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CERTIFICATION AND REQUEST FOR PRIORITIZED EXAMINATION UNDER 37 CFR 1.102(e) (Page 1 of 1)

First Named Inventor:	Erik Van Zijst	Nonprovisional Application Number (if known):	Filed herewith
Title of Invention:	LAYERED MULTICAST AND FAIR BANDWIDTH ALLOCATION AND PACKET PRIORITIZATION		

APPLICANT HEREBY CERTIFIES THE FOLLOWING AND REQUESTS PRIORITIZED EXAMINATION FOR THE ABOVE-IDENTIFIED APPLICATION.

1. The processing fee set forth in 37 CFR 1.17(i)(1) and the prioritized examination fee set forth in 37 CFR 1.17(c) have been filed with the request. The publication fee requirement is met because that fee, set forth in 37 CFR 1.18(d), is currently $0. The basic filing fee, search fee, and examination fee are filed with the request or have been already been paid. I understand that any required excess claims fees or application size fee must be paid for the application.

2. I understand that the application may not contain, or be amended to contain, more than four independent claims, more than thirty total claims, or any multiple dependent claims, and that any request for an extension of time will cause an outstanding Track I request to be dismissed.

3. The applicable box is checked below:

I. ☑ **Original Application (Track One) - Prioritized Examination under § 1.102(e)(1)**

i. (a) The application is an original nonprovisional utility application filed under 35 U.S.C. 111(a). This certification and request is being filed with the utility application via EFS-Web.

---OR---

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ii. An executed inventor's oath or declaration under 37 CFR 1.63 or 37 CFR 1.64 for each inventor, **or** the application data sheet meeting the conditions specified in 37 CFR 1.53(f)(3)(i) is filed with the application.

II. ☐ **Request for Continued Examination - Prioritized Examination under § 1.102(e)(2)**

i. A request for continued examination has been filed with, or prior to, this form.
ii. If the application is a utility application, this certification and request is being filed via EFS-Web.
iii. The application is an original nonprovisional utility application filed under 35 U.S.C. 111(a), or is a national stage entry under 35 U.S.C. 371.
iv. This certification and request is being filed prior to the mailing of a first Office action responsive to the request for continued examination.

...st for continued examination has been gra...ed prioritized examination status ...FR 1.102(e)(2).

Signature /Stephen Y. Liu/	Date January 29, 2016
Name (Print/Typed) Stephen Y. Liu	Practitioner Registration Number 62883

Note: *This form must be signed in accordance with 37 CFR 1.33. See 37 CFR 1.4(d) for signature requirements and certifications. Submit multiple forms if more than one signature is required.**

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Plano, TX 75025

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Signature	Thomas L. Crism Reg. 24,846	Date	11/18/2014
Name	Thomas Crisman	Telephone	214-744-5008
Title	MEMBER & ATTORNEY FOR BLITZ STREAM VIDEO LLC		

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LAYERED MULTICAST AND FAIR BANDWIDTH ALLOCATION AND PACKET PRIORITIZATION

CROSS-REFERENCE TO RELATED APPLICATIONS

[0001] This application is a divisional of pending U.S. Patent Application No. 13/619,969, filed September 14, 2012, which is a continuation of U.S. Patent Application No. 12/486,656, filed June 17, 2009, now U.S. Patent No. 8,514,718, which is a divisional of U.S. Patent Application No. 11/342,167, filed January 26, 2006, now U.S. Patent No. 7,733,868, which claims priority to U.S. Provisional Application No. 60/647,601, filed January 26, 2005, all of which are incorporated by reference as if set forth in full herein.

FIELD OF THE INVENTION

[0002] The invention relates to network management. Specifically, the invention relates to the management of data packets to support multicasting.

BACKGROUND

[0003] Despite the versatility in digital communication, interoperability and internationally accepted communication protocols of the Internet, its fundamental design has not changed much since its conception and does not excel in everything. Watching live TV for example is something which is not typically done over the Internet, even though television has been around almost twice as long as the Internet Protocol and represents a huge market. The reasons for this are based on the design of the Internet and Internet Protocol (IP).

ternet is a packet-switching network where data is exchanged in small units or packets that are independently transported over the network and concatenated again at the receiver into its original form. A strength of packet-switching is that it allows for very flexible use of the physical network wires. When two communicating parties have no data to exchange

for a certain period of time, no packets are sent and the wires can carry packets from other parties. On the Internet, bandwidth is not reserved; but available to and shared by everyone. The consequence is that it cannot guarantee a minimum amount of end-to-end bandwidth, making live video streams often appear jerky because frames are skipped due to congestion that delays or prevents delivery.

[0005] Even though with help from specialized protocols such as Distance Vector Multicase Routing Protocol (DVMRP) or Protocol Independent Multicast (PIM), the Internet Protocol allows for data packets to be multicast to a large number of receivers simultaneously, using this feature to successfully realize a live video broadcast is a challenge. A video stream is transmitted at a fixed high rate and not all parts of the network are likely to have sufficient bandwidth available to forward the stream.

[0006] When a bandwidth bottleneck is reached, the router discards the packets that cannot immediately be forwarded. This causes two problems. The data stream that is eventually received by one or more receivers further down the network is corrupt and the congestion also has a negative impact on communication sessions of other nodes that communicate through the bottleneck router. The only way to avoid this problem using the Internet Protocol and standard multicast is to find a transmission rate that is supported by all parts of the network. However, since the network is available to anyone, this rate will continuously change. A transmission rate is selected and the packet loss is accepted. However, when packets are dropped randomly by

ters the data stream will suffer packet loss. If additional packets are sent through the bottleneck router, there is a larger chance that the router will choose one of these packets when ready to send another packet, implicitly rewarding heavy streams during congestion. This encourages sending redundant data thereby exacerbating the problem.

[0007] A more fundamental problem of flow control using the Internet Protocol is that slowing down the data may not be an option for certain types of live data streams. However, packet loss is unavoidable using the Internet Protocol and while data types such as audio and video data can usually withstand some packet loss without becoming too corrupted to play, this does not apply to all types of live data. Real-time financial data, for example, will become useless and even dangerous to use if random packets of trades are lost.

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BRIEF DESCRIPTION OF THE DRAWINGS

[0008] Embodiments of the invention are illustrated by way of example and not by way of limitation in the figures of the accompanying drawings in which like references indicate similar elements. It should be noted that references to "an" or "one" embodiment in this discussion are not necessarily to the same embodiment, and such references mean at least one.

[0009] **Figure 1** is a diagram of one embodiment of an overlay multicast network.

[0010] **Figure 2** is a diagram of one embodiment of the basic components of a router daemon in the overlay multicast system.

[0011] **Figure 3A** is diagram of one example embodiment of a layered multicast system network divided into logical clusters.

[0012] **Figure 3B** is a diagram of a routing table for the network of **Figure 3A**.

[0013] **Figure 4** is a diagram of an example network with hierarchical structure.

[0014] **Figure 5A** is a diagram demonstrating summarization of an example network with an S2 domain.

[0015] **Figure 5B** is a diagram demonstrating the stretch factor for the example network of **Figure 5A** and shows the inner network of the S2 domain.

[0016] **Figure 6** is a diagram of an example embodiment of a multicast distribution or

[0017] **Figure 7** is a diagram of an overlay multicast network with a failed link.

[0018] **Figure 8A** is a flowchart of one embodiment of a process or managing congestion in the overlay multicast system.

[0019] **Figure 8B** is a flowchart of one embodiment of a process for handling layer repair.

[0020] **Figure 9A** is a diagram of two router daemons connected by a link.

[0021] **Figure 9B** is a diagram of two router daemons connected by a link where a retransmission is requested.

[0022] **Figure 10** is a diagram that shows the process of selective packet repair and ordering.

[0023] **Figure 11A** is a diagram showing thinning of a stream of data.

[0024] **Figure 11B** is a diagram of a multi-level token ring structure.

[0025] **Figure 11C** is a diagram of a multi-level token ring structure showing thinning of data.

DETAILED DESCRIPTION

[0026] To provide multicast services a network needs to support one-to-many communication that can send data packets from a data source to more than one receiver, ideally without putting extra stress on the network or source when the number of receivers increases. Multicast routing can be offered by different methods. One method is to let receivers tell the network, but not necessarily the source, which data streams they want to receive and let the network compute data distribution paths that deliver just the right packets to each receiver. Multicasting can also be done by letting the source encode the list of receivers in each data packet, thereby freeing the network from the potentially computationally intensive task of maintaining multicast distribution paths. However, this method does not scale to handle a large number of receivers. A third method relies on logic at the receivers by letting the network apply a broadcast mechanism whereby each packet is delivered to every connected node and letting the receivers filter out only those packets that are interesting. This method may also generate a heavy load on a larger network but it is simple.

[0027] In one embodiment, a multicast network is constructed as an overlay network. In one embodiment, the overlay network includes a number of software implemented routers connected by normal IP or TCP connections. A mesh is created in which every software router is connected to one or more other software routers by means of virtual connections that appear to be direct connections to the other software routers, but are likely implemented by a number of intervening routers situated between the software routers that are connected this way are adjacent in the perspective of the overlay, but in reality are many physical hops away from one another. Also, a software router that has three software router neighbors has three

independent virtual links. However, it is possible that this software router only has a single physical network connection that is shared by the three virtual links.

[0028] To the underlying network, the overlay network is nothing more than a collection of applications that send data between static pairs. Beyond each router pair (identifiable by their TCP connection) there is no relation between the individual daemons. As such, the overlay network can easily work through firewalls, NAT (IP masquerading), proxies and VPN's. Firewalls cannot control which software router POPs can talk to each other. The system shares some principals with a HTTP proxy server tunneling traffic to and from web browsers. In an example HTTP proxy server system, an intranet has two web browser machines. They both browse the Internet using the proxy server also on the local intranet. Although the individual web browser machines can surf the net individually, they never make a direct connection with a remote webserver, but only with the local proxy server that acts like a relay point. As such, the firewall that sits between the proxy server and the remote webserver can only choose to either allow the proxy server to talk to the webserver or deny it. It is generally unable to enforce unique policies for individual browser machines. It generally cannot tell on behalf of which client the proxy server is fetching a webpage. Returning to the overlay network, because each software router is a relay point that tunnels data traffic for different senders and receivers similar to a HTTP proxy server, firewalls have no fine-grained control over communication over the overlay network. As soon as a firewall allows for only a single TCP connection between an internal and [illegible], all software routers connected to the internal one can talk to all POPs connected to the external one, and vice versa, without restriction.

[0029] In one embodiment, the overlay multicast routing system also manages flow control and timely delivery. Non interactive live data streams do not actively anticipate network

congestion. To manage congestion the network manages the available bandwidth to allow for fair or equal division among the streams. Without management, high volume streams are assigned a larger capacity percentage on overloaded links resulting in little benefit in keeping the bandwidth requirements of a stream low, as the packet loss percentage is determined by how much the network is overloaded by all streams combined and not by the requirements of the individual streams. An alternative to letting the network handle the flow control and congestion is to put the responsibility at the source and receivers. However, letting data streams anticipate network conditions requires a form of feedback information from the network or the receivers. In this case it is beneficial that the amount of feedback does not grow linearly with the size of the audience, as that would reduce the scalability of the multicast. Even when a scalable form of feedback information can be realized and the data stream adapts its transmission rate according to the network conditions, the problem remains that live streams loose their value when they are slowed down and delivered late.

[0030] In one embodiment, an overlay multicast system also implements or manages delivery of packets including an option for guaranteed delivery. It would be ideal if every receiver would receive the data stream without loss or corruption. However, when the content is 'live' and cannot be slowed down, but the network has insufficient capacity, packet loss is difficult to avoid. In fact, even when there is sufficient bandwidth at all times, store-and-forward packet-switched networks are not able to guarantee the delivery of all packets. For example, when a router crashes, all packets in its buffers may be irrecoverably lost. If a network uses dynamic routing, packets are also dropped when routing policies change or packets are trapped in an occasional, temporary routing loop. In cases where packets are 'accidentally' lost, an end-to-end mechanism of retransmissions can be applied that can compensate for the loss. However,

since this requires a form of feedback information, it is beneficial for reasons of scalability that the overhead involved with retransmissions is not linearly related to the size of the audience.

[0031] End-to-end retransmission feedback may be avoided in at least two ways. First, it is possible to let the network components keep a copy of the most recently forwarded packets and let them participate in retransmissions by intercepting the retransmission requests and servicing them locally. This approach often utilizes greater storage and increased processing power requirements at the network components.

[0032] The second alternative to end-to-end retransmission requests is that of encoding redundant information in the data packets. If enough redundancy is encoded, a lost packet's content may be entirely recovered from the extra information in the other packets. The downside of this system is that it comes with a constant level of bandwidth overhead that is related to the level of packet loss tolerance, regardless of whether packets are actually lost. Each of these approaches to packet loss fail in the case of a live data stream that produces more bytes than the network can forward. When local or end-to-end retransmission requests are used, the problem may even be increased as the retransmission requests use extra bandwidth, causing more data packets to be lost.

[0033] Embodiments of the overlay multicast system are designed to offer an end-to-end solution to efficiently multicast live data including stock market data to any machine connected to the Internet. The system is capable of giving certain hard g[illegible] what is delivered to [illegible] packets must be dropped due to congestion or other irrecoverable problems, it is done in a fully deterministic way that does not corrupt the data. Where a receiver of data \ such as a viewer of a film may accept the random loss of one or two video frames, this type of data loss may wreak havoc in financial data when the missed packet contains an important trade. The

system supports deterministically delivery designated parts of a data stream when the network lacks sufficient capacity. The system utilizes a layered multicast with data streams subdivided into individual streams or layers. This allows receivers to subscribe to only those layers that the network can handle, so that random packet loss can largely be avoided.

[0034] In another embodiment, an enhanced form of layered multicast is used that guarantees complete delivery for certain layers to avoid random loss altogether, making it suitable for certain types of critical data such as market data. The system is characterized as controlling two primary activities. The first activity is running and managing a robust and scalable overlay network that uses its own routing algorithms and supports multicast, while the second activity is managing flow control and congestion when five streams overload the network and ensuring layered multicast can be offered with guarantees.

[0035] **Figure 1** is a diagram of one example embodiment of an overlay multicast system. In one embodiment, the system includes an overlay network 100 that includes a number of software router daemons 109, 115, that are interconnected by normal TCP connections 119 or similar reliable communication connections. The overlay multicast system forms an intricate web of routers and virtual links similar to the Internet, but operating on the application layer. This system operating at the application layer 'overlays' the physical network and other layers of the Internet providing its own system of routing, controlling the lower levels of the network. Any number or routers and client applications may be a part of the system. In the example, two of the 15 are in communication with local client applications 101, 111.

[0036] In one embodiment, each of the routers have a unique, string-based addresses 117. In another embodiment, the routers may have other types of unique addresses such as numerical addresses. Each of the router daemons 109, 115 executes a routing algorithm that computes the

shortest paths that allows each router to send data packets to any other router in the overlay multicast system as well as any machine in communication with these routers.

[0037] In one embodiment, the system includes a runtime client library 103, 113, application programming interface (API) or similar system for sending and receiving packets that can be utilized by user applications to communicate over the system. This library connects an application to a nearby system router daemon 109, 115 through a TCP connection 107, 121, native inter-process communication or similar communication method. The native inter-process communication method may be used if both the router and the client application run on the same physical machine. When connected to a router daemon, the client application 101, 111 can send and receive data packets from the network under the router's unique address. Router daemons have a unique address, while user applications connected to a router daemon are identified through logical ports.

[0038] In one embodiment, the topology of the overlay network may be configured at router daemon startup. The topology may be relatively static after configuration. In one embodiment, the relationship of router daemons to applications may be one to many, with a single router daemon serving multiple applications. In another embodiment, the relationship may be many to many.

[0039] In one embodiment, the .data packets in the overlay network may be addressed to a single communication endpoint or to abstract multicast addre[illegible]munication endpoints via network address used by user applications. Access to them is exclusive such that only one user application can use them for sending and receiving packets at a time.

[0040] In one embodiment, a unique network address in the overlay network that is used by an application is the combination of a logical node address assigned to the overlay router and the

port name chosen by the application. If an application that is connected to a overlay router with the logical node address "n1.mxx.1" wants to use a port named "myport" for receiving unicast data packets, the fully qualified network address becomes "n1.mxx.1:myport." Other applications connected to the overlay network that want to send packets to this application can use this as a destination address. Ports may be referred to as *tports*. Overlay network addresses must be bound, prior to sending or receiving packets. When a data packet is sent from an endpoint address it will contain this information as its source, allowing other routers or the receiving application to send a response.

[0041] In one embodiment, the overlay network may support packets being sent to abstract multipoint destinations, or multicast addresses. An overlay network multicast address is a destination that each machine on the network may subscribe to. The overlay network software routers ensure that a copy of each data packet published to this multicast address is delivered to every user application that subscribed to it. In one embodiment, multipoint destinations are single sourced. Only one application can publish data packets to the multicast address, making the overlay network suitable for one to many, but not many to many communications. Because a tport session of a multicast address can be freely chosen, each multicast address explicitly contains the location of the source on the overlay network. This makes multicast communication less flexible because publishing is not anonymous, but it greatly simplifies subscription management. In another embodiment, multiple applications may publish to a multicast address

,,,,,/ communication is supported.

[0042] In one embodiment, both unicast and multicast addresses are syntactically equal. For example, the address "n1.mxs .1:timeserice" could be a unicast address used and bound by an application that provides the local date and time in response to any data packet it receives.

However, it could also be a multicast address that anyone can subscribe to. For example, subscription to the address may provide periodic date and time broadcasts from the user application that bound this address as a multicast address. In one embodiment, each packet may contain a flag or similar indicator that indicates whether its destination address should be interpreted as a unicast or multicast address.

[0043] In one embodiment, the overlay network provides a set of higher layered protocols as services over the packet oriented base layer. These services may be referred to as protocol endpoints. Any number of protocol endpoints may be defined and supported. In one embodiment, five protocol endpoints may be supported. A unreliable unicast protocol (UUP), a reliable unicast protocol (RUP), unreliable multicast protocol (UMP), ordered layered multicast protocol (OLMP) and reliable ordered layered multicast protocol (ROLMP) may be supported. The UUP offers a best effort unicast datagram service to applications. The RUP offers reliable unicast communication between peers. The UMP offers best effort multicast datagram services to applications on the overlay network. The OLMP offers multicast communication with receiver driven rate control. Complete delivery is not guaranteed, but the packets that are received are guaranteed to be in their original order. The ROLMP offers reliable multicast communication with receiver driven rate control. Stream layering allows each subscriber to receive the data stream in the highest possible quality, while the source never has to slow down.

[0044] **Figure 2** is a diagram of one embodiment of the basic components of a router overlay multicast system. Each overlay network router 109 includes a packet switching core or kernel 201. In one embodiment, packets that are received, either from a connection to a neighbor router or from a connected application, pass through the kernel 201. The kernel 201 may not handled some specialized control packets. The task of the kernel 201 is

to inspect the destination of each packet and use a set of routing tables to determine how to forward the packet.

[0045] In one embodiment, the kernel 201 forwards packets to neighbors through a set of interface modules 203, 205. In one embodiment, the kernel 201 may execute on its own thread and be event driven. The kernel 201 remains idle until it is notified by an interface module 203, 205 of an incoming packet or by an application 227 that is sending a packet. The kernel thread is woken up, reads the packet from the interface module or application and processes it. If the kernel decides that the packet must be sent out through an interface, it passes it to that interface and waits for the next notification.

[0046] In one embodiment, each router daemon in the network is connected to one or more neighbors. This is done by establishing connections between the routers. In one embodiment, the connections may be long lived TCP connections or similar communication connections. In one embodiment, a router daemon 109 runs one interface module 203, 205 instance for each configured neighbor or configured communication channel. In another embodiment, the router 109 may run multiple interface modules for a configured neighbor or communication channel or a single interface module for multiple configured neighbors or communication channels. The responsibility of an interface module 203, 205 is to establish a connection, e.g., a TCP connection, and to pass packets from the kernel 201 to the connection and vice versa. Packets of the first kind are referred to as outbound packets, while the latter are inbound packets.

In one embodiment, the kernel 201 maintains a unicast routing table that is used for packet switching. To make it possible for the network to find shortest paths as well as adjusting these paths when the underlying physical network's characteristics change, each interface module may measure the quality of its virtual connection. These measurements are passed on to the

kernel 201 when the link quality is found to have changed. Inside the kernel 201, the measurements may be fed to a routing algorithm to determine if the changed link quality alters any of the entries in the routing table. If the routing table is changed, it is advertised to the neighbors by encoding the entire table or a portion of the table in a data packet and passing it to each interface. In one embodiment, if this type of packet is received from a neighbor and propagated to the kernel 201 through the receiving interface, the kernel inspects the message and analyzes it through the routing algorithm. If the new information leads to changes in routing entries, the router sends its own related routing entries to all its neighbors.

[0048] In one embodiment, if a neighbor router crashes, the interface detects this through an error on the virtual link and passes an indicator to the routing algorithm. The routing algorithm then changes the cost indication associated with the link to an infinite value or similar value indicating that the link should not be utilized. In one embodiment, the kernel 201 does not distinguish between a crashed neighbor and an infinitely slow link. The kernel 201 only detects that the link is not to be utilized when reading the entry in the routing table.

[0049] In one example, when a source application connected to router S wants to publish a live data stream to multicast group s:mygroup, where applications on node p and *q* want to receive it, the source may first bind the group in publish-mode . Binding a multicast group in publish-mode means that the group is owned by the binding process. Only the owner of the group is able to publish to it. The receiving or "sink" applications connected to routers p and q

ip s:mygroup in subscribe-mode. Binding a multicast group in subscribe-mode results in the router node being connected to a virtual multicast distribution tree. The subscribers receive a copy of any packet published by the source.

[0050] In one embodiment, multicast groups do not necessarily need to be bound in publish-mode first. Any application may subscribe to any multicast group at any time. If there is no source, there will be no data to receive. Binding a multicast group either in publish-mode or subscribe-mode are distinct operations. When the source also wants to receive a copy of its own stream, it binds its group in subscribe-mode and uses the resulting tsocket to read the data back. Data packets carry both the address of their source application, as well as the network address of their destination.

[0051] For unicast packets, the router uses its unicast routing table to find the preferred next hop, while multicast packets are routed according to a subscription list in a multicast subscription table. In one embodiment, a uncast data packet contains the node address of the source router, the tport of the source application, the address of the destination router and the tport of the destination application. A multicast data packet contains the multicast group it was published to, represented by the node address of the source router and the group identifier that was bound by the source application. It does not contain any other addressing information.

[0052] In one embodiment, the overlay multicast system determines unicast routing tables and multicast subscription tables for each router. The system utilizes any type of routing algorithms or protocols to determine routing tables. Protocols that are utilized include distance-vector protocols and link-state protocols. Link-state protocols take the relatively simple approach of propagating the state of the entire network as a list of all known links in the network with their

-vector protocols are based on the Bellman-Ford protocol. They work by letting every router periodically advertise its own routing table to its neighbors. In one embodiment, the Extended Bellman-Ford protocol, hereafter referred to as ExBF, is used as the basis of the overlay multicast network. For sake of convenience, the embodiments of overlay multicast

system are described that utilize the ExBF, however, those of ordinary skill in the art would understand that other routing algorithms may also be utilized.

[0053] In the ExBF protocol, a slight increase in the amount of information is kept for every destination. Instead of storing just the distance to each destination for every neighbor, ExBF also stores the address of the pre-final node of each path. Hence, instead of storing the collection of distances *{Di(j)}* where j represents a destination, D the distance between this node and *j*, while *i* ranges over the node's neighbors, each router also stores {Pi(j)} where P is the pre-final, or so-called 'head-of-path' node in the shortest path to destination *j*. Now because the router knows the pre-final node of every destination, it can backtrack the full path to any destination by recursively looking at the head-of-path of a destination and treating it as a new destination.

[0054] In one embodiment, the interface modules 203, 205 implement a simple algorithm to establish the communication connection to the neighbor machine or device. The interface module 203, 205 attempts to connect to the configured neighbor router by actively trying to connect to its network address, e.g., its TCP/IP network address. If the connection is aborted with a connection refused or other error, it is assumed that the neighbor is not yet running and the interface module 203, 205 starts to listen on its configured IP port so that the neighbor can connect to it when it is started up. The interface module 203, 205 waits for an incoming connection request for a brief period of time. After this time period expires, the interface module 203, 205 returns to actively connecting to its neighbor. To avoid a situation where both

time to switch between the listen and active states at the same time, the duration of the listening state is influenced by a random factor. An advantage of allowing each neighbor to switch between the active connection and listening states when establishing a connection is that it allows the connection of routers even if one of the routers is on a masquerade IP network. A

router on a masqueraded IP network is unable to accept incoming connections such as TCP connections.

[0055] In one embodiment, interface modules 203, 205 can be configured at router deployment through configuration files. This will make the router daemon automatically create the specified interfaces at startup. If the configured neighbor routers are online, all long-lived connections will automatically be established. It is also possible to add new neighbor connections and interface module instances dynamically at runtime. This way the overlay network topology can be changed flexibly and new routers can be added to the network. If an interface module 203, 205 manages to establish a connection with a neighbor router, the interface modules of the routers exchange overlay network node address information to inform each other of their presence. When the interface module 203, 205 receives the node address of its neighbor, it passes this information, together with an announcement that a connection has been made to the kernel 201. This information allows the kernel routing algorithms to build dynamic routing tables.

[0056] In one embodiment, a role of the interface modules 203, 205 is to establish the connection with a configured neighbor router and to send data packets from the router kernel 201 to the neighbor connection and vice versa. The interface module 203, 205 incorporates a framework that allows custom software plug-ins to influence the stream of packets that flows between the network and the kernel 201. This mechanism is referred to as the interceptor 257. In one embodiment, each software plug in component that needs to control the packet flow is a class that implements a simple programming interface. In another embodiment, the software plug ins may have any implementation structure including objected oriented structures. This interface allows interceptor instances to be chained together, forming a

packet processing pipeline. The contract of an interceptor is that it receives a packet, applies it operations and then passes the modified packet to the next interceptor in the chain.

[0057] In one embodiment, an interceptor pipeline 251, 257 sits between the router switching core and the network connection. When the router kernel 201 delivers a packet to the interface module 203, 205 for transmission to the neighbor router, the interface module runs the packet through the interceptor pipeline, giving the interceptors the chance to modify the packet. Each packet that comes out of the pipeline is transmitted to the neighbor router.

[0058] In one embodiment, each interface module 203, 205 has two interceptor pipelines 251, 257. The first interceptor pipeline 257 is used to process outbound packets. The second interceptor pipeline 251 is used for inbound packets. These pipelines are independent of one another, the ordering of the interceptors and the number of processing steps can be different for inbound and outbound packets. Each pipeline can be configured uniquely. Interceptor pipelines may have any number of interceptors, provided they do not add too much latency.

[0059] In one embodiment, an example of an interceptor is one that filters packets from a specific overlay network router. When this interceptor is implemented as a manageable component that can be configured dynamically at runtime, it can be used to implement basic firewall functionality on the overlay network. If it receives a packet that matches its rejection pattern, it discards the packet by not passing it to the next interceptor in the pipeline. Another type of interceptor that may be used is a traffic monitor that counts the size of each packet that passes by and uses this to log traffic activity and compute bandwidth statistics. This plug in mechanism allows an overlay network router to be extended with additional functionality without modifications to the underlying software.

[0060] In one embodiment, the interceptor pipelines 251, 257 act as a packet buffer between the router kernel 201 and the network. The interface modules 203, 205 temporarily buffer inbound and outbound packets. This ensures that the kernel 201 is not blocked when sending a packet to an interface. In one embodiment, the interface modules have a separate thread or set of threads that continuously dequeue and serialize packets from the buffer and write them to the communication connection or enqueue into a buffer received packets.

[0061] In one embodiment, the interceptor pipeline provides a temporary packet buffer and offers inter-thread communication between the kernel thread and the interface thread. Interceptors may be divided into two categories: interceptors that block and interceptors that return immediately. The first category is referred to as blocking or synchronous interceptors. In one embodiment, to avoid situations where a router kernel 201 is blocked for an arbitrarily long time, an interceptor pipeline may contain one non-blocking interceptor. A non-blocking interceptor guarantees immediate return of control to a caller by storing the packet in an internal buffer. The packets in the buffer may be discarded if the buffer exceeds a certain threshold size.

[0062] In one embodiment, a maximum size limit is placed on the packet buffers to prevent them from exhausting the router's memory. Storing packets before processing them means that their delivery will be delayed. The larger the buffer gets, the longer the packets are delayed. Because of this, the interceptor drops packets when the buffer reaches its maximum size. In one embodiment, the system uses reliable TCP or similar connections for transmitting packets

rs in which packets are only dropped inside the buffer interceptors of the interface modules.

[0063] In one embodiment, the interceptor plays a role in making packet loss deterministic. Packets are not explicitly dropped in any other part of the layered multicast system network.

except the buffer interceptor in the interface pipelines. However, that may not guarantee that a packet that successfully makes it through all buffer interceptors of the network's routers is delivered at its destination. Aside from controllable packet loss, the network may also occasionally lose packets in a non-deterministic way, for example when a router crashes with pending packets in its buffers, or when a connection between adjacent routers is unexpectedly closed during transmission of a packet.

[0064] In one example, an inbound interceptor pipeline 251 may be structured such that traffic throughput monitor interceptors 253 are positioned after a buffer interceptor 255 and a firewall interceptor 265 may be positioned before for the buffer interceptor 255. In an example outbound interceptor pipeline 257, a throughput limiting interceptor 259 may be positioned before a buffer interceptor 261 and a traffic monitor interceptor 267 may be positioned after the buffer interceptor 261.

[0065] In one embodiment, the overlay network is accessible to applications through the use of the client-side programming library 221. This library connects to a router daemon 109 at application startup and communicates with it using remote procedure protocol or similar communication protocols. The communication between the library 221 and the router daemon 109 is driven by the application 227. The application 227 invokes functions in the router 109 through the library 221. In one embodiment, when the kernel 201 receives packets addressed to a tport that is bound by the application 227, it stores them until the client actively picks them up.

n 227 through the library 221 continuously polls the router for packets. To minimize the overhead of the polling mechanism, the router poll function does not return until there is at least one packet delivered by the kernel 201. If more than one packet is waiting, the poll function returns all waiting packets at the time it is invoked. In another embodiment, the

router may send an indication such as an invoking a marshaled stub from the client, event notification or similar indicator to the application 227 through the library 221 to indicate the reception of a data packet for the application 227.

[0066] In one embodiment, a user application 227 uses a client library 221 through instances of the overlay multicast system communication sockets 225. If a user wants to be able to receive packets, they reserve a network address. In one embodiment, a network address in the overlay multicast system is represented as a combination of the address of the router and a unique port identifier, reserved for the socket of the router. For sake of convenience, a port in the layered multicast system will be referred to as a tport and a socket as a *tsocket*. When a user application creates a tsocket for receiving normal unicast packets, the tsocket automatically binds a local tport at the router daemon through a remote call to the client IO multiplexor 207 at the router. In one embodiment tports are bound exclusively and other clients may not use it concurrently.

[0067] In one example embodiment, communication sockets 225 communicate with a local input output (IO) multiplexor 229 that coordinates the handling of communication between the sockets and the router kernel through the formation of RPC calls, native inter-process communication or similar systems. A local IO multiplexor 229 utilizes an RPC stub 223 or similar program to communicate with the router via an RPC skeleton 219 and client adaptor 209. A client IO multiplexor 207 at the router manages the relay of these socket requests to the kernel 201.

[0068] In one embodiment, the router daemon 109 uses a packet buffer to temporarily store packets for each connected client application until they are picked up. An interceptor pipeline in the client adaptor 209 or similar process may be utilized for this buffering function.

[0069] To overcome the problem of growing routing tables, computation time and excessive advertisement overhead, large networks can be partitioned into smaller sub sections, connected by gateways. While the gateway routers maintain routing information necessary to route packets to nodes in other network sections, nodes inside a section or domain only maintain information for those nodes inside the same domain. By substituting logical ranges of hosts in the routing table by one single condensed entry, the size of the routing table is reduced. This is called address summarization. This mechanism introduces a form of hierarchy that allows the network as a whole to grow far beyond the practical limits of standard distance- vector or link-state algorithms. The farther a destination host is away, the more efficient it can be condensed together with other remote hosts. The more summarization is applied, though, the less efficient the paths become on average. The factor by which the actual data paths on summarized networks differ from the optimal paths, is known as the *stretch-factor*: the maximum ratio between the length of a route computed by the routing algorithm and that of a shortest path connecting the same pair of nodes.

[0070] In one embodiment, the overlay multicast system takes a relatively straightforward approach to address summarization. An administrator decides at deploy time which nodes form clusters and which clusters form aggregations of clusters. This is done by encoding hierarchy in the layered multicast system node addresses using a dotted or similar notation. Node addresses may be ASCII strings. In one embodiment, the strings are limited to at most 127 characters. In [illegible], only [a-z] and [0-9] are available. In another embodiment any characters, numbers of similar symbols may be utilized. Addressing may be case-sensitive or case-insensitive.

