rates which match the ratio of the generation of audio and video data, there will commonly be sufficient audio data stored in the receiver audio buffer for reproduction by the audio visual application, but insufficient video data in the receiver video buffer for reproduction at the same time as the audio data. Further problems arise from the separate application of the transmission rate control algorithm to each respective stream, and in particular from the multiplicative decrease nature of the standard TCP transmission rate control algorithm. Consider the case where an audio stream is being transmitted over a TCP connection separately to a video stream, which is also being transmitted using TCP. Usually, as explained previously, the average throughput of each connection would be substantially the same, but due to the multiplicative decrease in transmission rate when a packet loss on one of the streams occurs, at any particular moment in time there can in fact be large differences in the respective transmission rates of the two streams. These potentially large short term variations in transmission rate between the two streams introduce uncertainties into the data transmission, and can cause problems with the data buffers in the receiver, in that where temporary large differences occur, the audio buffer, for example, might fill and overflow thereby losing data, whereas the corresponding video buffer may have emptied therefore preventing AV reproduction from taking place. Urzaiz, [0011]-[0012]. In another variation, the server receives information relating to how full the buffers are, and performs step or continuous changes in the transmission rate to prevent the buffers from overflowing. There are many possible algorithms which could be applied in this case, such as, for example, the data rate being inversely related to the percentage of filling of the buffers (i.e. the greater the percentage the lower the data rate), or by achieving step changes using thresholding techniques (e.g. in a simple case: If buffer<x% full then transmit at a first higher rate, else if buffer>x% full then transmit at a second lower rate. Algorithms with more than one threshold can equally be envisaged). Step changes in transmission rate can be achieved

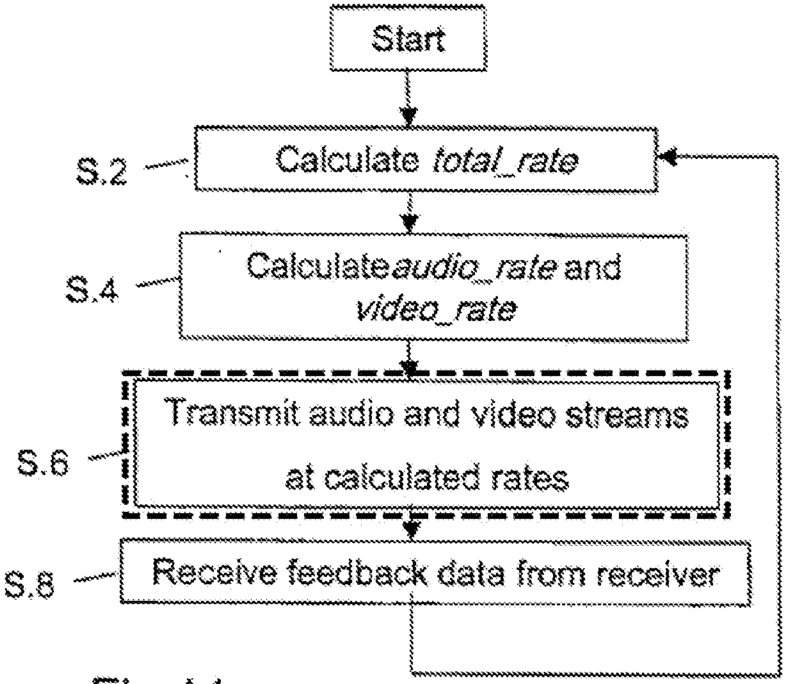
Claims	Relevant Disclosures in the Prior Art
Claim 9 (Ground 3)	
	by controlling the encoding of the source data to give a higher (better quality) or lower (poorer quality) encoding rate. Urzaiz, [0103]. In accordance with the above, the present invention also has application in the transmission of multiple data streams from a transmitter to one or more receivers which may be the same or different. Preferably, the transmitted data stream contains audio or video data. Where two data streams are transmitted simultaneously, preferably one of the streams contains audio data and the other of the streams contains video data. Preferably the audio aid [*sic*] video data is related in teat [*sic*] it is intended for reproduction at the receiver simultaneously, for example where the video data is a TV programme or film and the audio data is the soundtrack thereto. The invention is particularly intended for the transmission of audio and video data in streams, where the sending rate of each stream can be controlled by the invention so as to be substantially smooth, and so as to prevent the receiver buffers from overflowing. Preferably, the sending rate is controlled so as to match the read-out rate from the receiver buffers. Urzaiz, [0032]-[0033]. *See also* Urzaiz, [0129], [0130] (teaching that one can specify a ratio x:y between the audio and video rates, i.e., the respective rates at which the audio and video buffers in the receiver fill with data). * * * Requester provides the following disclosures from Gupta relevant to this claim limitation: Although the example give [*sic*] with reference to FIG. 11 is relatively specific, this method of bandwidth utilization can be generalized to include other types of media streams. **Each stream is assigned a priority. Audio will generally have a high priority. The high-priority streams are given priority when allocating bandwidth. Thus, in the example above, the audio stream is streamed to the client at its full quality,**

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Ground 3)	
	while the video stream is reduced in quality to fit within the remaining bandwidth. Gupta, 13:4-12. When a client 314 requests the multimedia content from server 310, **the server determines or notes both the speed factor** designated by the user **and the available bandwidth**. It then **selects the video stream that has best available quality while also requiring no more bandwidth (at the requested speed factor) than the difference between the available bandwidth and the bandwidth consumed by the selected audio stream**. Again, this allows the system to compensate for various available bandwidths. Gupta, 12:12-20. When a client 324 requests the multimedia content from server 320, the s**erver determines or notes both the speed factor** designated by the user **and the available bandwidth**. **It then selects an audio stream that most closely accords with the specified speed factor. It then selects the video stream that has best available quality while also requiring no more bandwidth than the difference between the available bandwidth and the bandwidth consumed by the selected audio stream**. Again, this allows the system to compensate for various available bandwidths. Gupta, 12:31-41. * * * Requester provides the following disclosures from Pogrebinsky relevant to this claim limitation: In FIG. 5, there is shown the method of the present invention. This method may be performed for example, to include stages of bit rate adjustment in accordance with the network state. A first stage is a first or coarse adjustment of the bit rate, while the second stage is a second or fine adjustment of the bit rate. This two stage bit rate adjustment, for example, is performed by algorithms.

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Ground 3)	
	A first algorithm is employed to detect the network state and perform a first or coarse adjustment of the bit rate in correspondence thereto. A second algorithm serves to cause a second or fine bit rate adjustment, to increase or decrease bit rate upon detection of congestion in the network. 1. **The bit rate, having been subject to a first (coarse) adjustment, and a second (fine) adjustment, if necessary, is then allocated among the audio and video channels by the allocator 21, such that the audio transmission is made with a proper bit rate allocated thereto in the preferred embodiment.** It is preferred that this exemplary method be performed in intervals of 5 seconds. Pogrebinsky, 6:29-48. The continuation of this first algorithm is such that the audio and video bit rate controls 19, 20 respectively, are queried for the total bit rate between the audio and video channels, at step 240. During this step 240, at least one, and preferably both the audio and video channels, are sampled by the audio sampling device 2 and video sampling devise 23 respectively, in communication with their respective bit rate controllers 19, 20. The allocator 21 includes hardware and software that can detect the total bit rate by querying the bit rate in the audio and video bit rate controllers 19, 20 (as per the audio and video channels respectively) and combining the bit rates to find the total bit rate at step 240. Accordingly, the alocator [*sic*] 21 will know the quality of the audio transmission in accordance with the Table of FIG. 4. Also, in accordance with the Table of FIG. 4, **the allocator 21 will know the total bit rate available, such that it can allocate bit rate between the audio and video bit rate controllers 19, 20, at step 242. In making the allocation, priority will always be given to the audio channel, such that the minimum bit rate for the audio is in accordance with the bit rates of the table FIG. 4.** Pogrebinsky, 8:44-64.
[9d] encoding audio and video media data according to the optimal audio bitrate and the optimal video bitrate; and	Urzaiz discloses, or at least renders obvious, *encoding audio and video media data according to the optimal audio bitrate and the optimal video bitrate* (e.g., Urzaiz's disclosure that audio and video data must be encoded prior to transmission), as claimed.

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Ground 3)	
	First, the *optimal audio bitrate* and *optimal video bitrate* claim terms in this limitation are disclosed by Urzaiz for the reasons set forth above in limitations [9a] and [9b]; this limitation merely requires that the method perform the *encoding* of the *audio and video media data* according to those rates. As to the step of *encoding audio and video media data*, Urzaiz describes that "[w]hatever the source of the data, however, the respective **<u>audio and video feed data</u> must first be suitably digitally <u>encoded</u> in order to compress the audio and video data signals to a size suitable for transmission over a network**. Commonly, audio and video encoding is performed in accordance with one of the various MPEG standards." Urzaiz, [0004]. Specifically, a POSITA would have understood that the encoding (i.e., the compression rate) in Urzaiz should be set in accordance with the transmission rate. *Id.*; Karam Decl., ¶ 74 (citing EX1008, [0051]-[0057]; and specifically, that it was known that "the variability of network bandwidth over time calls for a system that is dynamic in nature and capable of real time changes to the encoder settings." EX1008, [0055].). Urzaiz also teaches that "[f]ollowing **<u>encoding of the audio and video data</u>, the <u>encoded</u> data is passed to a network server**, where it is stored in separate audio and video buffers prior to transmission over the network to a client." Urzaiz, [0005]. For the reasons set forth above, Urzaiz discloses *encoding audio and video media data according to the optimal audio bitrate and the optimal video bitrate*, as claimed. * * * Requester provides the following disclosures from Urzaiz relevant to this claim limitation: Commonly, the **data to be streamed is multi-media data such as, for example, audio and video data.** The audio and video data may be from a live audio visual broadcast such as a news or sports event, or may be sourced from, for example, a video-on-demand service which permits subscribers to watch television programmes and films of their choice as and when they choose. Whatever the source of the data, however, **<u>the respective audio and video feed data must first be suitably digitally encoded in order to compress the audio and video</u>**

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Ground 3)	
	> **<u>data signals to a size suitable for transmission over a network</u>. Commonly, audio and video encoding is performed in accordance with one of the various MPEG standards.** > Following encoding of the **audio and video data, the encoded data is passed to a network server, where it is stored in separate audio and video buffers prior to transmission over the network to a client**. Urzaiz, [0004]-[0005]. > In order to provide for data communications between the server computer and the or [*sic*] each client computer a first user datagram protocol (UDP) connection **10** is provided between the server **40** and the or each client **50** along which **encoded video data is transmitted from the server 40.** Similarly, a second UDP connection **20** is also provided from the server **40** to the or each client **50** along which **encoded audio data is transmitted.** The transmission rates of the respective UDP connections **10** and **20** are controlled by the server in a manner to be described later for each embodiment of the invention. Urzaiz, [0073]. > In another variation, the server receives information relating to how full the buffers are, and performs step or continuous changes in the transmission rate to prevent the buffers from overflowing. There are many possible algorithms which could be applied in this case, such as, for example, the data rate being inversely related to the percentage of filling of the buffers (i.e. the greater the percentage the lower the data rate), or by achieving step changes using thresholding techniques (e.g. in a simple case: If buffer<x% full then transmit at a first higher rate, else if buffer>x% full then transmit at a second lower rate. Algorithms with more than one threshold can equally be envisaged). **Step changes in transmission rate can be achieved by controlling the encoding of the source data to give a higher (better quality) or lower (poorer quality) encoding rate**. Urzaiz, [0103].

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Ground 3)	
[9e] providing the encoded audio and video data for transmittal to a terminal.	Urzaiz discloses, or at least renders obvious, *providing the encoded audio and video data for transmittal to a terminal* (e.g., Urzaiz's disclosure that the encoded audio and video data is transmitted over the network to a client), as claimed. Urzaiz discloses a "data transmission method and system . . . in which one or more data streams **are transmitted at respective transmission rates**." Urzaiz, Abstract. Transmission in Urzaiz is commonly referred to as occurring "over a network," e.g., "**<u>[f]ollowing encoding of the audio and video data</u>**, the encoded data is passed to a network server, where it is stored in separate audio and video buffers prior to **<u>transmission over the network to a client</u>**." Urzaiz, [0005]. This is further described by the Figure 11 example described throughout this chart, "at step S6 the network connection 47 in the **server transmits the audio and video streams as separate UDP data streams,** at the calculated audio and video sending rates." Urzaiz, [0133].  Fig.11 Urzaiz, FIG. 11 (annotated).

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Ground 3)	
	Accordingly, in Urzaiz, after encoding the audio and video data (i.e., as described in limitation [9d]), the data is passed to a network server and then the client—in precisely the same way as claimed by the ’285 Patent. For the reasons set forth above, Urzaiz discloses *providing the encoded audio and video data for transmittal to a terminal*, as claimed. * * * Requester provides the following disclosures from Urzaiz relevant to this claim limitation: A **data transmission method and system is disclosed in which one or more data streams are transmitted at respective transmission rates** which are controlled to prevent data buffers in the receiver from overflowing. In some embodiments feedback data concerning the state of each buffer in a receiving client is received at the transmitting server, and used to adapt the sending rates to achieve the effect. Information indicative of the data decode rates or the fill extent of each buffer is communicated to the server as the feedback data. In other embodiments the server makes an open-loop estimate of the remaining space in the buffer, and controls the transmission rate accordingly. A data receiving method and system adapted to receive the data streams is also disclosed. Urzaiz, Abstract. Commonly, the **data to be streamed is multi-media data such as, for example, audio and video data.** The audio and video data may be from a live audio visual broadcast such as a news or sports event, or may be sourced from, for example, a video-on-demand service which permits subscribers to watch television programmes and films of their choice as and when they choose. Whatever the source of the data, however, the respective audio and video feed data must first be suitably digitally encoded in order to compress the audio and video data signals to a size suitable for <u>**transmission over a network**</u>. Commonly, audio and video encoding is performed in accordance with one of the various MPEG standards.

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Ground 3)	
	> **Following encoding of the audio and video data, the encoded data is passed to a network server, where it is stored in separate audio and video buffers prior to transmission over the network to a client**. Urzaiz, [0004]-[0005]. > In accordance with the above, the present invention also has application in the transmission of multiple data streams from a transmitter to one or more receivers which may be the same or different. > Preferably, the transmitted data stream contains audio or video data. Where two data streams are transmitted simultaneously, preferably one of the streams contains audio data and the other of the streams contains video data. Preferably the audio aid [*sic*] video data is related in teat [*sic*] it is intended for reproduction at the receiver simultaneously, for example where the video data is a TV programme or film and the audio data is the soundtrack thereto. The invention is particularly intended for the transmission of audio and video data in streams, where the sending rate of each stream can be controlled by the invention so as to be substantially smooth, and so as to prevent the receiver buffers from overflowing. Preferably, the sending rate is controlled so as to match the read-out rate from the receiver buffers. Urzaiz, [0032]-[0033]. > In order to provide for data communications between the server computer and the or [*sic*] each client computer a first user datagram protocol (UDP) connection **10** is provided between the server **40** and the or each client **50** along which **encoded video data is transmitted from the server 40.** Similarly, a second UDP connection **20** is also provided from the server **40** to the or each client **50** along which **encoded audio data is transmitted.** The transmission rates of the respective UDP connections **10** and **20** are controlled by the server in a manner to be described later for each embodiment of the invention. Urzaiz, [0073].

Claims	Relevant Disclosures in the Prior Art
Claim 9 (Ground 3)	
	Returning to FIG. 11, after the calculation of the audio and video sending rates for each stream, **at step S6 the network connection 47 in the server transmits the audio and video streams as separate UDP data streams, at the calculated audio and video sending rates**. It should be noted that as the audio and video steams [*sic*] are continuously transmitted, the steps of FIG. 11, although depicted sequentially, are actually performed in parallel, such the transmission rates of the audio and video streams are in reality updated once new values for the audio and video transmission rates have been calculated. While the new calculations are being performed, however, these streams continue to be transmitted at the previously calculated rate. Urzaiz, [0133].

Claims	Relevant Disclosures in the Prior Art
Claim 10 (Ground 3)	
[10P] The method of claim 9, further comprising	To the extent the preamble is limiting, Urzaiz in view of Gupta and Pogrebinsky discloses, or at least renders obvious, *the method of claim 9.* *See* claim 9 above.
[10a] dropping frames of the encoded video data.	Urzaiz in view of Gupta, discloses, or at least renders obvious, *dropping frames of the encoded video data* (e.g., Urzaiz's disclosure of adjusting video bit rate encoding; supplemented by Gupta's disclosure that an "easy way to reduce bandwidth is to simply drop lower-level dependent frames from the video stream"), as claimed. The '285 Patent itself describes that "frame dropping can be executed, when needed, by frame dropper 226." EX1001, 8:36-27. Further, "[w]hen 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," and that "[f]rame dropper 226 can drop the frames accordingly to deliver the optimal session bitrate." EX1001, 8:43-57. Urzaiz discloses an example where "the video data in particular the calculated rate will not satisfy the transmission rate requirements for the particular encoding rate used," and that in this situation, "to prevent the video buffer at the receiver from emptying," a lower

Claims	Relevant Disclosures in the Prior Art
Claim 10 (Ground 3)	
	bitrate video stream may be provided to the client. Urzaiz, [0142]. Thus, although not explicitly described, a POSITA would have been aware that one common method of lowering the bitrate of a video stream is to decrease the framerate (i.e., *dropping frames of the encoded video data*, as claimed). To the extent not disclosed in Urzaiz itself, Gupta explicitly describes this concept in a system similar to Urzaiz and explains why this limitation would have nonetheless been obvious to a POSITA: Furthermore, a stream such as a video stream can sometimes be timeline-modified dynamically at the server without incurring significant overhead. Accordingly, **the server can adjust the timeline and quality of the video stream dynamically to match the available bandwidth**, eliminating the need to store multiple video streams at the server. As an example of a situation where this might be easily accomplished, an MPEG (Motion Picture Expert Group) video stream contains independent frames and several levels of dependent frames. **One easy way to reduce bandwidth is to simply drop lower-level dependent frames from the video stream**. Gupta, 13:13-23. Accordingly, in Urzaiz's system—which already contemplates situations where video bitrate needs to be lowered (i.e., to reduce the bandwidth across the connection)—it would have been obvious to a POSITA in view of Gupta's disclosure that an "easy way to reduce bandwidth is to simply drop lower-level dependent frames from the video stream," i.e., thereby achieving Urzaiz's goal of reducing video bitrate. *Id.*; Karam Decl., ¶ 75. Thus, when Urzaiz's system describes lowering the video bitrate, in view of Gupta, a POSITA would have been well aware that dropping frames of encoded video data was a common method for decreasing the bit rate of a stream and would have done the same in Urzaiz. Karam Decl., ¶¶ 76-77. Gupta's teachings would have been combined with Urzaiz for the reasons discussed in the Request at Section I.E.3.d. Karam Decl., ¶¶ 64-68. For the reasons set forth above, Urzaiz in view of Gupta discloses *dropping frames of the encoded video data*, as claimed.