[0071] **Figure 3A** is diagram of one example embodiment of a layered multicast system network divided into logical clusters. The example illustrates a network of eight nodes, divided into three clusters. Assigning nodes to clusters may be based on geographical properties, administrative boundaries, wide area links and similar considerations. For example, nodes inside a corporate network are all assigned the same logical domain, whereas a network that connects nodes from different corporations, would usually assign a separate domain to each corporate network. Another criterion for assigning nodes is that nodes that often disconnect and reconnect again later, possibly because the overlay multicast system routers run on personal computers, are placed in a subdomain, to avoid the routing updates triggered by their changing state to propagate far into the network.

[0072] Given the logical domains that cluster groups of nearby nodes, each node can treat domains other than its own as a single entity and use a wildcard address that matches every host inside that domain. This reduces the size of the routing table, as well as the amount of routing information that needs to be exchanged between the nodes when the topology changes inside a domain. For example, when a new host is added to domain S1, there is no need to propagate that information to the other domains, as they already have a wildcard entry that will match the new address.

[0073] **Figure 3B** is a diagram of a routing table for the network of **Figure 3A**. The example routing table shows the effect of address summarization in this network on the routing

S2.C. The cost value that is listed in the third column represents the cost of the shortest path to the nearest node inside that domain. In the fourth routing entry, the number in the cost column is the cost to reach S1.R from S2.C. If it is assumed that both interdomain links S2.

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C-S1.R and S2.A-S1.R have the same weight or cost, S2.C will route all traffic for domain S1 through neighbor S1.R.

[0074] In one embodiment, when nodes exchange distance vectors, either because of a link change, or as a normal, periodic exchange, the receiving node first summarizes the destination address of each distance vector relative to its own address. Summarization is done by matching the destination address with its own address field- by- field and when a field does not match, the rest of the address is substituted by a wildcard. For example, when S2.C receives a distance-vector from S1.R that contains a path to destination S1.P, it is immediately summarized to S1.* upon arrival at node S2.C. This is because the first field differs from the first field of the local address and as such the remaining fields are replaced by a wildcard. This wildcard value is then fed to the distance vector algorithm that checks whether the new cost or path-length is shorter than the cost of the entry that was already in the routing table. In the present example there already is an S1.* wildcard entry in the routing table that was derived from neighbor destination S1.R. Since the path to S1.P runs through S1.R, the path to S1.R will always be shorter than the path to S1.P, so the entry in the routing table will not be changed and no further routing updates will be propagated to S2.C's neighbors.

[0075] In general, when a local address is 1.2.3.4 and an incoming distance-vector advertises destination 1.2.2.2, it will be summarized to 1.2.2.*. Destination 2.1.6 becomes 2.*, destination 1.2.3.4.5.6 becomes 1.2.3.4.5.*, destination 1.2.3.5 stays 1.2.3.5 and destination stays 1.2.3.4.5. Note that 1.2.3 and 1.2.3.* are two different addresses. The first only matches the exact address 1.2.3 while the second is a wildcard that matches everything that starts with 1.2.3 and has at least 4 address fields. This includes 1.2.3.4 and 1.2.3.4.5.6, but not 1.2.3. If this mechanism of address summarization is used in an ExBF implementation that

carries tuples containing destination, cost and head-of-path attributes in its vectors, then the destination address is summarized, as well as the head-of-path address.

[0076] **Figure 4** is a diagram of an example network with a hierarchical structure. When the example network first starts to converge using the ExBF algorithm, node B.X advertises the following routing information to neighbor A.S: $DV_{B.X,A.S}$: {(B.X,*,0), (B.Y, B.X, 1), (B.Z, B.X,1)}. When A.S receives the vectors, it summarizes its entries. What remains is the single vector DVax.As: {(B.*,*,0)}. The routing table of node A.S now contains $RT_{A.S}$: {(A.S, *, *, 0), (A.R, A.R, A.S, 1), (B.*, B.*, A.S, 1)}. All addresses are summarized before processing. This includes neighbor addresses. The consequence of this is that when a node has more than one connection with a foreign domain, both neighbor addresses will be summarized into the same wildcard. This leads to ambiguities and nondeterministic routing when this wildcard is listed as the preferred hop in a routing entry, as -it cannot identify a single outgoing link . This problem is solved by assigning a local identifier to each link and using these numbers in the preferred hop column, rather than the addresses of the links peers. The routing update A. S sends to A.S.Y contains $DV_{A.S,A.S.y}$: { (A.S, *, 0), (A.R, A.S, 1), (B.*, A.S, 1)} and leads to A.S.Y's routing table $RT_{A.S.v}$: {(A.S.Y, *, *, 0), (A.S.X, A.S.X, A.S.Y, 1), (A.S.Z, A.S.Z, A.S.Y, 1), (A.S, A.S, A.S.Y, 1), (A.R, A.S, A.S, 2), (B.*, A.S, A.S, 2)}. When A.S.Y has finished updating its routing table, it advertises $DV_{A.s.v,A.S}$: {(A.S.Y, *, 0), (A.S.X, A.S.Y, 1), (A.S.Z, A.S.Y, 1), (B.*, *, *), (A.S, *, *), (A.R, *, *)}back to neighbor .A. S where the asterisk indicates unreachable in the last three records to avoid long-lived loops. These loops were detected through the normal back trace mechanism of ExBF.

[0077] In one embodiment, summarization means that only a single path for a range of remote nodes is maintained. The consequence of this is that packets will not always be routed

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according to the real shortest path between source and destination. To illustrate this effect in the multicast network, consider the example network of **Figure 4.** Node S1.R receives the distance-vectors of both S2.A and S2.C. And after summarization learns that bo th neighbors offer a route for wildcard S2.*. If it is assumed that both interdomain links (S1.R - S2.A and S1.R- S2.C) have equals costs, then S1.R will choose S2.A to be the preferred hop for S2.* because of the fact that its address logically comes first. When S1.R needs to forward a data packet to S2.B, it uses the S2.* wildcard entry and forwards the packet to neighbor S2.A. Unfortunately this is not the shortest path to S2.B, as S2. A first has to route the packet through S2.C. Instead, S1.R should have sent the packet directly through neighbor S2.C. This ratio between the length of the actual path and the length of the optimal path between the endpoints are called the stretch factor.

[0078] In one embodiment, path stretching on the overlay multicas.t system occurs when packets are forwarded between nodes that are in different logical domains. An entire subnet is treated as a single node with several outgoing links. Because a virtual node often contains a large number of nodes, connected by its own internal network structure, it is sometimes better to choose a different interdomain link when sending packets to the domain.

[0079] Since the overlay network runs its own adaptive routing algorithms, the content streams through the network are constantly rerouted to avoid the network's hot spots and congestion. This can be particularly useful on wide area networks that are used for very different types of applications and hot spots are dynamic (i.e., moving around). Another advantage of

i routing algorithms is. The freedom of substituting them with others in the future.

[0080] In one embodiment, the software routers implement load balancing inside the network. Traditionally routing algorithms seek for "best paths" through the network and send data streams over these. However, it is sometimes much more desriable not to just send streams

over this optimal path, but to also select several sub-optimal paths and divide the stream over all of them. This also avoids the optimal path from getting congested when the stream requires more bandwidth than this single path can provide.

[0081] **Figure 5A** is a diagram demonstrating summarization of an example network with an 82 domain. When node S1.P needs to forward a packet for destination 82.F, it will send the packet directly down its interdomain link to the 82 domain. In this example both interdomain links are assumed to have equal costs. **Figure 5B** is a diagram demonstrating the stretch factor for the example network of **Figure 5A** and shows the inner network of the 82 domain. **Figure 5B** clearly shows that the stretch-factor is quite high for packets from S1.P to 82.F, as the optimal path runs through S1.R instead. Although the overlay multicast system address summarization technique can not guarantee a maximum upper bound on the stretch-factor, it can manipulate the stretch-factor by changing the summarization opacity.

[0082] By default, an address is summarized after the first field that differs from the local address. However, if that is changed to the second field, the overlay multicast system can look inside other domains for one level. A node with an address 1.2.3.4 will then summarize 2.3.4.5 into 2.3.*, rather than 2.* and 1.3.4.5 into 1.3.4.*, rather than 1.3.*. Doing this at least at the border nodes in the overlay multicast network that have the interdomain links reduces the stretch factor under certain circumstances. As the overlay multicast network was designed to be administrated by independent parties, administrators are free to experiment with different opacity levels without jeopardizing the other subnets or domains.

[0083] One advantage of the summarization of addresses in routing tables is that the number of links that are present between nodes from different logical domains is irrelevant with respect to the number of routing entries in the routing table of a distance-vector protocol or similar

protocols. Since these tables only contain destinations with a single forwarding strategy, the number of interdomain links does affect the size of the routing tables. If y is identified to be the number of entries in a node's routing table, X to be the total number of nodes in the entire network, n to be the number of entities (nodes or nested domains) inside a domain and m to be the depth of the hierarchy. From this it follows that the total number of nodes in the network can be calculated with:

$$\forall\, x = n^m \text{ where } n \in \mathbb{N} \wedge n > 1 \text{ and } m \in \mathbb{N} \wedge m > 0$$

[0084] The relation between domain depth, number of nodes and -domain density can be expressed by the following formula for a network with a uniform topology:

$$y = \left(\frac{\ln(x)}{\ln(n)}\right) * (n-1) + 1$$

[0085] Given this formula of the address summarization's effectiveness, it can be shown that routing entries in any network node remains under 100 even when the network as a whole grows to well over 10 million nodes. It can also be shown that the domain size n has no spectacular effect on the scalability. While a small value n yields a relatively deeply nested network hierarchy, which implies more routing entries, it also means that each -hierarchy level only contains a small number of entities. A large value *n* in a network of the same size yields a relatively flat hierarchy with few levels, each level contains a large number of entities. As a routing table can only contain a natural number of entries, its number remains constant as nodes are added to domains that. Are already captured by the existing wildcard entries.

[0086] An advantage of using a string based node addresses scheme is its large address space and flexible naming. New subdomains may be added at any time as there is no need for predefined or fixed hierarchical levels. All nodes inside a flat domain can be reconfigured into

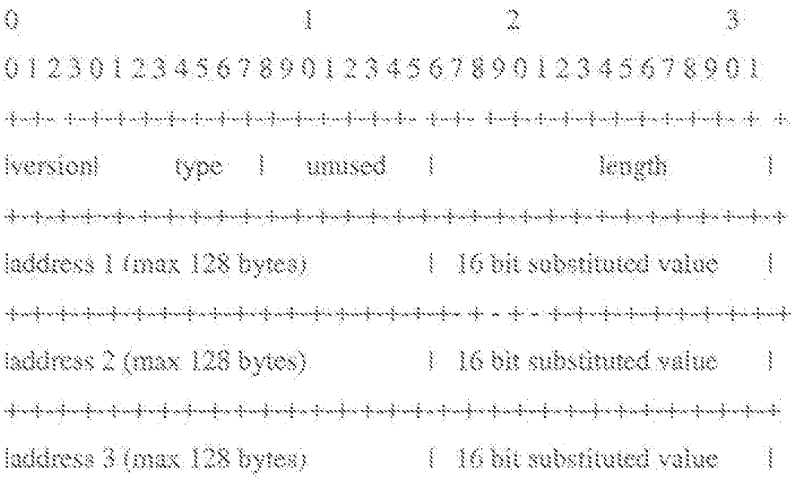
nested subdomains without the need to reconfigure routers or compute netmasks. Also, as address fields are not bound to a limited number of characters, they contain location or company names to make administration of the network easier and more transparent. An example might be the address nl.amsterdam.level3.marketxs.node4, or simply nl.ams.13.mxs.n3 to keep the address relatively short. The drawback of these kinds of unrestricted addresses is the amount of bytes they require. Because every data packet always contains a source and a destination address, the amount of overhead per packet is excessive at times.

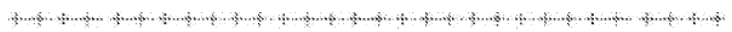
[0087] In one embodiment, the string based node addresses scheme is implemented by letting the first bytes of each address header specify the length of the address in bytes, followed by the address itself as a normal 8 bit ASCII string with a maximum of, for example, 256 bytes. In one alternative, null- terminated strings are used.

[0088] In another embodiment, a sufficiently large address space is used and divided it into logical subsets, similar to IPv4 and IPv6 addresses. In one embodiment, a 64 bit address space is used and divided into 8 bit fields (identical to IPv6) used to identify logical subnets. In this embodiment, addresses might look like 120-23-61-201-43-146-128-132. To make distinct addresses, IPv4 uses the dot \\ *.* for separation, IPv6 uses the colon ":" and the present system uses a hyphen "-". When each address field represents exactly one logical subnet level, this scheme provides a simple, but at the same time somewhat limited, way of addressing. It means that the complete network can address as many individual hosts as IPv6, while grouping them in ls of sub domains, where each sub domain can contain a maximum of 256 hosts or sub domains. The sub domains can be utilized to indicate a network hierarchy in the topology, including a geographical or similarly based heirarchy.

[0089] In one embodiment, to reduce the amount of bandwidth wasted on this overhead, a mechanism is used for substituting the address strings for small 16 bit values or similar small sized value when transmitting a packet to a neighbor. Also, a translation table is kept at both sides of the connection that is used to put the original address string back in the packet before passing it to the router's packet switching kernel. This optimization works on a per- link basis and may be totally independent from the communication with other neighbors. When it is used on more than one interface, each interface maintains its own translation table.

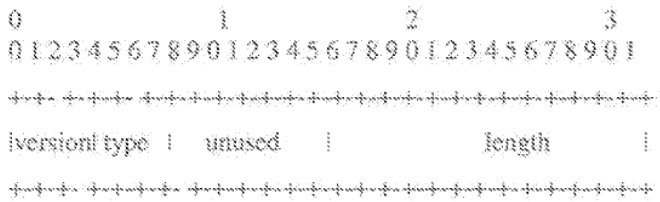
[0090] If a packet is transmitted to a neighbor, all address strings are stripped from the packet header. For each address string a 16 bit or similar value is generated and stored in the translation table. These 16 bits or similar values replace the original addresses and before the packet is transmitted, its protocol version field in the first header of its wire-level representation is changed to a higher version. This version number is reserved for optimized packets that would be unreadable for routers without support for address substitution. To ensure proper translation by the peer, the interface first sends a special packet that contains the new address substitutions. Such a packet is tagged with the higher protocol version number and contains one or more tuples of address strings and their substitute values. In one embodiment, such a packet takes the form:





Type 0x1 is used for issuing new substitutions and type 0x2 is used to invalidate an earlier or unknown substitution. The system attempts to ensure that this (or similar) packet is received by the peer prior to the data packet itself, which is something that cannot be guaranteed by every transport layer. TCP or similar guaranteed communication method may be used for communication between neighbor nodes in the overlay multicast network.

[0091] In one embodiment, configuring an interface for address substitution can be done manually, or automatically. In the latter case the interface uses a special handshake packet that is part of the higher protocol version. In one embodiment, the and shake packet takes the form:



The version field has the higher protocol version. The type for the handshake packet is 0x0 and the length is 4. A peer transmits such a (or similar) packet when the connection is first established. When the packet is echoed back by the peer, the interface knows that the neighbor also supports substitution and goes into substitution mode. After a connection with a neighbor has been lost and is re-established, the handshake is performed again as the peer could have been replaced by software that does not support substitution.

[0092] In one embodiment, entries in the translation table all have a time-out or similar tracking mechanism to ensure timelines and accuracy of entries. After this period the entry is removed from the table and any packet still using the substituted value will not be processed and cause the receiving node to respond with a special packet containing the substituted values that

weren't recognized, allowing the peer to synchronize translation tables by sending a packet containing the substitution tuples. Since the invalidation of timed-out entries is automatic, both peers use the same timeout or similar values in tracking mechanisms. In one embodiment, when the translation table is full, entries are removed according to a last recently used scheme. The address substitution is implemented in a substitution interceptor that is the lowest interceptor or a part of the lowest interceptor or other interceptor reconstructing packets immediately on arrival in an interface module of a router. Substitution introduces additional processing overhead and enabling it is .optional for each node.

[0093] In one embodiment, the overlay multicast system may be used solely for routing. The overlay multicast system offers multicast support over existing IP networks. The overlay multicast system goes beyond offering plain multicast functionality and additionally focuses on supporting live multicast streams with limited jitter and delay through the combination of overlay multicast, multicast congestion control and layered multicast.

[0094] In the overlay multicast system the ExBF protocol or similar routing algorithm implicitly contains information on which links in the network are child links. For example, this is a by-product of the long-lived loop avoidance mechanism in ExBF. In ExBF, a node will inform its neighbor about the fastest route it offers to a destination S, except the neighbor that is in the path towards S. Instead, an infinite distance is advertised to this neighbor. This information is used by a router to conclude that each neighbor advertising that node S is unreachable, is actually ir parent node towards S and hence will immediately mark those links as child links in the shortest path tree rooted at S.

[0095] Neighbors that do advertise reachable routes to destination S are not flagged as child links and therefore not used when forwarding multipoint packets from S. In another embodiment,

this implicit information is available through any distance-vector protocol that uses poisoned reverse or similar techniques. Both methods are used to quickly deliver a copy of any multipoint packet to all nodes in the network. A disadvantage of both mechanisms however is that a copy of every packet is delivered to every node, regardless of whether or not that node is actually interested in multipoint packets from that source.

[0096] In one embodiment, the overlay multicast system is equipped with a sparse forwarding mechanism. The overlay multicast system network is used for a single source multicast for high volume market data, video streams or similar data to be sent to large numbers receivers. In these types of uses it is not necessary that receivers themselves or other nodes be capable of publishing data packets to the multicast group. Instead, for security reasons it is better to know that only the real source can publish to the group. Given these requirements, together with the fact that multi-source protocols are more complex than their single-source alternatives, in one embodiment, the overlay multicast system preferably uses a custom single- source protocol for multicast distribution on the network. In another embodiment, the overlay multicast system supports multiple sources. For sake of convenience an embodiment of a single source multicast network is discussed. One of ordinary skill in the art would understand that the principles of operation can also be applied to a multi-source application as well.

[0097] **Figure 6** is a diagram of an example embodiment of a multicast distribution or "sink" tree. In the example embodiment, the multicast system starts with a source q that is active,

mple, a sparse-mode protocol, with no receiver yet subscribed, no distribution tree exists. When the first receiver application p_0 subscribes, its router initiates the reverse computation of one branch of the sink tree rooted at source q. It does so by first marking that p_0 has a local subscriber for q:G by setting the bit in LSp_0 [q,S] (LSp[] is a local array at node p that

keeps track of all subscriptions of the local user applications connected to router p_j to true and then sending a join (e.g., a <join (q:G)>) packet to the preferred hop towards source q. This neighbor $p1$ receives the <join (q:G)> packet and marks the link with p_n as a child link for multicast group q:G. It then forwards the packet to the next hop $p2$ in the shortest path towards q. Eventually node p_n sends the <join (q:G)> packet to q. On receipt, q marks the link on which the packet was received as a child link for its local multicast group G and starts forwarding all packets addressed to G over the link to node p_n. The multipoint packets addressed to G are inserted into the overlay multicast network at node q by the user application that previously bound the local multicast group $q:G$ in publish mode. Nodes p_n, p_{n-1}, ... p_1 all forward the packets to their neighbors from which a <join (q:G)> packet was received earlier. Node p_n has no child links for $q:G$, but does have $LS_q[Q:G]$ set, so it delivers the packets to the local application. When another node p_j becomes interested in q:G while $LS_{pi}[q:G)$ is not already set, it sends a <join (q:G)> packet to its preferred hop towards q. Let p_{i+n}be in the path from p_i to q that receives the <join (q:G)> packet and suppose that $Pi+n$ is also in the path between p_o and q. In this case, p_{i+n} *is* already in the established part of the sink tree of q, so it does not need to forward the <join (q:G)> packet further to q. Instead, it only marks the link on which it received the packet as a child link for q:G. In general, a node u only forwards a <join (S:G)> packet towards S if it is not already subscribed to S :G (hence, $LS_u[S:G]$ is not set and the collection of child links for S:G is empty).

An example sparse-mode distribution tree generated when processes subscribe to a S:Q group originally equals a part or all of the optimal sink tree rooted at S. However, when the underlying unicast routing algorithm detects changes in the network performance and updates some of the routing table entries, the optimal sink tree changes accordingly and may no longer be

matched by the multicast distribution tree, rendering the tree as well as the multicast performance suboptimal. An extreme example of this is when a link that is part of the multicast distribution fails entirely.

[0099] **Figure 7** is a diagram of an example overlay multicast network with a failed link. In the example a node *u* currently in the distribution tree for q;G only has a single route to *q* via *W*. The link between *u* and *w* fails. The node *u* concludes that it can no longer be part of the distribution tree (as u and q are no longer connected) and invalidates all child link subscription information. In case $LS_u[q;G]$ is set (i.e., the subscription table indicates that *u* is subscribed to group G from node *q*), *u* will rejoin the distribution tree as soon as a new path to q is found. In another embodiment, the algorithm sends a notification to the locally subscribed application to indicate that U is no longer connected to *q*.

[00100] In the example, it is assumed that all links have weight 1, while the link between p and *w* has weight 100. The result is that the link connecting *p* and *w* is not in the optimal sink tree rooted at *q*. Since only t and V have a local subscriber ($LS_t[q;G]$ and $LS_v[$ q;G] are set), the sparse distribution tree in this example equals the dotted arrows of **Figure 7.**

[00101] In the example, nodes *W, u, r* and *s* all have only a single route to *q*. Node *p* has a path through both sand W, while Sis the preferred hop towards *q*. When the link between *u* and *w* fails, *u* becomes disconnected from *q* and invalidates the subscription information for link *u-r*. Node *u* informs neighbor *r* of this fact as part of an immediate distance-vector exchange. This routing update implicitly tells node *r* that its join for q;G through *u is* no longer valid and that it should look for an alternative neighbor to rejoin.

[00102] In the example, an explicit leave packet to *u* is unnecessary. Since *r* is also left with no alternative path to *q* and has no locally subscribed applications, it invalidates its q;G

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subscription and is no longer part of the distribution tree. Neighbors (and *S* are implicitly informed about this through the ExBF routing update or similar routing update sent by r. Eventually w, u, r and s all leave the distribution tree, while nodes t and v schedule a rejoin when a new path to q is found. Again, they inform their local subscriber applications about this. When p receives the routing update from s, it looses its preferred hop for q and switches to neighbor w. The new cost to q is now 100 plus distance (w, q) and neighbor s is informed. Upon receipt, s discovers the new path to q and informs its neighbor's r and v. Node v then sends the <join (q;G)> packet to s. Noes s, p and 2 then construct the first part of the new, optimal distribution tree rooted at q. When the new path reaches r and *t*, *t* sends a join packet to r and r to *S*, reconnecting all receivers to the data stream.

[00103] In one embodiment, recovering from link failure or similar communication errors is divided in three steps. In the first step the underlying ExBF or similar routing protocol starts a wave that propagates a link failure through the network. Depending on the actual topology, this takes up to N-1 packets, where N represents the number of nodes *in* the network. The worst case time complexity for recovering from a link failure is 3N when ExBF or similar algorithm is used as the underlying unicast routing algorithm. A best case time complexity would be 2N.

[00104] The example discusses a scenario where the system recovers from a failure of a link that is part of the multicast distribution tree. More often however, links will not fail completely, but rather fluctuate in quality, causing the underlying routing algorithm to reassign preferred

his example case it is not a requirement that the distribution tree is changed to reflect these changes, as it is not partitioned. However, since the tree becomes suboptimal, it is changed.

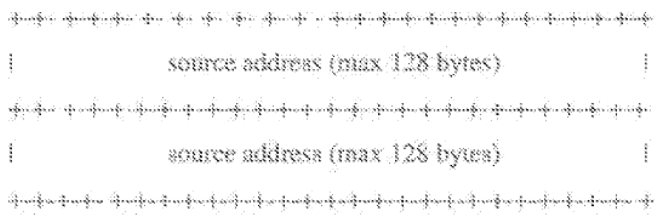
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[00105] In one embodiment, the decision whether or not to recompute a distribution tree after a routing table update is made according to the quality of the used path. When a node *p* in the tree detects that the neighbor in the optimal path towards the multicast source has a cost that is only marginally lower than the neighbor that is currently *p*'s parent in the actual distribution tree, the overhead of recomputation and the risk of packet loss outweighs the increased performance of the new tree. Also, since the quality of each link is continuously re- evaluated, the updated sink tree may only be temporal. Two straightforward solutions to this can be used. Either the algorithm sets a threshold on recomputation so that subscriptions are only moved to the neighbor with the currently optimal path towards the source if the decrease in cost is at least a ratio *X*, where x > 1. A larger value of *x* then postpone tree adjustments until a substantial improvement can be gained·, while a smaller x makes the tree actively follow the changing characteristics of the underlying network.

[00106] Another solution is to postpone recomputation of the tree for a period inversely proportional to the cost decrease ration *X*. The latter has the advantage that the optimal distribution tree is always guaranteed to be reached in finite time since the last routing table change.

[00107] In one embodiment, to run the tree building protocol, two new messages are introduced: the join message and the leave message. Both messages use a GroupData control message of the following format:

```
 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|version| type  |  unused |               length            |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
| control type  |num groups     |  unused                   |
```



This message is used to send a list of subscriptions to a neighbor node or to cancel an aggregated list of subscriptions with a neighbor. The action field (9th byte) is used to distinguish between join and leave. The 10th and 11th byte indicate the number of (S, G) groups in the message. The action field is either "join" (0) or "leave" (1). In an alternative embodiment, this is extended to "stale", "dead", etc., to indicate the state of the publishing application.

[00108] Many multicast applications require some level of reliable communication. Examples of this are uploading files to several receivers simultaneously or replicating web server caches . Without guaranteed delivery, a multicast transport service has limited applicability. In one embodiment, the overlay multicast network utilizes standard delivery control based on moderating the sending of data based on the lowest bandwidth available on the route to a destination. However, this delivery system is unsuitable for certain applications.

[00109] One example is financial data distribution, it needs a transport service that can multicast live data without packet loss or corruption and without substantial end-to-end delay when parts of the network are temporarily or permanently congested. While live market data cannot tolerate random packet loss, it can be thinned. Under certain circumstances it is [illegible] omit certain stock quote updates. An example is a desktop application that displays real-time updates for hundreds or thousands of financial instruments. If all quote updates were delivered to this application, this would require a substantial amount of bandwidth and would cause the value of the more volatile instruments to change faster than a human can read. The data

may be thinned through a process that involves inspection of the individual stock quotes and encoding them in individual data packets, tagging each with a priority value. For this application, as long as all packets labeled with the highest priority number are received, the partial stream can be considered intact. Additionally, when all updates of the second highest priority are also received, the quality of the partial stream is increased usually in the form of less latency.

[00110] In one embodiment of the overlay multicast system, priority numbers are associated with data packets. The priority numbers represent a logical layer inside a data stream. Data provided by a multicast application may be in the form of a stream of data. This data is subdivided into categories or priorities based on the nature of the data. When packets in a data stream are labeled with priorities in the range 0 to 3, the stream is said to have 4 layers. Also, the convention is to treat 0 as the highest priority and 3 as the lowest. Any other system of identifying priority levels may be utilized including alpha numeric indicators or similar identifiers. If a stream only contains a single layer, all packets are labeled with priority 0.

[00111] To software router daemons, a priority value of a packet is relative to the stream and becomes relevant when a decision to discard data at a congested router is made. The priority value has no meaning other than as a criterion for discarding packets on congested links. When an outgoing link of a router has insufficient bandwidth to transmit all pending packets, it forwards only those packets with a designated priority or the highest priority. This technique is also applied to any data type, including audio/video data and financial data. Using this system, priority numbers cannot be misused by sources to give their data packets a greater chance of prioritized transmission by giving them the highest priority to gain advantage over other sources. Packet priorities are only compared between packets that are part of the same data stream.

[00112] In one embodiment, knowing that routers will use the packet priority numbers when making forwarding selections on congested parts of the network, a source carefully divides its data packets over different layers or priorities, in a way that a subset of the layers still contain a useful, uncorrupted representation of the data. This system is especially useful when multicasting a live data stream with a high data rate to a large number of receivers, scattered over a heterogeneous wide area network. Receivers that are on congested parts of the network will then receive the highest priority parts of the data only. This eliminates the need to lower the publishing rate to match the slowest receiver, while still being able to offer live, uncorrupted, thinned data streams to clients suffering from insufficient bandwidth. Example applications of the system include audio and video codecs that divide live multimedia content over layers to enhance the user experience over wide area networks.

[00113] In one embodiment, incoming messages are sorted by unicast or multicast sender address. As described above, a sender address is the combination of source router address and application session. An example of a source address is n1.mxs.office. erik:video.an2 which could be used by a user application broadcasting a video channel. In this case, only the source address is relevant, not the (uni- or multicast) destination address. Each incoming message is added to the queue that holds messages sent by that particular sender. If this is the first message from a sender, a new queue is automatically created to store it.

[00114] **Figure 8A** is a flowchart of one embodiment of a process for managing congestion in the overlay multicast system. The congestion can be managed at each individual router. In one embodiment, the congestion is managed at the interface module level in each router. Each interface module has an inbound pipeline and outbound pipeline discussed above for processing inbound and outbound data. Each pipeline buffers data that is awaiting further processing.

However, if either pipeline is unable to keep up with the pace of incoming data that needs to be processed some data must dropped.

[00115] In one embodiment, data is received as a set of data streams at each router (block 851). The data is then buffered in the inbound pipeline buffer (block 853). The same process applies to outbound data that is received from the kernel by the outbound pipeline. This data is stored in the outbound buffer. After the data has been stored a check is made of the inbound or outbound buffer to determine if it is full (block 855). In one embodiment, data is stored in the data structures in the buffers that organize the data packets into a set of queues. Each source address, data stream or layer in a data stream has a separate queue. A queue is sorted with highest priority and oldest data packets at the front of the queue. If the buffer is full then a decision is made to drop a designated amount of data in the form of packets from the buffer to make room for incoming data packets (block 857).

[00116] In one embodiment, a queue is chosen randomly or in a round robin to have data dropped. In another embodiment, the queue with the most data is chosen to have data dropped. In a further embodiment, a weighting factor is calculated to determine which queue is selected to have data dropped. The weighting factor is based on the amount of data in a queue, size of packets in a queue and similar factors. The weighting factor counteracts unfair distribution that is caused by selecting a queue by a round robin, random or similar method of selection. Data streams with large packets are unfairly affected by other methods because a disproportionate amount of data is dropped in comparison with other queues with smaller packets. Queue selection is also influence by the size and the amount of data that is intended to be dropped. A large queue that can drop close to the amount of data desired is weighted for selection.

[00117] In one embodiment, data is dropped if a total amount of data stored in all queues exceeds a threshold value. This threshold value is set by an administrator or is a set value. These systems enforce the fair allotment of bandwidth between data streams. Also, this system implements the prioritization of logical layers by dropping lower priority level layers when congestion occurs. If the buffer is not full then the pipelines may continue to store data in the buffers (block 851).

[00118] In one embodiment, while the queues grow in size as packets are received, a background thread constantly dequeues packets from the queues and transmits them over the network.

[00119] In one embodiment, the system that manages in- and output for the queues is divided in two parts. The first part is run by the background thread that constantly dequeues packets from the queues and transmits them over the network, while the second part is in charge of queuing new packets that are to be transmitted. The latter also implements the logic that defines when and which packet should be discarded (due to a buffer overflow).

[00120] The dequeuing part of the system, when selecting a packet for transmission, only looks at the first packet (most urgent) of each queue. To make sure each stream gets an equal share of bandwidth, it takes the individual packet sizes into account. When all packets (note that only the first packet of each queue is observed) have equal size, the dequeuing thread simply selects a random queue with equal probability and dequeues one of its packets. Since each queue has the same probability of being selected, each queue will deliver an equal amount of packets per time unit. Hence, each source will transmit an equal amount of bytes per second.

[00121] Since individual packets can have any size between 1 and some determined maximum number of bytes, their size must be considered when the transmit thread selects

queues. Queues with many large packets should generally have a smaller probability of getting selected than queues with lots of small packets in order to keep the bandwidth division fair.

[00122] In one embodiment, only the first packet of each queue is considered to represent its queue and give it a probability of setting selected that is reversely proportional to its size in bytes. In an example where there are three queues, P, Q and R where the first packet in P is 100 bytes, the first packet from sender Q is 300 bytes, while the packet from R is 500 bytes, to calculate the selection probabilities, first their respective selection weights are defined. For example, the weight of the packet in P is computed by dividing the total size of P, Q and R by P's size: (100 + 300 + 500) / 100 = 9. Q's weight is 3 and R's weight is 9/5. The weights are converted into selection probabilities by dividing them by the sum of all weights. This gives P a probability of 9 / (9 + 3 + 1. 8) = 0.65, Q a probability of 0. 22 and R of 0.13. When a queue becomes empty after dequeuing, it is removed.