Claims	Relevant Disclosures in the Prior Art
Claim 10 (Ground 3)	
	* * * Requester provides the following disclosures from Urzaiz relevant to this claim limitation: Within the server, the actual transmission rates of each stream are controlled by the network controller 48 and the network connection 47 in combination by actually releasing packets on to the network in accordance with the calculated rates. However, in the special case of the transmission of audio and video data described in the second embodiment, as in the first embodiment it may be that for the video data in particular the calculated rate will not satisfy the transmission rate requirements for the particular encoding rate used. In this case, if it appears that the calculated transmission rate for the video stream has to drop such that at the present video encoding rate it will not be possible to transmit sufficient data in the video stream to prevent the video buffer at the receiver from emptying, then the network controller 48 controls the network connection 47 to take encoded video data from the low rate encoding video buffer 43 which has been encoded with a lower quality, which is more suitable for transmission across the network at the lower calculated transmission rate. At the receiver, the low rate encoded video data is placed in the video buffer and the video decoder 55 detects the lower rate of encoding and changes its own decoding rate to a lower rate, this reducing the rate at which video data is being read from the video buffer. Such measures prevent the video buffer from emptying completely, thereby permitting continuous video reproduction at the client computer. Urzaiz, [0142]. * * * Requester provides the following disclosures from Gupta relevant to this claim limitation: Furthermore, a stream such as a video stream can sometimes be timeline-modified dynamically at the server without incurring significant overhead. Accordingly, the server can adjust the timeline and quality of the video stream dynamically to match the available bandwidth, eliminating the need to store

Claims	Relevant Disclosures in the Prior Art
Claim 10 (Ground 3)	
	multiple video streams at the server. As an example of a situation where this might be easily accomplished, an MPEG (Motion Picture Expert Group) video stream contains independent frames and several levels of dependent frames. One easy way to reduce bandwidth is to simply drop lower-level dependent frames from the video stream. Gupta, 13:13-23.

Claims	Relevant Disclosures in the Prior Art
Claim 15 (Ground 3)	
[15P] 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:	To the extent the preamble is limiting, Urzaiz discloses, or at least renders obvious, *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* (e.g., Urzaiz's adaptive bandwidth transmission system for transmitting multiple data streams, such as video and audio streams, in a network), for the reasons discussed above in claim 9. *See* claim 9 above. In addition, Urzaiz makes clear that its disclosures are performed by a computer executing instructions, e.g.: The present invention relates to a method and system providing for data communications, and in particular to a method and system for transmitting one or more data streams across a network, as well as a method and system for receiving such transmitted data. Furthermore, **the present invention also relates to a computer readable storage medium storing a computer program which when run on a computer controls the computer to perform the aforementioned methods of data transmission and receipt**. Urzaiz, [0001]. Preferably, the computer readable storage medium is any of an optical disk, a magnetic disk, a magneto-optical disk, a solid state computer memory, or any other suitable data storage medium. Urzaiz, [0037].

Claims	Relevant Disclosures in the Prior Art
Claim 15 (Ground 3)	
	By **providing feedback from the receiver to the server it is possible to provide the server with up to date information** indicative of, for example, congestion conditions on the network resulting in packet losses. **The server then becomes capable of calculating the maximum transmission rate available for the stream dependent upon the present conditions on the network, <u>thereby optimising the transmission rate at which the stream is transmitted</u>.** Urzaiz, [0035].
[15a] receiving the optimal session bitrate;	Urzaiz discloses, or at least renders obvious, *receiving the optimal session bitrate* for the same reasons as discussed above with respect to element [9a] of claim 9 (reciting "receiving an optimal session bitrate"). *See* limitation [9a] above.
[15b] allocating the optimal session bitrate between audio and video media to produce an optimal audio bitrate and an optimal video bitrate,	Urzaiz discloses, or at least renders obvious, *allocating the optimal session bitrate between audio and video media to produce an optimal audio bitrate and an optimal video bitrate* for the same reasons as discussed above with respect to element [9b] of claim 9 (reciting an identical limitation to [15b]). *See* limitation [9b] above.
[15c] 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;	Urzaiz in view of Gupta and Pogrebinsky discloses, or at least renders obvious, *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* for the same reasons as discussed above with respect to element [9c] of claim 9 (reciting an identical limitation to [15c]). *See* limitation [9c] above.
[15d] encoding audio and video media data according to the optimal audio bitrate and the optimal video bitrate; and	Urzaiz discloses, or at least renders obvious, *encoding audio and video media data according to the optimal audio bitrate and the optimal video bitrate* for the same reasons as discussed above with respect to element [9d] of claim 9 (reciting an identical limitation to [15d]). *See* limitation [9d] above.

Claims	**Relevant Disclosures in the Prior Art**
Claim 15 (Ground 3)	
[15e] providing the encoded audio and video data for transmittal to a terminal.	Urzaiz discloses, or at least renders obvious, *providing the encoded audio and video data for transmittal to a terminal* for the same reasons as discussed above with respect to element [9e] of claim 9 (reciting an identical limitation to [15e]). *See* limitation [9e] above.

EXHIBIT BB

Claim Chart

Comparing Claims 1, 6, 11, and 14 of the '285 Patent

to Yano and Ogawa

I. GROUNDS OF UNPATENTABILITY

Ground	Claims	Statutes	Prior Art
4	1, 6, 11, and 14	35 U.S.C. § 103	Yano, Ogawa

A. Prior Art Relied Upon

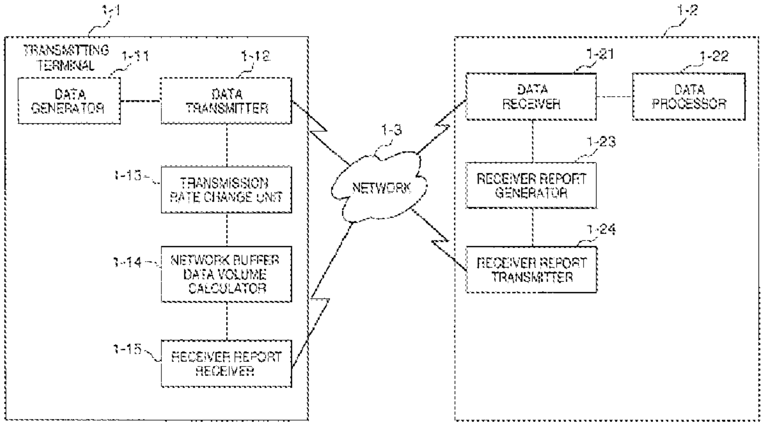
The ’285 Patent was filed on July 9, 2008 and claims priority to provisional application 60/948,917, filed July 10, 2007. Accordingly, the earliest possible priority date for the ’285 Patent is **July 10, 2007**.

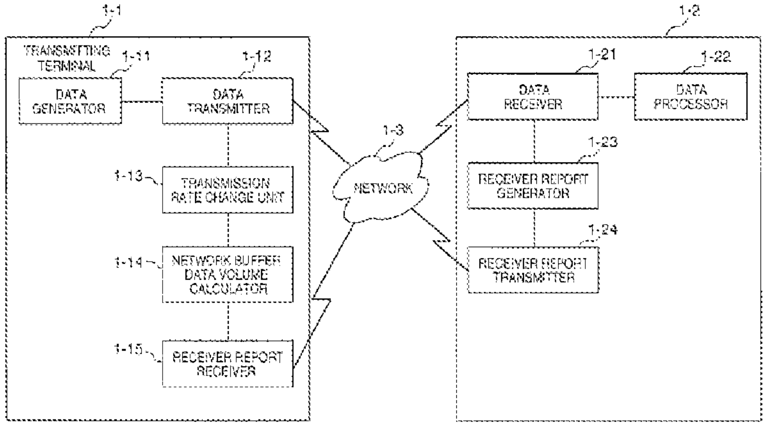
Prior Art
EX1011 (“Yano”) U.S. Patent Publication 2003/0037158 to Koichi Yano et al. was filed on August 20, 1998 and published on February 20, 2003. Accordingly, Yano qualifies as prior art to the ’285 Patent at least under 35 U.S.C. §§ 102(a) and (b) (pre-AIA).
EX1012 (“Ogawa”) U.S. Patent Publication 2006/0218264 to Akimichi Ogawa et al. was filed on March 22, 2006 and published on September 28, 2006. Accordingly, Ogawa qualifies as prior art to the ’285 Patent at least under 35 U.S.C. §§ 102(a) and (e) (pre-AIA).

B. Claim Charts

The claim charts below first include an argument portion explaining why Yano and Ogawa disclose and render obvious the pertinent limitation, followed by additional citations to Yano and Ogawa. Unless otherwise noted, all emphasis has been added by Requester.

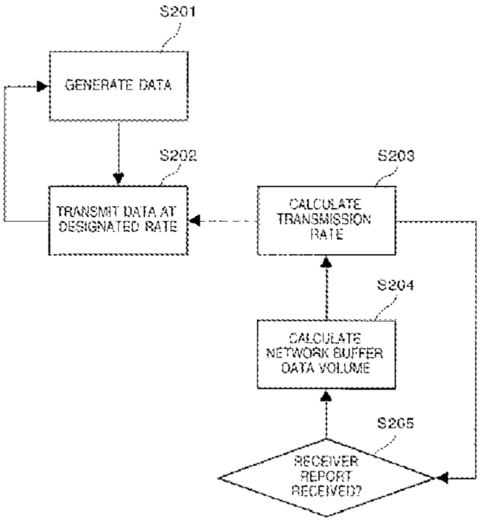
Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
[1P] A method comprising:	To the extent the preamble is limiting, Yano discloses, or at least renders obvious, the method described in claim 1 (e.g., Yano's transmission system). * * * Yano discloses: **This invention can make data communications at an optimal transfer rate on the basis of the unarrived data volume on a network between two end terminals**. For this purpose, a transmitting terminal (**1-1**) adds sequence number information to data generated by a data generator (**1-11**), and transmits the data to a receiving terminal (**1-2**) via a data transmitter (**1-12**). Since the receiving terminal (**1-2**) transmits data including the sequence number in the received data, the transmitting terminal determines that data (buffer capacity) corresponding to the difference between the current sequence number and the received sequence number remain on the network, and calculates that volume using a network buffer data volume calculator (**1-14**). The transmitting terminal determines the transmission rate on the basis of the calculation result, and controls the data transmitter (**1-12**) to transfer data at that transmission rate. Yano, Abstract. **It is another object of the present invention to provide a <u>data communication</u> apparatus, method, system, and storage medium, <u>which perform data communications at an optimal transfer rate on the basis of the volume of unarrived data on the network present between two end terminals</u>**. It is another object of the present invention to provide a data communication apparatus, receiving apparatus, control method, storage medium, and data communication system, which can realize optimal data transfer by dynamically controlling to change the transfer rate in correspondence with the conditions on the network in data communications via the network. Yano, [0007]-[0008].

Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	> FIG. 1 shows the connection relationship and structure of the respective devices when data transmitted by a transmitting terminal **1-1** is received by a receiving terminal **1-2** via a network **1-3**. Note that the network **1-3** includes anything from LANs used in corporations to a collection of many and unspecified networks coupled to each other (e.g., the internet), and is not particularly limited. The arrangements and operations of the terminals shown in FIG. 1 will be explained below. Yano, [0031]. FIG. 1  Yano, FIG. 1.
[1a] receiving a receiver report from a terminal;	Yano discloses, or at least renders obvious, *receiving a receiver report from a terminal* (e.g., Yano's receiving terminal 1-2 sending a receiver report to the receiver report receiver 1-15 of terminal 1-1), as claimed. Figure 1 of Yano depicts and describes a transmitting terminal (1-1) and a receiving terminal (1-2). See below:

Claims	**Relevant Disclosures in the Prior Art**
Claim 1 (Ground 4)	
	FIG. 1  Yano, FIG. 1. Yano is further explicit that "**the receiving terminal 1-2 sends back a receiver report**. A receiver report receiver 1-15 receives the receiver report, and sends the report contents to a network buffer data volume calculator 1-14." Yano, [0034]. Yano further explains that, as for the transmitting terminal, "[u]pon **reception of the receiver report**, the transmitting terminal calculates the volume of data which has been output from the transmitting terminal onto the network but has not reached the receiving terminal (step S204)." Yano, [0041]. Further, "**the receiving terminal periodically transmits receiver reports to the transmitting terminal** while transmitting/receiving data between the transmitting and receiving terminals." Yano, [0051]. FIG. 2, related to the transmitting terminal, also includes a step of noting whether or not the "receiver report received?" – e.g.:

Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	

F I G. 2



Yano, FIG. 2.

For all of these reasons, Yano discloses, or at least renders obvious, receiving a receiver report from a terminal.

* * *

Yano discloses:

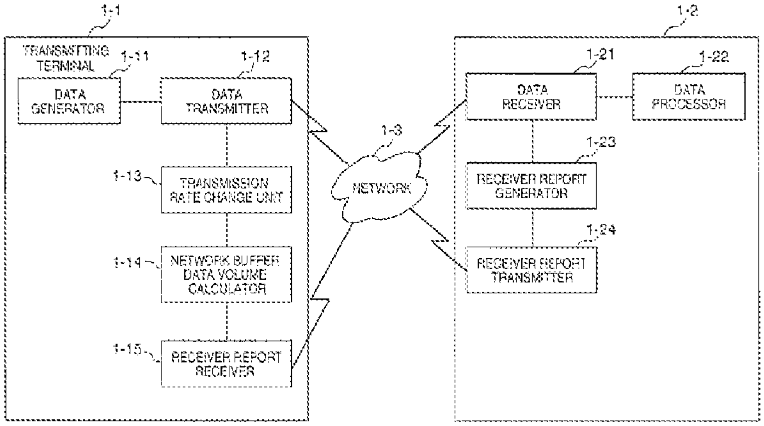
> **After such transmission, the receiving terminal 1-2 sends back a receiver report. A receiver report receiver 1-15 receives the receiver report, and sends the report contents to a network buffer data volume calculator 1-14.** The transmission rate change unit **1-13** determines the transmission rate on the basis of the data volume calculated by the network buffer data volume calculator **1-14**, and designates the transmission rate to the data transmitter **1-12**.

Yano, [0034].

> On the other hand, the transmitting terminal waits for a receiver report sent from the receiving terminal simultaneously with data transmission (step S **205**). FIG. 5 shows an example of the format of the receiver report. **The receiver report includes the reception sequence number and reception rate. Note that the reception sequence number is the one included in the last**

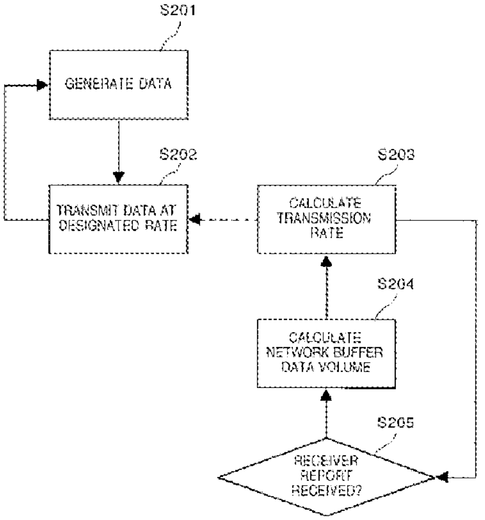
Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	> **packet received by the receiving terminal upon generating a receiver report, and the reception time is the time at which the receiving terminal received that packet. Upon reception of the receiver report, the transmitting terminal calculates the volume of data which has been output from the transmitting terminal onto the network but has not reached the receiving terminal (step S204)**. This data volume will be referred to as a network buffer data volume hereinafter. The method of calculating the network buffer data volume is as follows. Yano, [0041]. > The data receiver **1-21** measures information pertaining to the sequence number of the received data, data reception time, the received data volume, and the like, and sends that information to a receiver report generator **1-23**. The receiver report generator **1-23** calculates the reception rate that must be included in the receiver report, and supplies it to a receiver report transmitter **1-24** together with the sequence number. > **The receiver report transmitter 1-24 transmits the receiver report to the receiver report receiver 1-14 of the transmitting terminal 1-1 via the network 1-3**. Yano, [0036]-[0037]. > By repeating the above-mentioned steps, **the receiving terminal periodically transmits receiver reports to the transmitting terminal while transmitting/receiving data between the transmitting and receiving terminals**. On the other hand, the transmitting terminal determines the transmission rate based on the receiver report to make the network buffer a constant volume of data. > FIG. 6 shows an example of variations of the transmission and reception rates and network buffer data volume actually measured in data communications of this embodiment. The transmission and reception rates stabilize at an identical value around the available band of the network, and the network data buffer volume changes in the neighborhood of the target value. Yano, [0051]-[0052].

<table>
<tr><th>Claims</th><th>Relevant Disclosures in the Prior Art</th></tr>
<tr><td colspan="2">Claim 1 (Ground 4)</td></tr>
<tr><td></td><td>
FIG. 1



Yano, FIG. 1.