[00123] In one embodiment, to keep the delay introduced by buffering packets under control, the second part of the system enforces a maximum total queue size. The sum of all packets in all queues may never exceed this threshold. When a new packet comes in, it is always accepted at first. If necessary a new queue is created for it, or it *is* added to its designated, existing queue. After adding a new packet, the total size of all queues is checked. If it is larger than the configured maximum, the algorithm runs a removal round in which it first selects a queue and then tells that queue to remove one packet. If the queues are still to big after removal of one

ocess of selecting a queue and removing a packet is repeated iteratively until the total queue size is smaller than or equal to the configured maximum.

[00124] In one embodiment, shrinking the queues is always done after a packet was added, never preemptively. The reason for this is to let the new packet immediately participate in the

removal selection process, rather than discarding it or making room for it at the expense of the other queues. Contrary to de-queuing packets, where the algorithm tries to select the "most urgent" packet, we now need to select the "least urgent" one. This packet is determined by looking at the size of the individual queues (the largest queue should generally be shrunk to keep resource division among the streams fair) and which queue can match the required size most accurately. This comes from the fact that individual packets can differ greatly in size, so removing the "least urgent" packet from queue P could result in freeing 8 kilobytes, while removing the "least urgent" packet from queue Q yields 40 bytes of space for example. Now if the queues in total exceed the maximum size by only a couple of bytes, it seems logical to remove the small packet from Q. This policy comes with a consequence, namely that it implicitly promotes the use of larger packets. After all, streams with lots of very small packets are more likely to accurately match the amount of buffer space that must be freed than stream queues with few very large packets. This property may encourage developers to use larger packets, increasing the efficiency and throughput of the network. In short, when selecting a queue for shrinking, the system favors queues that are will expunge the least amount of bytes in order to match the total buffer capacity and are large compared to the others.

[00125] In one embodiment, before the selection can be made, the absolute weight factor for each queue is computed. The weight factor combines the queue's size and ability to accurately match the overcapacity of the buffer by removing packets. The latter is exposed in v. It represents the sum of the sizes of the packets that a queue must remove in order to eliminate the buffer's overcapacity. For example, if a queue P has 10 packets of 100 bytes each while the buffer currently has an overcapacity of 170 bytes (suppose the maximum is 10000 bytes, while all queues combined add up to 10170 bytes), P would need to remove at least 2 packets (2 * 100

bytes). In this case, v_p is 200. The formula to compute the absolute weight factor of queue m is defined:

$$w_m = \frac{(\sum_{i=0}^{n} v_i) \cdot s_m}{V_m}$$

In this formula, n represents the total number of queues, v represents the number of bytes that would be removed if the queue was selected and s represents the total size of a particular queue. The formula captures the direct relation to queue size and the inverse relation to the amount of bytes that would be removed by the queue if it was selected.

[00126] When the absolute weight factors of all queues have been derived, they are converted to weighted selection probabilities. This is done by dividing the individual weight values by the sum of all weights. The selection probability of queue m is expressed in P_m.

$$P_m = \frac{w_m}{\sum_{i=0}^{n} w_i}$$

After the selection probabilities have been computed, a roulette-wheel algorithm is run to select a queue and let that queue remove its v bytes. If v is smaller than the current overcapacity of the buffer in total (note that this implies that the selected queue became empty and was thus removed automatically), another selection round is done until the buffers shrunk sufficiently. Note that the queue selection process is implemented as an atomic operation. During selection rounds no new [illegible] added to or removed from the queues.

[00127] In one embodiment, the overlay multicast system is a layered multicast in combination with a scalable, selective packet retransmission mechanism to offer a service that can meet the demands of real-time financial data and similar applications. Selective packet retransmissions are crucial so that the overlay multicast system is guaranteed that subscribers

will receive one or more layers that are completely intact. When an occasional packet in a high priority layer is missed, it is repaired. However, the receiver may decide not to attempt to recover missing packets from low priority layers that are missing due to network congestion. Trying to repair these layers would require additional bandwidth, resulting in additional packet loss. Yet, as more end-to-end bandwidth becomes available, the receiver detects this and repairs additional layers, so more intact layers are delivered to the application.

[00128] **Figure 8B** is a flowchart of one embodiment of a process for handling layer repair. In one embodiment, the overlay multicast system supports a receiver- side multicast socket that receives the raw packets from the multicast stream and is in charge of repairing damaged stream layers before forwarding them to the user application (block 801). This method includes a retransmission request packet. When the receiver detects a missing packet in a layer that should be repaired (block 803), it sends a retransmission request for this packet towards the source (block 805).

[00129] In one embodiment, detecting packet loss is done in the conventional way by sequencing each packet with an incrementing number. In another embodiment, each layer has its own sequence number, so all packets in one layer are sequenced independent of the other layers. This way, packet loss is detected in each layer, regardless of the conditions and packet loss in other layers.

[00130] To avoid a cascade of retransmission requests which could be stopped close to the source, a NAK based negative acknowledgment is used. This technique provides localized repair without communication with the source, because an intermediate router that has a copy of the requested data packet responds by sending it again and suppressing the retransmission request. When the next router closer in the distribution tree receives the retransmission request it checks

its buffer to determine if the requested packet is still stored therein (block 807). If the requested packet is there, it is retransmitted (block 809). If the packet is not in the buffer then the retransmission request is forwarded to the next router in the distribution tree toward the source (block 811).

[00131] In one embodiment, when layers are dropped due to congestion, the programming API that is exposed to user applications notifies the user by means of an exception or special return value when it is reading packets. These notifications may not be fatal and only serve to inform the user that the quality of the stream has changed.

[00132] In one embodiment, localized repair works by having each router daemon store a window of transmitted packets. Packets are either stored for a fixed period of time, after which they are discarded, or a fixed amount of buffer space is reserved to store the most recently forwarded packets. Since the overlay multicast network by default use reliable connections between router daemons, packets never get lost while they are in transit between nodes. The only place where packets are purposely discarded is in the interface buffer interceptors. Thus, there is no need to buffer packets that were actually transmitted over the reliable connection, as those packets are guaranteed to have reached the neighbor node. In one embodiment, a router daemon copies packets to its retransmission packet store before those packets reach a buffer interceptor that discards packets.

[00133] The router's buffer interceptors copy packets to b[illegible]nsmission. In one [illegible], localized packet repair in the overlay multicast system are implemented by adding an interceptor to both interceptor pipelines of each interface and letting those interceptors simply store a copy of each packet. Additionally, when a retransmission request passes through the router on its way to the multicast source, the interceptor inspects the request packet and if it has

the requested packet in its temporal packet store, it does not forward the retransmission request to the next interceptor. The requested data packet is injected into the network again.

[00134] **Figure 9A** is a diagram of two r outer daemons connected by a link. There are two interceptor pipelines 903, 905 inside each interface 907. Each interface contains both an outbound 903 and an inbound 905 intereceptor pipeline. The latter processes packets received from the network, while the outbound pipeline processes packets just before they leave the router daemon. In one embodiment, the pipelines contain two different interceptors; the CCI (Congestion Control Interceptor] 909 and the PRI (Packet Retransmission Interceptor] 911. In another embodiment, a router daemon contains additional interceptor instances per pipeline.

[00135] The CCI 909 implements bandwidth allocation rules and is responsible for discarding packets. During normal operation the source 913 publishes its multicast packet stream to its router daemon 901 (path 1) that sends the packets to all interfaces that lead to interested receivers according to the multicast routing table. In the present example, the packets are passed to the first outbound interceptor (PRI) of the interface (path 2). The PRI is responsible for storing a copy of each recoverable packet that is transmitted. It stores the copies in the private temporary packet store 915 of its interface (path 3). Then the interceptor passes the packets to the next interceptor (CCI) (path 4) where it is buffered until the connection link with the adjacent router daemon has time to transfer it.

[00136] In one embodiment, if the packets are not dropped by the CCI, they are received by the neighbor router daemon (paths 5 and 6) and sent through that router's inbound interceptor pipeline 921 (path 7). They first enter the PRI 923 (path 7), which stores copies in its inbound buffer 924 (path 8) and then passes them on to the next interceptor (CCI) 925 (path 9) where they are temporarily parked until the router's kernel 927 thread picks them up (path 10)

and delivers the packets to the subscribed multicast tsocket which forwards them to their next destination 929 (path 11).

[00137] **Figure 9B** is a diagram of two router daemons connected by a link where a retransmission is requested. If a packet is lost in the outbound packet interceptor pipeline of the source router because the connection between the routers was not fast enough to transmit all the packets, the receiving tsocket notices the loss and send a retransmission request (path 1). The destination address of this unicast retransmission request packet is that of the multicast group. The network packet switching kernels recognize the retransmission packets and use the multicast group to route the packet towards the source of the group. The retransmission request packet of the example passes the router kernel 927 and reaches the outbound packet interceptor pipeline 931 and goes into the PRI 933 (path 2). This interceptor scans specifically for retransmission requests and tries to answer them locally. To this end it inspects the packet, looks up the multicast group, the stream layer identifier and the layer's sequence number and then checks the inbound packet buffer 935 to see if it contains the packet (path 3). If this is not the case, the packet is passed on to the rest of the interceptors (path 4) and eventually transmitted to the adjacent router (paths 5 and 6) where it is fed to the inbound interceptor pipeline 941 (path 7).

[00138] In one embodiment, the PRI 943 checks the interface's outbound packet buffer 945 (path 8) and finds the multicast packet that was dropped by the congested outbound CCI 909 earlier. It is then re-inserted into the outbound packet stream via the outbound PRI 911 (path 9). The retransmission request packet is then dropped and not forwarded any further. Assuming the packet is not dropped again, it travels the normal way towards the receiver (paths 10, 11, 12, 13). In one embodiment, the receiver's PRI sees the multicast packet for the first time and store a copy in its outbound packet buffer (path 14). The packet then continues on to destination 929

(paths 15, 16, 17). This mechanism of unicast retransmission packets and interceptors that offer localized repair is flexible in the way that not all router daemons need to support local packet repair. A router that does not support it simply forwards the requests and let the upstream routers handle them. The network as a whole becomes more efficient when more routers support it.

[00139] In one embodiment, because storing a copy of every multicast data packet requires storage capacity, the overlay multicast system links the packet buffers of the packet retransmission interceptors to a central data structure that will only store unique packets. This is done because in multicast transmissions a router that is a branch point in the multicast distribution tree otherwise ends up storing at least three copies of every data packet. This is because the packet is received on one interface and sent out over at least two other interfaces. If every interface would individually store its packets, the inbound packet buffer of the receiving interface as well as the outbound packet buffers of the forwarding interfaces contain the same packets.

[00140] In one embodiment, the central packet store eliminates duplicate packets by storing each packet only once and by keeping reference tables that point to packets for every interface. When an interface buffer stores a new packet in the central store, while that same packet was already stored by another interface, the central packet store merely adds a pointer to that packet instance to the interface's packet reference table.

[00141] In one embodiment, each entry in the reference table has a time-out attached to it. This is because packets are only temporarily stored. In one embodiment, the same packet could pass different interfaces at different times. The time-outs are not attached to the packets in the central store but rather to the packet references in the interface reference tables. In one embodiment, packets are only expunged from the central store when all references to the packet

have timed out. Another optimization that can substantially reduce the amount of required storage space is to only store packets that were explicitly discarded by the Congestion Control Interceptors (CC is) because of bandwidth constraints. Storing other packets has little value, as those are guaranteed to have arrived at the next router. No retransmissions may be requested for them unless some non-deterministic packet loss occurs as a result of a crashing node.

[00142] In one embodiment, if a packet was lost close to the multicast source and because of this and all receivers simultaneously send a retransmission request, the upstream routers still apply conventional negative acknowledgement "nack" suppression techniques to combine all concurrent retransmission requests from its child links into a single request that is forwarded upstream towards the source. Applying local packet stores at every router reduces the latency of packet recovery.

[00143] In one embodiment, time to live values are adjustable per router, set system wide or similarly configured. A short time-out minimizes storage requirements, but will also lead to more overhead towards the source as more retransmission requests need to be propagated further upstream. Large timeouts will be more tolerant to late retransmissions, but require more storage. When a time-out of 60 seconds is used by the routers that means a lost packet can still be recovered after a minute. However, when the data stream contains real-time data such as stock quotes, such a delay in delivery cannot be tolerated. Problems get worse if the data packets also have to be delivered to the user applications in their original order. In that case the user does not receive any data for up to a minute while the missed packet is recovered and all newer packets are waiting for it. In one embodiment, a ten second timeout setting is utilized.

[00144] If all routers in the network use the same time-out value, there is no use in propagating these requests as the upstream routers are likely to have purged their copies as well.

To avoid nack-implosions from slow receivers right after a packet is purged, each router interface remembers which packets it has purged. When a retransmission request is received for such a packet, the interface may respond by sending a special packet to the receiver that indicates that the requested packet had timed-out and cannot be recovered anymore. The retransmission request in this scenario is not propagated further upstream. When a receiver gets this timeout notification, it knows it will not have to wait any longer for the missed packet. It will notify the user application of the fact that data was lost and continue to deliver the next packets. Whether or not an application can tolerate packet loss will depend on the type of content. Audio and video will usually not be severely impacted by the loss of an occasional packet, while real-time stock quotes become dangerous to use when it is not known what quote was lost.

[00145] In one embodiment, where layered multicast is combined with localized packet retransmission, a service is realized that applies packet recovery only to those layers that can be delivered with the current bandwidth capacity. When the network has insufficient resources to deliver all layers of a stream, it is beneficial if the receivers know this and will not attempt to repair all missed packets from all layers. Instead, the receivers may use a mechanism that provides them feedback about the current network capacity and use that to decide for which layers it will recover missed packets. The algorithm monitors the status of the total stream and from this information derives how many layers of the stream can be reliably delivered to the user without stressing the network. It then marks these layers as being intact, reliable or similarly ... retransmission requests when an occasional packet is lost from these layers.

[00146] In one embodiment, the algorithm only delivers packets from the reliable layers to the application, but not before they have been put back into their original global order. In order for the algorithm to decide which layers are safely marked as intact, it requires some status

information about the reception of the stream as a whole as well as the network conditions. To provide this, every packet contains the number of the most recent packet from all other layers or similar sequence data. Aside from carrying its own sequence number, each packet contains the current sequence number of all the other layers as well. By inspecting these sequence numbers each time a packet arrives, the receiver determines whether it has missed any packets from the layers if marked as reliable. If that is the case, these packets may be recovered. Each time a missed packet is detected, a countdown timer is started for it. If the packet is recovered before the time-out expired, the timer is canceled and removed.

[00147] If however the timer manages to expire, the layer is considered impossible to repair and may be, together with all higher reliable layers (lower in priority), removed from the list of reliable layers. How long the time-out interval should be can be application or content dependent. The interval determines how long delay is tolerated by the user. Setting the interval to a low value means a temporary congestion may corrupt a layer long enough to cause the receivers to drop it. Since packets occasionally get lost when unicast routing tables converge, changing the shape of multicast distribution trees, or when a router crashes that had pending packets in its interface buffers, appropriate time- out values are adjustable by an administrator according to network conditions to be found experimentally.

[00148] In a overlay multicast network that uses packet prioritization, routers explicitly introduce out of order delivery when a burst of packets queues in the outbound interceptor outer interface with a slow connection. The time- out may be large enough to allow for this. Setting the time- out to a long period makes the stream much more resilient to congestion, but also increases the time for the receiver to discover that a layer must be dropped due to bandwidth constraints. Until congestion is finally detected, delivery of the previously

received packets that causally depend on missed packets is postponed. Starting timers when packet loss is detected allows for congestion detection. However, it cannot be used to guarantee low end-to end latency.

[00149] When no packets are lost and all are received according to their global order, the receiver cannot measure the total transmission delay. As such, a receiver cannot distinguish between an idle and a crashed source. In one embodiment, aside from removing layers from the reliable-list, the algorithm is also able to detect when more bandwidth becomes available and new higher layers can be added to this list, so that a higher quality stream can be delivered to the user. This is done by passively or actively monitoring the layers that are not currently in the reliable-list and not under repair. Every time a packet is received from these layers, this fact is stored for a period that is equal to the repair time-out discussed earlier. When a packet from a layer is received with a sequence numbers that shows that a packet has been lost from a higher layer that is not in the reliable list, no retransmission request is sent, but the fact that this packet should have been received is recorded and stored for the same repair time-out period.

[00150] If during the time-out period the packet is received after all, possibly because it was delayed by a router, the stored record is marked as received. Because the state of each packet from every layer is recorded, the overlay multicast system builds a packet arrival history that is used to determine whether the reception quality was high enough to add a higher layer to the reliable-list and repair any further packet loss. In the overlay multicast system the reception history or similar data may be used to calculate a moving average that describes the amount of packet loss over the last x seconds, where *x* is equal to the repair timeout. With a small moving average history, the receiver will quickly respond to increased network capacity, while a longer history will only add layers when the network ' s capacity was sufficiently high for a longer

period of time. The receiver may keep track of the reception quality of each layer by inspecting the list of sequence number that is attached to each data packet.

[00151] This causality information adds overhead to each packet, linearly related to the number of layers used in the stream. For example, when sequence numbers are 32 bits and the publisher uses all 256 available layers, each packet comes with a kilobyte of causality information, which contributes to at least 12.5% for an 8 Kb packet that is filled to the brim. To ease layer administration in the receiver socket, an overlay multicast stream only supports a static number of layers. When a publisher binds a multicast group address for reliable, layered communication, it may explicitly specify the number of layers that will be used.

[00152] In one embodiment, by default, packets are delivered to the user application in their original order. Not only are the packets inside the individual layers restored to their natural order using their sequence numbers, but the total ordering across the layers may also be restored. When the source sends three packets with different priority (each in a different layer), all three are received by the user application in the exact same order. A prioritized, layered packet is described as $P_{l[9,4,6]}$ where l represents the packet's priority or layer, 9 the sequence number of the last packet of layer 0 (the highest priority layer) that was sent at the time this packet was published, 4 represents the sequence number of this packet and 6 represents the sequence number of the last packet of layer 2. Also, the packet tells that the stream uses 3 layers in total: layer 0 up to and including layer 2. It is concluded that this packet causally depends on packet 9 from layer [illegible] from layer 2 and will only be delivered to the user application after those packets have been delivered. In another embodiment, the overlay multicast system does not reorder the packets, but may leave this task to the application using the network.

[00153] **Figure 10** is a diagram that shows one embodiment of a process of selective packet repair and ordering. For purposes of explanation the notation {0 (9,2,6) ; 1(9,3,6), . . . } to describe a sequence of packets where Po<9.2.6J was sent prior to Pl<9.3.6J, is used for sake of convenience. The first segment 1001, shows the sequence of packets originally published by the source. It reads from left to right, so Po<9.2.6J was published first, followed by Pl<9.3.6J, etc. In this example the source has published eight packets, divided over three priority levels or layers (0, 1 and 2, where 0 is 0 is the lowest layer with the highest priority). The second segment 1003 shows the packet stream as received by one of the subscribers. It shows two lost packets and an incorrect ordering. Before the packets are delivered to the user application, they are stored in internal buffers during the repair and reordering process. This state is depicted in segment 1005. Here the packets are in separate buffers, each representing a layer. The packets are ordered inside the buffers. The illustration shows the missing packets $P_{(0,4,7)}$ and $P_{(0,4,8)}$.

[00154] If the receiver currently only has layers 0 and 1 in its reliable-list, it will attempt to repair the hole in layer 1 by sending a retransmission request. Note that if the given sequence was just a snapshot of a running stream, the receiver would have detected the missing packet $P_{(0,4,7)}$ when $P_{(0,4,8)}$ was received, because this packet says it depends on packet #4 rom layer 1. So even before $P_{(0,3,9)}$ from layer 1 was received in our example, the receiver already detected loss in layer 1 and immediately scheduled a repair timeout for the missing packet and sent a retransmission request. In fact, if it is assumed that the receiver had received packet #2 from a our snapshot of **Figure 10,** then the conclusion would be that packet #4 as well as packet #3 were lost.

[00155] In the example, shortly after $P_{(0,4,8)}$ was received, packet $P_{(0,3,6)}$ is received. The receiver p l aces the delayed packet in the appropriate receiver buffer and cancels the repair time-

out it started earlier when it detected that the packet was missing. This is illustrated in segment 1007. Note that the hole in layer 2 may also be detected when $P_{(0,0,4,8)}$ is received, as that packet claims to be sent after packet #8 of layer 2 was published, so either packet #8 from layer 2 got lost in the network, or was delayed . However, since layer 2 is not in the reliable-list, a retransmission request is not sent. However, as for all packets, a timer is started for packet #8 of layer 2. When the user application is ready to read packets, the algorithm returns only packets from layers that are in the reliable-list. Even though some of the layer 2 packets were received, they are discarded and not delivered. The resulting stream of packets that is delivered to the user is equal to the stream originally published by the source, with all layer 2 packets removed.

[00156] In this example, despite the fact that the network has insufficient capacity, having dropped packets from every layer and delivered the packets out of order, the user received a deterministic subset of the stream that is uncorrupted. Because the overlay multicast system sources are live data streams that cannot slow down or pause and because real-time data are not buffered by the source too long, receivers will only attempt to recover lost packets for a limited period of time. Whether a packet is received, recovered or lost, its buffer slot will be freed after this timeout, resulting in a natural size limit of the receive buffer. How large the buffer can get is related to the average speed of the stream and the length of the repair time-out.

[00157] In one embodiment, the algorithm implementation does not enforce a hard size limit during packet repairs. A more troublesome situation occurs when the user application does not

the socket fast enough. When this happens, the amount of pending data builds up in the socket and both storage requirements and the transmission delay increases. In one embodiment, this may be handled by removing a layer from the reliable-list, causing those packets to be discarded from the buffers, while decreasing the amount of data that is delivered to

the application. If a clean feedback algorithm that can keep the number of layers in balance with the application's reading speed is not used, the overlay multicast system throws a fatal exception to the user or provide a similar indicator to the user application and close the socket when the receive buffer reaches a certain maximum size, or when the total time between arrival of packets and their actual delivery reaches a threshold.

[00158] Although restoring the original global packet ordering before delivering the data to the user application is assumed to be appropriate for most types of data, there is content for which each packet invalidates all previous packets. This is the case for stock quotes and similar time sensitive data. For example, when a new stock quote update is received for a financial ticker symbol, it renders the previous updates useless. For most applications that process real-time financial data, only the most recent information is interesting. When global ordering is restored, the algorithm will postpone the delivery of the most recent data until all prior packets have been received also. For a typical application, such as a market data terminal, that merely displays quote updates to the screen, this postponing adds little value. When the burst of pending updates is finally delivered by the tsocket, the application updates the symbol's last value on the screen, thereby leaving only the last and most recent update visible and overwriting all pending updates immediately.

[00159] In one embodiment, because the reordering process adds additional delay to the data delivery, it can be switched off by applications that do not benefit from it. Without reordering, relations between packets from different layers) and local ordering (order of individual packets inside a single layer) is ignored and packets are delivered to the application immediately after they have been received. Disabling reordering has no impact on the reliability. Lost or delayed packets may still be recovered for all layers the reliable- list, only the recovered

packets may be delivered with additional delay. Whether or not this makes them useless is up to the application to decide.

[00160] For example, with market data it is useful to know whether an update is older or newer than the one previously received. This is because a stock quote may be overwritten by a newer one, but not the other way around. In this case, the source could add a logical time stamp or similar sequence indicator to each stock quote, so the receiver can decide how to handle the update. In one embodiment, aside from configuring a layered multicast isocket to restore global ordering or no ordering at all, a receiver can configure a isocket to only restore local ordering. Whether or not this is useful will depend on the type of application and the content, but it offers the advantage that delivery of packets from lower (high priority) layers are not delayed during recovery of packets from higher, lower priority layers.

[00161] In one embodiment, an example market data publishing application tracks market financial updates in a linked list or similar data structure. Every time a new update is received, it is stored in the appropriate slot, overwriting the previous update. This way, the linked list always contains the most recent trade for every symbol. A virtual token traverses the list at a fixed speed. If the stream is to be thinned to one update per second, the token will visit one slot every second. When the token visits a slot, it removes the quote update and forwards it, leaving an empty slot. Empty slots will be skipped by the token without any delay. When one of the symbols has two updates per second in the incoming stream, the second incoming update after

f the token will overwrite the update pending in the slot and the older pending update is dropped. This allows the leaky bucket algorithm to always provide the thinned clients with the most recent, live update of every symbol. While the virtual token visits the slots that have a recent quote update pending to be sent out, a thread receives the incoming market data

stream and inserts new trades in the slots. To limit the outgoing transmission rate to one quote per second, the token thread sleeps for one second after it moved and sent an update from a slot. When the token is able to complete one full circle between two incoming updates of every symbol, there will be no data loss. Instead of limiting the outgoing bandwidth to a fixed maximum, the dequeuing thread could also be configured to run as fast as possible, relying on external feedback such as a blocking write call from the network, to reduce publishing speed. This could be particularly useful when the algorithm is used on the server side of a point-to-point TCP connection to a client application. In one embodiment, the server uses the algorithm to send the data at original speed, until network saturation slows down the links. A potential problems with this technique is that the original order of the updates is lost. In fact, when every symbol updates more than once while the token circles the list, the token will find every slot filled with a quote update, causing the order of the output stream to match the fixed order of the slots in the linked list. For quote updates from different symbols, this is usually not a problem, as they are not related to each other.

[00162] In another embodiment, an alternative algorithm may be used that provides another way of thinning a stream of data, such as live market data, with varying bandwidth to a fixed maximum rate, without the risk of providing stale data, for example stale market symbols. This algorithm may be used to produce enhancement layers that form a layered data stream, such as a layered market data stream. One example method of doing this is to have several instances of the algorithm running that each receive a copy of the input stream, letting their token run at different speeds and tagging their updates with a layer number. The layers may then be combined into a single stream. Unfortunately, this may introduce redundant information *in* the stream because the higher layers contain updates that are also available *in* the lower layers. This technique of

encoding an entire stream in different qualities and sending them to clients concurrently to meet their individual bandwidth capacities may be referred to as simulcasting.

[00163] In one embodiment, the thinning algorithm may be modified to have each layer contain only information that is not already present in the lower layers, thereby eliminating redundancy among the layers. To achieve this the leaky bucket algorithm may be extended to a multi-level ring as illustrated in **Figure 11B** , creating a virtual cylinder of vertically stacked, circular lists, where each ring represents a layer of the output stream. Each ring has its own token. In one embodiment, the tokens all circulate their ring at the same speed. In other embodiments, the tokens in each ring may be set to circulate at different speeds. This data structure also identifies columns. A column is the collection of vertically stacked slots that can all store an update for the same symbol.

[00164] **Figure 11A** is a diagram of an example method of thinning a stream of live market data. The gray slots represent symbols for which a recent quote update was received, while the transparent slots are empty. **Figure 11B** is a diagram illustrating the token ring data structure. Multiple instances of the thinning algorithm and data structure may be conceived as being vertically stacked, each instance represents a stream layer and each symbol has a column that can store more than one update. **Figure 11C** is a diagram of one embodiment of a representation of the stacked thinning algorithm of **Figure 11A** at work. In the example, the gray slots contain a pending quote update, while the transparent slots are empty. The illustration shows how the olatile symbol are spread out over several higher layers.

[00165] In one embodiment, the thinning algorithm has a single input thread that receives the original stream. Newly received updates may be inserted in the lowest ring. If a new update arrives for a slot in the lowest ring that already contains an update for that symbol, this older

update is vertically shifted into the next ring, so that the lower slot can always store the latest update. Once an update is shifted upwards, it cannot return to a lower ring anymore, even if all other slots beneath it were emptied.

[00166] In one example embodiment, symbols that have more than one trade during the time it takes the token to complete a full rotation of the ring will be spread out to two or more higher layers, while symbols that have very infrequent trades will only occupy a single slot in the bottom ring. An example result of this process is depicted in **Figure 11C**. **Figure 11C** shows the thinning algorithm using three layers to thin a market data stream that contains updates for eight symbols. Four symbols are included in data structure and assigned to the four visible columns in the front. The figure also shows that all four symbols have pending updates in the bottom ring, while the 'MSFT' symbol is so volatile that it shifted updates into all three layers before the tokens could dequeue any of the pending updates.

[00167] In a case where a symbol is so actively traded that it manages to shift an update out of the top ring, that update is irrecoverably lost. If required, loss of updates can be avoided by letting the token in the top ring run without dequeuing delay, letting it empty every slot immediately after an update was queued. The result is that the transmission rate of the upper layer will equal the transmission rate of the original stream, minus the combined rate of all lower layers. Because the transmission rate of the lower layers may have a fixed maximum, the upper layer's rate may fluctuate and follow the variations of the original stream.

[00168] In one embodiment, if the tokens all rotate at the same speed, this does not imply that they visit the same symbol slots at the same time. Instead, they will usually be at different positions in their ring. This is a result of empty slots that are skipped without waiting.

[00169] In one embodiment, the layered thinning algorithm has out of order transmission and may cause additional delay. Every ring can contain an update for the same symbol at a certain time and because the tokens can have different column locations inside their ring, it is undefined which ring will dequeue and transmit its update first, which will be second and so on. This means that not only the order of symbols is shuffled, but even the updates for a single symbol are no longer guaranteed to be transmitted chronologically.

[00170] While new updates for a symbol are queued, shifted and sent out, in some cases an update can survive in the rings substantially longer than its younger successors. This can occur when an update that is just about to be dequeued by the token, gets shifted to the next layer by a more recent update that entered through the lower ring. In this case it is likely that the more recent update is dequeued sooner than the older update that now has to wait for the token of its new ring to reach its slot. To address this issue, the system provides a way of putting the outgoing, layered updates back in the original, chronological order. This is done by labeling incoming updates with sequential numbers and having the tokens pass their dequeued, labeled updates to a transmit window that uses the sequence numbers to restore the chronological order. When updates are shifted out of the upper ring and therefore considered lost the transmit window is notified not to wait for their sequence numbers.

[00171] When applying order reconstruction across all rings, the transmit window must have the same capacity as all rings combined, $size_w = N \cdot j$, where $size_w$ is the maximum capacity of

indow, N the number of symbols in the original stream (and therefore the number of slots per ring) and j the number of layers used. The potential size of the transmit window, together with its bursty behavior when a delayed quote update releases a substantial amount of pending updates from the window, may justify a mechanism that applies order reconstruction to

achieve chronologic transmission per symbol only, similar to the output of the base layer encoder.

[00172] Like the algorithm depicted in Figure 11A, the layered system introduces delay. The model is essentially a collection of instances of the first algorithm, thus, its worst case delay is the sum of each ring's worst case delay:

$$t_{max} = \sum_{i=0}^{j} \frac{N}{f_{update}(i)}$$

[00173] where tmax is the maximum delay in seconds that the algorithm could introduce for a single quote update, *N* the number of symbols in the original stream, *j* the number of used layers and $f_{update}(i)$ the update frequency of the token in layer *i*.

[00174] In the foregoing specification, the invention has been described with reference to specific embodiments thereof. It will however, be evident that various modification and changes can be made thereto without departing from the broader spirit and scope of the invention as set forth in the appended claims. The specification and drawings are, accordingly, to be regarding in an illustrative rather than a restrictive sense.

What is claimed is:

1. A video signal distribution system comprising:

a source endpoint, wherein the source endpoint scalably encodes video signals into layers to form a layered data stream, wherein the layered data stream comprises a base layer and a set of enhancement layers;

a scalable video coding router, wherein the scalable video coding router forwards the base layer from the source endpoint to a video receiver endpoint, and wherein the scalable video coding router selectively forwards one or more of the set of enhancement layers based upon information received by the scalable video coding router; and

wherein the layered data stream is transmitted via a communications channel configured for a selected network protocol.

2. The video signal distribution system of claim 1, wherein the information received by the scalable video coding router relates to throughput in a portion of the communications channel linking the scalable video coding router and the video receiver endpoint.

3. The video signal distribution system of claim 1, wherein the information received by the scalable video coding router is directly related to video processing capabilities of the video

4. The video signal distribution system of claim 1, wherein the scalable video coding router divides available bandwidth among a plurality of data streams, and wherein the scalable video coding router prioritizes forwarding of the set of enhancement layers.

5. The video signal distribution system of claim 1, wherein each layer of the layered data stream comprises data packets, each of which is encoded with a sequence number and a priority identifier, and wherein the sequence number and the priority identifier for each data packet is based upon a layer to which the packet belongs.

6. The video signal distribution system of claim 2, wherein each layer of the layered data stream comprises data packets, each of which is encoded with a sequence number and a priority identifier,

wherein the sequence number and the priority identifier for each data packet is based upon a layer to which the packet belongs,

wherein the information received by the scalable video coding router is related to throughput in a portion of the communication channel linking the scalable video coding router and the video receiver endpoint, and

wherein the information includes a request for retransmission of one or more packets from the layered data stream.

7. The video signal distribution system of claim 1, wherein the selected network protocol is TCP/IP.

8. A method for transmitting video signals in a video signal distribution system comprising a source endpoint, a scalable video coding router, and a set of video receiver endpoints, the method comprising:

encoding video signals into layers to form a layered data stream, wherein the layered data stream comprises a base layer and a set of enhancement layers;

receiving information relating one or more video receiver endpoints;

forwarding the base layer to the one or more video receiver endpoints; and

selectively forwarding one or more of the set of enhancement layers based upon the information relating to the one or more receiver endpoints;

wherein the layered data stream is transmitted via a communications channel configured for a selected network protocol.

9. The method of claim 8, wherein the information received relates to throughput in a portion of the communications channel linking the scalable video coding router and the one or more video receiver endpoints.

10. The method of claim 8, wherein the information received is directly related to video processing capabilities of the one or more video receiver endpoints.

d of claim 8, further comprising:

dividing available bandwidth among a plurality of data streams; and

prioritizing the set of enhancement layers for the forwarding step.

12. The method of claim 8, wherein each layer of the layered data stream comprises data packets, the method further comprising:

encoding a sequence number and a priority identifier in each of the data packets, wherein the sequence number and the priority identifier for each data packet is based upon a layer to which the packet belongs.

13. The method of claim 9, wherein each layer of the layered data stream comprises data packets, the method further comprising:

encoding a sequence number and a priority identifier in each of the data packets, wherein the sequence number and the priority identifier for each data packet is based upon a layer to which the packet belongs,

wherein the information received is related to throughput in a portion of the communication channel linking the scalable video coding router and the video receiver endpoint, and

wherein the information includes a request for retransmission of one or more data packets from the layered data stream.

14. The method of claim 8, wherein the selected network protocol is TCP/IP.

Abstract

Embodiments include an overlay multicast network. The overlay multicast network may provide a set of features to ensure reliable and timely arrival of multicast data. The embodiments include a congestion control system that may prioritize designated layers of data within a data stream over other layers of the same data stream. Each data stream transmitted over the network may be given an equal share of the bandwidth. Addressing in routing tables maintained by routers in the may utilize summarized addressing based on the difference in location of the router and destination address. Summarization levels may be adjusted to minimize travel distances for packets in the network. Data from high priority data stream layers may also be retransmitted upon request from a destination machine to ensure reliable delivery of data.

SCALE VIDEO CODING, LLC
v. CISCO SYSTEMS, INC.
IPR2024-01371
Ex. 1002, p. 73 of 830

Electronic Patent Application Fee Transmittal				
Application Number:				
Filing Date:				
Title of Invention:	LAYERED MULTICAST AND FAIR BANDWIDTH ALLOCATION AND PACKET PRIORITIZATION			
First Named Inventor/Applicant Name:	Erik Van Zijst			
Filer:	Stephen Y. Liu/Carolyn Bridwell			
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Filed as Small Entity				
Filing Fees for	**Track I Prioritized Examination - Nonprovisional Application under 35 USC 111(a)**			
Description	**Fee Code**	**Quantity**	**Amount**	**Sub-Total in USD($)**
Basic Filing:				
Utility filing Fee (Electronic filing)	4011	1	70	70
Utility Search Fee	2111	1	300	300
Utility Examination Fee	2311	1	360	360
or Prioritized Examination	2817	1	2000	2000
Pages:				
Claims:				
Miscellaneous-Filing:				

Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
Publ. Fee- Early, Voluntary, or Normal	1504	1	0	0
PROCESSING FEE, EXCEPT PROV. APPLS.	2830	1	70	70
Petition:				
Patent-Appeals-and-Interference:				
Post-Allowance-and-Post-Issuance:				
Extension-of-Time:				
Miscellaneous:				
		Total in USD ($)		**2800**

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Deposit Account	500392
Authorized User	LIU, STEPHEN Y
The Director of the USPTO is hereby authorized to charge indicated fees and credit any overpayment as follows: Charge any Additional Fees required under 37 CFR 1.16 (National application filing, search, and examination fees) Charge any Additional Fees required under 37 CFR 1.17 (Patent application and reexamination processing fees)	

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File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Application Data Sheet	US9_ADS_AsFiled.pdf	1561686 a93e48b4a427735e0be86876d93858931f68acaf	no	7
Warnings:					
Information:					
2	Drawings-only black and white line drawings	US9Dwgs_AsFiled.pdf	4541058 654cddd3f3da5561085454 58bb863e87375f6fd7	no	12
Warnings:					
Information:					
3	TrackOne Request	US9Track1Form_AsFiled.pdf	157064 171b890219c1df6e2093d8c2c2ac282f8f02375b	no	2
Warnings:					
Information:					
4	Power of Attorney	Stmt373c.pdf	120940 494dad2734a213a750f124b57d3ad9056fa12214	no	3
Warnings:					
Information:					
5	Power of Attorney	Signed_POAA.pdf	44591 6c8821c19035d5122bd3f81c5105594b153dd74e	no	1
Warnings:					
Information:					
6		US9Application_AsFiled.pdf	10236926 dc899fe6e26bb	yes	70

Multipart Description/PDF files in .zip description

Document Description	Start	End
Specification	1	65
Claims	66	69
Abstract	70	70

Warnings:					
Information:					
7	Fee Worksheet (SB06)	fee-info.pdf	41192 d4855385143083 7bcefcbdf90b82f8aff6615ed8	no	2
Warnings:					
Information:					
			Total Files Size (in bytes):	16703457	

This Acknowledgement Receipt evidences receipt on the noted date by the USPTO of the indicated documents, characterized by the applicant, and including page counts, where applicable. It serves as evidence of receipt similar to a Post Card, as described in MPEP 503.

New Applications Under 35 U.S.C. 111
If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.

National Stage of an International Application under 35 U.S.C. 371
If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.

New International Application Filed with the USPTO as a Receiving Office
If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.

Electronic Acknowledgement Receipt	
EFS ID:	24765743
Application Number:	15010573
International Application Number:	
Confirmation Number:	9731
Title of Invention:	LAYERED MULTICAST AND FAIR BANDWIDTH ALLOCATION AND PACKET PRIORITIZATION
First Named Inventor/Applicant Name:	Erik Van Zijst
Customer Number:	122200
Filer:	Stephen Y. Liu/Carolyn Bridwell
Filer Authorized By:	Stephen Y. Liu
Attorney Docket Number:	GPB1-US9
Receipt Date:	29-JAN-2016
Filing Date:	
Time Stamp:	15:40:25
Application Type:	Utility under 35 USC 111(a)

ion:

nt	yes
Payment Type	Credit Card
Payment was successfully received in RAM	$2800
RAM confirmation Number	2042
Deposit Account	500392
Authorized User	LIU, STEPHEN Y
The Director of the USPTO is hereby authorized to charge indicated fees and credit any overpayment as follows: Charge any Additional Fees required under 37 CFR 1.16 (National application filing, search, and examination fees) Charge any Additional Fees required under 37 CFR 1.17 (Patent application and reexamination processing fees)	

Charge any Additional Fees required under 37 CFR 1.19 (Document supply fees)

Charge any Additional Fees required under 37 CFR 1.20 (Post Issuance fees)

Charge any Additional Fees required under 37 CFR 1.21 (Miscellaneous fees and charges)

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Application Data Sheet	US9_ADS_AsFiled.pdf	1561686 a93e48b4a427735e0be86876d93858931f68acaf	no	7
Warnings:					
Information:					
2	Drawings-only black and white line drawings	US9Dwgs_AsFiled.pdf	4541058 654cddd3f3da5561085454 58bb863e87375f6fd7	no	12
Warnings:					
Information:					
3	TrackOne Request	US9Track1Form_AsFiled.pdf	157064 171b890219c1df6e2093d8c2c2ac282f8f02375b	no	2
Warnings:					
Information:					
4	Power of Attorney	Stmt373c.pdf	120940 494dad2734a213a750f124b57d3ad9056fa12214	no	3
Warnings:					
Information:					
5	Power of Attorney	Signed_POAA.pdf	44591 6c8821c19035d5122bd3f81c5105594b153dd74e	no	1
Warnings:					
Information:					
6		US9Application_AsFiled.pdf	10236926 dc899fe6e26bb	yes	70

Multipart Description/PDF files in .zip description

Document Description	Start	End
Specification	1	65
Claims	66	69
Abstract	70	70

Warnings:					
Information:					
7	Fee Worksheet (SB06)	fee-info.pdf	41192 d4855385143083 7bcefcbdf90b82f8aff6615ed8	no	2
Warnings:					
Information:					
			Total Files Size (in bytes):	16703457	

This Acknowledgement Receipt evidences receipt on the noted date by the USPTO of the indicated documents, characterized by the applicant, and including page counts, where applicable. It serves as evidence of receipt similar to a Post Card, as described in MPEP 503.

New Applications Under 35 U.S.C. 111
If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.

National Stage of an International Application under 35 U.S.C. 371
If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.

New International Application Filed with the USPTO as a Receiving Office
If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.

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Application Data Sheet 37 CFR 1.76	Attorney Docket Number	GPB1-US9
	Application Number	
Title of Invention	LAYERED MULTICAST AND FAIR BANDWIDTH ALLOCATION AND PACKET PRIORITIZATION	

The application data sheet is part of the provisional or nonprovisional application for which it is being submitted. The following form contains the bibliographic data arranged in a format specified by the United States Patent and Trademark Office as outlined in 37 CFR 1.76.
This document may be completed electronically and submitted to the Office in electronic format using the Electronic Filing System (EFS) or the document may be printed and included in a paper filed application.

Secrecy Order 37 CFR 5.2

☐ Portions or all of the application associated with this Application Data Sheet may fall under a Secrecy Order pursuant to 37 CFR 5.2 (Paper filers only. Applications that fall under Secrecy Order may not be filed electronically.)

Inventor Information:

Inventor 1					Remove
Legal Name					
Prefix	**Given Name**	**Middle Name**	**Family Name**	**Suffix**	
	Erik		Van Zijst		
Residence Information (Select One)	● US Residency	Non US Residency	Active US Military Service		
City	Belmont	**State/Province**	CA	**Country of Residence**	US

Mailing Address of Inventor:			
Address 1	c/o Blitz Stream Video, LLC		
Address 2	P.O. Box 250649		
City	Plano	**State/Province**	TX
Postal Code	75025	**Country**	US

All Inventors Must Be Listed - Additional Inventor Information blocks may be generated within this form by selecting the **Add** button. Add

Correspondence Information:

Enter either Customer Number or complete the Correspondence Information section below.
For further information see 37 CFR 1.33(a).

☐ **An Address is being provided for the correspondence Information of this application.**

Customer Number	122200

dd Email | Remove Email

Application Information:

Title of the Invention	LAYERED MULTICAST AND FAIR BANDWIDTH ALLOCATION AND PACKET PRIORITIZATION		
Attorney Docket Number	GPB1-US9	Small Entity Status Claimed	☒
Application Type	Nonprovisional		
Subject Matter	Utility		
Total Number of Drawing Sheets (if any)	12	Suggested Figure for Publication (if any)	

Filing By Reference :

EFS Web 2.2.11

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Application Data Sheet 37 CFR 1.76	Attorney Docket Number	GPB1-US9
	Application Number	
Title of Invention	LAYERED MULTICAST AND FAIR BANDWIDTH ALLOCATION AND PACKET PRIORITIZATION	

Only complete this section when filing an application by reference under 35 U.S.C. 111(c) and 37 CFR 1.57(a). Do not complete this section if application papers including a specification and any drawings are being filed. Any domestic benefit or foreign priority information must be provided in the appropriate section(s) below (i.e., "Domestic Benefit/National Stage Information" and "Foreign Priority Information").

For the purposes of a filing date under 37 CFR 1.53(b), the description and any drawings of the present application are replaced by this reference to the previously filed application, subject to conditions and requirements of 37 CFR 1.57(a).

Application number of the previously filed application	Filing date (YYYY-MM-DD)	Intellectual Property Authority or Country

Publication Information:

☐ Request Early Publication (Fee required at time of Request 37 CFR 1.219)

☐ **Request Not to Publish.** I hereby request that the attached application not be published under 35 U.S.C. 122(b) and certify that the invention disclosed in the attached application **has not and will not** be the subject of an application filed in another country, or under a multilateral international agreement, that requires publication at eighteen months after filing.

Representative Information:

Representative information should be provided for all practitioners having a power of attorney in the application. Providing this information in the Application Data Sheet does not constitute a power of attorney in the application (see 37 CFR 1.32).
Either enter Customer Number or complete the Representative Name section below. If both sections are completed the customer Number will be used for the Representative Information during processing.

Please Select One:	● Customer Number	US Patent Practitioner	○ Limited Recognition (37 CFR 11.9)
Customer Number	122200		

Domestic Benefit/National Stage Information:

This section allows for the applicant to either claim benefit under 35 U.S.C. 119(e), 120, 121, or 365(c) or indicate National Stage
Providing this information in the application data the specific reference required
37 CFR 1.78.
rrent application, please leave the application number blank.

Prior Application Status	Pending		Remove
Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)
	Division of	13619969	2012-09-14
Prior Application Status	Patented		Remove

Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)	Patent Number	Issue Date (YYYY-MM-DD)
13619969	Continuation of	12486656	2009-06-17	8514718	2013-08-20
Prior Application Status	Patented				Remove

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Application Data Sheet 37 CFR 1.76	Attorney Docket Number	GPB1-US9
	Application Number	
Title of Invention	LAYERED MULTICAST AND FAIR BANDWIDTH ALLOCATION AND PACKET PRIORITIZATION	

Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)	Patent Number	Issue Date (YYYY-MM-DD)
12486656	Division of	11342167	2006-01-26	7733868	2010-06-08

Prior Application Status	Expired	Remove

Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)
11342167	Claims benefit of provisional	60647601	2005-01-26

Additional Domestic Benefit/National Stage Data may be generated within this form by selecting the **Add** button. Add

Foreign Priority Information:

This section allows for the applicant to claim priority to a foreign application. Providing this information in the application data sheet constitutes the claim for priority as required by 35 U.S.C. 119(b) and 37 CFR 1.55(d). When priority is claimed to a foreign application that is eligible for retrieval under the priority document exchange program (PDX)[i] the information will be used by the Office to automatically attempt retrieval pursuant to 37 CFR 1.55(h)(1) and (2). Under the PDX program, applicant bears the ultimate responsibility for ensuring that a copy of the foreign application is received by the Office from the participating foreign intellectual property office, or a certified copy of the foreign priority application is filed, within the time period specified in 37 CFR 1.55(g)(1).

Remove

Application Number	Country[i]	Filing Date (YYYY-MM-DD)	Access Code[i] (if applicable)

Additional Foreign Priority Data may be generated within this form by selecting the **Add** button. Add

Statement under 37 CFR 1.55 or 1.78 for AIA (First Inventor to File) Transition Applications

This application (1) claims priority to or the benefit of an application filed before March 16, 2013 and (2) also at any time, a claim to a claimed invention that filing date on or after March

NOTE: By providing this statement under 37 CFR 1.55 or 1.78, this application, with a filing date on or after March 16, 2013, will be examined under the first inventor to file provisions of the AIA.

Authorization to Permit Access:

☒ Authorization to Permit Access to the Instant Application by the Participating Offices

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Application Data Sheet 37 CFR 1.76	Attorney Docket Number	GPB1-US9
	Application Number	
Title of Invention	LAYERED MULTICAST AND FAIR BANDWIDTH ALLOCATION AND PACKET PRIORITIZATION	

If checked, the undersigned hereby grants the USPTO authority to provide the European Patent Office (EPO), the Japan Patent Office (JPO), the Korean Intellectual Property Office (KIPO), the World Intellectual Property Office (WIPO), and any other intellectual property offices in which a foreign application claiming priority to the instant patent application is filed access to the instant patent application. See 37 CFR 1.14(c) and (h). This box should not be checked if the applicant does not wish the EPO, JPO, KIPO, WIPO, or other intellectual property office in which a foreign application claiming priority to the instant patent application is filed to have access to the instant patent application.

In accordance with 37 CFR 1.14(h)(3), access will be provided to a copy of the instant patent application with respect to: 1) the instant patent application-as-filed; 2) any foreign application to which the instant patent application claims priority under 35 U.S.C. 119(a)-(d) if a copy of the foreign application that satisfies the certified copy requirement of 37 CFR 1.55 has been filed in the instant patent application; and 3) any U.S. application-as-filed from which benefit is sought in the instant patent application.

In accordance with 37 CFR 1.14(c), access may be provided to information concerning the date o f filing this Authorization.

Applicant Information:

Providing assignment information in this section does not substitute for compliance with any requirement of part 3 of Title 37 of CFR to have an assignment recorded by the Office.

Applicant 1 — Remove

If the applicant is the inventor (or the remaining joint inventor or inventors under 37 CFR 1.45), this section should not be completed. The information to be provided in this section is the name and address of the legal representative who is the applicant under 37 CFR 1.43; or the name and address of the assignee, person to whom the inventor is under an obligation to assign the invention, or person who otherwise shows sufficient proprietary interest in the matter who is the applicant under 37 CFR 1.46. If the applicant is an applicant under 37 CFR 1.46 (assignee, person to whom the inventor is obligated to assign, or person who otherwise shows sufficient proprietary interest) together with one or more joint inventors, then the joint inventor or inventors who are also the applicant should be identified in this section.

Clear

● Assignee	Legal Representative under 35 U.S.C. 117	Joint Inventor
Person to whom the inventor is obligated to assign.	Person who shows sufficient proprietary interest	

If applicant is the legal representative, indicate the authority to file the patent application, the inventor is:

Name of the Deceased or Legally Incapacitated Inventor :

Organization check here. ☒

Organization Name	Blitz Stream Video, LLC		
Mailing Address Information For Applicant:			
Address 1	P.O. Box 250649		
Address 2			
City	Plano	**State/Province**	TX
Country	US	Postal Code	75025
Phone Number		Fax Number	

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Application Data Sheet 37 CFR 1.76	Attorney Docket Number	GPB1-US9
	Application Number	
Title of Invention	LAYERED MULTICAST AND FAIR BANDWIDTH ALLOCATION AND PACKET PRIORITIZATION	

Email Address	

Additional Applicant Data may be generated within this form by selecting the Add button. Add

Assignee Information including Non-Applicant Assignee Information:

Providing assignment information in this section does not subsitute for compliance with any requirement of part 3 of Title 37 of CFR to have an assignment recorded by the Office.

Assignee 1

Complete this section if assignee information, including non-applicant assignee information, is desired to be included on the patent application publication . An assignee-applicant identified in the "Applicant Information" section will appear on the patent application publication as an applicant. For an assignee-applicant, complete this section only if identification as an assignee is also desired on the patent application publication.

Remove

If the Assignee or Non-Applicant Assignee is an Organization check here. ☐

Prefix	**Given Name**	Middle Name	**Family Name**	Suffix

Mailing Address Information For Assignee including Non-Applicant Assignee:

Address 1			
Address 2			
City		**State/Province**	
Country i		Postal Code	
Phone Number		Fax Number	
Email Address			

Additional Assignee or Non-Applicant Assignee Data may be generated within this form by selecting the Add button. Add

Remove

NOTE: This form must be signed in accordance with 37 CFR 1.33. See 37 CFR 1.4 for signature requirements and certifications.

Signature	/Stephen Y. Liu/			Date (YYYY-MM-DD)	2016-01-29
First Name	Stephen	Last Name	Liu	Registration Number	62883

Additional Signature may be generated within this form by selecting the Add button. Add

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Application Data Sheet 37 CFR 1.76	Attorney Docket Number	GPB1-US9
	Application Number	
Title of Invention	LAYERED MULTICAST AND FAIR BANDWIDTH ALLOCATION AND PACKET PRIORITIZATION	

This collection of information is required by 37 CFR 1.76. The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.14. This collection is estimated to take 23 minutes to complete, including gathering, preparing, and submitting the completed application data sheet form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**

Privacy Act Statement

The Privacy Act of 1974 (P.L. 93-579) requires that you be given certain information in connection with your submission of the attached form related to a patent application or patent. Accordingly, pursuant to the requirements of the Act, please be advised that: (1) the general authority for the collection of this information is 35 U.S.C. 2(b)(2); (2) furnishing of the information solicited is voluntary; and (3) the principal purpose for which the information is used by the U.S. Patent and Trademark Office is to process and/or examine your submission related to a patent application or patent. If you do not furnish the requested information, the U.S. Patent and Trademark Office may not be able to process and/or examine your submission, which may result in termination of proceedings or abandonment of the application or expiration of the patent.

The information provided by you in this form will be subject to the following routine uses:

1. The information on this form will be treated confidentially to the extent allowed under the Freedom of Information Act (5 U.S.C. 552) and the Privacy Act (5 U.S.C. 552a). Records from this system of records may be disclosed to the Department of Justice to determine whether the Freedom of Information Act requires disclosure of these records.

2. A record from this system of records may be disclosed, as a routine use, in the course of presenting evidence to a court, magistrate, or administrative tribunal, including disclosures to opposing counsel in the course of settlement negotiations.

3. A record in this system of records may be disclosed, as a routine use, to a Member of Congress submitting a request involving an individual, to whom the record pertains, when the individual has requested assistance from the Member with respect to the subject matter of the record.

4. A record in this system of records may be disclosed, as a routine use, to a contractor of the Agency having need for the information in order to perform a contract. Recipients of information shall be required to comply with the requirements of the Privacy Act of 1974, as amended, pursuant to 5 U.S.C. 552a(m).

5. A record related to an International Application filed under the Patent Cooperation Treaty in this system of records may be disclosed, as a routine use, to the International Bureau of the World Intellectual Property Organization, pursuant to the Patent C o o p eration Treaty.

6. A record in this system of records may be disclosed, as a routine use, to another federal agency for purposes of National Security review (35 U.S.C. 181) and for review pursuant to the Atomic Energy Act (42 U.S.C. 218(c)).

7. A record from this system of records may be disclosed, as a routine use, to the Administrator, General Services, or his/her designee, during an inspection of records conducted by GSA as part of that agency's responsibility to recommend improvements in records management practices and programs, under authority of 44 U.S.C. 2904 and 2906. Such disclosure shall be made in accordance with the GSA regulations governing inspection of records for this purpose, and any other relevant (i.e., GSA or Commerce) directive. Such disclosure shall not be used to make determinations about individuals.

8. A record from this system of records may be disclosed, as a routine use, to the public after either publication of the application pursuant to 35 U.S.C. 122(b) or issuance of a patent pursuant to 35 U.S.C. 151. Further, a record may be disclosed, subject to the limitations of 37 [illegible] use, to the public if the record was filed in an application which became abandoned or in which the proceedings were terminated and which application is referenced by either a published application, an application open to public inspections or an issued patent.

9. A record from this system of records may be disclosed, as a routine use, to a Federal, State, or local law enforcement agency, if the USPTO becomes aware of a violation or potential violation of law or regulation.

EFS Web 2.2.11

FIG. 1

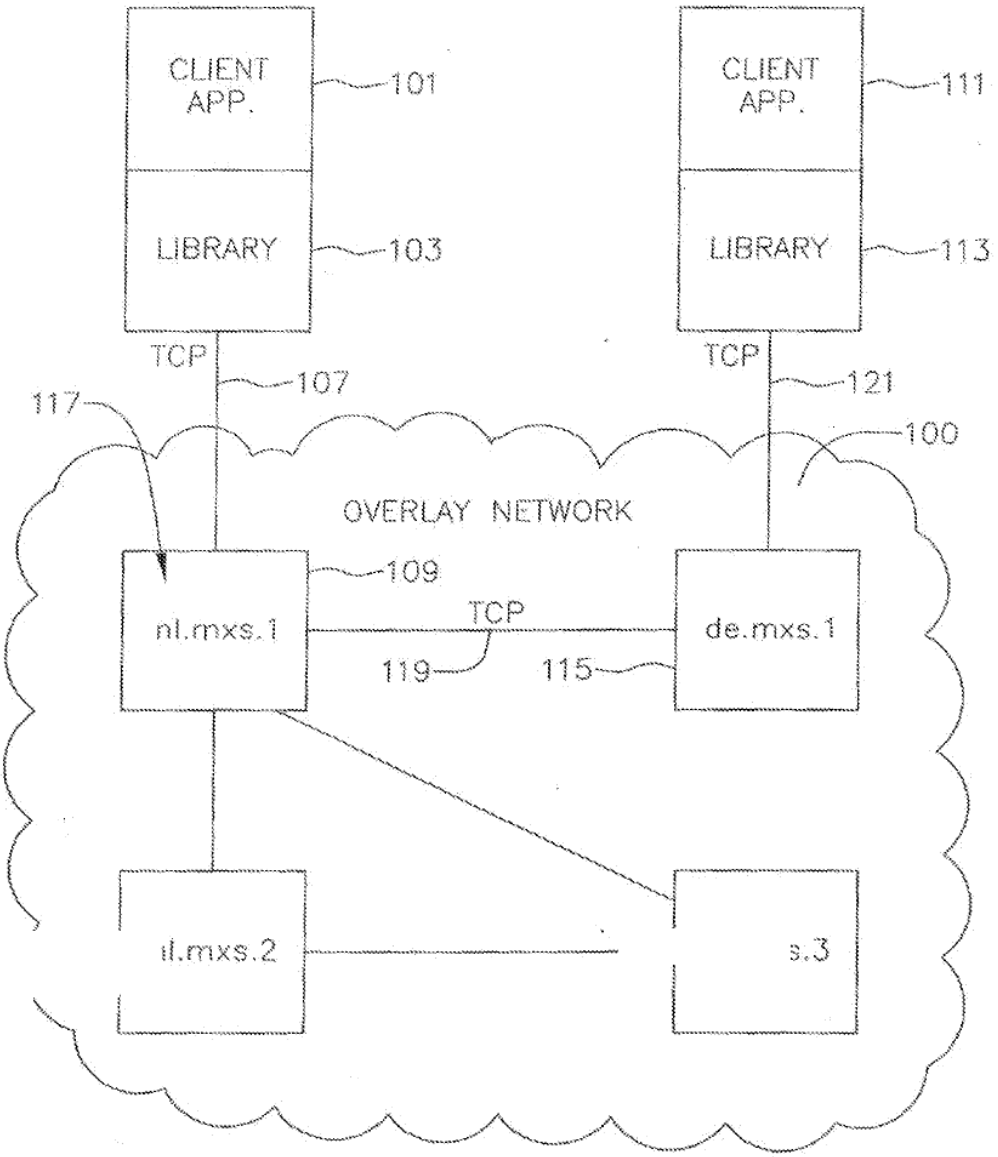

CLIENT APP.
101
LIBRARY
103
TCP
107
CLIENT APP.
111
LIBRARY
113
TCP
121
117
100
OVERLAY NETWORK
109
nl.mxs.1
TCP
119
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de.mxs.1
il.mxs.2
s.3

FIG. 2

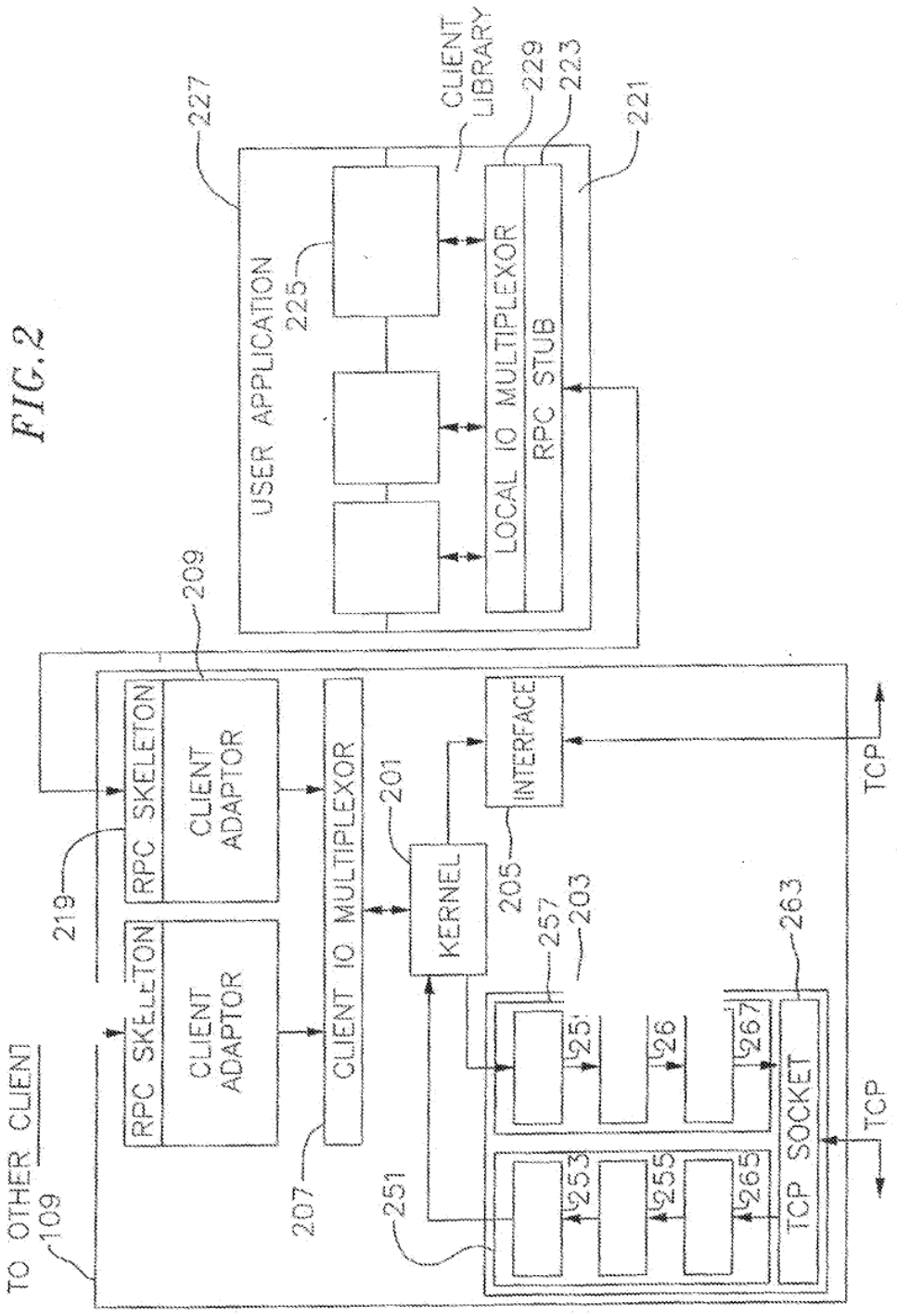

TO OTHER CLIENT
109
RPC SKELETON
CLIENT ADAPTOR
RPC SKELETON
CLIENT ADAPTOR
219
209
207
CLIENT IO MULTIPLEXOR
201
KERNEL
205
INTERFACE
TCP
TCP
251
253
255
265
257
203
251
261
263
TCP SOCKET
227
USER APPLICATION
225
CLIENT LIBRARY
229
223
221
LOCAL IO MULTIPLEXOR
RPC STUB

FIG. 3A

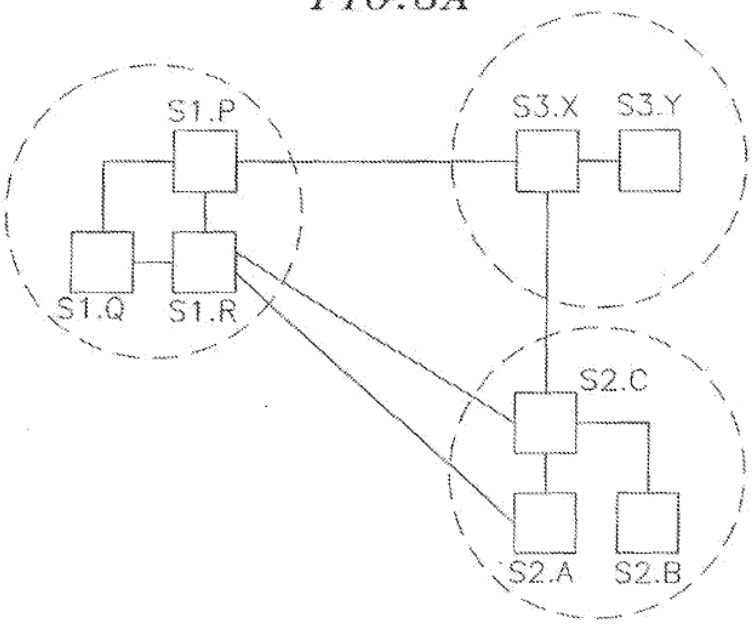


FIG. 3B

ROUTING TABLE OF NODE S2.C

DEST	HOP	COST
S2.C	—	0
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S2.B	S2.B	1
S1.*	S1.R	1
S3.*	S3.X	1