FIG. 2



Yano, FIG. 2.
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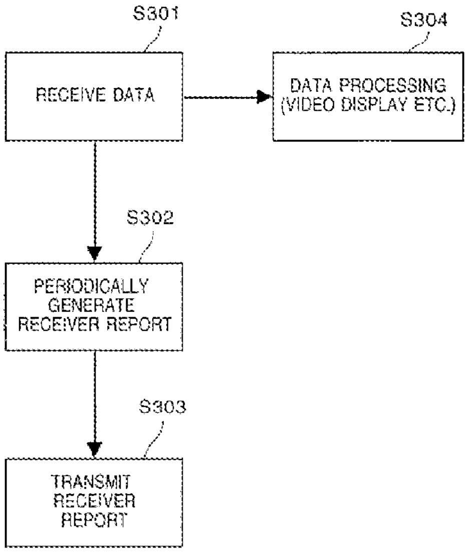
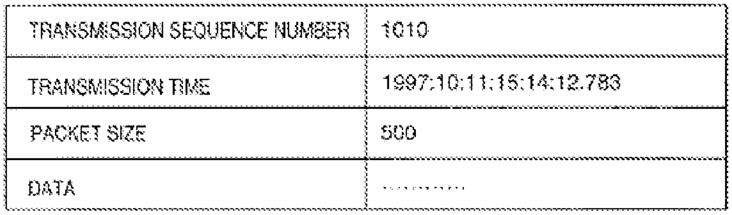
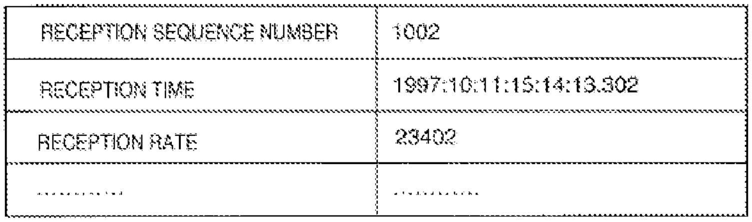
Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	FIG. 3  Yano, FIG. 3. FIG. 4  TRANSMISSION SEQUENCE NUMBER: 1010 TRANSMISSION TIME: 1997:10:11:15:14:12.783 PACKET SIZE: 500 DATA: .... Yano, FIG. 4.

<table>
<tr><th>Claims</th><th>Relevant Disclosures in the Prior Art</th></tr>
<tr><td colspan="2">Claim 1 (Ground 4)</td></tr>
<tr><td></td><td>FIG. 5

<table>
<tr><td>RECEPTION SEQUENCE NUMBER</td><td>1002</td></tr>
<tr><td>RECEPTION TIME</td><td>1997:10:11:15:14:13.302</td></tr>
<tr><td>RECEPTION RATE</td><td>23402</td></tr>
<tr><td>...........</td><td>...........</td></tr>
</table>
Yano, FIG. 5.</td></tr>
<tr><td>[1b] estimating one or more network conditions of a media network using the receiver report;</td><td>Yano in view of Ogawa discloses, or at least renders obvious, <i>estimating one or more network conditions of a media network using the receiver report</i> (e.g., Yano’s transmitting terminal determining transmission rate “on the basis of a receiver report”), as claimed.

Yano explicitly discloses that the considered network is a media network, e.g., “[t]he present invention relates to an apparatus and system for transmitting and/or receiving steadily generated data such as video data, audio data, and the like via a network.” Yano, [0001].

Additionally, Yano explicitly discloses estimating one or more network conditions using the receiver report, e.g., “the <u>transmitting terminal determines the transmission rate on the basis of a receiver report</u> sent from the receiving terminal as in the first embodiment.” Yano, [0081].

This is also depicted in Yano’s Figure 2:</td></tr>
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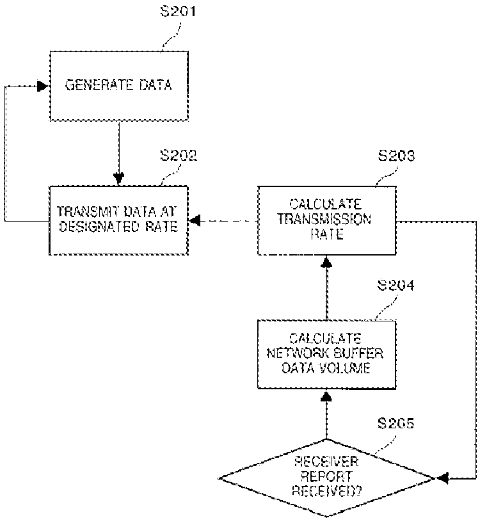
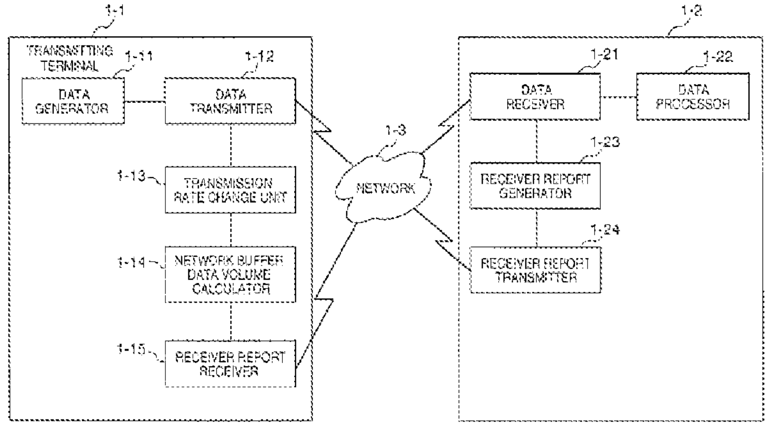
Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	FIG. 2  Yano, FIG. 2. For example, as shown there, the system at Step S205 looks to see if the receiver report has been received, and if it has, the system proceeds to calculate the network buffer data volume and the transmission rate in Steps S204 and S203, respectively. This process is also described in other portions of Yano, e.g., "[a] receiver report receiver 1-15 receives the receiver report, and sends the report contents to a network buffer data volume calculator 1-14. The transmission rate change unit 1-13 determines the transmission rate on the basis of the data volume calculated by the network buffer data volume calculator 1-14, and designates the transmission rate to the data transmitter 1-12." Yano, [0034]. For these reasons, Yano discloses, or at least renders obvious, *estimating one or more network conditions of a media network using the receiver report.* To the extent further disclosure is necessary (e.g., as it relates to explicit disclosure that it was obvious to perform *estimating one or more network conditions of a media network*), Ogawa discloses that: More specifically, it is desirable to provide a communication processing apparatus, a data communication system, and a communication processing method with **which a server predicts an optimal value of bitrate of data transmitted in consideration of factors such as congestion on a**

Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	communication path or disturbance on a communication link and with which the bitrate is dynamically controlled on the basis of the predicted value so that data streaming is carried out in an optimal data transmission mode.” Ogawa, [0016]. The teachings of Yano and Ogawa would have been considered together and combined for the reasons discussed in the Request at Section I.E.4.c. Karam Decl. (EX1003), ¶¶ 90-93. * * * Yano discloses: The present invention relates to an apparatus and system for **transmitting and/or receiving steadily generated data such as video data, audio data, and the like via a network**. Yano, [0001] It is another object of the present invention to provide a data communication apparatus, method, system, and storage medium, which perform data communications at an optimal transfer rate on the basis of the volume of unarrived data on the network present between two end terminals. It is another object of the present invention to provide a data communication apparatus, receiving apparatus, control method, storage medium, and data communication system, **which can realize optimal data transfer by dynamically controlling to change the transfer rate in correspondence with the conditions on the network in data communications via the network**. Yano, [0007]-[0008]. After such transmission, **the receiving terminal 1-2 sends back a receiver report. A receiver report receiver 1-15 receives the receiver report, and sends the report contents to a network buffer data volume calculator 1-14**. The **transmission rate change unit 1-13** **determines the transmission rate on the basis of the data volume calculated by the network buffer**

Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	data volume calculator **1-14**, and designates the transmission rate to the data transmitter **1-12**. Yano, [0034]. **As described above, according to the fourth embodiment, the transfer rate is dynamically changed in correspondence with the network condition in data communications via the network, thus realizing optimal data transfer**. Hence, the present **invention is particularly effective for real-time processing, e.g., transferring live images captured by a camera**. Yano, [0126]. As a consequence, data transfer between two terminals via the network can be optimally done in correspondence with the buffer capacity of that network. **Even when the network traffic is very smooth and a high transfer rate may be set, the data transfer is controlled to make constant the data volume which stays as buffer data on the network without increasing the transfer rate. Hence, the load on the network can be reduced, and use of the network by other parties is not disturbed**. Yano, [0064]. The fourth embodiment will be described below. The arrangement of the transmitting and receiving terminals of the fourth embodiment is the same as that in the first embodiment. **Also, the transmitting terminal determines the transmission rate on the basis of a receiver report sent from the receiving terminal as in the first embodiment**. Hence, the overall operations are the same as those shown in FIGS. 2 and 3, and the format of transmission data is the same as that shown in FIG. 4. However, the format of the receiver report sent from the receiving terminal 1-2 in the fourth embodiment is as shown in FIG. 10. Yano, [0081].

<table>
<tr><th>Claims</th><th>Relevant Disclosures in the Prior Art</th></tr>
<tr><td colspan="2">Claim 1 (Ground 4)</td></tr>
<tr><td></td><td>

FIG. 1



Yano, FIG. 1.

</td></tr>
<tr><td>[1c] determining an optimal session bitrate using the estimated one or more network conditions, wherein determining the optimal session bitrate further comprises:</td><td>

Yano in view of Ogawa, discloses, or at least renders obvious, *determining an optimal session bitrate using the estimated one or more network conditions, wherein determining the optimal session bitrate further comprises* (e.g., Yano's determining an "optimal transfer rate" on the basis of network conditions; supplemented by Ogawa's disclosure of a system for controlling transmission bitrate to transmit "at an optimal bitrate"), as claimed.

Yano discloses that its purpose is to make data communications at "an optimal transfer rate." Yano, Abstract. Based on network conditions determined using a receiver report, Yano's "transmitting terminal determines the transmission rate on the basis of the calculation result, and controls the data transmitter (1-12) to transfer data at that transmission rate." *Id.*

Yano later describes that it performs "data communications at **an optimal transfer rate on the basis of the volume of unarrived data on the network present between two end terminals**." Yano, [0007]. Yano further describes that "optimal data transfer" can be realized "**by dynamically controlling to change the transfer rate in correspondence with the conditions on the network in data communications via the network**." Yano, [0008].

Yano further describes that its "**transmitting terminal determines the transmission rate on the basis of a receiver report**," where the

</td></tr>
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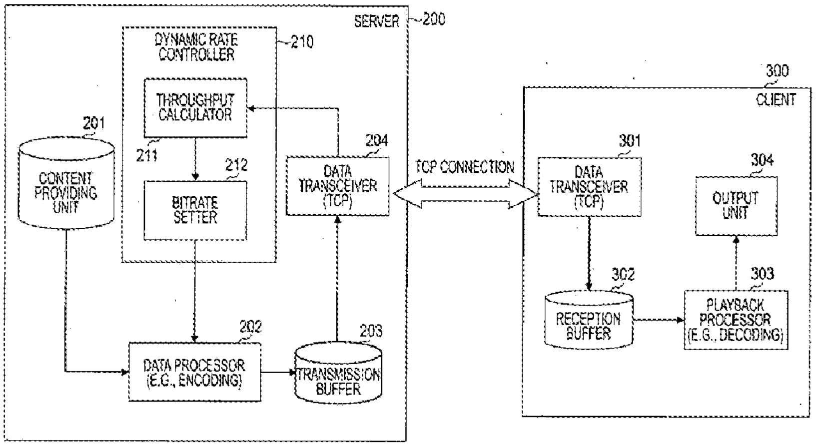
Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	receiver report (as described above in limitation [1a]) relates to network conditions. For these reasons, Yano discloses *determining an optimal session bitrate using the estimated one or more network conditions, wherein determining the optimal session bitrate further comprises*, as claimed. But to the extent it is found not to, Yano's disclosure would be supplemented by Ogawa, which also discloses this limitation. Ogawa discloses "a data communication system, and a **communication processing method that allow data to be transmitted in an optimal transmission mode** in transmission and reception of streaming data." Ogawa, [0015]. Ogawa explains: More specifically, it is desirable to provide a communication processing apparatus, a data communication system, and a communication processing method with **which a server predicts an optimal value of bitrate of data transmitted in consideration of factors such as congestion on a communication path or disturbance on a communication link and with which the bitrate is dynamically controlled on the basis of the predicted value so that data streaming is carried out in an optimal data transmission mode**." Ogawa, [0016]. FIG. 2 **dynamically controls the transmission bitrate** so that the **streaming data will be transmitted at an optimal bitrate**. The process will be described later more specifically. When the transmission of all the streaming data is finished, in step S14, the connection between the server and the client is closed, and the process is then exited. Ogawa, [0102].

<table>
<tr><th>Claims</th><th>Relevant Disclosures in the Prior Art</th></tr>
<tr><td>Claim 1 (Ground 4)</td><td></td></tr>
<tr><td></td><td>

FIG. 2



Ogawa, FIG. 2.

The teachings of Yano and Ogawa would have been considered together and combined for the reasons discussed in the Request at Section I.E.4.c. Karam Decl., ¶¶ 90-93.

* * *

Yano discloses:

> **This invention can make data communications at an optimal transfer rate on the basis of the unarrived data volume on a network between two end terminals**.
>
> For this purpose, a transmitting terminal (**1-1**) adds sequence number information to data generated by a data generator (**1-11**), and transmits the data to a receiving terminal (**1-2**) via a data transmitter (**1-12**). Since the receiving terminal (**1-2**) transmits data including the sequence number in the received data, the transmitting terminal determines that data (buffer capacity) corresponding to the difference between the current sequence number and the received sequence number remain on the network, and calculates that volume using a network buffer data volume calculator (**1-14**). **The transmitting terminal determines the transmission rate on the basis of the calculation result, and controls the data transmitter (1-12) to transfer data at that transmission rate.**

Yano, Abstract.

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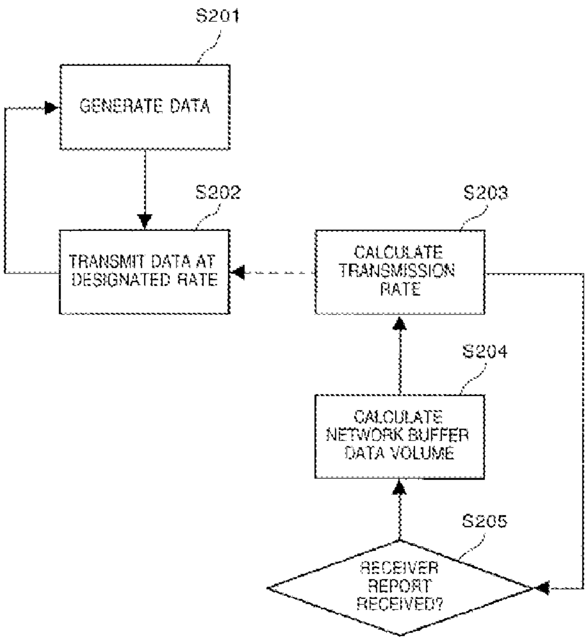
Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	It is another object of the present invention to provide a data communication apparatus, method, system, and storage medium, **which perform data communications at an optimal transfer rate on the basis of the volume of unarrived data on the network present between two end terminals**. It is another object of the present invention to **provide a data communication apparatus, receiving apparatus, control method, storage medium, and data communication system, which can realize optimal data transfer by dynamically controlling to change the transfer rate in correspondence with the conditions on the network in data communications via the network.** Yano, [0007]-[0008]. Then, **a transmission rate Rnew is determined so that the calculated network buffer data volume BUFcur approaches a target value BUFdes of the network buffer data volume** (step S **203**). The calculation formula is as follows: $Rnew=Rcur+C\times(BUFdes-BUFcur)$ where **Rcur is the current transmission rate, and Rnew is the new transmission rate to be determined**. C is an appropriate constant. The transmission rate R determined by this processing is supplied to the data transmitter to designate the transmission rate in the data transmission step (S **202**). With this operation, in the transmitting terminal **1-1**, the network buffer data volume calculator **1-14** calculates the buffer capacity of the network present between the two terminals in accordance with the receiver report from the receiving terminal **1-2**, and the transmission rate change unit **1-13** sets the data transfer rate in the data transmitter **1-12** in accordance with the calculation result. Yano, [0046]-[0048]. By repeating the above-mentioned steps, **the receiving terminal periodically transmits receiver reports to the transmitting terminal while transmitting/receiving data between the transmitting and receiving terminals**. On the other hand, the **transmitting terminal determines the transmission rate based**

Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	on the receiver report to make the network buffer a constant volume of data. FIG. 6 shows an example of variations of the transmission and reception rates and network buffer data volume actually measured in data communications of this embodiment. The transmission and reception rates stabilize at an identical value around the available band of the network, and the network data buffer volume changes in the neighborhood of the target value. Yano, [0051]-[0052]. The fourth embodiment will be described below. The arrangement of the transmitting and receiving terminals of the fourth embodiment is the same as that in the first embodiment. **Also, the transmitting terminal determines the transmission rate on the basis of a receiver report sent from the receiving terminal as in the first embodiment**. Hence, the overall operations are the same as those shown in FIGS. 2 and 3, and the format of transmission data is the same as that shown in FIG. 4. However, the format of the receiver report sent from the receiving terminal 1-2 in the fourth embodiment is as shown in FIG. 10. Yano, [0081]. **In this embodiment, a transmission rate Rnew is determined so that the calculated network buffer data volume BUFcur approaches the target value BUFdes of the network buffer data** volume (step S **203**). The transmission rate Rnew is calculated by: Rnew=Rcur+C*(BUFdes−BUFcur) where **Rcur is the current transmission rate, and Rnew is the new transmission rate to be determined**. C is an appropriate constant. The transmission rate Rnew determined by this processing is supplied to the data transmitter to designate the transmission rate in the data transmission step (step S 202 in FIG. 2). In this embodiment, the transmission rate is determined with reference to the transmission rate Rcur but may be determined with reference to the reception rate Rrecv. If the reception interval (Interval) of the receiver reports is known, the transmission rate

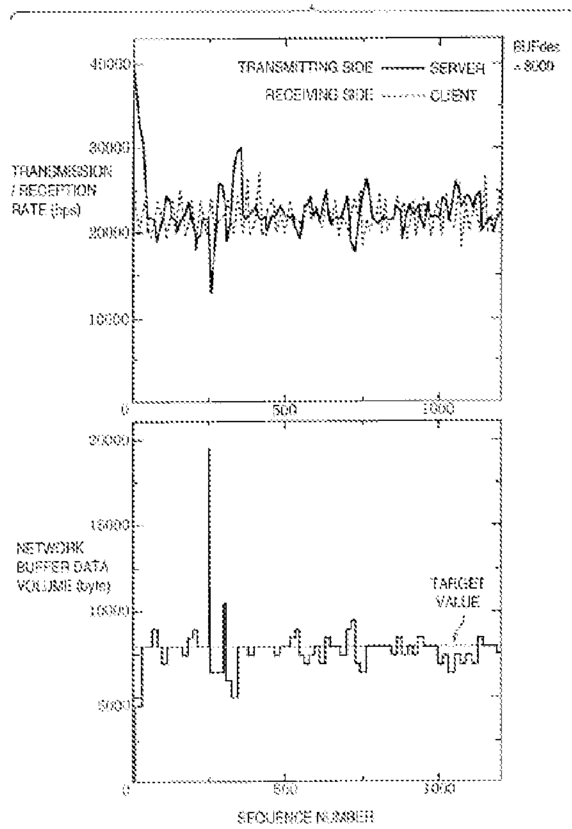
Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	Rnew may be calculated so that the network buffer data volume matches the target value upon arrival of the next receiver report: Rnew=Rrecv+{(BUFdes−BUFcur)/Interval} Yano, [0096]-[0098]. To restate, according to the fourth embodiment, **the transmission rate is controlled to maintain constant the data volume (network buffer data volume) that has been output from the transmitting terminal onto the network but has not reached the receiving terminal yet, in accordance with an increase in data round-trip time. With this control, since the data volume which is en route to the destination on the network can be accurately calculated, and is adjusted, a transmission delay can be suppressed to fall within an allowable range**. Since data is transmitted/received to always save data in a buffer, the available band of the network can be sufficiently used to transmit/receive the data. **Since the data volume buffered in the network is used as a parameter for controlling the transmission rate without directly using the delay time, a sufficient buffer volume which is to exist on the network can be defined as a target buffer data volume, and data can be prevented from being excessively output onto the network**. Yano, [0123]. As described above, according to the fourth embodiment, **the transfer rate is dynamically changed in correspondence with the network condition in data communications via the network, thus realizing optimal data transfer**. Hence, the present invention is particularly effective for real-time processing, e.g., transferring live images captured by a camera. Yano, [0126]. A transmitting apparatus which transmits data at a predetermined transmitting rate to a network and receives information related to data transmission condition on said network from said network, wherein said predetermined transmitting rate is controlled on the basis of said information related to data transmission condition. Yano, claim 44.

Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	FIG. 2  Yano, FIG. 2.
[1d] determining stability criterion using the estimated one or more network conditions, wherein determining stability criterion includes at least one of:	Yano in view of Ogawa, discloses, or at least renders obvious, *determining stability criterion using the estimated one or more network conditions, wherein determining stability criterion includes at least one of* (e.g., Yano's use of network transmission and reception rates to stabilize data buffers; supplemented by Ogawa's disclosures), as claimed. As will be described in greater detail in the two following limitations (i.e., limitations [1e] and [1f]), Yano's system (supplemented by Ogawa) *determin[es] stability criterion using the estimated one or more network conditions*. For example, as will be described in greater detail below in limitation [1e], Yano describes transfer time and reception times for data and receiver reports, together with Yano's disclosure of data round-trip times; supplemented by Ogawa's disclosure of calculating data transmission and reception periods and round trip times. *See* limitation [1e].

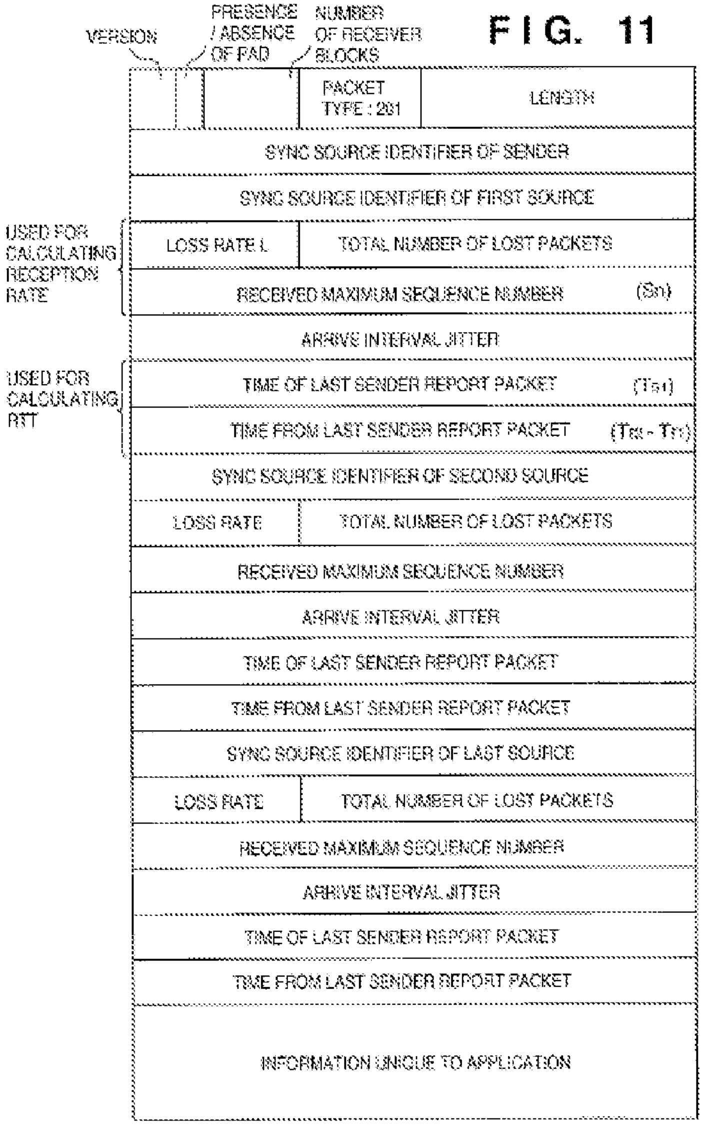
Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	And as will be described in greater detail below in limitation [1f], Yano described comparing of current transmission rates, new transmission rates, and reception rates; supplemented by Ogawa's disclosure of a bitrate setter comparing a current transmission-data bitrate with the maximum throughput. *See* limitation [1f]. * * * Yano discloses: The data receiver **1-21** measures information pertaining to the sequence number of the received data, data reception time, the received data volume, and the like, and sends that information to a receiver report generator **1-23**. **The receiver report generator 1-23 calculates the reception rate that must be included in the receiver report, and supplies it to a receiver report transmitter 1-24 together with the sequence number**. The receiver report transmitter **1-24** transmits the receiver report to the receiver report receiver **1-14** of the transmitting terminal **1-1** via the network **1-3**. Yano, [0036]-[0037]. A network buffer data volume BUFcur is obtained by multiplying a packet size Psize by the difference between a sequence number SEQsend of the last packet output from the transmitting terminal and a reception sequence number SEQrecv included in the receiver report sent back from the receiving terminal: $\text{BUFcur}=\text{Psize}\times(\text{SEQsend}-\text{SEQrecv})$ **Note that the time (timer) of the receiving terminal does not always perfectly match that of the transmitting terminal. For this reason, the transmitting terminal may calculate:** $(\text{Ts2}-\text{Ts1})-(\text{Tr2}-\text{Tr1})$ where **Ts1 is the <u>transfer time of data</u> of a given sequence number, Tr1 is the <u>reception time of that data</u> at the receiving side, Tr2 is the <u>transfer time of that receiver report</u>, and Ts2 is the <u>reception time of the receiver report</u> including the sequence number**. This difference may be divided by "2" to

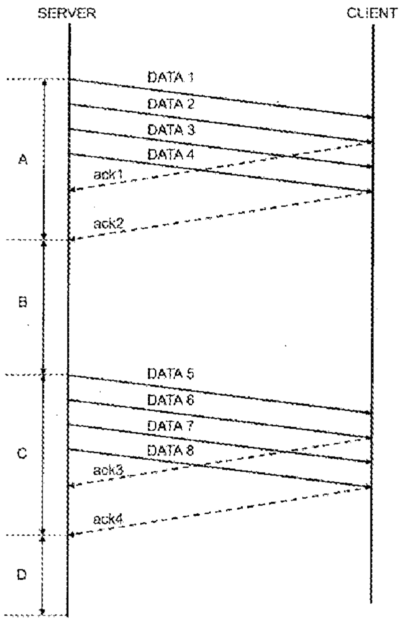
Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	obtain the time required for transfer. The reason why the difference is divided by "2" is that the difference corresponds to a round trip. In this case, the receiving side adds the reception time and the output time of the receiver report. If the processing time on the receiving time is negligibly short, since the round-trip time alone need be considered, the receiving side need not insert the reception time and output time in the receiver report. Then, a transmission rate Rnew is determined so that the calculated network buffer data volume BUFcur approaches a target value BUFdes of the network buffer data volume (step S **203**). The calculation formula is as follows: Rnew=Rcur+C×(BUFdes−BUFcur) where Rcur is the current transmission rate, and Rnew is the new transmission rate to be determined. C is an appropriate constant. The transmission rate R determined by this processing is supplied to the data transmitter to designate the transmission rate in the data transmission step (S **202**). With this operation, in the transmitting terminal **1-1**, the network buffer data volume calculator **1-14** calculates the buffer capacity of the network present between the two terminals in accordance with the receiver report from the receiving terminal **1-2**, and the transmission rate change unit **1-13** sets the data transfer rate in the data transmitter **1-12** in accordance with the calculation result. Yano, [0042]-[0048]. By repeating the above-mentioned steps, the receiving terminal periodically transmits receiver reports to the transmitting terminal while transmitting/receiving data between the transmitting and receiving terminals. On the other hand, the transmitting terminal determines the transmission rate based on the receiver report to make the network buffer a constant volume of data. FIG. 6 shows an example of variations of the transmission and reception rates and network buffer data volume actually measured in data communications of this embodiment. **<u>The transmission and reception rates stabilize at an identical value around the</u>**

Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	available band of the network, and the network data buffer volume changes in the neighborhood of the target value. Yano, [0051]-[0052]. As a consequence, **data transfer between two terminals via the network can be optimally done in correspondence with the buffer capacity of that network. Even when the network traffic is very smooth and a high transfer rate may be set, the data transfer is controlled to make constant the data volume which stays as buffer data on the network without increasing the transfer rate**. Hence, the load on the network can be reduced, and use of the network by other parties is not disturbed. Yano, [0064]. In this embodiment, **a transmission rate Rnew is determined so that the calculated network buffer data volume BUFcur approaches the target value BUFdes of the network buffer data volume (step S 203)**. The transmission rate Rnew is calculated by: Rnew=Rcur+C*(BUFdes−BUFcur) **where Rcur is the current transmission rate, and Rnew is the new transmission rate to be determined**. C is an appropriate constant. The transmission rate Rnew determined by this processing is supplied to the data transmitter to designate the transmission rate in the data transmission step (step S 202 in FIG. 2). In this embodiment, the transmission rate is determined with reference to the transmission rate Rcur but may be determined with reference to the reception rate Rrecv. If the reception interval (Interval) of the receiver reports is known, the transmission rate Rnew may be calculated so that the network buffer data volume matches the target value upon arrival of the next receiver report: Rnew=Rrecv+{(BUFdes−BUFcur)/Interval} Yano, [0096]-[0098].

Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	FIG. 6  Yano, FIG. 6.
[1e] comparing a media time in transit and a round trip time estimate; and	Yano in view of Ogawa, discloses, or at least renders obvious, *comparing a media time in transit and a round trip time estimate* (e.g., Yano's disclosure of transfer time and reception times for data and receiver reports, together with disclosure of data round-trip times; supplemented by Ogawa's disclosure of calculating data transmission and reception periods and round trip times), as claimed. Yano discloses *comparing a media time in transit and a round trip time estimate* by describing that its "transmitting terminal may calculate: (Ts2−Ts1)−(Tr2−Tr1) where Ts1 is the transfer time of data of a given sequence number, Tr1 is the reception time of that data at the receiving side, Tr2 is the transfer time of that receiver report, and Ts2 is the reception time of the receiver report including the sequence number." Yano, [0042]-[0044].

Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	Yano further discloses in paragraphs 89-94: The transmitting terminal (the terminal that provides a video distribution service) in the fourth embodiment holds the minimum value of the **calculated data round-trip times** as a reference round-trip time RTTbase. The **reference round-trip time RTTbase** uses the **round-trip time RTTcur calculated upon reception of the first receiver report after the beginning of data transmission/reception**. That is, at the beginning of transmission, RTTbase =RTTcur After that, every time a receiver report is received, RTTcur is calculated. The calculated round-trip time RTTcur is compared with RTTbase, and if RTTcur is smaller than RTTbase, the RTTbase is updated by that RTTcur: RTTbase=RTTcur (IF RTTcur<RTTbase) The network buffer data volume is calculated on the basis of the difference between the reference value RTTbase of the data round-trip time and the latest measured round-trip time RTTcur. Yano, [0089]-[0094]. This information is also shown in Figure 11, which is a format for a receiver report packet:

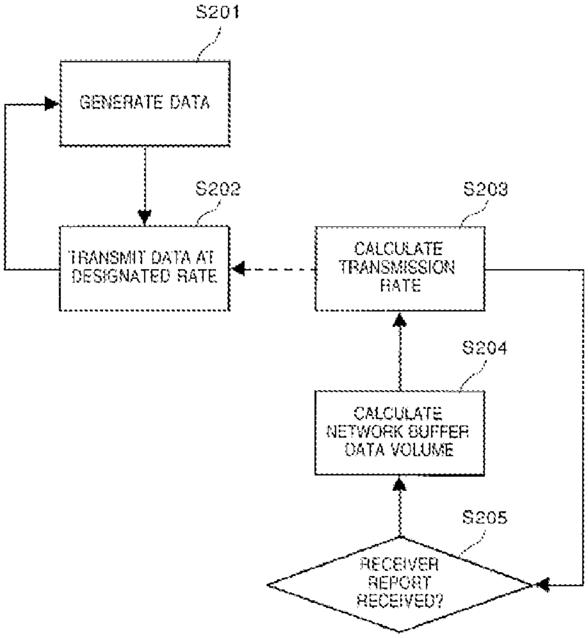
Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	 Yano, FIG. 11. To the extent not disclosed or rendered obvious by Yano, Ogawa provides further support. Ogawa describes *comparing a media time in transit and a round trip time estimate* as follows: In this embodiment, as described with reference to FIG. 4, a **maximum throughput is calculated using measured values**

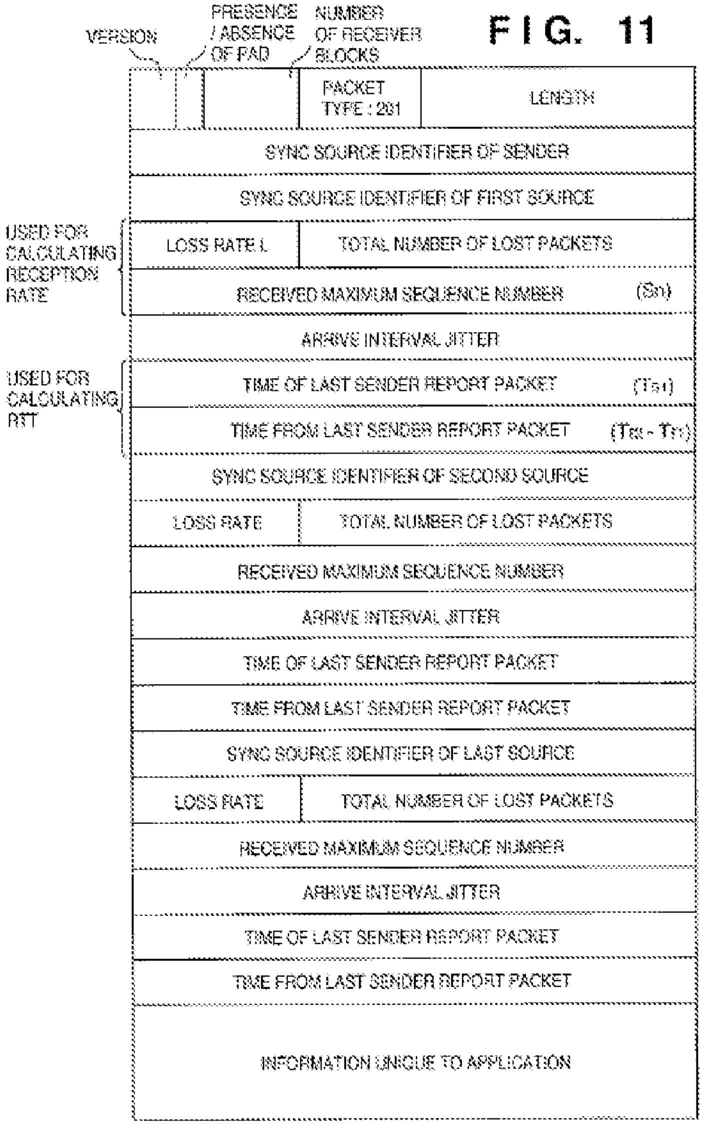
Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	associated with "<u>effective data transmission and reception periods</u>", i.e., the periods of transmission of successive data and reception of reception acknowledgements, such as the periods A and C shown in FIG. 4, and the transmission bitrate is controlled within a range of up to an upper limit of a maximum allowable bitrate corresponding to the maximum throughput. That is, a maximum throughput is calculated on the basis of an effective data transmission and reception period not including an ineffective data transmission and reception period that does not contribute to data transmission and reception, such as the period B shown in FIG. 4, and the bitrate is controlled within a range of up to an upper limit of a maximum allowable bitrate corresponding to the maximum throughput calculated. According to this embodiment, it is possible to control the bitrate in consideration of an actual transmission rate. Thus, data can be transmitted within a range of up to an upper limit corresponding to a maximum bitrate at which data can be transmitted reliably, without excessively increasing or decreasing the bitrate Ogawa, [0147]. FIG. 4  Ogawa, FIG. 4. Furthermore, in the communication processing apparatus, the **communication-bandwidth information includes, for**

Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	example, information that is generated on the basis of data relating to at least one of a round trip time, a received signal strength indicator, and a transmission rate of communications between the access point and the client. Ogawa, [0027]. According to another embodiment of the present invention, there is provided a communication processing apparatus that acts as a client for receiving data from a server. The communication processing apparatus includes a data transceiver configured to carry out communications with the server; and a **packet-interval measurer** configured to measure a reception interval of data packets received from the server. The **packet-interval measurer is configured to measure a reception interval of packets that are received successively, according to identification information included in packets received from the server, and to transmit reception-interval information representing the reception interval or throughput information calculated on the basis of the reception interval to the server via the data transceiver**. Ogawa, [0039]. The teachings of Yano and Ogawa would have been considered together and combined for the reasons discussed in the Request at Section I.E.4.c. Karam Decl., ¶¶ 90-93. * * * Yano discloses: A network buffer data volume BUFcur is obtained by multiplying a packet size Psize by the difference between a sequence number SEQsend of the last packet output from the transmitting terminal and a reception sequence number SEQrecv included in the receiver report sent back from the receiving terminal: $BUFcur=Psize\times(SEQsend-SEQrecv)$ **Note that the time (timer) of the receiving terminal does not always perfectly match that of the transmitting terminal. For this reason, the transmitting terminal may calculate:**

Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	(Ts2−Ts1)−(Tr2−Tr1) where **Ts1 is the transfer time of data of a given sequence number, Tr1 is the reception time of that data at the receiving side, Tr2 is the transfer time of that receiver report, and Ts2 is the reception time of the receiver report including the sequence number**. This difference may be divided by "2" to obtain the time required for transfer. The reason why the difference is divided by "2" is that the difference corresponds to a round trip. In this case, the receiving side adds the reception time and the output time of the receiver report. If the processing time on the receiving time is negligibly short, since the round-trip time alone need be considered, the receiving side need not insert the reception time and output time in the receiver report. Then, a transmission rate Rnew is determined so that the calculated network buffer data volume BUFcur approaches a target value BUFdes of the network buffer data volume (step S **203**). The calculation formula is as follows: Rnew=Rcur+C×(BUFdes−BUFcur) where Rcur is the current transmission rate, and Rnew is the new transmission rate to be determined. C is an appropriate constant. The transmission rate R determined by this processing is supplied to the data transmitter to designate the transmission rate in the data transmission step (S **202**). With this operation, in the transmitting terminal **1-1**, the network buffer data volume calculator **1-14** calculates the buffer capacity of the network present between the two terminals in accordance with the receiver report from the receiving terminal **1-2**, and the transmission rate change unit **1-13** sets the data transfer rate in the data transmitter **1-12** in accordance with the calculation result. Yano, [0042]-[0048]. When the data round-trip time is calculated using such method, an accurate round-trip time required for data transmission can be calculated even when the times of the timers of the receiving and transmitting terminals do not match each other. Since the transmitting terminal need only obtain the time required for the

Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	internal processing of the receiving terminal, the receiving terminal may add that time in place of the reception and transmission times. Also, if the transmitting terminal stores the sequence numbers and transmission times of previous transmission data, the "data transmission time" in FIG. 10 may be omitted. **The transmitting terminal (the terminal that provides a video distribution service) in the fourth embodiment holds the minimum value of the calculated data round-trip times as a reference round-trip time RTTbase. The reference round-trip time RTTbase uses the round-trip time RTTcur calculated upon reception of the first receiver report after the beginning of data transmission/reception**. That is, at the beginning of transmission, RTTbase =RTTcur After that, **every time a receiver report is received, RTTcur is calculated. The calculated round-trip time RTTcur is compared with RTTbase, and if RTTcur is smaller than RTTbase, the RTTbase is updated by that RTTcur**: RTTbase=RTTcur (IF RTTcur<RTTbase) **The network buffer data volume is calculated on the basis of the difference between the reference value RTTbase of the data round-trip time and the latest measured round-trip time RTTcur.** In practice, if Rrecv represents the reception rate (bits/sec) included in the receiver report, the current network buffer data volume BUFcur is calculated by: BUFcur=Rrecv*(RTTcur−RTTbase) Yano, [0089]-[0094]. The method described in this embodiment can be implemented in various kinds of networks, e.g., the internet based on the IP protocol, LAN, and the like. A method of practicing the present invention using an RTP (Real Time Transport Protocol) standardized as RFC1889 in January, '96 will be described below. Note that the RTP is a protocol suitable for real-time applications,

Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	and is used in place of the TCP in the transport layer. FIG. 11 shows an example of a receiver report packet determined in such RTP. **The aforementioned RTT (data round-trip time) and Rrecv (reception rate) are obtained from the RTCP receiver report** packet shown in FIG. 11 by: RTTcur=(Ts2−Ts1)−(Tr2−Tr1) Yano, [0111]-[0112]. FIG. 2  Yano, FIG. 2.

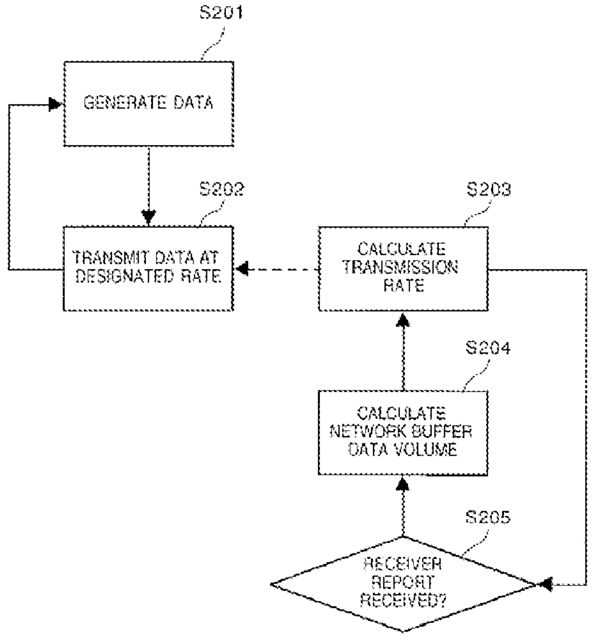
Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	 FIG. 11 Yano, FIG. 11.
[1f] comparing a bitrate received with a current bitrate session; and	Yano in view of Ogawa, discloses, or at least renders obvious, *comparing a bitrate received with a current bitrate session* (e.g., Yano's comparison of current transmission rates, new transmission rates, and reception rates; supplemented by Ogawa's disclosure of a bitrate setter comparing a current transmission-data bitrate with the maximum throughput), as claimed.

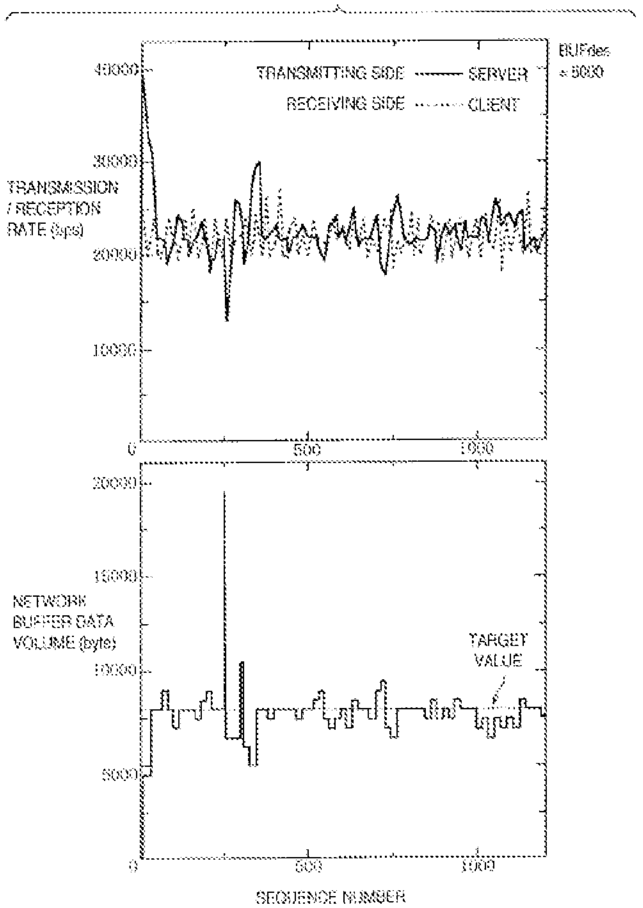
Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	Yano discloses *comparing a bitrate received with a current bitrate session* in paragraphs 69-71: The second embodiment is substantially the same as the first embodiment, except for the sequence (S 203 in FIG. 2) for calculating the transmission rate. In the first embodiment, a new transmission rate is determined with reference to the current transmission rate. However, in this embodiment, **<u>the transmission rate is determined with reference to a rate reported by the receiver report</u>**: Rsend=Rrecv+C×(BUFdes−BUFcur) where **Rsend is the transmission rate**, and **<u>Rrecv is the reception rate calculated based on the receiver report</u>**. **In this fashion, when the <u>transmission rate is determined based on the reception rate</u>, high followability for the reception rate is expected when the transmission rate is corrected upon variation of an available band due to changes in network condition**. Yano, [0069]-[0071]. In this example, Rrecv (i.e., the *bitrate received*) is used in calculating the transmission rate, Rsend (i.e., the *current session bitrate*). To the extent not disclosed or rendered obvious by Yano, Ogawa provides further support. Ogawa discloses *comparing a bitrate received with a current bitrate session*, e.g.: When the buffer amount is greater than the predetermined threshold Th2, in step S204, **<u>the bitrate setter 212 compares the current transmission-data bitrate with the maximum throughput calculated by the throughput calculator 211</u> in the process described above. When the current transmission-data bitrate is less than the bitrate corresponding to the maximum throughput calculated by the throughput calculator 211, in step S205, the bitrate setter 212 increases the transmission bitrate**. However, the bitrate corresponding to the maximum throughput calculated by the throughput calculator 211, i.e., a maximum allowable bitrate, serves as an upper limit.

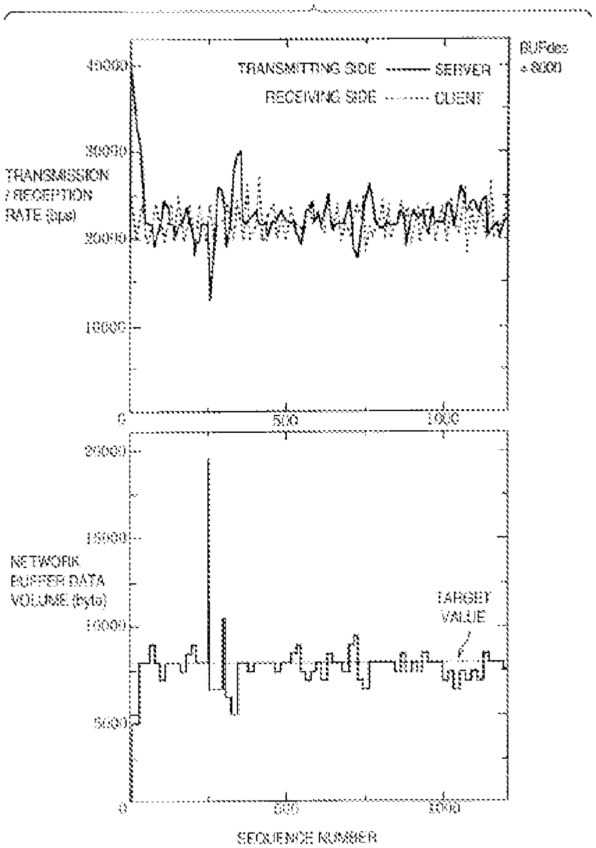
Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	When the current transmission-data bitrate is equal to the maximum allowable bitrate corresponding to the maximum throughput calculated by the throughput calculator 211, the process returns to step S201 without changing the bitrate. This occurs when the buffer amount is in a range Bc of the buffer-amount information 250 shown in FIG. 6. Ogawa, [0145]. The checking algorithm involves, for example, **comparison of the communication-bandwidth information obtained from the access point (AP) 400 with thresholds held in the bitrate setter 212**. More specifically, for example, the bitrate is decreased when parameters included in the communication-bandwidth information received from the access point (AP) 400, such as weighted averages or standard deviations of RTT, RSSI, and transmission rate, exceed the corresponding thresholds held in the bitrate setter 212 N times in succession. The comparison against the thresholds is executed a number of times in order to prevent decreasing the bitrate meaninglessly, for example, when the parameters accidentally exceed the corresponding thresholds only once. Ogawa, [0186]. Furthermore, in the communication processing apparatus, the **communication-bandwidth information includes, for example, information that is generated on the basis of data relating to at least one of a round trip time, a received signal strength indicator, and a transmission rate of communications** between the access point and the client. Ogawa, [0027]. The teachings of Yano and Ogawa would have been considered together and combined for the reasons discussed in the Request at Section I.E.4.c. Karam Decl., ¶¶ 90-93. * * * Yano discloses: Then, **a transmission rate Rnew is determined so that the calculated network buffer data volume BUFcur approaches a**

Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	> **target value BUFdes of the network buffer data volume (step S 203). The calculation formula is as follows**: > Rnew=Rcur+C×(BUFdes−BUFcur) > **<u>where Rcur is the current transmission rate, and Rnew is the new transmission rate to be determined</u>**. C is an appropriate constant. The transmission rate R determined by this processing is supplied to the data transmitter to designate the transmission rate in the data transmission step (S **202**). > With this operation, in the transmitting terminal **1-1**, the network buffer data volume calculator **1-14** calculates the buffer capacity of the network present between the two terminals in accordance with the receiver report from the receiving terminal **1-2**, and the transmission rate change unit **1-13** sets the data transfer rate in the data transmitter **1-12** in accordance with the calculation result. Yano, [0046]-[0048]. > By repeating the above-mentioned steps, the receiving terminal periodically transmits receiver reports to the transmitting terminal while transmitting/receiving data between the transmitting and receiving terminals. On the other hand, the transmitting terminal determines the transmission rate based on the receiver report to make the network buffer a constant volume of data. > **FIG. 6 shows an example of <u>variations of the transmission and reception rates and network buffer data volume actually measured in data communications</u> of this embodiment**. The transmission and reception rates stabilize at an identical value around the available band of the network, and the network data buffer volume changes in the neighborhood of the target value. Yano, [0051]-[0052]. > The second embodiment is substantially the same as the first embodiment, except for the sequence (S 203 in FIG. 2) for calculating the transmission rate. In the first embodiment, a new transmission rate is determined with reference to the current transmission rate. However, in this embodiment, **<u>the transmission rate is determined with reference to a rate reported by the receiver report</u>**:

Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	Rsend=Rrecv+C×(BUFdes−BUFcur) where Rsend is the transmission rate, and **Rrecv is the reception rate calculated based on the receiver report**. **In this fashion, when the transmission rate is determined based on the reception rate, high followability for the reception rate is expected when the transmission rate is corrected upon variation of an available band due to changes in network condition**. Yano, [0069]-[0071]. The operation sequence of the third embodiment is also nearly the same as those in the first and second embodiments, except for the sequence (S **203** in FIG. 2) for calculating the transmission rate. In the first and second embodiments, for the constant C used in control for determining the transmission rate, a value that assures stable operation must be empirically found. However, in the third embodiment, the transmission rate is determined without using such constant. Yano, [0073]. In this embodiment, **a transmission rate Rnew is determined so that the calculated network buffer data volume BUFcur approaches the target value BUFdes of the network buffer data volume (step S 203). The transmission rate Rnew is calculated by**: Rnew=Rcur+C*(BUFdes−BUFcur) **where Rcur is the current transmission rate, and Rnew is the new transmission rate to be determined**. C is an appropriate constant. The transmission rate Rnew determined by this processing is supplied to the data transmitter to designate the transmission rate in the data transmission step (step S 202 in FIG. 2). In this embodiment, the transmission rate is determined with reference to the transmission rate Rcur but may be determined with reference to the reception rate Rrecv. If the reception interval (Interval) of the receiver reports is known, the transmission rate

Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	Rnew may be calculated so that the network buffer data volume matches the target value upon arrival of the next receiver report: Rnew=Rrecv+{(BUFdes−BUFcur)/Interval} Yano, [0096]-[0098]. FIG. 2  Yano, FIG. 2.

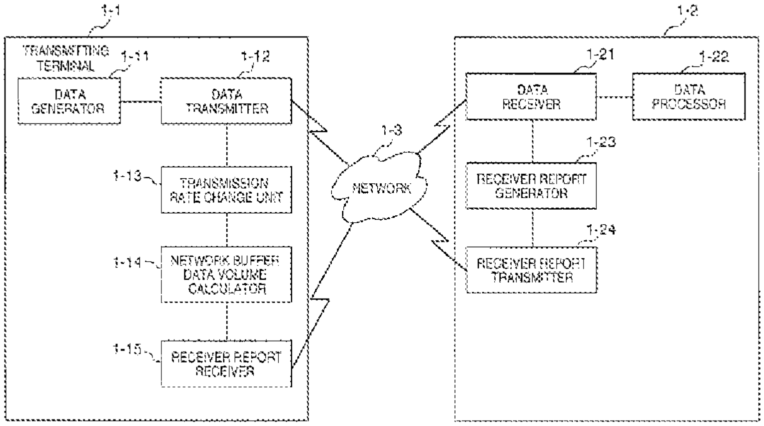
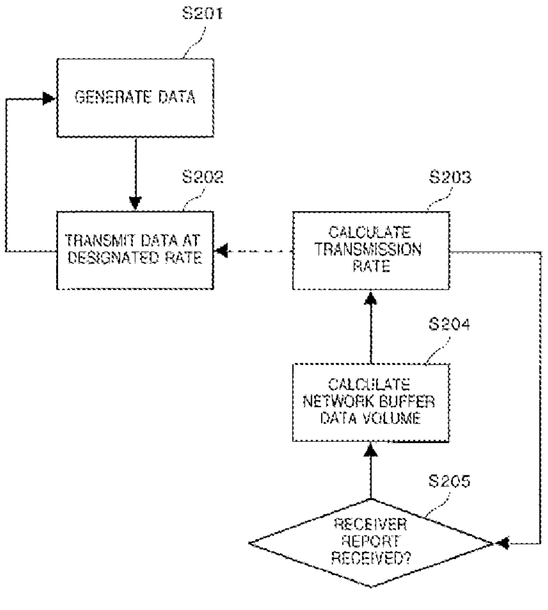
Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	FIG. 6  Yano, FIG. 6.
[1g] determining the stability of the media network; and	Yano in view of Ogawa, discloses, or at least renders obvious, *determining the stability of the media network* (e.g., Yano's disclosure of stabilizing transfer and reception rates "around the available band of the network"; supplemented by Ogawa's disclosure of a communication-bandwidth monitor for increasing or decreasing bitrates based on network conditions), as claimed. Yano discloses *determining the stability of the media network* by its very nature—i.e., the purpose of the constant sending and receiving of receiver reports is to assess the current stability of the network, the data buffers, and whether or not bitrates need to be adjusted by the transmitting terminal 1-1.