FIG. 4

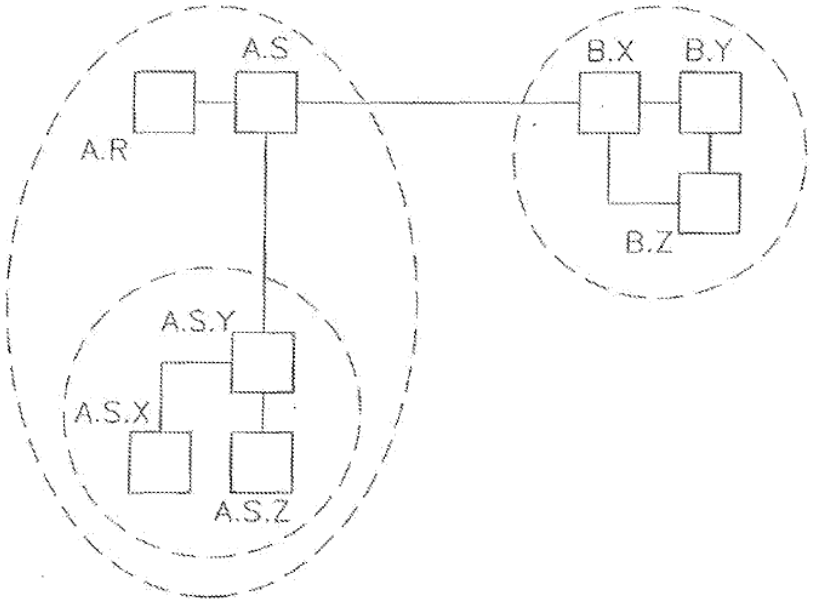


FIG. 5A

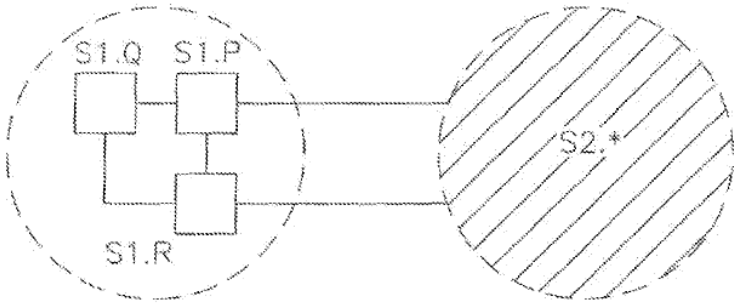


FIG. 5B

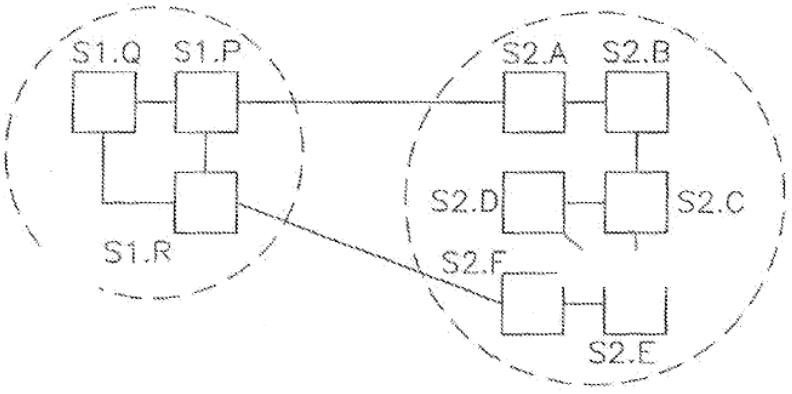


FIG. 6

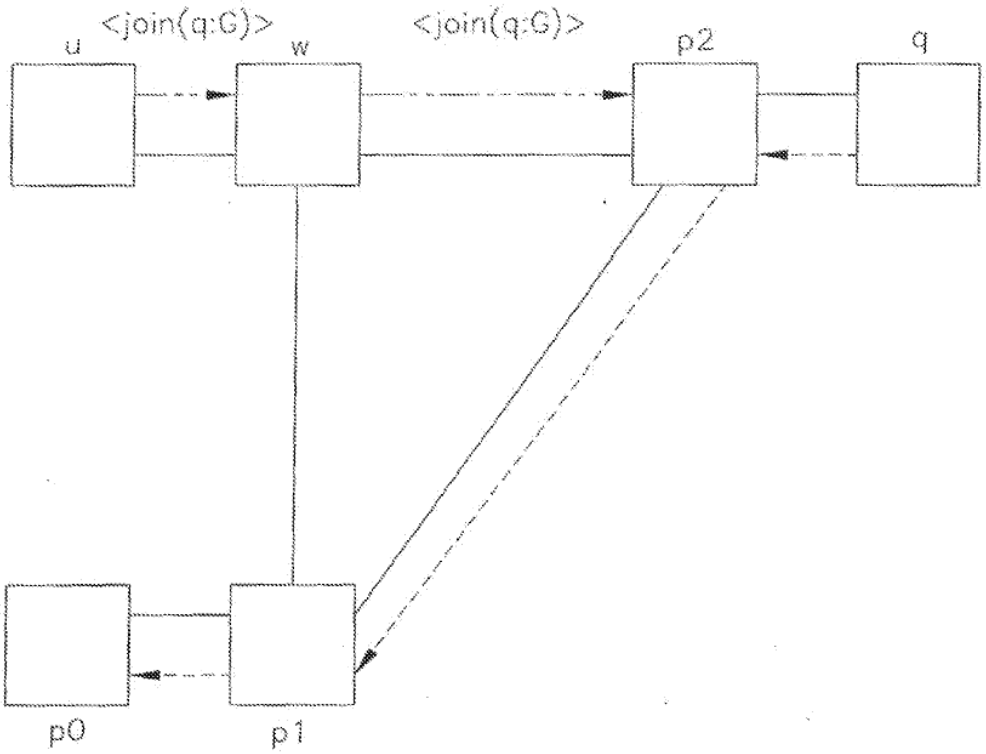


FIG. 7

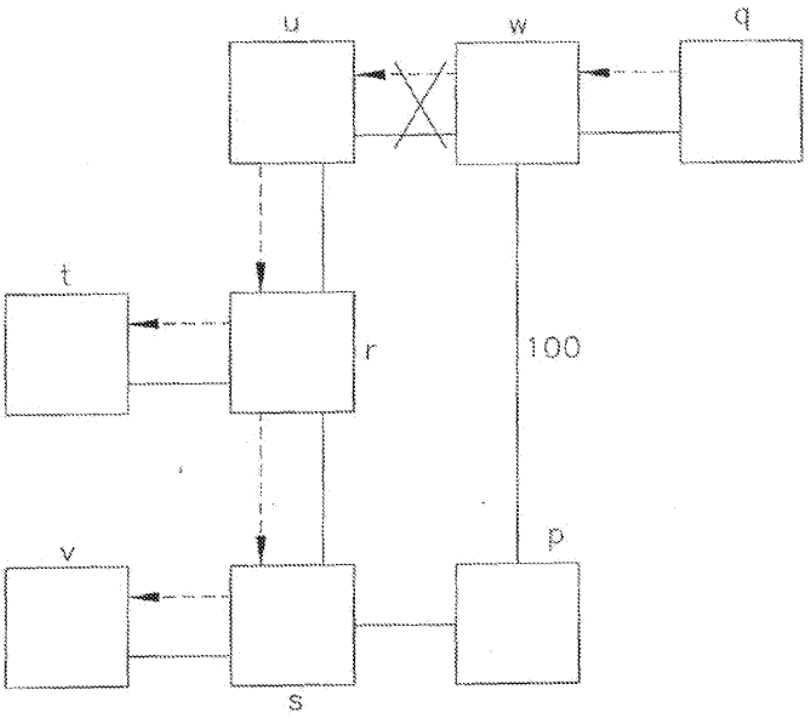


FIG. 8A

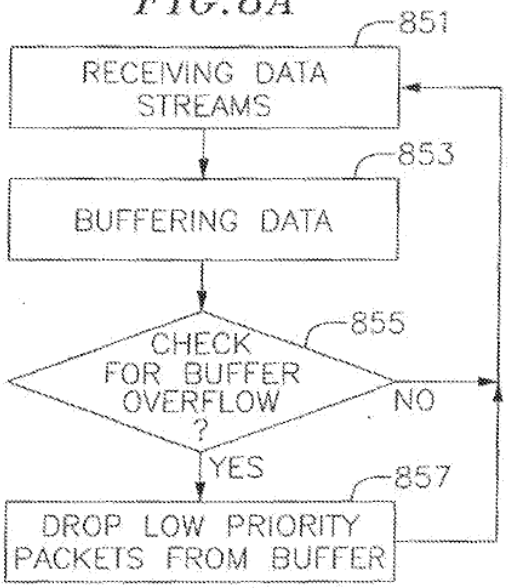
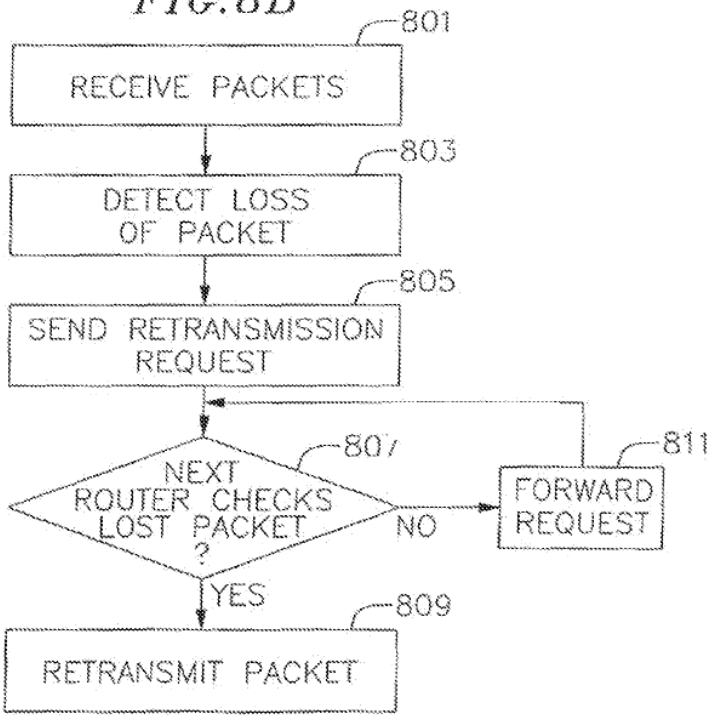

851
RECEIVING DATA STREAMS
853
BUFFERING DATA
855
CHECK FOR BUFFER OVERFLOW ?
NO
YES
857
DROP LOW PRIORITY PACKETS FROM BUFFER

FIG. 8B


801
RECEIVE PACKETS
803
DETECT LOSS OF PACKET
805
SEND RETRANSMISSION REQUEST
807
NEXT ROUTER CHECKS LOST PACKET ?
NO
811
FORWARD REQUEST
YES
809
RETRANSMIT PACKET

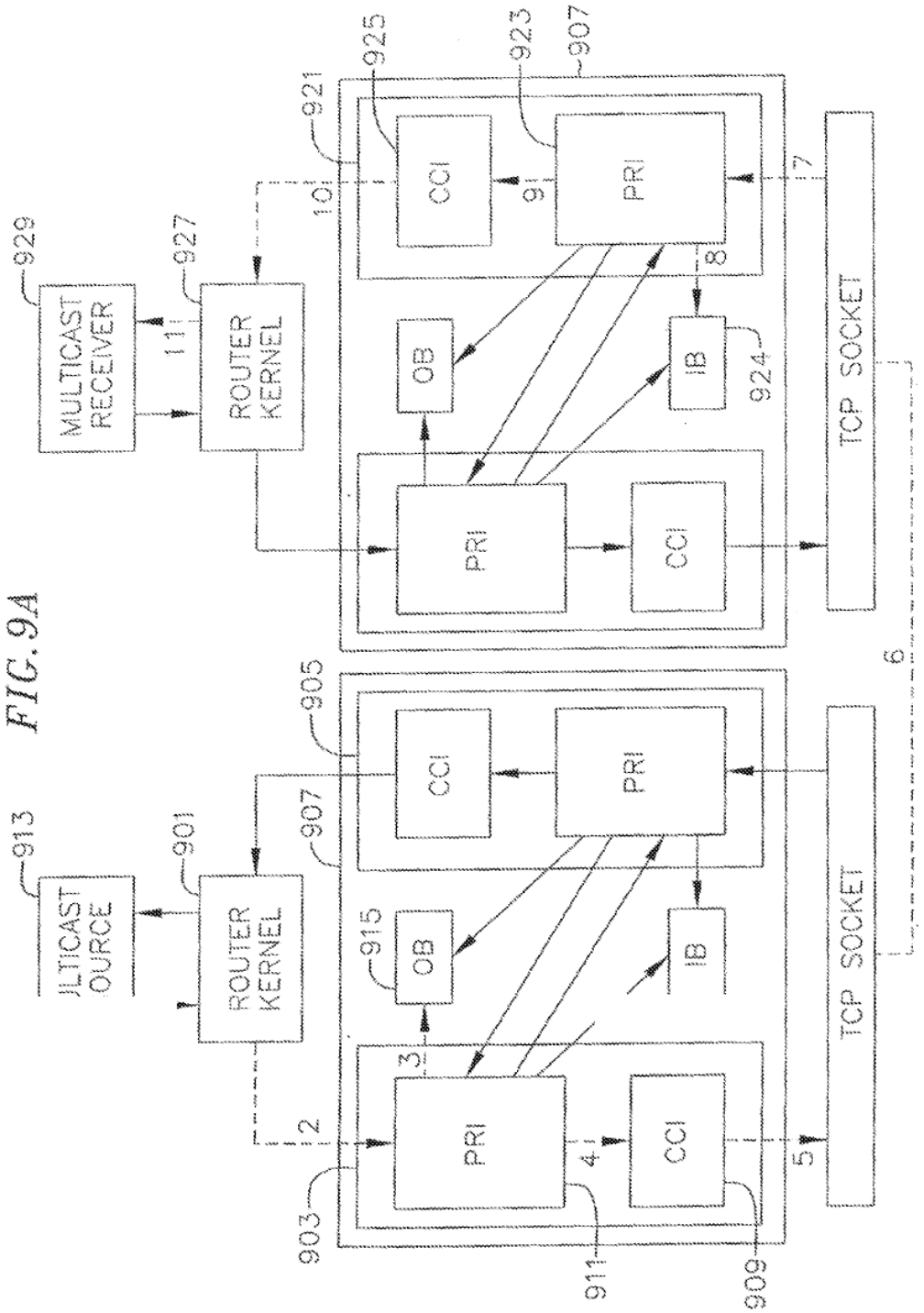

MULTICAST RECEIVER
ROUTER KERNEL
MULTICAST SOURCE
ROUTER KERNEL
CCI
PRI
OB
IB
TCP SOCKET
CCI
PRI
OB
IB
TCP SOCKET
929
927
11
10
921
925
923
907
9
8
924
7
6
913
901
907
905
915
3
2
4
5
903
911
909

FIG. 9A

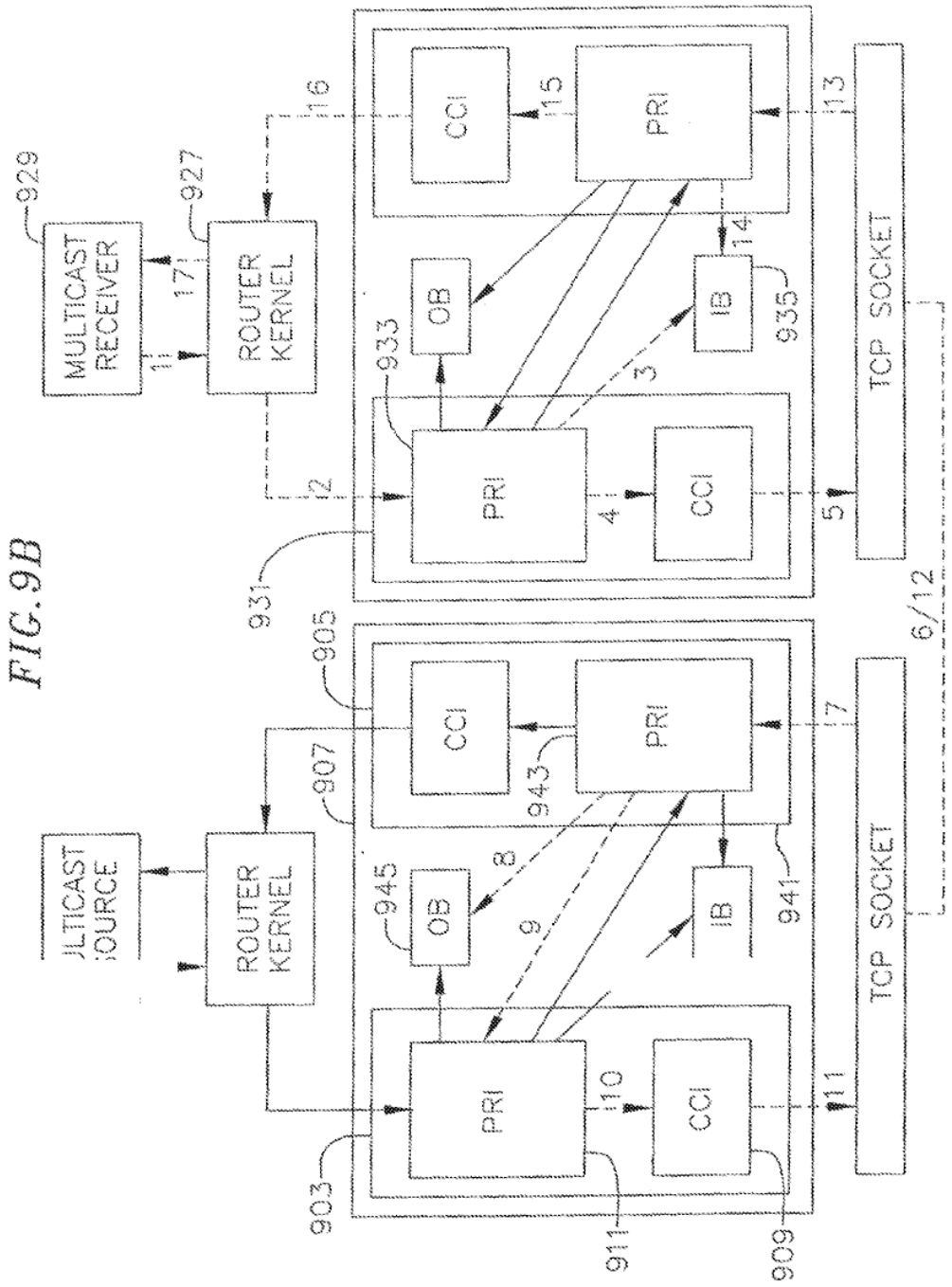
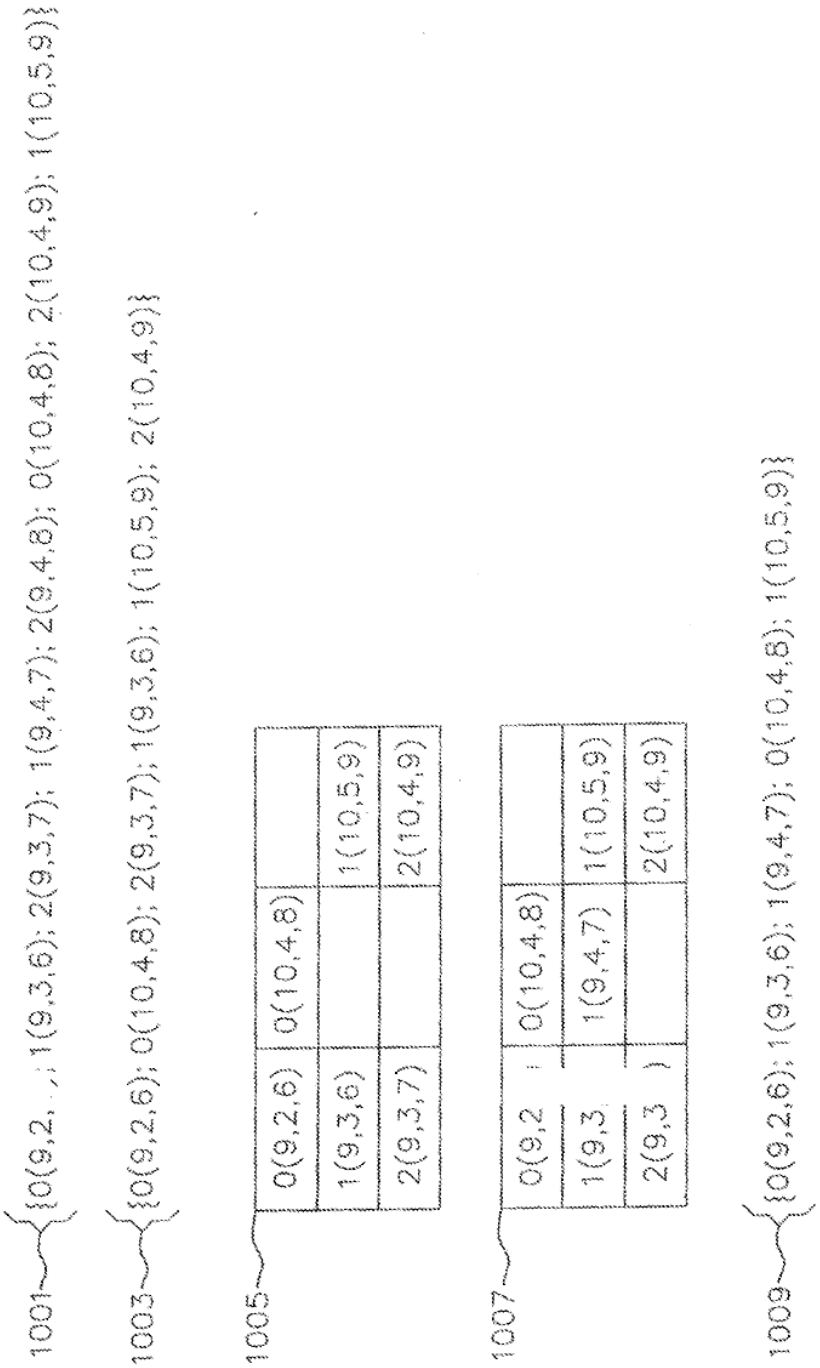

MULTICAST RECEIVER
MULTICAST SOURCE
ROUTER KERNEL
ROUTER KERNEL
TCP SOCKET
TCP SOCKET
CCI
PRI
OB
IB
929
927
931
933
935
905
907
943
945
941
903
911
909
6/12

FIG. 9B

FIG. 10


1001
{0(9,2, ,); 1(9,3,6); 2(9,3,7); 1(9,4,7); 2(9,4,8); 0(10,4,8); 2(10,4,9); 1(10,5,9)}
1003
{0(9,2,6); 0(10,4,8); 2(9,3,7); 1(9,3,6); 1(10,5,9); 2(10,4,9)}
1005
0(9,2,6)
0(10,4,8)
1(9,3,6)
1(10,5,9)
2(9,3,7)
2(10,4,9)
1007
0(9,2
0(10,4,8)
1(9,3
1(9,4,7)
1(10,5,9)
2(9,3)
2(10,4,9)
1009
{0(9,2,6); 1(9,3,6); 1(9,4,7); 0(10,4,8); 1(10,5,9)}

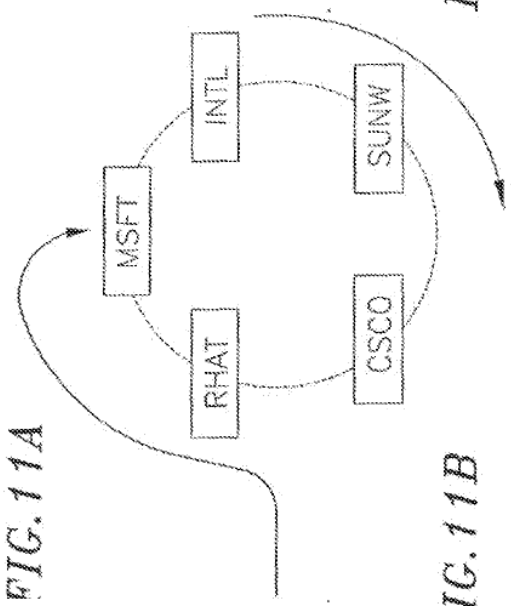


FIG. 11A

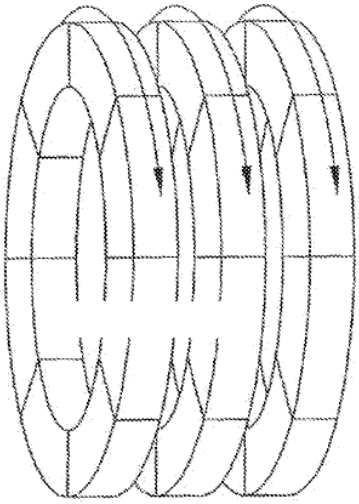


FIG. 11B

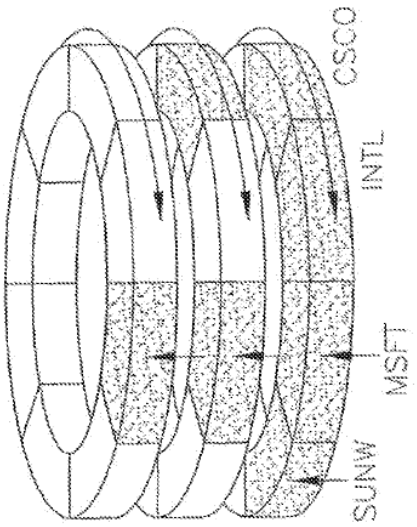


FIG. 11C

PTO/AIA/96 (08-12)
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U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE
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STATEMENT UNDER 37 CFR 3.73(c)

Applicant/Patent Owner: Blitz Stream Video, LLC

Application No./Patent No.: Filed herewith Filed/Issue Date: January 29, 2016

Titled: Layered Multicast and Fair Bandwidth Allocation and Packet Prioritization

Blitz Stream Video, LLC, a corporation

(Name of Assignee) (Type of Assignee, e.g., corporation, partnership, university, government agency, etc.)

states that, for the patent application/patent identified above, it is (choose **one** of options 1, 2, 3 or 4 below):

1. ☑ The assignee of the entire right, title, and interest.

2. ☐ An assignee of less than the entire right, title, and interest (check applicable box):

   ☐ The extent (by percentage) of its ownership interest is _____%. Additional Statement(s) by the owners holding the balance of the interest must be submitted to account for 100% of the ownership interest.

   ☐ There are unspecified percentages of ownership. The other parties, including inventors, who together own the entire right, title and interest are:

   Additional Statement(s) by the owner(s) holding the balance of the interest must be submitted to account for the entire right, title, and interest.

3. ☐ The assignee of an undivided interest in the entirety (a complete assignment from one of the joint inventors was made). The other parties, including inventors, who together own the entire right, title, and interest are:

   Additional Statement(s) by the owner(s) holding the balance of the interest must be submitted to account for the entire right, title, and interest.

4. ☐ The recipient, via a court proceeding or the like (*e.g.*, bankruptcy, probate), of an undivided interest in the entirety (a complete transfer of ownership interest was made). The certified document(s) showing the transfer is attached.

The interest identified in option 1, 2 or 3 above (not option 4) is evidenced by either (choose **one** of options A or B below):

A. ☐ An assignment from the inventor(s) of the patent application/patent identified above. The assignment was recorded in the United States Patent and Trademark Office at Reel _____, Frame _____, or for which a copy thereof is attached.

B. ☑ A chain of title from the inventor(s), of the patent application/patent identified above, to the current assignee as follows:

1. From: Erik Van Zijst To: Internet Broadcasting Corporation

   ument was recorded in the United States Patent anc at

   el 022738, Frame 0587, or for which a copy thereof is attached.

2. From: Internet Broadcasting Corporation To: Blitz Stream Video, LLC

   The document was recorded in the United States Patent and Trademark Office at

   Reel 028159, Frame 0105, or for which a copy thereof is attached.

This collection of information is required by 37 CFR 3.73(b). The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.11 and 1.14. This collection is estimated to take 12 minutes to complete, including gathering, preparing, and submitting the completed application form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450**.

If you need assistance in completing the form, call 1-800-PTO-9199 and select option 2.

PTO/AIA/96 (08-12)
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U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE
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STATEMENT UNDER 37 CFR 3.73(c)

3. From: _____ To: _____

The document was recorded in the United States Patent and Trademark Office at

Reel _____, Frame _____, or for which a copy thereof is attached.

4. From: _____ To: _____

The document was recorded in the United States Patent and Trademark Office at

Reel _____, Frame _____, or for which a copy thereof is attached.

5. From: _____ To: _____

The document was recorded in the United States Patent and Trademark Office at

Reel _____, Frame _____, or for which a copy thereof is attached.

6. From: _____ To: _____

The document was recorded in the United States Patent and Trademark Office at

Reel _____, Frame _____, or for which a copy thereof is attached.

☐ Additional documents in the chain of title are listed on a supplemental sheet(s).

☑ As required by 37 CFR 3.73(c)(1)(i), the documentary evidence of the chain of title from the original owner to the assignee was, or concurrently is being, submitted for recordation pursuant to 37 CFR 3.11.

[NOTE: A separate copy (i.e., a true copy of the original assignment document(s)) must be submitted to Assignment Division in accordance with 37 CFR Part 3, to record the assignment in the records of the USPTO. See MPEP 302.08]

The undersigned (whose title is supplied below) is authorized to act on behalf of the assignee.

/Stephen Y. Liu/	January 29, 2016
Signature	Date
Stephen Y. Liu	62,883
Printed or Typed Name	Title or Registration Number

Privacy Act Statement

The **Privacy Act of 1974 (P.L. 93-579)** requires that you be given certain information in connection with your submission of the attached form related to a patent application or patent. Accordingly, pursuant to the requirements of the Act, please be advised that: (1) the general authority for the collection of this information is 35 U.S.C. 2(b)(2); (2) furnishing of the information solicited is voluntary; and (3) the principal purpose for which the information is used by the U.S. Patent and Trademark Office is to process and/or examine your submission related to a patent application or patent. If you do not furnish the requested information, the U.S. Patent and Trademark Office may not be able to process and/or examine your submission, which may result in termination of proceedings or abandonment of the application or expiration of the patent.

The information provided by you in this form will be subject to the following routine uses:

1. The information on this form will be treated confidentially to the extent allowed under the Freedom of Information Act (5 U.S.C. 552) and the Privacy Act (5 U.S.C 552a). Records from this system of records may be disclosed to the Department of Justice to determine whether disclosure of these records is required by the Freedom of Information Act.
2. A record from this system of records may be disclosed, as a routine use, in the course of presenting evidence to a court, magistrate, or administrative tribunal, including disclosures to opposing counsel in the course of settlement negotiations.
3. A record in this system of records may be disclosed, as a routine use, to a Member of Congress submitting a request involving an individual, to whom the record pertains, when the individual has requested assistance from the Member with respect to the subject matter of the record.
4. A record in this system of records may be disclosed, as a routine use, to a contractor of the Agency having need for the information in order to perform a contract. Recipients of information shall be required to comply with the requirements of the Privacy Act of 1974, as amended, pursuant to 5 U.S.C. 552a(m).
5. A record related to an International Application filed under the Patent Cooperation Treaty in this system of records may be disclosed, as a routine use, to the International Bureau of the World Intellectual Property Organization, pursuant to the Patent Cooperation Treaty.
6. A record in this system of records may be disclosed, as a routine use, to another federal agency for purposes of National Security review (35 U.S.C. 181) and for review pursuant to the Atomic Energy Act (42 U.S.C. 218(c)).
7. A record from this system of records may be disclosed, as a routine use, to the Administrator, General Services, or his/her designee, during an inspection of records conducted by GSA as part of that agency's responsibility to recommend improvements in records management practices and programs, under authority of 44 U.S.C. 2904 and 2906. Such disclosure shall be made in accordance with the GSA regulations governing inspection of records for this purpose, and any other relevant (*i.e.*, GSA or Commerce) directive. Such disclosure shall not be used to make determinations about individuals.
8. A record from this system of records may be disclosed, as a routine use, to the public after either publication of the application pursuant to 35 U.S.C. 122(b) or issuance of a patent pursuant to 35 U.S.C. 151. Further, a record may be disclosed, subject to the limitations of 37 CFR 1.14, as a routine use, to the public if the record was filed in an application which became abandoned or in which the proceedings were terminated and which application is referenced by either a published application, an application open to public inspection or an issued patent.
9. A record from this system of records may be disclosed, as a routine use, to a Federal, State, or local law enforcement agency, if the USPTO becomes aware of a violation or potential violation of law or regulation.

PATENT ASSIGNMENT COVER SHEET

Electronic Version v1.1
Stylesheet Version v1.2

EPAS ID: PAT3718343

SUBMISSION TYPE:	NEW ASSIGNMENT
NATURE OF CONVEYANCE:	ASSIGNMENT

CONVEYING PARTY DATA

Name	Execution Date
INTERNET BROADCASTING CORPORATION B.V.	05/06/2012

RECEIVING PARTY DATA

Name:	BLITZ STREAM VIDEO, LLC
Street Address:	2101 CEDAR SPRINGS ROAD
City:	DALLAS
State/Country:	TEXAS
Postal Code:	75201

PROPERTY NUMBERS Total: 1

Property Type	Number
Application Number:	15010573

CORRESPONDENCE DATA

Fax Number: (972)367-2002

Correspondence will be sent to the e-mail address first; if that is unsuccessful, it will be sent using a fax number, if provided; if that is unsuccessful, it will be sent via US Mail.

Phone: 9723672001
Email: cbridwell@cclaw.com
Correspondent Name: CARSTENS & CAHOON LLP
Address Line 1: PO BOX 802334
Address Line 4: DALLAS, TEXAS 75380

ATTORNEY DOCKET NUMBER:	GPB1-US9
NAME OF SUBMITTER:	STEPHEN Y. LIU
	/STEPHEN Y. LIU/
DATE SIGNED:	01/29/2016
	This document serves as an Oath/Declaration (37 CFR 1.63).

Total Attachments: 4
source=2012.05.04 Internet Broadcasting Corp to Blitz#page1.tif
source=2012.05.04 Internet Broadcasting Corp to Blitz#page2.tif
source=2012.05.04 Internet Broadcasting Corp to Blitz#page3.tif
source=2012.05.04 Internet Broadcasting Corp to Blitz#page4.tif

EXHIBIT A
Assignment of Patent Rights

For good and valuable consideration, the receipt of which is hereby acknowledged, Internet Broadcasting Corporation B.V., a corporation having an address at Sparrenlaan 27, 211 AE Aerdenhout, Netherlands, ("Assignor"), does hereby sell, assign, transfer, and convey unto Blitz Stream Video, LLC an, LLC, having an address at 2101 Cedar Springs Rd., Dallas, Texas 75201 ("Assignee"), or its designees, all right, title, and interest that exist today and may exist in the future in and to any and all of the following:

(a) the patent(s) and patent application(s) listed in the table below (the ***"Patents"***);

Patent/Application Number	Country	Filing Date	Title and first named inventor
60/647,601	US	01-26-2005	Live multicast on wide area IP networks (E. van Zijst)
7,733,868 (11/342,167)	US	01-26-2006 (issue date: 06-08-2010)	Layered multicast and fair bandwidth allocation and packet prioritization (E. van Zijst)
12/486,635	US	06-17-2009	Layered multicast and fair bandwidth allocation and packet prioritization (E. van Zijst)
12/486,667	US	06-17-2009	Layered multicast and fair bandwidth allocation and packet prioritization (E. van Zijst)
12/486,656	US	06-17-2009	Layered multicast and fair bandwidth allocation and packet prioritization (E. van Zijst)
12/486,615	US	06-17-2009	Layered multicast and fair bandwidth allocation and packet prioritization (E. van Zijst)

patents and patent applications: (i) to which any of or indirectly claims priority, (ii) for which any of the Patents directly or indirectly forms a basis for priority, and/or (iii) that were co-owned applications that incorporate by reference, or are incorporated by reference into, the Patents;

(c) all reissues, reexaminations, extensions, continuations, continuations-in-part, continuing prosecution applications, requests for continuing examinations, divisions, registrations of any item in any of the foregoing categories (a) and (b);

(d) all foreign patents, patent applications, and counterparts relating to any item in any of the foregoing categories (a) through (c), including, without limitation, certificates of invention, utility models, industrial design protection, design patent protection, and other governmental grants or issuances;

(e) all inventions, invention disclosures, and discoveries described in any of the Patents that: (i) are included in any claim in the Patents, (ii) are subject matter capable of being reduced to a patent claim in a reissue or reexamination proceedings brought on any of the Patents, and/or (iii) could have been included as a claim in any of the Patents;

(f) all rights to apply in any or all countries of the world for patents, certificates of invention, utility models, industrial design protections, design patent protections, or other governmental grants or issuances of any type related to any item in any of the foregoing categories (a) through (e), including, without limitation, under the Paris Convention for the Protection of Industrial Property, the International Patent Cooperation Treaty, or any other convention, treaty, agreement, or understanding;

(g) all causes of action (whether known or unknown or whether currently pending, filed, or otherwise) and other enforcement rights under, or on account of, the Patents and/or any item in any of the foregoing categories (b) through (f), including, without limitation, all causes of action and other enforcement rights for (i) damages, (ii) injunctive relief, and (iii) any other remedies of any kind for past, current, and future infringement; and

(h) all rights to collect royalties and other payments under or on account of the Patents and/or any item in any of the foregoing categories (a) through (g);

((a)-(h) collectively, the ***"Patent Rights"***).

Assignor hereby authorizes the respective patent office or governmental agency in each jurisdiction to issue any and all patents, certificates of invention, utility models or other governmental grants or issuances that may be granted upon any of the Patent Rights in the name of Assignee, as the assignee to the entire interest therein.

The terms and conditions of this Assignment of Patent Rights will inure to the benefit of Assignee, its successors, assigns, and other legal representatives and will be binding upon Assignor, its successors, or legal representatives.

IN WITNESS WHEREOF, this Assignment of Patent Rights is executed at Haarlem on 4 May 2012

ASSIGNOR:

Internet Broadcasting Corporation B.V.
By:

Name: Marc Vissen
Title: CEO

STATE OF (N. Holland)
) ss.
COUNTY OF N. Holland Haarlem

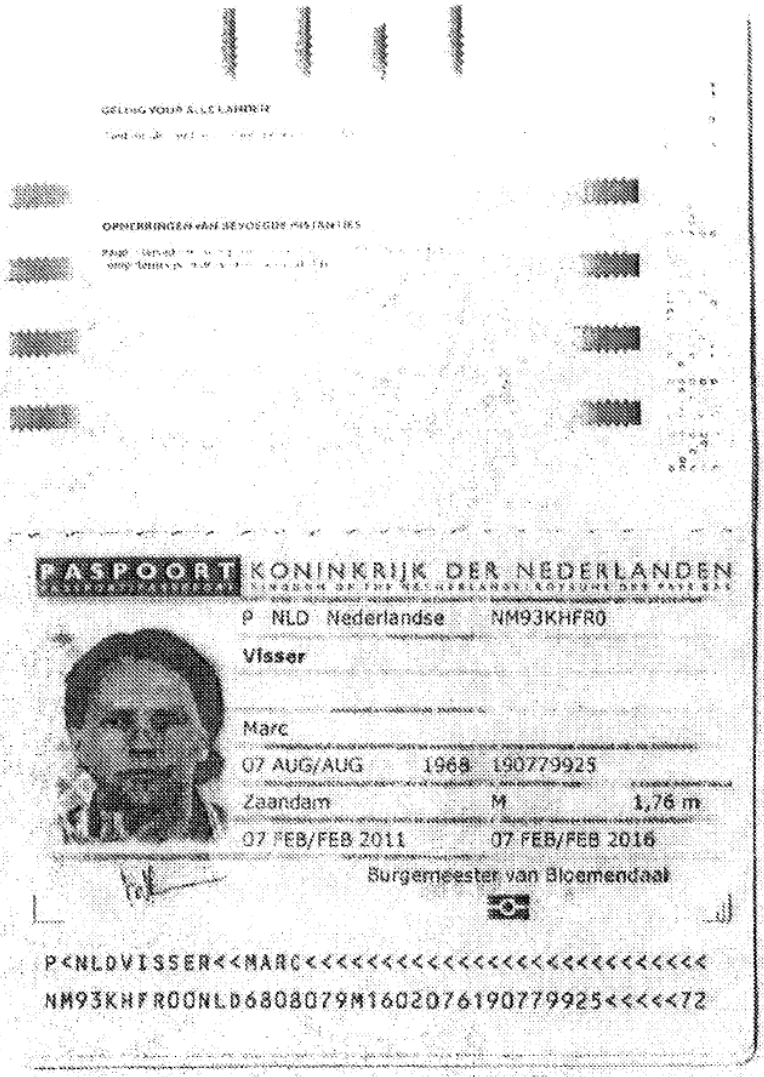
On 4 May 2012 before me, Paul Joseph Hubert Mahieu Notary Public in and for said State, personally appeared Marc Vissen, personally known to me (or proved to me on the basis of satisfactory evidence) to be the person whose name is subscribed to the within instrument and acknowledged to me that he/she executed the same in his/her authorized capacity, and that by his/her signature on the instrument the person, or the entity upon behalf of which the person acted, executed the instrument.

WITNESS my ~~hand and official~~ seal.

Signature _____ (Seal)

Gezien voor de legalisatie van de handtekening van:
Seen for legalization of the signature of:
Vu pour légalisation de la signature de:
Es wird hiermit beglaubigt dass die Unterschrift von:
Vista para la Legalización de La Firma de:
Marc Vissen
door mij, notaris, heden de
by me, civil-law notary, on this day of
par moi-même, notaire, aujourd'hui le
von dem unterzeichneten Notar anerkannt ist
heute den,
Por mi, N
4 May 2012

M.P.J.H.M. PREVOO * TE HAARLEM *



PASPOORT KONINKRIJK DER NEDERLANDEN

P NLD Nederlandse NM93KHFR0

Visser

Marc

07 AUG/AUG 1968 190779925

Zaandam M 1,76 m

07 FEB/FEB 2011 07 FEB/FEB 2016

Burgemeester van Bloemendaal

P<NLDVISSER<<MARC<<<<<<<<<<<<<<<<<<<<<<<<<<<<

NM93KHFR0ONLD6808079M1602076190779925<<<<<72

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Mr P.J. ... PREVOO

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01 FC : 2051 70.00 DA

Doc code: IDS
Doc description: Information Disclosure Statement (IDS) Filed

PTO/SB/08a (01-10)
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U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE
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INFORMATION DISCLOSURE STATEMENT BY APPLICANT
(Not for submission under 37 CFR 1.99)

Application Number	15010573
Filing Date	2016-01-29
First Named Inventor	Van Zijst, Erik
Art Unit	2465
Examiner Name	Not yet assigned
Attorney Docket Number	GPB1-US9

U.S.PATENTS Remove

Examiner Initial*	Cite No	Patent Number	Kind Code[1]	Issue Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear
	1	4799215		1989-01-17	Suzuki	
	2	5095480		1992-03-10	Fenner	
	3	5262906		1993-11-16	Mazzola	
	4	5331637		1994-07-19	Francis	
	5	5596574		1997-01-21	Perlman	
	6	5842224		1998-11-24	Fenner	
	7	5860136		1999-01-12	Fenner	
	8	5918074		1999-06-29	Wright	

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INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	15010573
	Filing Date	2016-01-29
	First Named Inventor	Van Zijst, Erik
	Art Unit	2465
	Examiner Name	Not yet assigned
	Attorney Docket Number	GPB1-US9

	9	6222841		2001-04-24	Taniguchi	
	10	6330238		2001-12-11	Ooe	
	11	6347090		2002-01-12	Ooms	
	12	6496477		2002-12-17	Perkins	
	13	6577609		2003-06-10	Sharony	
	14	6614900		2003-09-02	Champoux	
	15	6691192		2004-02-10	Ajanovic	
	16	6757294		2004-06-29	Maruyama	
				2005-01-18	Olsson	
	18	6904015		2005-06-07	Chen	
	19	6907037		2005-06-14	Tsuchiya	

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INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	15010573
	Filing Date	2016-01-29
	First Named Inventor	Van Zijst, Erik
	Art Unit	2465
	Examiner Name	Not yet assigned
	Attorney Docket Number	GPB1-US9

	20	6917984		2005-07-12	Tan	
	21	7076569		2006-07-11	Bailey	
	22	7085691		2006-08-01	Riess	
	23	7542417		2009-06-02	Babonneau	
	24	7733868		2010-06-08	Van Zijst	

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Examiner Initial*	Cite No	Publication Number	Kind Code[1]	Publication Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear
	1	20010014103		2001-08-16	Burns	
	2	20010048662		2001-12-06	Suzuki	
	3	20020009067		2002-01-24	Sachs	
	4	20020012667		2002-01-31	McMichael	

EFS Web 2.1.17

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	15010573
	Filing Date	2016-01-29
	First Named Inventor	Van Zijst, Erik
	Art Unit	2465
	Examiner Name	Not yet assigned
	Attorney Docket Number	GPB1-US9

	5	20020028687		2002-03-07	Sato	
	6	20020114282		2002-08-22	MeLampy	
	7	20020120769		2002-08-29	Ammitzboell	
	8	20020118692		2002-08-29	Oberman	
	9	20020126671		2002-09-12	Ellis	
	10	20020143951		2002-10-03	Khan	
	11	20020176406		2002-11-28	Tsukada	
	12	20020191631		2002-12-19	Couty	
		[illegible]		2003-02-06	Navas	
	14	20030035424		2003-02-20	Abdollahi	
	15	20030035425		2003-02-20	Abdollahi	

EFS Web 2.1.17

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	15010573
	Filing Date	2016-01-29
	First Named Inventor	Van Zijst, Erik
	Art Unit	2465
	Examiner Name	Not yet assigned
	Attorney Docket Number	GPB1-US9

	16	20030037132		2003-02-20	Abdollahi	
	17	20030041171		2003-02-27	Kobayashi	
	18	20030046425		2003-03-06	Lee	
	19	20030123479		2003-07-03	Lee	
	20	20030123438		2003-07-03	Li	
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Signature	/Stephen Y. Liu/	Date (YYYY-MM-DD)	2016-02-11
Name/Print	Stephen Y. Liu	Registration Number	62883

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International Application Number:	
Confirmation Number:	9731
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First Named Inventor/Applicant Name:	Erik Van Zijst
Customer Number:	122200
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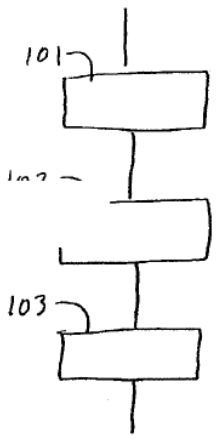
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(54) Title: ADAPTIVE DROPPING OF PRIORITIZED TRANSMISSION PACKETS



(57) **Abstract:** A series of information portions are provided to a transmitter, the information portions have critical times when they need to be available to a receiver. The transmitter estimates whether the information portions can be transmitted by the transmitter in time to be made available to the receiver before the critical time. Then the transmitter only transmits the information to the receiver if it is estimated that the information portions will be transmitted in time to be made available to the receiver before the critical time.

WO 2004/010 A2

Adaptive Dropping Of Prioritized Transmission Packets

The invention is related to the field of transmission of media packets over a variable delay network such as the internet or a packet lossy network such as wireless LAN (local area networks) for real time presentation.

For real-time multimedia presentations such as MPEG encoded broadcast or live performances, each video and audio frame portion (frame or field) has a presentation time when the presentation of the frame portion needs to begin in order to provide continuous audio and video. In order to present the frame portion on time, each frame portion has a decoding time when all the packets of the frame portion must be available for transfer to the decoder for decoding of the frame portion.

Networks such as the internet or a wireless LAN tend to have variable bandwidths resulting in variable delays. If bandwidth becomes too constrained then packets fail to arrive in time and frames can not be decoded and presented on time. Typically for video, late frames are simply dropped and the previous frame repeated resulting in loss in the human perceived quality of the presentation.

For wireless LAN, delays and transmission failures are common and bandwidth varies. First, received signal power, hence channel throughput, decreases as the distance between the sender and the receiver increases. Second, collisions between two transmitters in the same wireless LAN network occur and are handled by random delays before retransmission. Third, packet errors occur due to random interference as well as interference er devices emitting signals in t uency band. is particularly observed in a 802.11b network, which uses

the 2.4 GHz ISM band in which many other devices, such as microwave ovens, 2.4GHz cordless phones, and other 802.11b networks, operate. For wireless LAN, not only are there packet delays due to the variable bandwidth, but often packets fail to be transmitted due to retransmission limits.

The MAC (media access) layer of an 802.11b receiver does not do any error correcting. If a packet fails the error checking of the receiver MAC layer, then the receiver does not send an acknowledgement and the MAC layer of the 802.11b transmitter re-sends the packet. The transmitter continues re-sending the packet until an acknowledgement is received or until a predetermined retransmission limit is reached.

It has been proposed to add error correction for media transmission through wireless LAN to reduce the need for retransmissions. One proposed error correction technique involves separating the audio/video information into two layers and sending these layers as different streams. One stream will contain the more important information, and another stream will contain the less important information. More error protection is provided for the more important stream and less error protection is provided to the less important stream (this is called Unequal Error Protection). Although overhead is increased, the audio/video quality increases even more, so that for a given audio/video quality the overhead is actually less.

Another proposal is to transmit all the higher priority packets and drop all the lower priority packets when throughput is inadequate.

Those skilled in the art are directed to WO 01/65848 and [illegible]981 and which are incorporated h[illegible]ole. These [illegible]ences disclose schemes for splitting video streams into

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high and low priority streams and reducing the errors in the high priority stream with respect to the low priority stream.

In the invention, a series of information portions are provided to a transmitter, the information portions have critical times when they need to be available to a receiver. The transmitter estimates whether the information portions can be transmitted by the transmitter in time to be made available to the receiver before the critical time. Then the transmitter only transmits the information to the receiver if it is estimated that the information portions will be transmitted in time to be made available to the receiver before the critical time.

When the transmitter determines that bandwidth and delay requirements prevent all the information from being useful to the receiver then the transmitter drops the packets that will not be useful.

For example, in a video-layered solution, if the transmitter determines that the less important video information would not be useful (due to delay conditions and other requirements) then it is not transmitted at all. Even high priority packets may be dropped if all the high priority packets cannot be transmitted in time.

In the invention the transmitter has to be aware of the delay requirements of the video stream in order to predict whether transmission of packets will meet the delay requirements. The delay requirements may be standardized or the receiver needs to transmit an indication of the delay requirements to the transmitter.

ere are two competing goals. Firs essful
pt of all base layer packets needs to be maximized.

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Second, the discarding of enhancement layer packets needs to be minimized. The proper balance between these goals increases throughput without substantially increasing overhead in the transmission. The invention balances these two goals so that the perceived quality of a presentation is increased with relation to the quality when trying to send all the packets.

Additional aspects and advantages of the invention will become readily apparent to those skilled in the art from the detailed description below with reference to the following drawings.

Figure 1 illustrates the method of the invention.

Figure 2, shows relevant portions of an example transmitter of the invention.

Figure 3, shows relevant portions of an example receiver of the invention.

Figure 4 illustrate some of the components of an example system of the invention.

Figure 5 shows the network stack of an example sender of the invention.

Figure 6 depicts the structure of an example receiver of the invention.

Figure 7 illustrates an example algorithm of the invention for adaptively dropping prioritized packets.

Figure 8 shows a variation of the decision module of the algorithm of figure 7.

Figure 1 illustrates an example method of the invention. In step 101, a series of multiple information portions is provided to a transmitter. The information portions have

l times when they need to be ava: receiver.

frame portion may have a presentation time and a decoding

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time when decoding needs to begin to guarantee that a decoded frame portion will be available at the presentation time, the decoding time and presentation time being relative to a playing time of the presentation.

The information portions may be packets of media frame portions (frames or fields of a frame) for a multimedia presentation. Such a presentation contains multiple frame portions and each frame portion contains multiple packets of media data.

Providing the information portions may include separating media information into high priority and low priority portions. The high priority portion may be a so called base layer that is required to decode the information portions and the low priority portion may be a so called enhancement layer that is used to enhance the decoded information portion.

For example, information portions for a media frame may include multiple high priority packets of the base layer and multiple low priority packets of the enhancement layer. All the high priority packets of the frame portion have to be available to the receiver in order to begin decoding the frame portion. If all the high priority packets are available, then whatever low priority packets are also available can be used by the decoder to enhance the frame portion. Packets with different priorities may be provided serially interleaved for a frame. Alternatively, all higher priority packets may be provided before providing any lower priority packets of a frame. This simplifies the determination of which packets to send if bandwidth is inadequate to send all the packets of the

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In step 102, the transmitter estimates whether the information portions can be transmitted in time to be made available to the receiver before the critical time.

The estimating may include determining a target time and estimating a transfer time and comparing the target time to the sum of a current time and the estimated transfer time. The target time is related to when an information portion needs to be available to the receiver for decoding the information portion (decoding time). The estimated transfer time is the time required for transmitting the information portion to the receiver and making the information portion available to the receiver for decoding the information portion.

The target time depends on the start-up buffering provided by the receiver. Receiver buffering is usually limited by a start-up limitation rather than by the cost of storage. Human viewers of presentations often object to start-up times exceeding a few seconds. Increasing the receiver buffering reduces the effects of short term restrictions on the throughput of the wireless channel. The receiver buffering may be standardized so the transmitter would know the buffer limits. Alternatively, the receiver can transmit the an indication of the buffering limits to the transmitter (e.g. start-up time, program play time, buffer size). For example, the viewer may input a start-up time preference depending on his aversion to slow start-up compared to his aversion to presentation quality degradations, and the receiver transmits the start-up preference to the transmitter.

The target time may be based on the time when an :ion portion is provided. For mec .t can be ıed that packets of frames are provided as needed for

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decoding so its reasonable to base the target time on the time a packet was provided. Alternatively, the decoding time can be read from the packets of a media frame and used as a target time. Since all the high priority packets have to be at the decoder in order to begin decoding a frame, the target time can be the same for all the high priority packets of the frame. Also, typically all the low priority packets have to be available at the decoding time in order to enhance the frame. Thus, the target time may be derived from the time that the first portion of a frame is provided, and the same target time can be used for all the packets of the frame.

The estimation may depend on the priority of the information portions, so that higher priority information portions are more likely to be transmitted than lower priority information portions. For example, the target time or estimated transfer time for an information portion may be adjusted (by adding a factor or multiplying by a factor) depending on the priority of the information portion. For example, the target time may be calculated for a media frame, by adding the time when the first packet of the media frame arrives to the buffering time of the receiver. Different target times may be calculated for high and low priority frames by multiplying the buffering time by a delay tolerance which is higher for high priority packets and lower for low priority packets.

The information portions may be packets of a video presentation encoded based on a GOP (group of pictures) such as MPEG. In such encoding there are different types of video frames (frames or fields of frames) including I-frames that are independent of the decoding of a[illegible]ame, and P-[illegible]s that are decoded based on the decoding of the previous

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I-frame, and B-frames that are decoded based in the previous and the following I or P-frame. Usually there are two B-frames for each P-frame. Thus, the I-frames are the most important frames because the decoding of all the other frames in a GOP is dependent on decoding the I-frame of the GOP. The P-frames are also important because the decoding of up to four B-frames depends on the decoding of each P-frame.

For GOP encoded video frames, the estimation may depend on the type of video frame. In a manor similar to that described above for packets with different priorities (base layer vs. enhancement layer packets).

The estimated transfer time can be fixed and based on the maximum packet size and retransmission limit. Alternatively, the estimated transmission time may be estimated dynamically, for example, based on retransmission counts when previous packets were transmitted. For example, an average transmission time of a previous predetermined number of packets can be used or the average can be a time weighted average in which more weight is assigned to more recently transmitted packets.

For variable size packets, the estimated transfer time of a packet may depend on the size of a packet. Larger packets take longer to transmit and tend to have more errors requiring more retransmissions. The adjustment for packet size can be predetermined or it can be based on averages or time weighted averages of a predetermined number of previous packets in the same size range.

In step 103, the packets are transmitted depending on the estimation. That is, packet which can be transmitted and ed by the receiver in time to be : the cal time are transmitted and packets that can not be

transmitted and processed by the receiver by the critical time are dropped.

For a media frame with packets having different priorities, all the higher priority packets of the frame can be transmitted prior to transmitting any lower priority packets. Alternatively, packets having different priorities can be transmitted interleaved as they are received. An advantage of transmitting higher priority packets earlier is that when bandwidth is reduced, the higher priority packets are automatically more likely to reach the receiver then the lower priority packets.

Typically in video encoding, all the packets of a frame portion must be available in order to decode a frame portion. In that case, if one packet of a frame portion is dropped, for example, because the retransmission limit is exhausted, then all the remaining packets of the frame portion may be dropped because they cannot be used by the receiver.

On the other hand, when the video information is separated into a base layer and an enhancement layer then only all the base layer packets are required to decode a frame portion. If an enhancement layer packet is dropped then any remaining base layer packets should still be transmitted.

The estimating may include multiple estimations during the transmission of packets of a media frame portion, whether all the packets of the media frame portion can be transmitted by the transmitter in time to be made available to the receiver before the critical time; and the transmitting depends on the multiple estimations for all the packets of the frame portion.

hen it is estimated that it is l ome of the packets for the frame portion will not be transmitted in time,

then the remaining packets for the frame portion are not transmitted.

When the information portions for a frame include information portions having different priorities, the estimating may include multiple estimations during the transmission of packets of a media frame portion, whether all the packets of the media frame portion of a priority can be transmitted by the transmitter in time to be made available to the receiver before the critical time, and the transmitting of packets of the same or lower priority depends on the multiple estimations for all the packets of the frame portion of the priority. Thus, when it is estimated that it is likely that some of the packets for the frame portion of the priority will not be transmitted in time for decoding the frame, then the remaining packets for the frame portion for the same or lower priority are not transmitted.

Figure 2 shows relevant portions of an example of a transmitter of the invention. Source 110 provides a series of information portions that need be available to a receiver at critical times. The source may include an input 111 from an antenna or cable connection for receiving satellite or terrestrial broadcast signals. In addition or in the alternative to input 111, source 110 may include a source unit 112 which may include, for example, a hard drive, a media player such as a DVD player, or electronic storage unit; or the source unit may include a camera for capturing a live performance.

Controller 111 controls the operation of the transmitter ng the transmission of the infor ons to a receiver. The controller estimates whether the information

portions can be transmitted by the transmitter in time to be made available to the receiver before the critical time.

Transmission unit 114 transmits the information portions to the receiver (shown below in figure 3) depending on the estimating. The controller sends a control signal to the transmission unit to transmit an information portion. That is, the information portions are only transmitted by the transmitter unit if the information portions are likely to be made available to the receiver before the critical time.

The information portions may include high priority and low priority information portions or source 110 may contain a partitioner 115 to separate provided information into high priority information portions and low priority information portions.

Buffer 116 is provided to store the information portions and allow processing of the information portions by the controller. For example, partitioner 115 may packetize the information into serially interleaved high priority packets and low priority packets which are stored in the buffer, and then the controller controls the buffer to send all the high priority packets to the transmitter before sending any of the low priority packets to the transmitter.

Receiver unit 117 receives control information from the receiver (shown below in figure 3). Such information may include a request to receive the information portions and information indicating the start-up time (buffer limit) of the receiver of figure 3.

Figure 3 illustrates relevant portions of an example : of the invention. Receiver unit :s the mation portions. Controller 132 controls a buffer 133 to

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buffer the information portions and at critical times, to send the information portions to decoder 134. Buffer 133 will store information portions for a length of time up to a start-up delay time, and then pass the information portions to decoder 134. For example, when the information portions include encoded video information then at a decoding time the decoder receives the information portions for a frame portion and at a presentation time the decoder provides the decoded frame portion to video display 135. During a presentation, video frames are normally displayed at a constant predetermined frame rate, but decoding time may vary depending on frame type, frame size, and frame complexity.

User input 136 may be used to initiate reception of information portions and may be used to select a start-up delay. Controller 132 controls transmission unit 137 to transmit the request to receive information portions and the start-up delay of the receiver to the transmitter of figure 2.

The following describes an example system of the invention. A layered multimedia streaming system is implemented using a client-server model. The encoded bit stream has several layers including a base layer that is required to decode video stream and enhancement layers that are used to enhance the decoding. An application operating on the client initiates a streaming session by sending a request to the server to start transmitting. The protocol used to exchange management information may be RTSP (Real Time Streaming Protocol). The server responds by transmitting a data stream to the client. The protocol used for the data stream is RTP (Real Time .). RTSP streams rely on TCP (Tra ntrol col) as the transport protocol while RTP streams employ

UDP (User Data Protocol). IP (internet protocol) is the protocol used at the network layer, and the link layer uses IEEE 802.11 wireless technology.

In figure 4, a broadcast information stream is received at 150 and separated by partitioner 151 into a base layer stream 152 and enhancement layer stream 153. A streaming server 154 passes the base and enhancement layer streams to a MAC (Media Access) layer 155 communicating through a wireless transmitter 156. The base and enhancement streams travel through the wireless medium 157 to wireless receiver 158. Then the base and enhancement streams are passed to a client application 159 which provides them to a decoder 160which merges the streams and decodes them to provide a decoded multimedia presentation at 161.

Figure 5 illustrates parts of the sender network stack. In an application layer 170, a video streaming server 171 provides a layered multimedia stream. The main tasks of the streaming server are to build RTP packets and to manage the streaming session (including packet scheduling). The server bases its operation on the hinting information available in the video file. The server reads the hinting information and creates two streams of RTP packets. One stream contains base layer packets and the other stream contains the enhancement layer packets. Each packet includes an RTP payload field and the RTP payload field values of the base layer packets are different from the RTP payload field values of the enhancement layer packets. Thus, the base layer packets can be distinguished from the enhancement layer packets based on their different values for payload field.

The server also adds an RTP timestamp to each packet. The value of the RTP timestamp is the same for all the packets of a frame, and indicates the time when the packet was created.

The streaming server delivers the streams to an operating system kernel 173 using a UDP socket. At certain instants the server forewords all packets corresponding to the same video frame to the system kernel. First the server sends all the base layer (high priority) packets for the frame and then the server sends all the enhancement layer (low priority) packets for the frame. Thus, the streams are forwarded in alternating cycles of base and enhancement layer packets. The system Kernel passes the streams to an 802.11 device driver 174. Operating system Kernel 173 and 802.11 device driver 174 are portions of an operating system layer. The 802.11 device driver passes the streams to an wireless transmission device 175 that includes an 802.11 MAC layer 176 and an 802.11 physical layer 177 for radio frequency (RF) wireless transmission.

Since the system server (as described above) sends alternating cycles of base and enhancement layer packets, and the base and enhancement layer packets have different values in their RTP payload fields, the beginning of a frame is easily detected. An alternative way to detect frame boundaries is to use the RTP timestamp since it has the same value for all the packets of a frame and different value from the packets of adjacent frames.

Sending the base layer packets first is a good way to prioritize their transmission to the detriment of the enhancement layer packets. After the base layer packets are all fully sent we can decide whether ancement

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layer packets depending on delay limitations and on system conditions.

Figure 6 illustrates the structure of an example receiver of the invention. Packets are received by a 802.11 network device 190 through a wireless link at 191. The packets are passed from the 802.11 network device 190 through 802.11 device driver 192 and through operating system kernel 193. 802.11 device driver 192 and through operating system kernel 193 are parts of the operating system layer 194 of the receiver. Then the packets are passed from operating system kernel 193 into client buffer 195 of client application layer 195. The RTP timestamp is used to identify which packets belong to a particular frame. At a decoding time for the frame, all the packets for the frame are passed to decoder 196.

Figure 7 illustrates an example algorithm of the invention for adaptively dropping prioritized packets. The dashed horizontal lines separate steps that are performed in different portions of the server. In step 210, a streaming video server creates the base layer and enhancement layer packets and in step 211 the server timestamps all the packets for a frame with a timestamp time and sends all the packets for the frame to an operating system kernel. The server sends all the base layer packets for a frame before sending any of the enhancement layer packets for the frame. In step 212, the packets for the frame are stored in socket buffering of the system kernel. In step 213, the 802.11 device driver reads a packet type.

In a detection module 214 which detects the packet type and whether the packet is the start of a new video frame. In ection module, in step 215, the c nines ıer the packet is a base layer packet. If the packet is a

base layer packet then in step 216 the driver determines whether the driver is already in a base layer cycle state. If the driver is not in the base layer cycle state then this is the first packet of a frame and in step 217 the driver enters the base layer cycle state. Also, in detection module 214, if in step 215 its determined that the packet is not a base layer packet, then the packet is an enhancement layer packet and in step 218 the driver is in an enhancement layer state.

In the target time module 220 for estimating target time, at the beginning of the transmission of a new frame, the driver estimates a target time which is the latest time that any packet of the frame should arrive at the receiver. In the target time module, in step 221, a target time for base layer packets and a different target time for enhancement layer packets is calculated. The target time for base layer packets is the sum of the RTP timestamp time of the first packet of the frame and the buffering time of the receiver multiplied by a delay tolerance for base layer packets. The target time for enhancement layer packets is the sum of the RTP timestamp time of the first packet of the frame and the buffering time of the receiver multiplied by a delay tolerance for enhancement layer packets.

The delay tolerance for base layer packets should be much greater than the delay tolerance for enhancement layer packets. The values of these delay tolerances may be constant or they may be tuned depending on the specific application, the specific network, and on network conditions.

The buffering time of the receiver is essentially the
› delay for the stream in the rec may be a
:termined fixed amount. Alternatively, the buffering time

may be a variable amount that has to be received from the receiver. The transmission of the buffering time can be done using RTSP (real time streaming protocol), which was also the protocol used by the receiver to initiate the streaming session.

In this embodiment, the same base layer target time is used for all the base layer packets and the same enhancement layer target time is used by all the enhancement layer packets.

The adjustment of these delay tolerances may be used to balance the two competing goals in the transmission of media packets to maximize perceived quality. The first goal is to maximize the successful receipt of all base layer packets. The second goal is to minimize the discarding of enhancement layer packets.