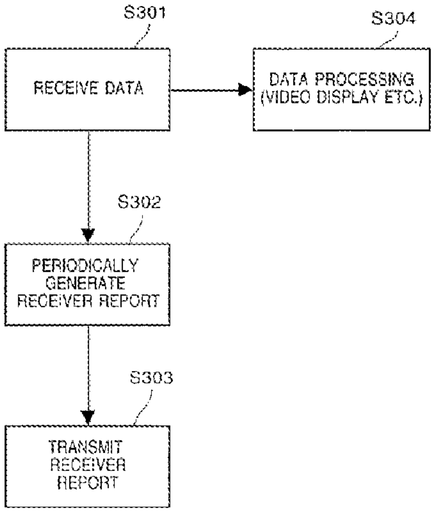
Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	For example, as described by Yano in discussing Figure 6, the "transmission and reception rates **<u>stabilize</u> at an identical value around the available band of the network**, and the network data buffer volume changes in the neighborhood of the target value." Yanno, [0052]. Yano acknowledged that "[e]ven when the network traffic is very smooth and a high transfer rate may be set, the data transfer is controlled **to make constant the data volume** which stays as buffer data on the network without increasing the transfer rate. Hence, the load on the network can be reduced, and use of the network by other parties is not disturbed." Yano, [0064]. Accordingly, when Yano's system stabilizes its transmission and reception rates around the available band of the network, and controls data transfer to make constant the data volume, it performs *determining the stability of the media network*. FIG. 6  Yano, FIG. 6. To the extent not disclosed or rendered obvious by Yano, Ogawa provides further support.

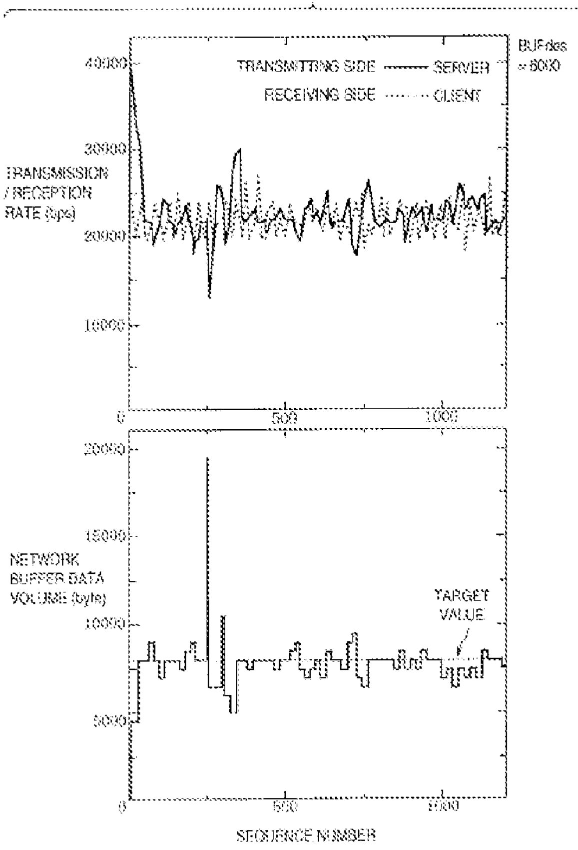
Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	Ogawa discloses *determining the stability of the media network* because it discloses increasing or decreasing bitrate in response to current network conditions, a process which involves *determining the stability of the media network. See, e.g.*, Ogawa: **In streaming distribution, it is desirable to distribute data at an optimal transmission rate.** For example, according to a scheme disclosed in Japanese Unexamined Patent Application Publication No. 2004-297565, **a data transmitting apparatus and a data receiving apparatus measure an upstream transmission rate and a downstream transmission rate, and a rate controlling apparatus exercises control so that data is transmitted and received at a smaller one of the rates measured, so that data can be transmitted stably.** Ogawa, [0009]. In step S503, it is checked on the basis of the result of analysis in step S502 whether the transmission bitrate is to be decreased. When it is determined that the transmission bitrate is to be decreased, in step S504, the bitrate setter 212 decreases the transmission bitrate. **Furthermore, in step S505, the bitrate setter 212 checks on the basis of the result of analysis in step S502 whether it is possible to increase the transmission bitrate**. When it is determined that it is possible to increase the transmission bitrate, in step S506, the bitrate setter 212 increases the transmission bitrate. However, it is not allowed to increase the bitrate beyond an upper limit of a maximum allowable bitrate corresponding to a maximum throughput calculated by the throughput calculator 211. The upper limit is a maximum allowable bitrate corresponding to a maximum throughput calculated by the throughput calculator 211 or by a client, described in the context of the first and second embodiments. Ogawa, [0190]. The teachings of Yano and Ogawa would have been considered together and combined for the reasons discussed in the Request at Section I.E.4.c. Karam Decl., ¶¶ 90-93. * * *

<table>
<tr><th>Claims</th><th>Relevant Disclosures in the Prior Art</th></tr>
<tr><td colspan="2">Claim 1 (Ground 4)</td></tr>
<tr><td></td><td>
<p>Yano discloses:</p>
<blockquote>
<p>The data receiver 1-21 measures information pertaining to the sequence number of the received data, data reception time, the received data volume, and the like, and sends that information to a receiver report generator 1-23. The receiver report generator 1-23 calculates the reception rate that must be included in the receiver report, and supplies it to a receiver report transmitter 1-24 together with the sequence number.</p>
<p>The receiver report transmitter 1-24 transmits the receiver report to the receiver report receiver 1-14 of the transmitting terminal 1-1 via the network 1-3.</p>
</blockquote>
<p>Yano, [0036]-[0037].</p>
<blockquote>
<p>By repeating the above-mentioned steps, the receiving terminal periodically transmits receiver reports to the transmitting terminal while transmitting/receiving data between the transmitting and receiving terminals. On the other hand, the transmitting terminal determines the transmission rate based on the receiver report to make the network buffer a constant volume of data.</p>
<p>FIG. 6 shows an example of variations of the transmission and reception rates and network buffer data volume actually measured in data communications of this embodiment. <u>The transmission and reception rates stabilize at an identical value around the available band of the network, and the network data buffer volume changes in the neighborhood of the target value</u>.</p>
</blockquote>
<p>Yano, [0051]-[0052].</p>
<blockquote>
<p>As a consequence, data transfer between two terminals via the network can be optimally done in correspondence with the buffer capacity of that network. Even when the network traffic is very smooth and a high transfer rate may be set, the data transfer is controlled to make constant the data volume which stays as buffer data on the network without increasing the transfer rate. Hence, the load on the network can be reduced, and use of the network by other parties is not disturbed.</p>
</blockquote>
<p>Yano, [0064].</p>
</td></tr>
</table>

Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	The operation sequence of the third embodiment is also nearly the same as those in the first and second embodiments, except for the sequence (S203 in FIG. 2) for calculating the transmission rate. In the first and second embodiments, for the constant C used in control for determining the transmission rate, a value that assures stable operation must be empirically found. However, in the third embodiment, the transmission rate is determined without using such constant. Yano, [0073]. To restate, according to the fourth embodiment, **the transmission rate is controlled to maintain constant the data volume (network buffer data volume) that has been output from the transmitting terminal onto the network but has not reached the receiving terminal yet, in accordance with an increase in data round-trip time. With this control, since the data volume which is en route to the destination on the network can be accurately calculated, and is adjusted, a transmission delay can be suppressed to fall within an allowable range.** Since data is transmitted/received to always save data in a buffer, the available band of the network can be sufficiently used to transmit/receive the data. **Since the data volume buffered in the network is used as a parameter for controlling the transmission rate without directly using the delay time, a sufficient buffer volume which is to exist on the network can be defined as a target buffer data volume, and data can be prevented from being excessively output onto the network**. Yano, [0123].

Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	FIG. 1  Yano, FIG. 1. FIG. 2  Yano, FIG. 2.

<table>
<tr><th>Claims</th><th>Relevant Disclosures in the Prior Art</th></tr>
<tr><td colspan="2">Claim 1 (Ground 4)</td></tr>
<tr><td></td><td>FIG. 3

Yano, FIG. 3.</td></tr>
</table>

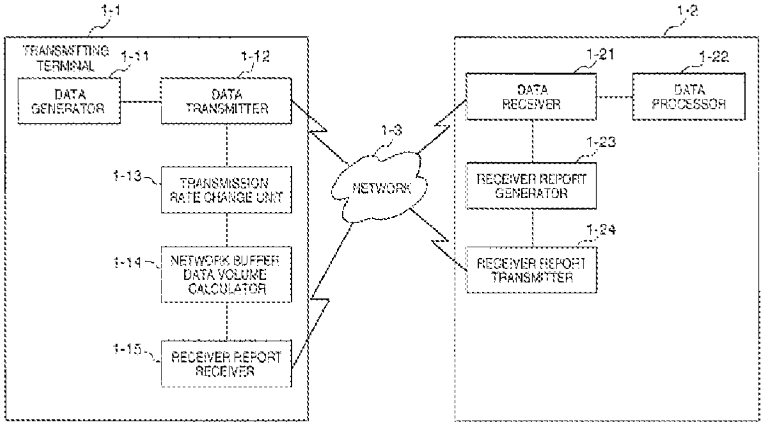
Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	FIG. 6  Yano, FIG. 6.
[1h] providing the optimal session bitrate based at least in part on the media-network-stability determination; and	Yano in view of Ogawa, discloses, or at least renders obvious, *providing the optimal session bitrate based at least in part on the media-network-stability determination* (e.g., Yano's disclosure of transmitting data "at an optimal transfer rate" based on current network conditions and network stability; supplemented as needed by Ogawa's disclosure of transmitting data at an "optimal bit rate" based on current network conditions and network stability), as claimed. This limitation is disclosed by Yano and Ogawa for the same reasons discussed above in limitations [1c] (i.e., discussing the *optimal session bitrate using the estimated one or more network conditions*) and limitation [1g] (i.e., discussing the *media-network-stability determination*).

Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	* * * Yano discloses: **This invention can make data communications at an optimal transfer rate on the basis of the unarrived data volume on a network between two end terminals**. For this purpose, a transmitting terminal (**1-1**) adds sequence number information to data generated by a data generator (**1-11**), and transmits the data to a receiving terminal (**1-2**) via a data transmitter (**1-12**). Since the receiving terminal (**1-2**) transmits data including the sequence number in the received data, the transmitting terminal determines that data (buffer capacity) corresponding to the difference between the current sequence number and the received sequence number remain on the network, and calculates that volume using a network buffer data volume calculator (**1-14**). **The transmitting terminal determines the transmission rate on the basis of the calculation result, and controls the data transmitter (1-12) to transfer data at that transmission rate**. Yano, Abstract. It is another object of the present invention to provide a data communication apparatus, method, system, and storage medium, **which perform data communications at an optimal transfer rate on the basis of the volume of unarrived data on the network present between two end terminals**. It is another object of the present invention to provide a data communication apparatus, receiving apparatus, **control method, storage medium, and data communication system, which can realize optimal data transfer by dynamically controlling to change the transfer rate in correspondence with the conditions on the network in data communications via the network**. Yano, [0007]-[0008]. The data receiver **1-21** measures information pertaining to the sequence number of the received data, data reception time, the received data volume, and the like, and sends that information to a receiver report generator **1-23**. The receiver report generator **1-**

Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	> **23** calculates the reception rate that must be included in the receiver report, and supplies it to a receiver report transmitter **1-24** together with the sequence number. > The receiver report transmitter **1-24** transmits the receiver report to the receiver report receiver **1-14** of the transmitting terminal **1-1** via the network **1-3**. Yano, [0036]-[0037]. > Then, a transmission rate Rnew is determined so that the calculated network buffer data volume BUFcur approaches a target value BUFdes of the network buffer data volume (step S **203**). The calculation formula is as follows: > Rnew=Rcur+C×(BUFdes−BUFcur) > where Rcur is the current transmission rate, and Rnew is the new transmission rate to be determined. C is an appropriate constant. The transmission rate R determined by this processing is supplied to the data transmitter to designate the transmission rate in the data transmission step (S **202**). > With this operation, in the transmitting terminal **1-1**, the network buffer data volume calculator **1-14** calculates the buffer capacity of the network present between the two terminals in accordance with the receiver report from the receiving terminal **1-2**, and the transmission rate change unit **1-13** sets the data transfer rate in the data transmitter **1-12** in accordance with the calculation result. Yano, [0046]-[0048]. > **By repeating the above-mentioned steps, the receiving terminal periodically transmits receiver reports to the transmitting terminal while transmitting/receiving data between the transmitting and receiving terminals**. On the other hand, the transmitting terminal determines the transmission rate based on the receiver report to make the network buffer a constant volume of data. > FIG. 6 shows an example of variations of the transmission and reception rates and network buffer data volume actually measured in data communications of this embodiment. The transmission and reception rates stabilize at an identical value around the

Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	available band of the network, and the network data buffer volume changes in the neighborhood of the target value. Yano, [0051]-[0052]. The fourth embodiment will be described below. The arrangement of the transmitting and receiving terminals of the fourth embodiment is the same as that in the first embodiment. **Also, the transmitting terminal determines the transmission rate on the basis of a receiver report sent from the receiving terminal as in the first embodiment**. Hence, the overall operations are the same as those shown in FIGS. 2 and 3, and the format of transmission data is the same as that shown in FIG. 4. However, the format of the receiver report sent from the receiving terminal 1-2 in the fourth embodiment is as shown in FIG. 10. Yano, [0081]. In this embodiment, a transmission rate Rnew is determined so that the calculated network buffer data volume BUFcur approaches the target value BUFdes of the network buffer data volume (step S **203**). The transmission rate Rnew is calculated by: $\text{Rnew}=\text{Rcur}+C*(\text{BUFdes}-\text{BUFcur})$ where Rcur is the current transmission rate, and Rnew is the new transmission rate to be determined. C is an appropriate constant. The transmission rate Rnew determined by this processing is supplied to the data transmitter to designate the transmission rate in the data transmission step (step S 202 in FIG. 2). In this embodiment, the transmission rate is determined with reference to the transmission rate Rcur but may be determined with reference to the reception rate Rrecv. If the reception interval (Interval) of the receiver reports is known, the transmission rate Rnew may be calculated so that the network buffer data volume matches the target value upon arrival of the next receiver report: $\text{Rnew}=\text{Rrecv}+\{(\text{BUFdes}-\text{BUFcur})/\text{Interval}\}$ Yano, [0096]-[0098].

Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	The operation sequence of the third embodiment is also nearly the same as those in the first and second embodiments, except for the sequence (S203 in FIG. 2) for calculating the transmission rate. In the first and second embodiments, for the constant C used in control for determining the transmission rate, a value that assures stable operation must be empirically found. However, in the third embodiment, the transmission rate is determined without using such constant. Yano, [0073]. To restate, according to the fourth embodiment, **the transmission rate is controlled to maintain constant the data volume (network buffer data volume) that has been output from the transmitting terminal onto the network but has not reached the receiving terminal yet, in accordance with an increase in data round-trip time. With this control, since the data volume which is en route to the destination on the network can be accurately calculated, and is adjusted, a transmission delay can be suppressed to fall within an allowable range.** Since data is transmitted/received to always save data in a buffer, the available band of the network can be sufficiently used to transmit/receive the data. **Since the data volume buffered in the network is used as a parameter for controlling the transmission rate without directly using the delay time, a sufficient buffer volume which is to exist on the network can be defined as a target buffer data volume, and data can be prevented from being excessively output onto the network**. Yano, [0123]. As described above, according to the fourth embodiment, **the transfer rate is dynamically changed in correspondence with the network condition in data communications via the network, thus realizing optimal data transfer.** Hence, the present invention is particularly effective for real-time processing, e.g., transferring live images captured by a camera. Yano, [0126].

Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	FIG. 1  Yano, FIG. 1.
[1i] providing media data to the terminal according to the optimal session bitrate.	Yano discloses, or at least renders obvious, *providing media data to the terminal according to the optimal session bitrate* (e.g., Yano's disclosure of transmitting data from data transmitter 1-12 according to an "optimal transmission rate" determined by the transmission rate change unit 1-13 and network buffer data volume calculator 1-14), as claimed. Yano discloses *providing media data to the terminal according to the optimal session bitrate* because it discloses generally that its system "can make **data communications at an optimal transfer rate** on the basis of the unarrived data volume on a network between two end **terminals**." Yano, Abstract. This data is *media data* because Yano describes that its system transmits "**video** data, **audio** data, and the like via a network." Yano, [0001]. * * * Yano discloses: **This invention can make data communications at an optimal transfer rate on the basis of the unarrived data volume on a network between two end terminals**. For this purpose, **a transmitting terminal (1-1) adds sequence number information to data generated by a data generator (1-11), and transmits the data to a receiving terminal (1-2) via a data transmitter (1-12)**. Since the receiving terminal **(1-2)**

Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	transmits data including the sequence number in the received data, the transmitting terminal determines that data (buffer capacity) corresponding to the difference between the current sequence number and the received sequence number remain on the network, and calculates that volume using a network buffer data volume calculator (**1-14**). **The transmitting terminal determines the transmission rate on the basis of the calculation result, and controls the data transmitter (1-12) to transfer data at that transmission rate**. Yano, Abstract. The present invention relates to an apparatus and system for **transmitting and/or receiving steadily generated data such as video data, audio data**, and the like via a network. Yano, [0001]. It is another object of the present invention to provide a data communication apparatus, method, system, and storage medium, **which perform data communications at an optimal transfer rate on the basis of the volume of unarrived data on the network present between two end terminals**. It is another object of the present invention to provide a data communication apparatus, receiving apparatus, **control method, storage medium, and data communication system, which can realize optimal data transfer by dynamically controlling to change the transfer rate in correspondence with the conditions on the network in data communications via the network**. Yano, [0007]-[0008]. The data receiver **1-21** measures information pertaining to the sequence number of the received data, data reception time, the received data volume, and the like, and sends that information to a receiver report generator **1-23**. The receiver report generator **1-23** calculates the reception rate that must be included in the receiver report, and supplies it to a receiver report transmitter **1-24** together with the sequence number.

Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	The receiver report transmitter **1-24** transmits the receiver report to the receiver report receiver **1-14** of the transmitting terminal **1-1** via the network **1-3**. Yano, [0036]-[0037]. Then, a transmission rate Rnew is determined so that the calculated network buffer data volume BUFcur approaches a target value BUFdes of the network buffer data volume (step S **203**). The calculation formula is as follows: $Rnew=Rcur+C\times(BUFdes-BUFcur)$ where Rcur is the current transmission rate, and Rnew is the new transmission rate to be determined. C is an appropriate constant. The transmission rate R determined by this processing is supplied to the data transmitter to designate the transmission rate in the data transmission step (S **202**). With this operation, in the transmitting terminal **1-1**, the network buffer data volume calculator **1-14** calculates the buffer capacity of the network present between the two terminals in accordance with the receiver report from the receiving terminal **1-2**, and the transmission rate change unit **1-13** sets the data transfer rate in the data transmitter **1-12** in accordance with the calculation result. Yano, [0046]-[0048]. **By repeating the above-mentioned steps, the receiving terminal periodically transmits receiver reports to the transmitting terminal while transmitting/receiving data between the transmitting and receiving terminals**. On the other hand, the transmitting terminal determines the transmission rate based on the receiver report to make the network buffer a constant volume of data. FIG. 6 shows an example of variations of the transmission and reception rates and network buffer data volume actually measured in data communications of this embodiment. The transmission and reception rates stabilize at an identical value around the available band of the network, and the network data buffer volume changes in the neighborhood of the target value. Yano, [0051]-[0052].

<table>
<tr><th>Claims</th><th>Relevant Disclosures in the Prior Art</th></tr>
<tr><td colspan="2">Claim 1 (Ground 4)</td></tr>
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<blockquote>The fourth embodiment will be described below. The arrangement of the transmitting and receiving terminals of the fourth embodiment is the same as that in the first embodiment. Also, the <u>transmitting terminal determines the transmission rate on the basis of a receiver report</u> sent from the receiving terminal as in the first embodiment. Hence, the overall operations are the same as those shown in FIGS. 2 and 3, and the format of transmission data is the same as that shown in FIG. 4. However, the format of the receiver report sent from the receiving terminal 1-2 in the fourth embodiment is as shown in FIG. 10.</blockquote>
<p>Yano, [0081].</p>
<blockquote>In this embodiment, a transmission rate Rnew is determined so that the calculated network buffer data volume BUFcur approaches the target value BUFdes of the network buffer data volume (step S 203). The transmission rate Rnew is calculated by:
<p>Rnew=Rcur+C*(BUFdes−BUFcur)</p>
<p>where Rcur is the current transmission rate, and Rnew is the new transmission rate to be determined. C is an appropriate constant. The transmission rate Rnew determined by this processing is supplied to the data transmitter to designate the transmission rate in the data transmission step (step S 202 in FIG. 2).</p>
<p>In this embodiment, the transmission rate is determined with reference to the transmission rate Rcur but may be determined with reference to the reception rate Rrecv. If the reception interval (Interval) of the receiver reports is known, the transmission rate Rnew may be calculated so that the network buffer data volume matches the target value upon arrival of the next receiver report:</p>
<p>Rnew=Rrecv+{(BUFdes−BUFcur)/Interval}</p></blockquote>
<p>Yano, [0096]-[0098].</p>
<blockquote>To restate, according to the fourth embodiment, the transmission rate is controlled to maintain constant the data volume (network buffer data volume) that has been output from the transmitting terminal onto the network but has not reached the receiving terminal yet, in accordance with an increase in</blockquote>
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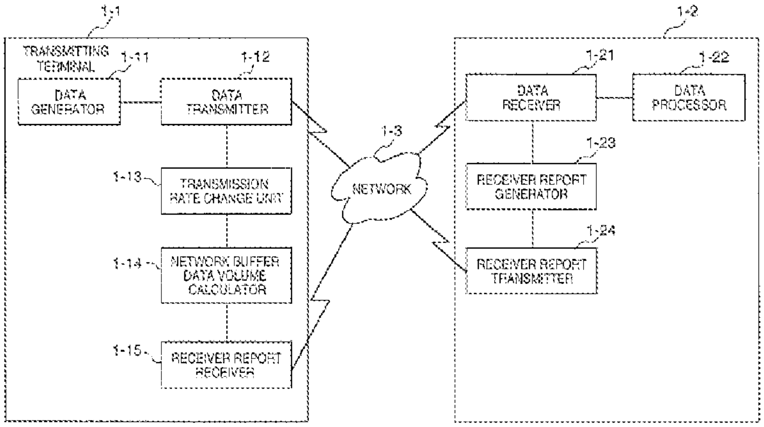
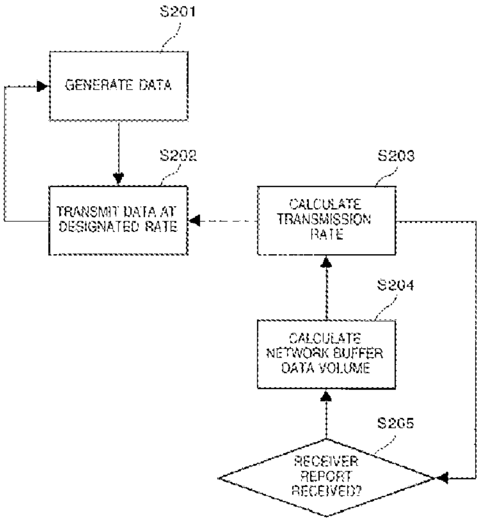
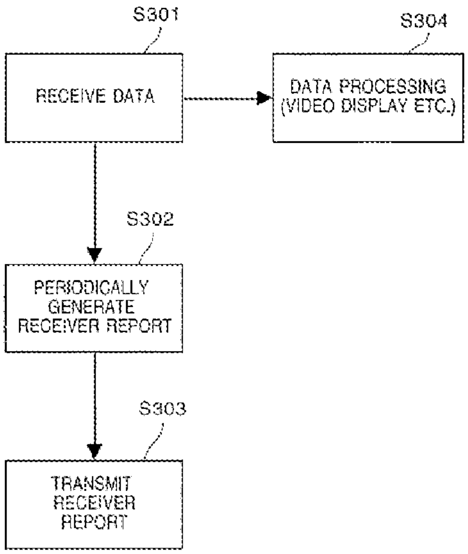
Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	data round-trip time. With this control, since the data volume which is en route to the destination on the network can be accurately calculated, and is adjusted, a transmission delay can be suppressed to fall within an allowable range. Since data is transmitted/received to always save data in a buffer, the available band of the network can be sufficiently used to transmit/receive the data. **Since the data volume buffered in the network is used as a parameter for controlling the transmission rate without directly using the delay time, a sufficient buffer volume which is to exist on the network can be defined as a target buffer data volume, and data can be prevented from being excessively output onto the network.** Yano, [0123]. As described above, according to the fourth embodiment, **the transfer rate is dynamically changed in correspondence with the network condition in data communications via the network, thus realizing optimal data transfer.** Hence, the present invention is particularly effective for real-time processing, e.g., transferring live images captured by a camera. Yano, [0126]. FIG. 1  Yano, FIG. 1.

Claims	Relevant Disclosures in the Prior Art
Claim 1 (Ground 4)	
	FIG. 2  Yano, FIG. 2. FIG. 3  Yano, FIG. 3.

<table>
<tr><th>Claims</th><th>Relevant Disclosures in the Prior Art</th></tr>
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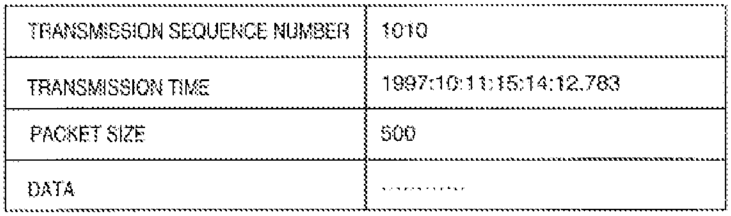
FIG. 4

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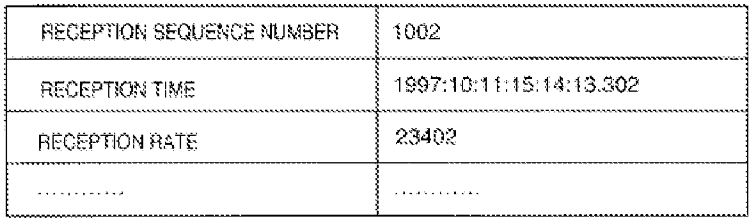
Yano, FIG. 4.

FIG. 5

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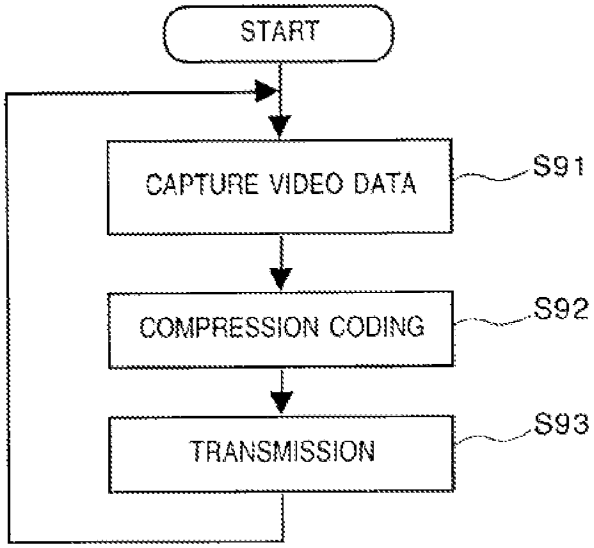
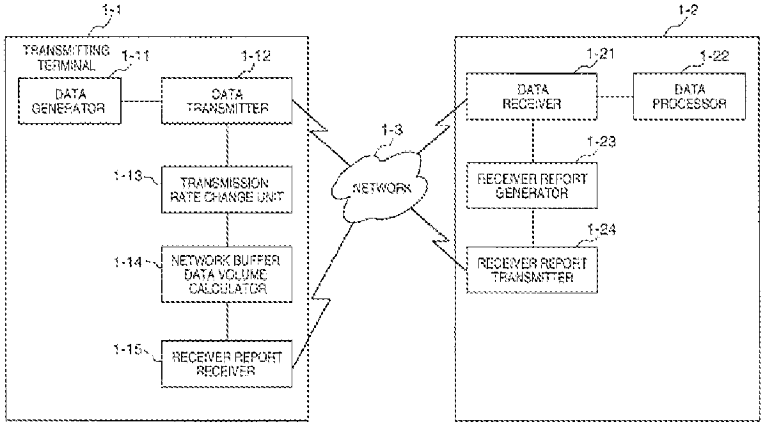
</table>
Yano, FIG. 5.
</td></tr>
</table>

Claims	Relevant Disclosures in the Prior Art
Claim 6 (Ground 4)	
[6P] A method comprising:	Yano discloses, or at least renders obvious, the method of claim 6 for the same reasons as discussed throughout claim 1 above. *See* claim 1 above (e.g., limitation [1P]).
[6a] receiving a receiver report from a terminal;	Yano discloses, or at least renders obvious, *receiving a receiver report from a terminal* for the same reasons as discussed above with respect to limitation [1a] of claim 1 (reciting the same limitation). *See* limitation [1a] above.
[6b] estimating one or more network conditions	Yano discloses, or at least renders obvious, *estimating one or more network conditions of a media network using the receiver report* for

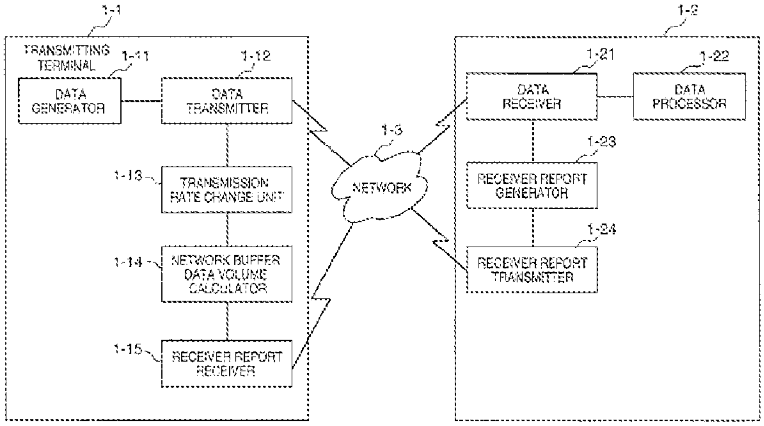
Claims	Relevant Disclosures in the Prior Art
Claim 6 (Ground 4)	
of a media network using the receiver report;	the same reasons as discussed above with respect to limitation [1b] of claim 1 (reciting the same limitation). *See* limitation [1b] above.
[6c] determining stability criterion, wherein determining stability criterion comprises at least one of:	Yano in view of Ogawa discloses, or at least renders obvious, "*determining stability criterion, wherein determining stability criterion comprises at least one of:*" for the same reasons as discussed above with respect to limitation [1d] of claim 1 (reciting "determining stability criterion using the estimated one or more network conditions, wherein determining stability criterion includes at least one of"). *See* limitation [1d] above.
[6d] comparing a media time in transit and a round trip time estimate; and	Yano in view of Ogawa discloses, or at least renders obvious, *comparing a media time in transit and a round trip time estimate* for the same reasons as discussed above with respect to limitation [1e] of claim 1 (reciting the same limitation). *See* limitation [1e] above.
[6e] comparing a bitrate received with a current bitrate session; and	Yano in view of Ogawa discloses, or at least renders obvious, *comparing a bitrate received with a current bitrate session* for the same reasons as discussed above with respect to limitation [1f] of claim 1 (reciting the same limitation). *See* limitation [1f] above.
[6f] determining the stability of the media network using the determined stability criterion;	Yano in view of Ogawa discloses, or at least renders obvious, *determining the stability of the media network using the determined stability criterion* for the same reasons as discussed above with respect to limitation [1g] of claim 1 (reciting "determining the stability of the media network"). *See* limitation [1g] above.
[6g] controlling a session bitrate based at least in part on the media-network-stability determination; and	This limitation is similar to limitation [1h] – however, while limitation [1h] recites "*providing the optimal session bitrate* based at least in part on" limitation [6g] recites "*controlling a session bitrate based at least in part on*." *See* limitation [1h] above.

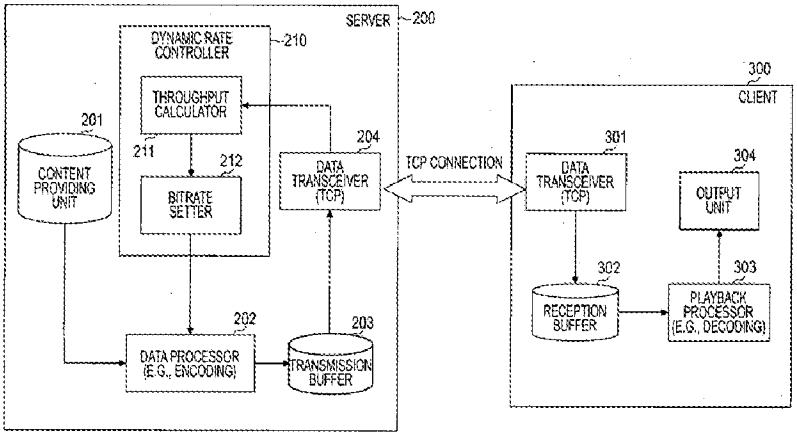
Claims	Relevant Disclosures in the Prior Art
Claim 6 (Ground 4)	
	Yano discloses *controlling a session bitrate* because it describes that: To restate, according to the fourth embodiment, **the transmission rate is controlled to maintain constant the data volume (network buffer data volume) that has been output from the transmitting terminal onto the network but has not reached the receiving terminal yet, in accordance with an increase in data round-trip time. With this control, since the data volume which is en route to the destination on the network can be accurately calculated, and is adjusted, a transmission delay can be suppressed to fall within an allowable range.** Since data is transmitted/received to always save data in a buffer, the available band of the network can be sufficiently used to transmit/receive the data. **Since the data volume buffered in the network is used as a parameter for controlling the transmission rate without directly using the delay time, a sufficient buffer volume which is to exist on the network can be defined as a target buffer data volume, and data can be prevented from being excessively output onto the network**. Yano, [0123]. As described above, according to the fourth embodiment, **the transfer rate is dynamically changed in correspondence with the network condition in data communications via the network, thus realizing optimal data transfer.** Hence, the present invention is particularly effective for real-time processing, e.g., transferring live images captured by a camera. Yano, [0126]. Yano also notes that the "**transmitting terminal determines the transmission rate** on the basis of the calculation result, and controls the data transmitter (1-12) to transfer data at that transmission rate." Yano, Abstract. Yano also mentions that "**the transmitting terminal determines the transmission rate on the basis of a receiver report** sent from the receiving terminal as in the first embodiment." Yano, [0081].