The above calculation of the target times for base and enhancement layer packets is based on the RTP timestamp time of the first packet of the frame. Another alternative is to use the frame rate to estimate the decoding time. However the RTP timestamp time and the frame rate are not accurate indicators of the decoding time - which is the true target time. However, the decoding time can be derived from the RTP time, and then the decoding time can be used in the calculation of the target time for a frame.

In a decision module 230 that decides whether to transmit packets, the driver estimates the transmission duration of each packet and decides whether to transmit the packet or not depending on the target time for the type of packet that was determined in the target time module in step 221. In the n module, if the packet is a bas et then in

step 231, the transfer time of the base layer packet is estimated.

The average transfer time of the packet may be derived based on network conditions and the size of the packet. A time weighted average can be used for network conditions. Alternately, actual network conditions may be ignored and worst case conditions used. In that case transfer time is fixed and does not need to be calculated. Worst case transfer time can be based on the maximum retransmission limit for the type of packet and the maximum packet size. The retransmission limit may be higher for base layer packets than for enhancement layer packets. A typical maximum packet size for video packets in 802.11 wireless transmission is 1500 bytes.

In step 232, the driver determines whether the sum of the current time and the transfer time exceeds the target time for the base layer packet. If the target time in step 232 is exceeded, then in step 233 the base layer packet is dropped. Since the target time is the same for all the base layer packets of the same frame then this means that all the following base layer packets will also be dropped. Also, the target time for all the following enhancement layer packets of the frame will be the same and will be lower so that all the following enhancement layer packets of the same frame will also be dropped.

Otherwise in step 232 if the target time is not exceeded then in step 234, the base layer packet is sent to the 802.11 device.

In the decision module, if the packet is an enhancement acket, then in step 235, the trar E the icement layer packet is estimated. The above discussion of

estimating of the transfer time for base layer packets is also generally applicable to the estimating of the transfer time of enhancement layer packets.

In step 236, the driver determines whether the sum of the current time and the transfer time exceeds the target time for the enhancement layer packet. If the target time in step 236 is exceeded, then in step 233 the enhancement layer packet is dropped. Since the target time is the same for all the enhancement layer packets of the same frame then this means that all the following enhancement layer packets will also be dropped.

Otherwise in step 236 if the target time is not exceeded then in step 234, the enhancement layer packet is sent to the 802.11 device.

In step 240, the 802.11 device stores the packet to be transferred in the MAC buffer, and then transmits the packet. If no ACK is received, then the packet is retransmitted until either an ACK is received or the retransmission limit is exhausted.

Figure 8 shows a modification of the decision module 230 of the algorithm of figure 7. In the algorithm of figure 8, the determination of whether to transmit or drop frames depends on the frame type. In modified decision module 250, if the packet is a base layer packet, then in step 251 a frame type is determined. If the frame type is an I-frame, then in step 252 a tolerance is set to an I-frame tolerance value for base layer packets for I-frames. Otherwise, if the frame type is an P-frame, then in step 253 the tolerance is set to an P-frame

:e value for base layer packets f Otherwise,

ıe frame type is an B-frame, then in step 254 the tolerance

is set to a B-frame tolerance value for base layer packets for B-frames. The tolerance value for I frames should be larger than the tolerance value for P-frames or B-frames. The tolerance value for B-frames should be, for example, zero.

In step 255, the driver determines whether the sum of the current time and the transfer time exceeds the sum of the target time and the tolerance for the base layer packet. If the sum of the target time and the tolerance in step 255 is exceeded, then in step 233 the base layer packet is dropped. Otherwise in step 255 if the sum of the target time and the tolerance is not exceeded then in step 234, the base layer packet is sent to the 802.11 device.

In modified decision module 250, if the packet is an enhancement layer packet, then in step 261 a frame type is determined. If the frame type is an I-frame, then in step 262 a tolerance is set to an I-frame tolerance value for enhancement layer packets for I-frames. Otherwise, if the frame type is an P-frame, then in step 263 the tolerance is set to an P-frame tolerance value for enhancement layer packets for P-frames. Otherwise, if the frame type is an B-frame, then in step 264 the tolerance is set to a B-frame tolerance value for enhancement layer packets for B-frames. Again, the tolerance value for I frames should be larger than the tolerance value for P-frames or B-frames. The tolerance value for B-frames should be, for example, zero.

In step 265, the driver determines whether the sum of the current time and the transfer time exceeds the sum of the target time and the tolerance for the enhancement layer packet.

sum of the target time and the t step 255 is
eded, then in step 233 the enhancement layer packet is

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dropped. Otherwise in step 255 if the sum of the target time and the tolerance is not exceeded then in step 234, the enhancement layer packet is sent to the 802.11 device.

The invention has been described above in relation to specific example embodiments. Those skilled in the art will know how to modify these example embodiments within the scope of the invention herein. The invention is only limited by the following claims.

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Claims:

1. A method comprising:

providing a series of information portions to a transmitter, the information portions having critical times when they need to be available to a receiver;

estimating whether the information portions can be transmitted by the transmitter in time to be made available to the receiver before the critical time; and

transmitting the information portions to the receiver depending on the estimating.

2. The method of claim 1, wherein the information portions are packets of media frame portions for a multimedia presentation, the presentation including multiple frame portions each frame portion including multiple high priority packets and multiple low priority packets.

3. The method of claim 2, wherein the multiple frame portions each have a decoding time before which all the high priority packets of the frame portion need to be available to the receiver to decode the frame portion in time for presentation at a predetermined presentation time of the frame portion, the decoding time and presentation time being relative to a playing time of the presentation.

4. The method of claim 3, wherein the low priority packets are used to enhance the frame portion during decoding when some all of the low priority packets ilable to eceiver at the critical time.

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5. The method of claim 1, wherein providing the information portions includes separating information into information portions having different priorities.

6. The method of claim 1, in which providing the information portions includes providing media frame portions and partitioning each frame portion into high priority information portions that are necessary to decode the frame portion and low priority information portions that are only needed to enhance the frame portions.

7. The method of claim 1, wherein the information portions include high priority and low priority packets of media frame portions and all the high priority packets for one frame portion are provided before any low priority packets for the frame portion are provided.

8. The method of claim 1, wherein: the estimating includes determining a target time when one information portion needs to be available to the receiver and determining an estimated transfer time for transmitting the information portion and making the information portion available to the receiver; the estimating includes determining whether the target time exceeds the sum of the estimated transfer time plus the current time of the transmission; and the information portion is transmitted depending on the determination.

: method of claim 1, wherein the lepends on a ring limit of the receiver.

10. The method of claim 9, wherein the receiver transmits an indication of the buffering limit to the transmitter.

11. The method of claim 1, wherein the estimating depends on a priority of the information portion being transmitted.

12. The method of claim 8, wherein the information portions include high priority packets and low priority packets and the target time depends on a predetermined delay tolerance that is larger for the high priority packets than for the low priority packets so that the high priority packets are more likely to be received by the receiver than the low priority packets.

13. The method of claim 1, wherein the information portions are packets of media frame portions and the estimating depends on a type of media frame of the packet.

14. The method of claim 8, wherein:

the information portions are portions of a video presentation encoded based on a group of pictures of different types of video frames including I-frames that are decoded independent of the decoding of any other frame, and P-frames that are decoded based on the decoding of the previous I or P-frame, and B-frames that are decoded based in the previous and the following I or P-frame; and

the target time depends on the type of video frame.

[illegible] method of claim 1, wherein the portions ackets of media frame portions and the estimating depends

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on a decoding time of the media frame portion of the information portion to be transmitted.

16. The method of claim 8, wherein the information portions are packets of media frame portions of a presentation, multiple media frame portions each having a decoding time when packets of the frame portion have to be available at a decoder of the receiver for decoding in time for presentation at a predetermined presentation time of the frame portion, the target time of one information portion depending on the decoding time of the frame portion of the information portion, the decoding time and presentation time of a frame portion being relative to a play time of the presentation.

17. The method of claim 1, wherein the estimating depends on the maximum size on an information portions.

18. The method of claim 1, wherein the transmission includes retransmitting information portions until a predetermined limit on the number of retransmissions is reached or an acknowledgement is received from the receiver that the portion has been successfully received without any uncorrectable errors, and the estimating depends on the retransmission limit.

19. The method of claim 1, wherein the estimating depends on the actual transfer time of previously transmitted information portions.

method of claim 1, wherein the portions
igh priority and low priority packets of media frame

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portions and transmitting all the high priority packets for one frame portion begins before the beginning of transmitting the low priority packets for the frame portion.

21. The method of claim 1, wherein transmitting the information portions depends on whether previous information portions were successfully transmitted.

22. The method of claim 1, wherein the information portions have different priorities and transmitting the information portions depends on whether previous information portions with the same or higher priority were successfully transmitted.

23. The method of claim 1, wherein the information portions are high priority and low priority packets of media frame portions and transmitting the high priority packets depends on whether previous high priority packets for the same frame portion have been transmitted and transmitting the low priority packets depends on whether previous high priority and previous low priority packets for the same frame portion have been transmitted.

24. The method of claim 1, wherein the method further comprises receiving a request from the receiver to initiate transmitting the information portions to the receiver.

25. The method of claim 2, wherein the estimating includes multiple estimations during the transmission of packets of a ame portion, whether all the pac media frame on can be transmitted by the transmitter in time to be

made available to the receiver before the critical time, and the transmitting depends on the multiple estimations for all the packets of the frame portion so that when it is estimated that it is likely that some of the packets for the frame portion will not be transmitted in time, then the remaining packets for the frame portion are not transmitted.

26. The method of claim 2, wherein the information portions for a frame include information portions having different priorities, the estimating includes multiple estimations during the transmission of packets of a media frame portion, whether all the packets of the media frame portion of a priority can be transmitted by the transmitter in time to be made available to the receiver before the critical time, and the transmitting of packets of the same or lower priority depends on the multiple estimations for all the packets of the frame portion of the priority so that when it is estimated that it is likely that some of the packets for the frame portion of the priority will not be transmitted in time, then the remaining packets for the frame portion for the same or lower priority are not transmitted.

27. A method comprising:

separating a media stream for a performance into multiple media streams with different priorities for transmission over a network having a variable conditions, the media stream having a predetermined schedule;

determining whether to transmit portions of the multiple streams depending on the priorities and network conditions so that the perceived quality of the performance is

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increased relative to attempting to transmit all of the multiple media streams; and

transmitting the portions of the multiple media streams depending on the determination.

28. The method of claim 27, wherein the network is a wireless network and packets are transmitted serially, and each packet is transmitted and retransmitted until an acknowledgement is received or a retransmission limit is reached.

29. A transmitter for transmitting information portions to a receiver, comprising:

means for providing a series of information portions that need be available to the receiver at critical times;

means for estimating whether the information portions can be transmitted by the transmitter in time to be made available to the receiver before the critical time; and

transmitting apparatus to transmit the information portions to the receiver depending on the estimating.

30. The transmitter of claim 29, wherein:

the transmitter further comprises means for receiving an indication of the size of buffering from the receiver; and

the estimating depends on the indication of the size of buffering in the receiver.

31. The transmitter of claim 29, wherein:

the transmitter further comprises means for determining a time of the information portion

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the estimating depends on the decoding time of the information portions.

32. The transmitter of claim 29, wherein:

the means for providing information portions includes means for partitioning information into information portions having different priorities; and

the estimating depends on the priorities of the information portions.

33. A receiver comprising:

means for requesting a transmitter to transmit information portions that have to be available to the receiver at critical times;

means for buffering information portions that arrive before the critical time; and

means for transmitting an indication of the size of the buffering to the transmitter.

34. The receiver of claim 33, wherein:

the information portions are packets for encoded media frames, each of multiple media frames containing multiple packets;

the receiver further comprises a buffer for storing the packets and means for decoding the packets at the critical time;

all the packets for a frame contain the same timestamp; and

l the packets for a frame are ide ed on the timestamp and transferred together to the decoder.

35. The receiver of claim 33, wherein:

the information portions are packets for encoded media frames, each of multiple media frames containing multiple packets;

the receiver further comprises a buffer for storing the packets and means for decoding the packets at the critical time;

the packets include high priority packets and low priority packets;

all the packets contain an indication of their priority;

the high priority packets for a frame are received before the low priority packets for the frame are received; and

all the packets for a frame are identified by the changes in the indication of packet priorities and transferred together to the decoder.

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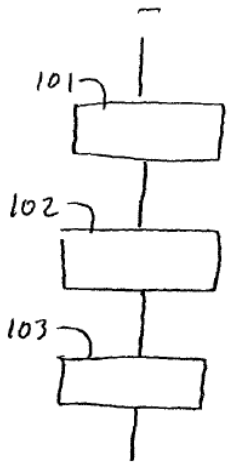


Fig 1

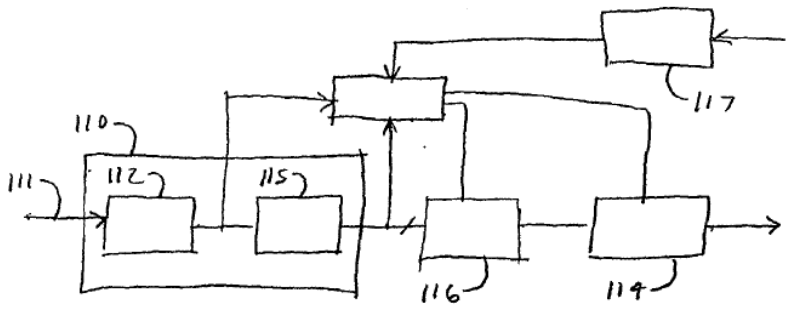


Fig 2

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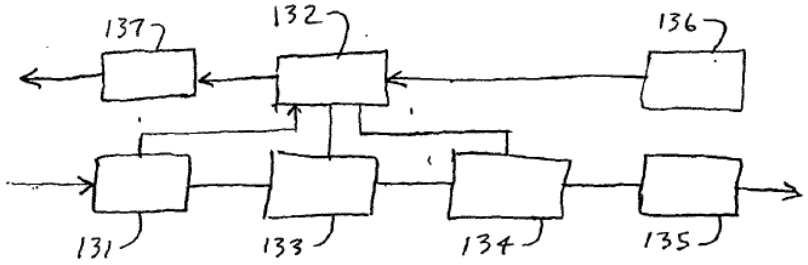

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Fig 3

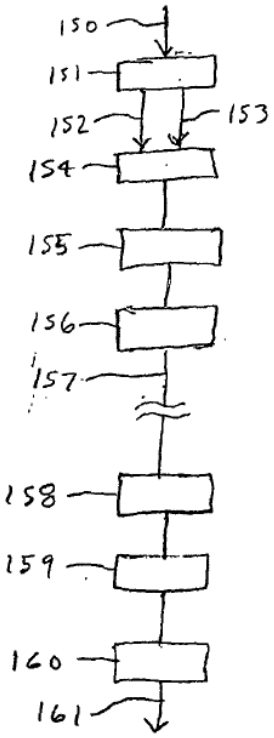

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Fig 4

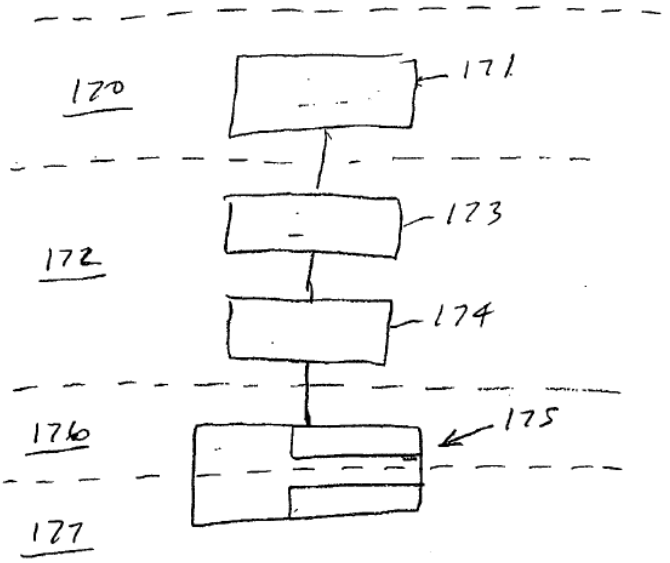


Fig 5

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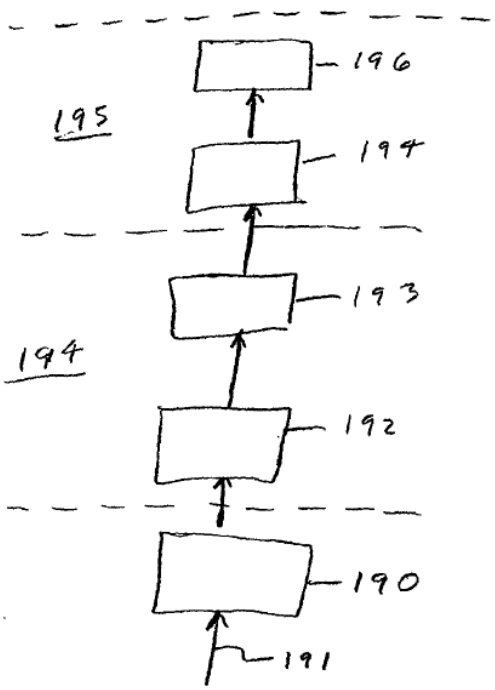


Fig 6

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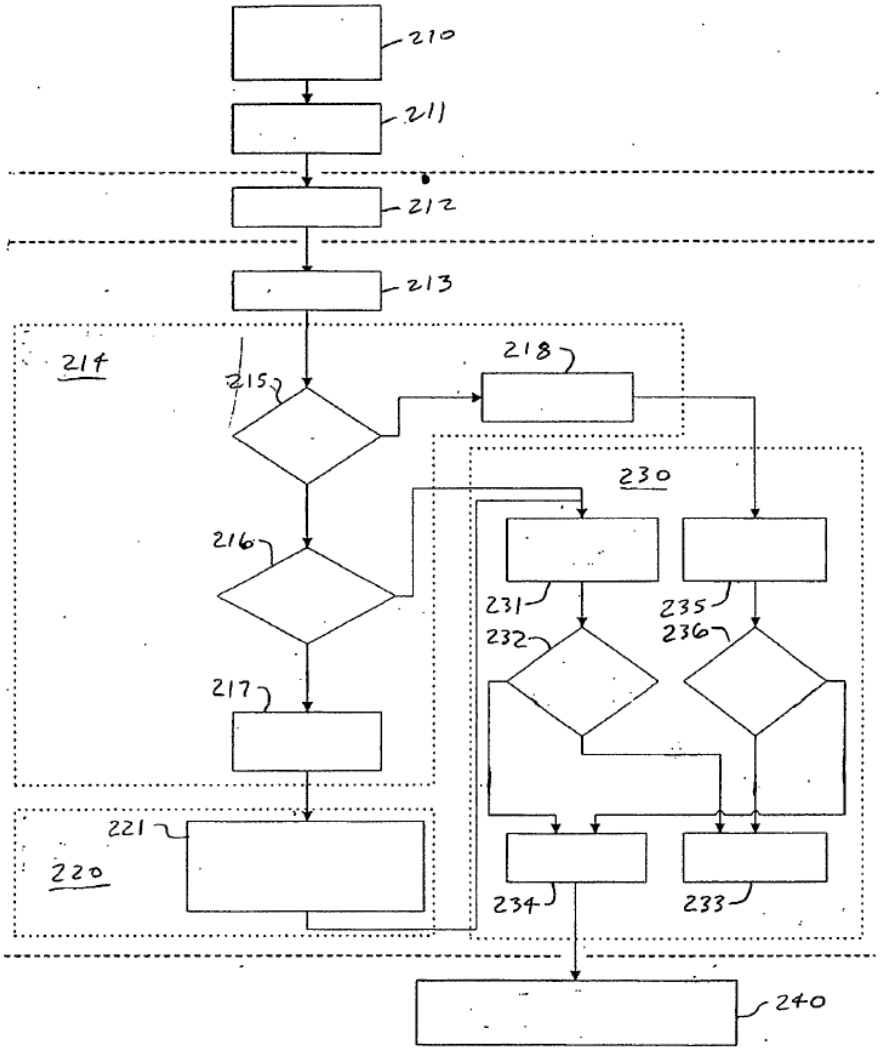


Fig 7

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(54) Title: LAYERED MULTICAST AND FAIR BANDWIDTH ALLOCATION AND PACKET PRIORITIZATION

(57) Abstract: Embodiments include an overlay multicast network. The overlay multicast network may provide a set of features to ensure reliable and timely arrival of multicast data. The embodiments include a congestion control system that may prioritize designated layers of data within a data stream over other layers of the same data stream. Each data stream transmitted over the network may be given an equal share of the bandwidth. Addressing in routing tables maintained by routers in the may utilize summarized addressing based on the difference in location of the router and destination address. Summarization levels may be adjusted to minimize travel distances for packets in the network. Data from high priority data stream layers may also be retransmitted upon request from a destination machine to ensure delivery of data.

WO 2006/081454 A3

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US06/02995

A. CLASSIFICATION OF SUBJECT MATTER

IPC: **G01R 31/08**(2006.01);**G06F 11/00**(2006.01);**H04J 1/16**(2006.01),**3/16**(2006.01),**3/18**(2006.01);**H04L 12/28**(2006.01),**12/54**(2006.01);**G06F 15/16**(2006.01),**15/173**(2006.01)

USPC: 370/229,230,231,235,238,394,389,428,468,469,477;709/223,230,238,245

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
U.S. : 370/229,230,231,235,238,394,389,428,468,469,477; 709/223,230,238,245

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched
IETF, ACM, IEEE

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)
EAST: US-PGPUB, US-PAT, EPO, JPO, DERWENT, IBM_TDB

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	WO 2004/010250 A2 (Ruiz et al) 29.01.04, pg. 5 lines 16-29, pg. 13 lines 4-14	1-17
Y	Gorinsky, S. et al "Addressing Heterogeneity and Scalability in Layered Multicast Congestion Control", Technical Report TR2000-31, 24 Nov 2000; pgs. 2, 4 and 5	1, 2, 5 and 17
Y	US 2003/0233464 A1 (Walpole et al), 18.12.2003, para. 0038-39 and para. 0059-60	3, 4, 6 and 11
Y	US 6,845,105 B1 (Olsson et al), 18.01.2005, col. 6 lines 11-31, col. 8 lines 17-27	7 and 8
Y	US 2004/0081197 A1 (Liu), 29.04.2004; para. 0014	9
Y	US 2005/0002363 A1 (Cheng et al), 06.01.2005, para. 0052	10 and 13
Y	US 6,691,192 B2 (Ajanovic et al), 10.02.2004; col. 26 lines 46-67, col. 27 lines 1-3	11

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Date of the actual completion of the international search	Date of mailing of the international search report
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Name and mailing address of the ISA/US Mail Stop PCT, Attn: ISA/US Commissioner of Patents P.O. Box 1450 Alexandria, Virginia 22313-1450 Facsimile No. (571) 273-3201	Authorized officer Xavier Szewai Wong Telephone No. 571-272-3050

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INTERNATIONAL SEARCH REPORT

C. (Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	US 2003/0123479 A1 (Lee et al), 03.07.2003; para. 0041	12
Y	US 4,799,215 A (Suzuki), 17.01.1989; col. 12 lines 22-38, fig. 16	14
Y	US 2004/0258094 A1 (Bashan et al), 23.12.2004; para. 0013	15
Y	US 2002/0028687 A1 (Sato et al), 07.03.2002; para. 0130	16
X	US 2002/0114282 A1 (McLampy et al), 22.08.2002; para. 0027 and 0074	18
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Y	US 2003/0026268 A1 (Navas), 06.02.2003; para. 0136, 0152, 0159 and 0182-184	19-22
X	US 2002/0114282 A1 (McLampy et al), 22.08.2002; para. 0027 and 0074	18
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X	US 2002/0114282 A1 (McLampy et al), 22.08.2002; para. 0027 and 0074	18
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Y	US 2003/0123438 A1 (Li et al), 03.07.2003; para. 0077, fig. 1	25
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Y	US 2002/0176406 A1 (Tsukada et al), 28.11.2002; para. 0041	27
X	US 2001/0048662 A1 (Suzuki et al), 06.12.2001; para. 0028-29	26
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Y	US 2003/0161311 A1 (Hiironniemi), 28.08.2003; para. 0006	28
X	US 2001/0048662 A1 (Suzuki et al), 06.12.2001; para. 0028-29	26
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Y	US 2004/0015591 A1 (Wang et al), 22.01.2004; para. 0079	29 and 30
X	US 5,596,574 A (Perlman et al), 21.01.1997; col. 5 lines 42-61, col. 8 lines 23-40	31 and 32
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Y	US 2002/0120769 A1 (Ammitzboell), 29.08.2002; para. 0032	37
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X	US 5,596,574 A (Perlman et al), 21.01.1997; col. 5 lines 42-61, col. 8 lines 23-40	31 and 32
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Y	US 2003/0123438 A1 (Li et al), 03.07.2003; para. 0077, fig. 1	33-36
X	US 2004/0010616 A1 (McCanne), 15.01.2004; para. 0027, 0045, 0046, 0048, 0122 and 0180; figs. 4 a, b and c	39, 40, 41, 45 and 46
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C. (Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

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X --- Y	US 2004/0010616 A1 (McCanne), 15.01.2004; para. 0027, 0045, 0046, 0048, 0122 and 0180; figs. 4 a, b and c Cheng, C. et al "A Loop-Free Extended Bellman-Ford Routing Protocol Without Counting Effect" ACM 089791-332-9/89/0009/0224, 1989; pg. 227 and 228 section 3.2	39, 40, 41, 45 and 46 ---------- 43
X --- Y	US 2004/0010616 A1 (McCanne), 15.01.2004; para. 0027, 0045, 0046, 0048, 0122 and 0180; figs. 4 a, b and c US 2003/0212816 A1 (Bender et al), 13.11.2003; para. 0029	39, 40, 41, 45 and 46 ---------- 44
X --- Y	US 2004/0010616 A1 (McCanne), 15.01.2004; para. 0027, 0045, 0046, 0048, 0122 and 0180; figs. 4 a, b and c US 2004/0258002 A1 (Tran et al), 23.12.2004; para. 0036	39, 40, 41, 45 and 46 ---------- 47
X --- Y	US 2004/0010616 A1 (McCanne), 15.01.2004; para. 0027, 0045, 0046, 0048, 0122 and 0180; figs. 4 a, b and c US 2003/0123453 A1 (Ooghe et al), 03.07.2003; para. 0038	39, 40, 41, 45 and 46 ---------- 48

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IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Applicant:	Erik Van ZIJST
Serial No.:	15/010,573
Filing Date:	January 29, 2016
Title:	Layered Multicast and Fair Bandwidth Allocation and Packet Prioritization
Confirmation No.:	9731
Examiner:	Not yet assigned
Art Unit:	2465
Attorney File No.:	GPB1-US9

INFORMATION DISCLOSURE STATEMENT

In accordance with 37 CFR §1.56, and in accordance with the provisions of 37 C.F.R. §§1.97 and 1.98, Applicant hereby makes disclosure of the patents, publications, and/or other information listed on the accompanying form PTO/SB/08A, which references are considered to be potentially material to the patentability of the invention disclosed in the above-referenced application. Copies of the non-US listed references are submitted herewith.

In accordance with 37 C.F.R. §1.97(h), the filing of this Information Disclosure Statement shall not constitute an admission that any information cited therein is, or is considered to be, material to patentability as defined in 37 C.F.R. §1.56(b). In the interest of full and complete disclosure to

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No fees are believed to be due at this time. However, the Commissioner is hereby authorized to charge any additional payment that may be due or credit any overpayment to Deposit Account 50-0392.

Respectfully submitted,

By: /Stephen Y. Liu/
Stephen Y. Liu
Registration No. 62,883
Attorney for Applicant

Date: February 11, 2016

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SCALE VIDEO CODING, LLC
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IPR2024-01371
Ex. 1002, p. 175 of 830

Doc code: IDS
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Application Number	15010573
Filing Date	2016-01-29
First Named Inventor	Van Zijst, Erik
Art Unit	2465
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	Application Number	15010573
	Filing Date	2016-01-29
	First Named Inventor	Van Zijst, Erik
	Art Unit	2465
	Examiner Name	Not yet assigned
	Attorney Docket Number	GPB1-US9

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	2	VICISANO et al., "TCP-like congestion control for layered multicast data transfer", IEEE INFOCOM '98 Proceedings, pp. 996-1003 Vol. 3, March 1998, Sections (1 pp. 1-3 of 25, 2.3 through 3.1 inclusive pp. 4-7 of 25, 3.4 pp. 8-10 of 25), IEEE Computer Society, 10662 Los Vaqueros Circle, Los Alamitos, California 90720-1314 USA.	
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INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	15010573
	Filing Date	2016-01-29
	First Named Inventor	Van Zijst, Erik
	Art Unit	2465
	Examiner Name	Not yet assigned
	Attorney Docket Number	GPB1-US9

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	Application Number	15010573
	Filing Date	2016-01-29
	First Named Inventor	Van Zijst, Erik
	Art Unit	2465
	Examiner Name	Not yet assigned
	Attorney Docket Number	GPB1-US9

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☒ See attached certification statement.

Fee set forth in 37 CFR 1.17 (p) has been submitted herewith.

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Signature	/Stephen Y. Liu/	Date (YYYY-MM-DD)	2016-02-11
Name/Print	Stephen Y. Liu	Registration Number	62883

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Application Number:	15010573
International Application Number:	
Confirmation Number:	9731
Title of Invention:	LAYERED MULTICAST AND FAIR BANDWIDTH ALLOCATION AND PACKET PRIORITIZATION
First Named Inventor/Applicant Name:	Erik Van Zijst
Customer Number:	122200
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New Applications Under 35 U.S.C. 111

If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.

National Stage of an International Application under 35 U.S.C. 371

If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.

New International Application Filed with the USPTO as a Receiving Office

If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US06/02995

A. CLASSIFICATION OF SUBJECT MATTER

IPC: **G01R 31/08**(2006.01);**G06F 11/00**(2006.01);**H04J 1/16**(2006.01),**3/16**(2006.01),**3/18**(2006.01);**H04L 12/28**(2006.01),**12/54**(2006.01);**G06F 15/16**(2006.01),**15/173**(2006.01)

USPC: 370/229,230,231,235,238,394,389,428,468,469,477;709/223,230,238,245

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
U.S. : 370/229,230,231,235,238,394,389,428,468,469,477; 709/223,230,238,245

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched
IETF, ACM, IEEE

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)
EAST: US-PGPUB, US-PAT, EPO, JPO, DERWENT, IBM_TDB

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	WO 2004/010250 A2 (Ruiz et al) 29.01.04, pg. 5 lines 16-29, pg. 13 lines 4-14	1-17
Y	Gorinsky, S. et al "Addressing Heterogeneity and Scalability in Layered Multicast Congestion Control", Technical Report TR2000-31, 24 Nov 2000; pgs. 2, 4 and 5	1, 2, 5 and 17
Y	US 2003/0233464 A1 (Walpole et al), 18.12.2003, para. 0038-39 and para. 0059-60	3, 4, 6 and 11
Y	US 6,845,105 B1 (Olsson et al), 18.01.2005, col. 6 lines 11-31, col. 8 lines 17-27	7 and 8
Y	US 2004/0081197 A1 (Liu), 29.04.2004; para. 0014	9
Y	US 2005/0002363 A1 (Cheng et al), 06.01.2005, para. 0052	10 and 13
Y	US 6,691,192 B2 (Ajanovic et al), 10.02.2004; col. 26 lines 46-67, col. 27 lines 1-3	11

☒ Further documents are listed in the continuation of Box C. ☐ See patent family annex.