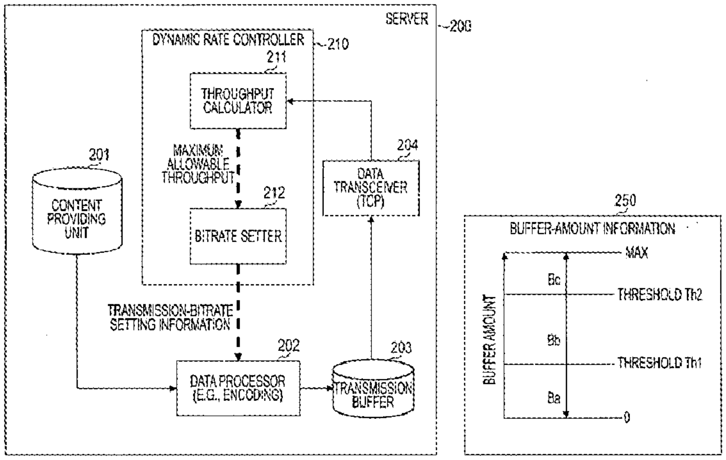
Claims	Relevant Disclosures in the Prior Art
Claim 6 (Ground 4)	
[6h] providing the session bitrate to an encoder for transmitting media data according to the provided session bitrate.	Yano discloses, or at least renders obvious, *providing the session bitrate to an encoder for transmitting media data according to the provided session bitrate largely* for the same reasons as discussed above with respect to limitation [1i] of claim 1 (reciting the same limitation). *See* limitation [1i] above. In addition, this limitation describes that the *session bitrate* is provided *to an encoder for transmitting media data*. Yano discloses this too because it describes that "each transmission data consists of the sequence number (information for specifying that data) of that data, transmission time (the time of the transmitting terminal), packet size, **and compressed and encoded image data**." Yano [0143]. *See also* Yano, [0040] ("when video data is to be transmitted, an image is captured by a video camera, and the captured video data is **compressed and coded**"). FIG. 9  Yano, FIG. 9.

Claims	Relevant Disclosures in the Prior Art
Claim 11 (Ground 4)	
[11P] A system comprising:	Yano discloses, or at least renders obvious, the system of claim 11 for the same reasons as discussed throughout claim 1 above. *See* claim 1 above (e.g., limitation [1P]).
[11a] a terminal, having a media player, configured to provide a receiver report; and	Yano discloses, or at least renders obvious, *a terminal, having a media player, configured to provide a receiver report* (e.g., Yano's receiving terminal 1-2), as claimed. As to *a terminal . . . configured to provide a receiver report*, Yano's receiving terminal 1-2 includes a "**receiver report transmitter** 1-24," as shown in Figure 1 below. As to *a terminal, having a media player*, Yano describes that "in the receiving terminal 1-2, the data transmitted via the network 1-3 are received by a data receiver 1-21. The received data are sent to and processed by a data processor 1-22. For example, **when the received data form video data, processing for displaying an image (decoding, display processing, and the like) is executed by the processor 1-22**." Yano, [0035]. Accordingly, Yano's data processor 1-22, which has the ability to decode, perform display processing, and display an image, constitutes *a terminal, having a media player*. *See also* Yano, [0056]. **FIG. 1**  Yano, FIG. 1. FIG. 1 shows the connection relationship and structure of the respective devices when data transmitted by a transmitting terminal **1-1** is received by a receiving terminal **1-2** via a

Claims	Relevant Disclosures in the Prior Art
Claim 11 (Ground 4)	
	network **1-3**. Note that the network **1-3** includes anything from LANs used in corporations to a collection of many and unspecified networks coupled to each other (e.g., the internet), and is not particularly limited. The arrangements and operations of the terminals shown in FIG. 1 will be explained below. Yano, [0031]. For this purpose, a transmitting terminal (**1-1**) adds sequence number information to data generated by a data generator (**1-11**), and transmits the data to a receiving terminal (**1-2**) via a data transmitter (**1-12**). Since the receiving terminal (**1-2**) transmits data including the sequence number in the received data, the transmitting terminal determines that data (buffer capacity) corresponding to the difference between the current sequence number and the received sequence number remain on the network, and calculates that volume using a network buffer data volume calculator (**1-14**). The transmitting terminal determines the transmission rate on the basis of the calculation result, and controls the data transmitter (**1-12**) to transfer data at that transmission rate. Yano, Abstract.
[11b] an adaptive bitrate manager configured to:	Yano in view of Ogawa discloses, or at least renders obvious, *an adaptive bitrate manager*, as claimed. Yano's transmission rate change unit 1-13 (used in conjunction with network buffer data volume calculator 1-14 and receiver report receiver 1-15) acts as an *adaptive bitrate manager*. As described above in limitation [1c] relating to the *optimal session bitrate*, Yano's purpose is to "realize optimal data transfer by **<u>dynamically controlling to change the transfer rate</u>** in correspondence with the conditions on the network in data communications via the network." Yano, [0008]. As described above, according to the fourth embodiment, **the <u>transfer rate is dynamically changed in correspondence with the network condition</u> in data communications via the network, thus realizing optimal data transfer.** Hence, the present invention is particularly effective for real-time processing, e.g., transferring live images captured by a camera.

Claims	**Relevant Disclosures in the Prior Art**
Claim 11 (Ground 4)	
	Yano, [0126]. Accordingly, Yano discloses *an adaptive bitrate manager* through transmission rate change unit 1-13 in transmitting terminal 1-1. FIG. 1  Yano, FIG. 1. To the extent not disclosed or rendered obvious by Yano, Ogawa provides further support. Ogawa discloses *an adaptive bitrate manager* by describing a "dynamic rate controller 210," including a "bitrate setter 212," that dynamically adjusts its bitrate based on current network conditions, e.g.: > Then, in step S13, the server distributes streaming data to the client. In this process of transmitting streaming data, the **dynamic rate controller 210** of the server 200 shown in FIG. 2 **dynamically controls the transmission bitrate so that the streaming data will be <u>transmitted at an optimal bitrate</u>**. The process will be described later more specifically. When the transmission of all the streaming data is finished, in step S14, the connection between the server and the client is closed, and the process is then exited. Ogawa, [0102].

Claims	**Relevant Disclosures in the Prior Art**
Claim 11 (Ground 4)	
	When the buffer amount is greater than the predetermined threshold Th2, in step S204, the **bitrate setter 212 compares the current transmission-data bitrate with the maximum throughput calculated by the throughput calculator 211 in the process described above**. When the current transmission-data bitrate is less than the bitrate corresponding to the maximum throughput calculated by the throughput calculator 211, in step S205, **the bitrate setter 212 increases the transmission bitrate**. However, the bitrate corresponding to the maximum throughput calculated by the throughput calculator 211, i.e., a maximum allowable bitrate, serves as an upper limit. When the current transmission-data bitrate is equal to the maximum allowable bitrate corresponding to the maximum throughput calculated by the throughput calculator 211, the process returns to step S201 without changing the bitrate. This occurs when the buffer amount is in a range Bc of the buffer-amount information 250 shown in FIG. 6. Ogawa, [0145]. FIG. 2  Ogawa, FIG. 2.

Claims	Relevant Disclosures in the Prior Art
Claim 11 (Ground 4)	
	FIG. 6  Ogawa, FIG. 6. The teachings of Yano and Ogawa would have been considered together and combined for the reasons discussed in the Request at Section I.E.4.c. Karam Decl., ¶¶ 90-93.
[11c] receive the receiver report,	Yano discloses, or at least renders obvious, *receive the receiver report* for the same reasons as discussed above with respect to limitation [1a] of claim 1 (reciting "receiving a receiver report from a terminal"). *See* limitation [1a] above.
[11d] estimate one or more network conditions using the receiver report,	Yano discloses, or at least renders obvious, *estimate one or more network conditions using the receiver report* for the same reasons as discussed above with respect to limitation [1b] of claim 1 (reciting "estimating one or more network conditions of a media network using the receiver report"). *See* limitation [1b] above.
[11e] determine stability criterion using the estimated one or more network conditions, wherein determine	Yano in view of Ogawa discloses, or at least renders obvious, "*determine stability criterion using the estimated one or more network conditions, wherein determine stability criterion includes at least one of*" for the same reasons as discussed above with respect to limitation [1d] of claim 1 (reciting the same limitation).

Claims	Relevant Disclosures in the Prior Art
Claim 11 (Ground 4)	
stability criterion includes at least one of:	*See* limitation [1d] above.
[11f] comparing a media time in transit and a round trip estimate, and	Yano in view of Ogawa discloses, or at least renders obvious, *comparing a media time in transit and a round trip time estimate* for the same reasons as discussed above with respect to limitation [1e] of claim 1 (reciting the same limitation). *See* limitation [1e] above.
[11g] comparing a bitrate received with a current bitrate session, and	Yano in view of Ogawa discloses, or at least renders obvious, *comparing a bitrate received with a current bitrate session* for the same reasons as discussed above with respect to limitation [1f] of claim 1 (reciting the same limitation). *See* limitation [1f] above.
[11h] determine the stability of the media network,	Yano in view of Ogawa discloses, or at least renders obvious, *determine the stability of the media network* for the same reasons as discussed above with respect to limitation [1g] of claim 1 (reciting essentially the same limitation; limitation [1g] recites "determining" while limitation [11h] recites "determine"). *See* limitation [1g] above.
[11i] determine an optimal session bitrate based at least in part on the media-network-stability determination, and	Yano in view of Ogawa discloses, or at least renders obvious, *determine an optimal session bitrate based at least in part on the media-network-stability determination* for the same reasons as discussed above with respect to limitations [1c] and [1h] of claim 1 (reciting essentially the same thing as limitation [1h]; limitation [1h] recites "providing the optimal…" while limitation [11i] recites "determine the optimal…"). *See* limitations [1c] and [1h] above.
[11j] provide media data to the terminal according to the optimal session bitrate.	Yano discloses, or at least renders obvious, *provide media data to the terminal according to the optimal session bitrate* for the same reasons as discussed above with respect to limitation [1i] of claim 1 (reciting essentially the same limitation; limitation [1i] recites "providing" while limitation [11j] recites "provide"). *See* limitation [1i] above.

Claims	Relevant Disclosures in the Prior Art
Claim 14 (Ground 4)	
[14P] 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:	As to the *non-transitory computer readable storage medium storing instruction that, when executed by a computer, cause the computer to perform a method* . . . portion of this preamble limitation, Yano discloses this because it discloses its invention can be performed by computers and as software on a computer executing instructions: In FIG. 7, reference numeral 10 denotes a camera server, which transfers video data captured by a camera 100 to a client 20 via a network 300. In case of the above embodiment, the camera Server 10 corresponds to the transmitting terminal 1-1, and the client 20 corresponds to the receiving terminal 1-2. The hardware difference between the camera server 10 and client 20 is the presence/absence of the camera and capture unit, and they can be implemented by, e.g.,personal computers. That is, components 103 to 109 and 203 to 209 form substantially the same arrangements, which can be implemented by versatile computers (e.g., personal computers). On the other hand, the software difference is as follows. That is, **<u>software</u>** (stored in an external storage device 106 and **is loaded onto a RAM 105 upon <u>execution</u>**) for transferring the captured video data to the client is running on the camera server 10, and **software** (stored in an external storage device 206 and is **loaded onto a RAM 205 upon execution**) for receiving video data and displaying the received video data is **running on the client 20**. Yano, [0054]-[0056]. As to the "*for processing a receiver report*" portion of this preamble limitation, limitation [1a] above describes how Yano's system receives and processes a receiver report in its system. *See* limitation [1a] above.
[14a] receiving the receiver report from a terminal;	Yano discloses, or at least renders obvious, *receiving a receiver report from a terminal* for the same reasons as discussed above with respect to limitation [1a] of claim 1 (reciting the same limitation).

Claims	Relevant Disclosures in the Prior Art
Claim 14 (Ground 4)	
	See limitation [1a] above.
[14b] estimating one or more network conditions of a media network using the receiver report;	Yano discloses, or at least renders obvious, *estimating one or more network conditions of a media network using the receiver report* for the same reasons as discussed above with respect to limitation [1b] of claim 1 (reciting the same limitation). *See* limitation [1b] above.
[14c] determining stability criterion, wherein determining stability criterion comprises at least one of:	Yano in view of Ogawa discloses, or at least renders obvious, "*determining stability criterion, wherein determining stability criterion comprises at least one of:*" for the same reasons as discussed above with respect to limitation [1d] of claim 1 (reciting "determining stability criterion using the estimated one or more network conditions, wherein determining stability criterion includes at least one of"). *See* limitation [1d] above.
[14d] comparing a media time in transit and a round trip time estimate; and	Yano in view of Ogawa discloses, or at least renders obvious, *comparing a media time in transit and a round trip time estimate* for the same reasons as discussed above with respect to limitation [1e] of claim 1 (reciting the same limitation). *See* limitation [1e] above.
[14e] comparing a bitrate received with a current bitrate session; and	Yano in view of Ogawa discloses, or at least renders obvious, *comparing a bitrate received with a current bitrate session* for the same reasons as discussed above with respect to limitation [1f] of claim 1 (reciting the same limitation). *See* limitation [1f] above.
[14f] determining the stability of the media network using the determined stability criterion;	Yano in view of Ogawa discloses, or at least renders obvious, *determining the stability of the media network using the determined stability criterion* for the same reasons as discussed above with respect to limitation [1g] of claim 1 (reciting "determining the stability of the media network") and limitation [6f] of claim 6 (reciting the same limitation). *See* limitation [1g] and [6f] above.
[14g] controlling a session bitrate based at least in part	This limitation is similar to limitation [1h] – however, while limitation [1h] recites "*providing the optimal session bitrate*

Claims	Relevant Disclosures in the Prior Art
Claim 14 (Ground 4)	
on the media-network-stability determination; and	*based at least in part* on" limitation [14g] recites "*controlling a session bitrate* at least in part on." *See* limitation [1h] above. Yano discloses *controlling a session bitrate* because it describes that: The operation sequence of the third embodiment is also nearly the same as those in the first and second embodiments, except for the sequence (S203 in FIG. 2) for calculating the transmission rate. In the first and second embodiments, **for the constant C used in control for determining the transmission rate, a value that assures stable operation must be empirically found.** However, in the third embodiment, the transmission rate is determined without using such constant. Yano, [0073]. To restate, according to the fourth embodiment, **the transmission rate is controlled to maintain constant the data volume (network buffer data volume) that has been output from the transmitting terminal onto the network but has not reached the receiving terminal yet, in accordance with an increase in data round-trip time. With this control, since the data volume which is en route to the destination on the network can be accurately calculated, and is adjusted, a transmission delay can be suppressed to fall within an allowable range.** Since data is transmitted/received to always save data in a buffer, the available band of the network can be sufficiently used to transmit/receive the data. **Since the data volume buffered in the network is used as a parameter for controlling the transmission rate without directly using the delay time, a sufficient buffer volume which is to exist on the network can be defined as a target buffer data volume, and data can be prevented from being excessively output onto the network**. Yano, [0123]. As described above, according to the fourth embodiment, **the transfer rate is dynamically changed in correspondence with the network condition in data**

Claims	Relevant Disclosures in the Prior Art
Claim 14 (Ground 4)	
	communications via the network, thus realizing optimal data transfer. Hence, the present invention is particularly effective for real-time processing, e.g., transferring live images captured by a camera. Yano, [0126]. Yano also notes that the "**transmitting terminal determines the transmission rate** on the basis of the calculation result, and controls the data transmitter (1-12) to transfer data at that transmission rate." Yano, Abstract. Yano also mentions that "**the transmitting terminal determines the transmission rate on the basis of a receiver report** sent from the receiving terminal as in the first embodiment." Yano, [0081].
[14h] providing the session bitrate to an encoder for transmitting media data according to the provided session bitrate.	Yano in view of Ogawa discloses, or at least renders obvious, *providing the session bitrate to an encoder for transmitting media data according to the provided session bitrate* largely for the same reasons as discussed above with respect to limitation [1i] of claim 1 (reciting the same limitation). *See* limitation [1i] above. In addition, this limitation describes that the *session bitrate* is provided *to an encoder for transmitting media data.* Yano in view of Ogawa discloses this too because Yano describes that "each transmission data consists of the sequence number (information for specifying that data) of that data, transmission time (the time of the transmitting terminal), packet size, **and compressed and encoded image data**." Yano [0143]. *See also* Yano, [0040] ("when video data is to be transmitted, an image is captured by a video camera, and the captured video data is **compressed and coded**").

<table>
<tr><th>Claims</th><th>Relevant Disclosures in the Prior Art</th></tr>
<tr><td colspan="2">Claim 14 (Ground 4)</td></tr>
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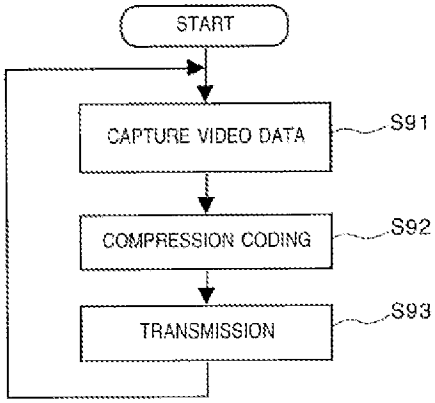
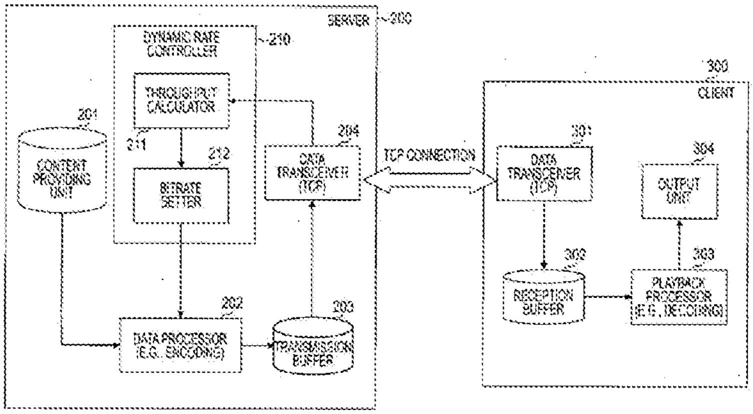
F I G. 9



Yano, FIG. 9.

Further, Ogawa describes "**a bitrate setter configured to set a transmission bitrate within a range of up to a maximum allowable bitrate corresponding to the maximum throughput calculated by the throughput calculator.**" Additionally, Ogawa teaches providing the session bitrate set by the bitrate setter to an encoder for transmitting media data according to the provided session bitrate. *See* Ogawa, Figs. 2, 6, 8, 13-14, [0017], [0097]-[0098].

FIG. 2



Ogawa, FIG. 2; *see also* Ogawa, FIGs. 6, 8, 13-14.

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