* Special categories of cited documents:

"A" document defining the general state of the art which is not considered to be of particular relevance

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"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

al disclosure, use, exhibition or other means

ior to the international filing date but later than the

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvio art

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
07 May 2008 (07.05.2008)	29 AUG 2008
Name and mailing address of the ISA/US Mail Stop PCT, Attn: ISA/US Commissioner of Patents P.O. Box 1450 Alexandria, Virginia 22313-1450 Facsimile No. (571) 273-3201	Authorized officer Xavier Szewai Wong Telephone No. 571-272-3050

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INTERNATIONAL SEARCH REPORT

C. (Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

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Y	US 2004/0258094 A1 (Bashan et al), 23.12.2004; para. 0013	15
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X	US 2002/0114282 A1 (MeLampy et al), 22.08.2002; para. 0027 and 0074	18
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Y	US 2003/0026268 A1 (Navas), 06.02.2003; para. 0136, 0152, 0159 and 0182-184	19-22
X	US 2002/0114282 A1 (MeLampy et al), 22.08.2002; para. 0027 and 0074	18
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Y	US 2004/0052251 A1 (Mehrotra et al), 18.03.2004; para. 0031, fig. 1	23
X	US 2002/0114282 A1 (MeLampy et al), 22.08.2002; para. 0027 and 0074	18
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Y	US 2004/0165536 A1 (Xu et al), 26.08.2004; para. 0043 and 0048	24
X	US 2002/0114282 A1 (MeLampy et al), 22.08.2002; para. 0027 and 0074	18
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Y	US 2003/0123438 A1 (Li et al), 03.07.2003; para. 0077, fig. 1	25
X	US 2001/0048662 A1 (Suzuki et al), 06.12.2001; para. 0028-29	26
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Y	US 2002/0176406 A1 (Tsukada et al), 28.11.2002; para. 0041	27
X	US 2001/0048662 A1 (Suzuki et al), 06.12.2001; para. 0028-29	26
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Y	US 2003/0161311 A1 (Hiironniemi), 28.08.2003; para. 0006	28
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Y	US 2003/0123438 A1 (Li et al), 03.07.2003; para. 0077, fig. 1	33-36
X	US 2004/0010616 A1 (McCanne), 15.01.2004; para. 0027, 0045, 0046, 0048, 0122 and 0180; figs. 4 a, b and c	39, 40, 41, 45 and 46
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Y	US 2004/0133631 A1 (Hagen et al), 08.07.2004; para. 0056, fig. 3	42

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INTERNATIONAL SEARCH REPORT

C. (Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X --- Y	US 2004/0010616 A1 (McCanne), 15.01.2004; para. 0027, 0045, 0046, 0048, 0122 and 0180; figs. 4 a, b and c Cheng, C. et al "A Loop-Free Extended Bellman-Ford Routing Protocol Without Councing Effect" ACM 089791-332-9/89/0009/0224, 1989; pg. 227 and 228 section 3.2	39, 40, 41, 45 and 46 ---------- 43
X --- Y	US 2004/0010616 A1 (McCanne), 15.01.2004; para. 0027, 0045, 0046, 0048, 0122 and 0180; figs. 4 a, b and c US 2003/0212816 A1 (Bender et al), 13.11.2003; para. 0029	39, 40, 41, 45 and 46 ---------- 44
X --- Y	US 2004/0010616 A1 (McCanne), 15.01.2004; para. 0027, 0045, 0046, 0048, 0122 and 0180; figs. 4 a, b and c US 2004/0258002 A1 (Tran et al), 23.12.2004; para. 0036	39, 40, 41, 45 and 46 ---------- 47
X --- Y	US 2004/0010616 A1 (McCanne), 15.01.2004; para. 0027, 0045, 0046, 0048, 0122 and 0180; figs. 4 a, b and c US 2003/0123453 A1 (Ooghe et al), 03.07.2003; para. 0038	39, 40, 41, 45 and 46 ---------- 48

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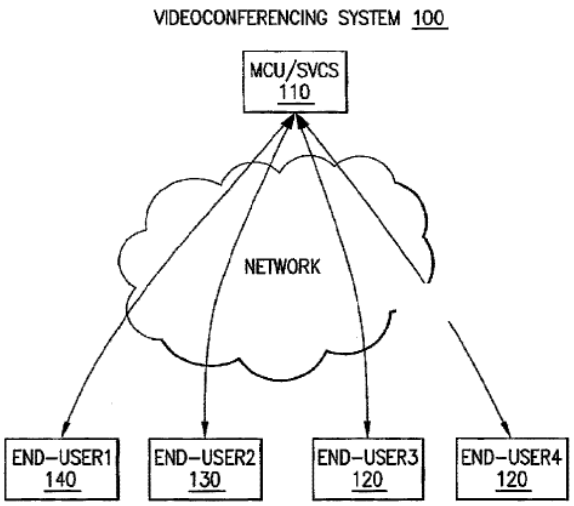
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(54) Title: SYSTEM AND METHOD FOR SCALABLE AND LOW-DELAY VIDEOCONFERENCING USING SCALABLE VIDEO CODING



(57) Abstract: Scalable video codecs are provided for use in videoconferencing systems and applications hosted on heterogeneous endpoints/receivers and network environments. The scalable video codecs provide a coded representation of a source video signal at multiple temporal, quality, and spatial resolutions.

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— *the filing date of the international application is within two months from the date of expiration of the priority period*

— *with information concerning one or more priority claims considered void*

SYSTEM AND METHOD FOR SCALABLE AND LOW-DELAY VIDEOCONFERENCING USING SCALABLE VIDEO CODING

SPECIFICATION

CROSS-REFERENCE TO RELATED APPLICATIONS

This application claims the benefit of United States provisional patent application Serial Nos. 60/701,108 filed July 20, 2005, 60/714,741 filed Spetember 7, 2005 and 60/723,392 filed October 4, 2005. Further, this application is related to co-filed United States patent application Serial Nos. [SVCSystem], [Trunk] and [Jitter]. All of the aforementioned priority and related applications are hereby incorporated by reference herein in their entireties.

FIELD OF THE INVENTION

The present invention relates to multimedia and telecommunications technology. In particular, the invention relates to systems and methods for videoconferencing between user endpoints with diverse access equipment or terminals, and over inhomogeneous network links.

BACKGROUND OF THE INVENTION

Videoconferencing systems allow two or more remote participants/endpoints to communicate video and audio with each other in real-time using both audio and video. When only two remote participants are involved, direct transmission of communications over suitable electronic networks between the two endpoints can be used. When more than two participants/endpoints are involved, a Multipoint Conferencing Unit (MCU), or bridge, is commonly used to connect to all the participants/endpoints. The MCU mediates communications between the multiple participants/endpoints, which may be connected, for example, in a star configuration.

For a videoconference, the partic ; or terminals are --pped with suitable encoding and decoding devices. An encoder formats local audio and video output at a transmitting endpoint into a coded form suitable for signal transmission over the electronic communication network. A decoder, in contrast, processes a received signal, which has encoded audio and video information, into a decoded form suitable for audio playback or image display at a receiving endpoint.

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Traditionally, an end-user's own image is also displayed on his/her screen to provide feedback (to ensure, for example, proper positioning of the person within the video window).

In practical videoconferencing system implementations over communication networks, the quality of an interactive videoconference between remote participants is determined by end-to-end signal delays. End-to-end delays of greater than 200ms prevent realistic live or natural interactions between the conferencing participants. Such long end-to-end delays cause the videoconferencing participants to unnaturally restrain themselves from actively participating or responding in order to allow in-transit video and audio data from other participants to arrive at their endpoints.

The end-to-end signal delays include acquisition delays (e.g., the time it takes to fill up a buffer in an A/D converter), coding delays, transmission delays (the time it takes to submit a packet-full of data to the network interface controller of an endpoint), and transport delays (the time a packet travels in a communication network from endpoint to endpoint). Additionally, signal-processing times through mediating MCUs contribute to the total end-to-end delay in the given system.

An MCU's primary tasks are to mix the incoming audio signals so that a single audio stream is transmitted to all participants, and to mix video frames or pictures transmitted by individual participants/endpoints into a common composite video frame stream, which includes a picture of each participant. It is noted that the terms frame and picture are used interchangeably herein, and further that coding of interlaced frames as individual fields or as combined frames (field-based or frame-based picture coding) can be incorporated as is obvious to persons skilled in the art. The MCUs, which are deployed in conventional communication networks systems, only offer a single common resolution (e.g., CIF or QCIF resolution) for all the individual pictures mixed into the common composite video frame distributed to all ants in a videoconferencing session. T ial communication ...orks systems do not readily provide customized videoconferencing functionality by which a participant can view other participants at different resolutions. Such desirable functionality allows the participant, for example, to view another specific participant (e.g., a speaking participant) in CIF resolution and view other, silent participants in QCIF resolution. MCUs can be configured to provide this desirable functionality by repeating the video mixing operation, as many times as the number of

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participants in a videoconference.' However, in such configurations, the MCU operations introduce considerable end-to-end delay. Further, the MCU must have sufficient digital signal processing capability to decode multiple audio streams, mix, and re-encode them, and also to decode multiple video streams, composite them into a single frame (with appropriate scaling as needed), and re-encode them again into a single stream. Video conferencing solutions (such as the systems commercially marketed by Polycom Inc., 4750 Willow Road, Pleasanton, CA 94588, and Tandberg, 200 Park Avenue, New York, NY 10166) must use dedicated hardware components to provide acceptable quality and performance levels.

The performance levels of and the quality delivered by a videoconferencing solution are also a strong function of the underlying communication network over which it operates. Videoconferencing solutions, which use ITU H.261, H.263, and H.264 standard video codecs, require a robust communication channel with little or no loss for delivering acceptable quality. The required communication channel transmission speeds or bitrates can range from 64 Kbps up to several Mbps. Early videoconferencing solutions used dedicated ISDN lines, and newer systems often utilize high-speed Internet connections (e.g., fractional T1, T1, T3, etc.) for high-speed transmission. Further, some videoconferencing solutions exploit Internet Protocol ("IP") communications, but these are implemented in a private network environment to ensure bandwidth availability. In any case, conventional videoconferencing solutions incur substantial costs associated with implementing and maintaining the dedicated high-speed networking infrastructure needed for quality transmissions.

The costs of implementing and maintaining a dedicated videoconferencing network are avoided by recent "desktop videoconferencing" systems, which exploit high bandwidth corporate data network connections (e.g., 100 Mbit, Ethernet). In these desktop videoconferencing solutions, common personal s (PCs), which are equipped with USB- ideo cameras and priate software applications for performing encoding/decoding and network transmission, are used as the participant/endpoint terminals.

Recent advances in multimedia and telecommunications technology involve integration of video communication and conferencing capabilities with Internet Protocol ("IP") communication systems such as IP PBX, instant messaging, web conferencing, etc. In order to effectively integrate video conferencing into such

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systems, both point-to-point and multipoint communications must be supported. However, the available network bandwidth in IP communication systems can fluctuate widely (e.g., depending on time of day and overall network load), making these systems unreliable for the high bandwidth transmissions required for video communications. Further, videoconferencing solutions implemented on IP communication systems must accommodate both network channel heterogeneity and endpoint equipment diversity associated with the Internet system. For example, participants may access videoconferencing services over IP channels having very different bandwidths (e.g., DSL vs. Ethernet) using a diverse variety of personal computing devices.

The communication networks on which videoconferencing solutions are implemented can be categorized as providing two basic communication channel architectures. In one basic architecture, a guaranteed quality of service (QoS) channel is provided via a dedicated direct or switched connection between two points (e.g., ISDN connections, T1 lines, and the like). Conversely, in the second basic architecture, the communication channels do not guarantee QoS, but are only "best-effort" packet delivery channels such as those used in Internet Protocol (IP) -based networks (e.g., Ethernet LANs).

Implementing video conferencing solutions on IP-based networks may be desirable, at least due to the low cost, high total bandwidth, and widespread availability of access to the Internet. As noted previously, IP-based networks typically operate on a best-effort basis, i.e., there is no guarantee that packets will reach their destination, or that they will arrive in the order they were transmitted. However, techniques have been developed to provide different levels of quality of service (QoS) over the putatively best-effort channels. The techniques may include protocols such as DiffServ for specifying and controlling network traffic by class so that certain types of traffic get precedence and RSVP. These protocols can ensure ndwidth and/or delays for portions of the idth. Techniques as forward error correction (FEC) and automatic repeat request (ARQ) mechanisms may also be used to improve recovery mechanisms for lost packet transmissions and to mitigate the effects of packet loss.

Implementing video conferencing solutions on IP-based networks requires consideration of the video codecs used. Standard video codecs such as the standard H.261, H.263 codecs designated for videoconferencing and standard MPEG-

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1 and MPEG-2 Main Profile codecs designated for Video CDs and DVDs, respectively, are designed to provide a single bitstream ("single-layer") at a fixed bitrate. Some of these codecs may be deployed without rate control to provide a variable bitrate stream (e.g., MPEG-2, as used in DVDs). However, in practice, even without rate control, a target operating bitrate is established depending on the specific infrastructure. These video codecs designs are based on the assumption that the network is able to provide a constant bitrate, and a practically error-free channel between the sender and the receiver. The H-series Standard codecs, which are designed specifically for person-to-person communication applications, offer some additional features to increase robustness in the presence of channel errors, but are still only tolerant to a very small percentage of packet losses (typically only up to 2-3%).

Further, the standard video codecs are based on "single-layer" coding techniques, which are inherently incapable of exploiting the differentiated QoS capabilities provided by modern communication networks. An additional limitation of the single-layer coding techniques for video communications is that even if a lower spatial resolution display is required or desired in an application, a full resolution signal must be received and decoded with downscaling performed at a receiving endpoint or MCU. This wastes bandwidth and computational resources.

In contrast to the aforementioned single-layer video codecs, in "scalable" video codecs based on "multi-layer" coding techniques, two or more bitstreams are generated for a given source video signal: a base layer and one or more enhancement layers. The base layer may be a basic representation of the source signal at a minimum quality level. The minimum quality representation may be reduced in the SNR (quality), spatial, or temporal resolution aspects or a combination of these aspects of the given source video signal. The one or more enhancement layers correspond to information for increasing the quality of the SNR (quality), ›r temporal resolution aspects of the base l video codecs have developed in view of heterogeneous network environments and/or heterogeneous receivers. The base layer can be transmitted using a reliable channel, i.e., a channel with guaranteed Quality of Service (QoS). Enhancement layers can be transmitted with reduced or no QoS. The effect is that recipients are guaranteed to receive a signal with at least a minimum level of quality (the base layer signal). Similarly, with heterogeneous receivers that may have different screen sizes, a small

picture size signal may be transmitted to, e.g., a portable device, and a full size picture may be transmitted to a system equipped with a large display.

Standards such as MPEG-2 specify a number of techniques for performing scalable coding. However, practical use of "scalable" video codecs has been hampered by the increased cost and complexity associated with scalable coding, and the lack of widespread availability of high bandwidth IP-based communication channels suitable for video.

Consideration is now being given to developing improved scalable codec solutions for video conferencing and other applications. Desirable scalable codec solutions will offer improved bandwidth, temporal resolution, spatial quality, spatial resolution, and computational power scalability. Attention is in particular directed to developing scalable video codecs that are consistent with simplified MCU architectures for versatile videoconferencing applications. Desirable scalable codec solutions will enable zero-delay MCU architectures that allow cascading of MCUs in electronic networks with no or minimal end-to-end delay penalties.

SUMMARY OF THE INVENTION

The present invention provides scalable video coding (SVC) systems and methods (collectively, "solutions") for point-to-point and multipoint conferencing applications. The SVC solutions provide a coded "layered" representation of a source video signal at multiple temporal, quality, and spatial resolutions. These resolutions are represented by distinct layer/bitstream components that are created by endpoint/terminal encoders.

The SVC solutions are designed to accommodate diversity in endpoint/receivers devices and in heterogeneous network characteristics, including, for example, the best-effort nature of networks such as those based on the Internet Protocol. The scalable aspects of the video coding techniques employed allow conferencing applications to adapt to different conditions, and also accommodate different end-user requirements (e.g., a user may elect to view another user at a high or low spatial resolution).

Scalable video codec designs allow error-resilient transmission of video in point-to-point and multipoint scenarios, and allow a conferencing bridge to provide continuous presence, rate matching, error localization, random entry and

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personal layout conferencing features, without decoding or recoding in-transit video streams and without any decrease in the error resilience of the stream.

An endpoint terminal, which is designed for video communication with other endpoints, includes video encoders/decoders that can encode a video signal into one or more layers of a multi-layer scalable video format for transmission. The video encoders/decoders can correspondingly decode received video signal layers, simultaneously or sequentially, in as many video streams as the number of participants in a videoconference. The terminal maybe implemented in hardware, software, or a combination thereof in a general-purpose PC or other network access device. The scalable video codecs incorporated in the terminal may be based on coding methods and techniques that are consistent with or based on industry standard encoding methods such as H.264.

In an H.264 based SVC solution, a scalable video codec creates a base layer that is based on standard H.264 AVC encoding. The scalable video codec further creates a series of SNR enhancement layers by successively encoding, using again H.264 AVC, the difference between the original signal and the one coded at the previous layer with an appropriate offset. In a version of this scalable video codec, DC values of the direct cosine transform (DCT) coefficients are not coded in the enhancement layers, and further, a conventional deblocking filter is not used.

In an SVC solution, which is designed to use SNR scalability as a means of implementing spatial scalability, different quantization parameters (QP) are selected for the base and enhancement layers. The base layer, which is encoded at higher QP, is optionally low-pass filtered and downsampled for display at receiving endpoints/terminals.

In another SVC solution, the scalable video codec is designed as a spatially scalable encoder in which a reconstructed base layer H.264 low-resolution signal is upsampled at the encoder and subtracted from the original signal. The e is fed to the standard encoder operati lution, after being t by a set value. In another version, the upsampled H.264 low-resolution signal is used as an additional possible reference frame in the motion estimation process of a standards-based high-resolution encoder.

The SVC solutions may involve adjusting or changing threading modes or spatial scalability modes to dynamically respond to network conditions and participants' display preferences.

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BRIEF DESCRIPTION OF THE DRAWINGS

Further features of the invention, its nature, and various advantages will be more apparent from the following detailed description of the preferred embodiments and the accompanying drawing in which:

FIGS. 1A and 1B are schematic diagrams illustrating exemplary architectures of a videoconferencing system, in accordance with the principles of the present invention.

FIG. 2 is a block diagram illustrating an exemplary end-user terminal, in accordance with the principles of the present invention.

FIG. 3 is a block diagram illustrating an exemplary architecture of an encoder for the base and temporal enhancement layers (i.e., layers 0 though 2), in accordance with the principles of the present invention.

FIG. 4 is a block diagram illustrating an exemplary layered picture coding structure for the base, temporal enhancement, and SNR or spatial enhancement layers, in accordance with the principles of the present invention.

FIG. 5 is a block diagram illustrating the structure of an exemplary SNR enhancement layer encoder, in accordance with the principles of the present invention.

FIG. 6 is a block diagram illustrating the structure of an exemplary single-loop SNR video encoder, in accordance with the principles of the present invention.

FIG. 7 is a block diagram illustrating an exemplary structure of a base layer for a spatial scalability video encoder, in accordance with the principles of the present invention.

FIG. 8 is a block diagram illustrating an exemplary structure of a spatial scalability enhancement layer video encoder, in accordance with the principles of the present invention.

FIG. 9 is a block diagram illustra ary structure of a spatial scalability enhancement layer video encoder with inter-layer motion prediction, in accordance with the principles of the present invention.

FIGS. 10 and 11 are block diagrams illustrating exemplary base layer and SNR enhancement layer video decoders, respectively, in accordance with the principles of the present invention.

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FIG. 12 is a block diagram illustrating an exemplary SNR enhancement layer, single-loop video decoder, in accordance with the principles of the present invention.

FIG. 13 is block diagram illustrating an exemplary spatial scalability enhancement layer video decoder, in accordance with the principles of the present invention.

FIG. 14 is block diagram illustrating an exemplary video decoder for spatial scalability enhancement layers with inter-layer motion prediction, in accordance with the principles of the present invention.

FIGS. 15 and 16 are block diagrams illustrating exemplary alternative layered picture coding structures and threading architectures, in accordance with the principles of the present invention.

FIG. 17 is a block diagram illustrating an exemplary Scalable Video Coding Server (SVCS), in accordance with the principles of the present invention.

FIG. 18 is a schematic diagram illustrating the operation of an SVCS switch, in accordance with the principles of the present invention.

FIGS. 19 and 20 are illustrations of exemplary SVCS Switch Layer and Network Layer Configuration Matrices, in accordance with the principles of the present invention.

Throughout the figures, unless otherwise stated, the same reference numerals and characters are used to denote like features, elements, components or portions of the illustrated embodiments. Moreover, while the present invention will now be described in detail with reference to the figures, it is done so in connection with the illustrative embodiments.

DETAILED DESCRIPTION OF THE INVENTION

The present invention provides systems and techniques for scalable ding (SVC) of video data signals for m int-to-point video rencing applications. The SVC systems and techniques (collectively "solutions") are designed to allow the tailoring or customization of delivered video data in response to different user participants/endpoints, network transmission capabilities, environments, or other requirements in a videoconference. The inventive SVC solutions provide compressed video data in a multi-layer format, which can be readily switched layer-by-layer between conferencing participants using convenient

zero- or low-algorithmic delay switching mechanisms. Exemplary zero- or low-algorithmic delay switching mechanisms - Scalable Video Coding Servers (SVCS), are described in co-filed U.S. Patent application No. __[SVCS]_____.

FIGS. 1A and 1B show exemplary videoconferencing system 100 arrangements based on the inventive SVC solutions. Videoconferencing system 100 may be implemented in a heterogeneous electronic or computer network environment, for multipoint and point-to-point client conferencing applications. System 100 uses one or more networked servers (e.g., an SVCS or MCU 110), to coordinate the delivery of customized data to conferencing participants or clients 120, 130, and 140. As described in co-pending U.S. Patent Application S/N ____, MCU 110 may coordinate the delivery of a video stream 150 generated by endpoint 140 for transmission to other conference participants. In system 100, a video stream is first suitably coded or scaled down using the inventive SVC techniques into a multiplicity of data components or layers. The multiple data layers may have differing characteristics or features (e.g., spatial resolutions, frame rates, picture quality, signal-to-noise ratio qualities (SNR), etc.). The differing characteristics or features of the data layers may be suitably selected in consideration, for example, of the varying individual user requirements and infrastructure specifications in the electronic network environment (e.g., CPU capabilities, display size, user preferences, and bandwidths). MCU 110 is suitably configured to select an appropriate amount of information (i.e., SVC layers) for each particular participant/recipient in the conference from a received data stream (e.g., SVC video stream 150), and to forward only the selected or requested amounts of information/layers to the respective participants/recipients 120-130. MCU 110 may be configured to make the suitable selections in response to receiving-endpoint requests (e.g., the picture quality requested by individual conference participants) and upon consideration of network conditions and policies.

This customized data selection anc heme exploits the nal structure of the SVC video stream, which allows clear division of the video stream into multiple layers having different resolutions, frame rates, and/or bandwidths, etc. FIG. 1B, which is reproduced from the referenced patent application ____[SVCS]____, shows an exemplary internal structure of SVC video stream 150 that represents a media input of endpoint 140 to the conference. The exemplary

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internal structure of SVC video stream 150 includes a "base" layer 150b, and one or more distinct "enhancement" layers 150a.

FIG. 2 shows an exemplary participant/endpoint terminal 140, which is designed for use with SVC-based videoconferencing systems (e.g., system 100). Terminal 140 includes human interface input/output devices (e.g., a camera 210A, a microphone 210B, a video display 250C, a speaker 250D), and a network interface controller card (NIC) 230 coupled to input and output signal multiplexer and demultiplexer units (e.g., packet MUX 220A and packet DMUX 220B). NIC 230 may be a standard hardware component, such as an Ethernet LAN adapter, or any other suitable network interface device.

Camera 210A and microphone 210B are designed to capture participant video and audio signals, respectively, for transmission to other conferencing participants. Conversely, video display 250C and speaker 250D are designed to display and play back video and audio signals received from other participants, respectively. Video display 250C may also be configured to optionally display participant/terminal 140's own video. Camera 210A and microphone 210B outputs are coupled to video and audio encoders 210G and 210H via analog-to-digital converters 210E and 210F, respectively. Video and audio encoders 210G and 210H are designed to compress input video and audio digital signals in order to reduce the bandwidths necessary for transmission of the signals over the electronic communications network. The input video signal may be live, or pre-recorded and stored video signals.

Video encoder 210G has multiple outputs connected directly to packet MUX 220A. Audio encoders 210H output is also connected directly to packet MUX 220A. The compressed and layered video and audio digital signals from encoders 210G and 210H are multiplexed by packet MUX 220A for transmission over the communications network via NIC 230. Conversely, compressed video and audio ignals received over the communications n 230 are forwarded acket DMUX 220B for demultiplexing and further processing in terminal 140 for display and playback over video display 250C and speaker 250D.

Captured audio signals may be encoded by audio encoder 210H using any suitable encoding techniques including known techniques, for example, G.711 and MPEG-1. In an implementation of videoconferencing system 100 and terminal 140, G.711 encoding is preferred for audio encoding. Captured video signals are

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encoded in a layered coding format by video encoder 210G using the SVC techniques described herein. Packet MUX 220A may be configured to multiplex the input video and audio signals using, for example, the RTP protocol or other suitable protocols. Packet MUX 220A also may be configured to implement any needed QoS-related protocol processing.

In system 100, each stream of data from terminal 140 is transmitted in its own virtual channel (or port number in IP terminology) over the electronics communication network. In an exemplary network configuration, QoS may be provided via Differentiated Services (DiffServ) for specific virtual channels or by any other similar QoS-enabling technique. The required QoS setups are performed prior to use of the systems described herein. DiffServ (or the similar QoS-enabling technique used) creates two different categories of channels implemented via or in network routers (not shown). For convenience in description, the two different categories of channels are referred to herein as "high reliability" (HRC) and "low reliability" (LRC) channels, respectively. In the absence of an explicit method for establishing an HRC or if the HRC itself is not reliable enough, the endpoint (or the MCU 110 on behalf of the endpoint) may (i) proactively transmit the information over the HRC repeatedly (the actual number of repeated transmissions may depend on channel error conditions), or (ii) cache and retransmit information upon the request of a receiving endpoint or SVCS, for example, in instances where information loss in transmission is detected and reported immediately. These methods of establishing an HRC can be applied in the client-to-MCU, MCU-to-client, or MCU-to-MCU connections individually or in any combination, depending on the available channel type and conditions.

For use in a multi-participant videoconferencing system, terminal 140 is configured with one or more pairs of video and audio decoders (e.g., decoders 230A and 230B) designed to decode signals received from the conferencing nts who are to be seen or heard at termina ; of decoders 230A ..... 230B may be designed to process signals individually participant-by-participant or to sequentially process a number of participant signals. The configuration or combinations of pairs of video and audio decoders 230A and 230B included in terminal 140 may be suitably selected to process all participant signals received at terminal 140 with consideration of the parallel and/or sequential processing design features of the encoders. Further, packet DMUX 220B may be configured to receive

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packetized signals from the conferencing participants via NIC 230, and to forward the signals to appropriate pairs of video and audio decoders 230A and 230B for parallel and/or sequential processing.

Further in terminal 140, audio decoder 230B outputs are connected to audio mixer 240 and a digital-to-analog converter (DA/C) 250B, which drives speaker 250D to play back received audio signals. Audio mixer 240 is designed to combine individual audio signals into a single signal for playback. Similarly, video decoder 230A outputs are combined in frame buffer 250A by a compositor 260. A combined or composite video picture from frame buffer 250A is displayed on monitor 250C.

Compositor 260 may be suitably designed to position each decoded video picture at a corresponding designated position in the composite frame or displayed picture. For example, monitor 250C display may be split into four smaller areas. Compositor 260 may obtain pixel data from each of video decoders 230A in terminal 140 and place the pixel data in an appropriate frame buffer 250A position (e.g., filling up the lower right picture). To avoid double buffering (e.g., once at the output of decoder 230B and once at frame buffer 250A), compositor 260 may, for example, be configured as an address generator that drives the placement of output pixels of decoder 230B. Alternative techniques for optimizing the placement of individual video decoder 230A outputs on display 210 C may also be used to similar effect.

It will be understood that the various terminal 140 components shown in FIG. 2 may be implemented in any suitable combination of hardware and/or software components, which are suitably interfaced with each other. The components may be distinct stand-alone units or integrated with a personal computer or other device having network access capabilities.

With reference to video encoders used in terminal 140 for scalable video coding, FIGS. 3-9 respectively show various scalable video encoders or codecs that may be deployed in terminal 140.

FIG. 3 shows exemplary encoder architecture 300 for compressing input video signals in a layered coding format (e.g., layers L0, L1, and L2 in SVC terminology, where L0 is the lowest frame rate). Encoder architecture 300 represents a motion-compensated, block-based transform codec based, for example, on a standard H.264/MPEG-4 AVC design or other suitable codec designs. Encoder architecture 300 includes a FRAME BUFFERS block 310, an ENC REF CONTROL

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block 320, and a DeBlocking Filter block 360 in addition to conventional "text-book" variety video coding process blocks 330 for motion estimation (ME), motion compensation (MC), and other encoding functions. The motion-compensated, block-based codec used in system 100/terminal 140 may be a single-layer temporally predictive codec, which has a regular structure of I, P, and B pictures. A picture sequence (in display order) may, for example, be "IBBPBBP". In the picture sequence, the 'P' pictures are predicted from the previous P or I picture, whereas the B pictures are predicted using both the previous and next P or I picture. Although the number of B pictures between successive I or P pictures can vary, as can the rate in which I pictures appear, it is not possible, for example, for a P picture to use as a reference for prediction another P picture that is earlier in time than the most recent one. Standard H.264 coding advantageously provides an exception in that two reference picture lists are maintained by the encoder and decoder, respectively. This exception is exploited by the present invention to select which pictures are used as references and also which references are used for a particular picture that is to be coded. In FIG. 3, FRAME BUFFERS block 310 represents memory for storing the reference picture list(s). ENC REF CONTROL block 310 is designed to determine which reference picture is to be used for the current picture at the encoder side.

The operation of ENC REF CONTROL block 310 is placed in context further with reference to an exemplary layered picture coding "threading" or "prediction chain" structure shown in FIG. 4. (FIGS. 8-9 show alternative threading structures). Codecs 300 utilized in implementations of the present invention may be configured to generate a set of separate picture "threads" (e.g., a set of three threads 410-430) in order to enable multiple levels of temporal scalability resolutions (e.g., L0-L2) and other enhancement resolutions (e.g., S0-S2). A thread or prediction chain is defined as a sequence of pictures that are motion-compensated using pictures either from the same thread, or pictures from a lower level thread. The arrows in FIG. 4 the direction, source, and target of pre : threads 410-430. ads 410-420 have a common source L0 but different targets and paths (e.g., targets L2, L2, and L0, respectively). The use of threads allows the implementation of temporal scalability, since any number of top-level threads can be eliminated without affecting the decoding process of the remaining threads.

It will be noted that in encoder 300, according to H.264, ENC REF CONTROL block may use only P pictures as reference pictures. However, B pictures

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also may be used with accompanying gains in overall compression efficiency. Using even a single B picture in the set of threads (e.g., by having L2 be coded as a B picture) can improve compression efficiency. In traditional interactive communications, the use of B pictures with prediction from future pictures increases the coding delay and is therefore avoided. However, the present invention allows the design of MCUs with practically zero processing delay. (See co-filed U.S. Patent Application No. SCVS). With such MCUs, it is possible to utilize B pictures and still operate with an end-to-end delay that is lower than state-of-the-art traditional systems.

In operation, encoder 300 output L0 is simply a set of P pictures spaced four pictures apart. Output L1 has the same frame rate as L0, but only prediction based on the previous L0 picture is allowed. Output L2 pictures are predicted from the most recent L0 or L1 picture. Output L0 provides one fourth (1:4) of the full temporal resolution, L1 doubles the L0 frame rate (1:2), and L2 doubles the L0+L1 frame rate (1:1). A lesser number (e.g., less than 3, L0-L2) or an additional number of layers may be similarly constructed by encoder 300 to accommodate different bandwidth/scalability requirements or different specifications of implementations of the present invention.

In accordance with the present invention, for additional scalability, each compressed temporal video layer (e.g., L0-L1) may include or be associated with one or more additional components related to SNR quality scalability and/or spatial scalability. FIG. 4 shows one additional enhancement layer (SNR or spatial). Note that this additional enhancement layer will have three different components (S0-S2), each corresponding to the three different temporal layers (L0-L2).

FIGS. 5 and 6 show SNR scalability encoders 500 and 600, respectively. FIGS. 7-9 show spatial scalability encoders 700-900, respectively. It will be understood that SNR scalability encoders 500 and 600 and spatial scalability encoders 700-900 are based on and may use the same processing blocks (e.g., blocks and 320) as encoder 300 (FIG. 3).

It is recognized that for the base layer of an SNR scalable codec, the input to the base layer codec is a full resolution signal (FIGS. 5-6). In contrast, for the base layer of a spatial scalability codec, the input to the base layer codec is a downsampled version of the input signal FIGS. 7-9. It is also noted that the SNR/spatial quality enhancement layers S0-S2 may be coded according to the forthcoming ITU-T H.264 Annex F standard or other suitable technique.

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FIG. 5 shows the structure of an exemplary SNR enhancement encoder 500, which is similar to the structure of layered encoder 300 based on H.264 shown in FIG. 3. It will, however, be noted that the input to the SNR enhancement layer coder 500 is the difference between the original picture (INPUT, FIG. 3) and the reconstructed coded picture (REF, FIG. 3) as recreated at the encoder.

FIG. 5 also shows use of encoder 500 based on H.264 for encoding the coding error of the previous layers. Non-negative inputs are required for such encoding. To ensure this, the input (INPUT-REF) to encoder 500 is offset by a positive bias (e.g., by OFFSET 340). The positive bias is removed after decoding and prior to the addition of the enhancement layer to the base layer. A deblocking filter that is typically used in H.264 codec implementations (e.g., Deblocking filter 360, FIG. 3) is not used in encoder 500. Further, to improve subjective coding efficiency, DC direct cosine transform (DCT) coefficients in the enhancement layer may be optionally ignored or eliminated in encoder 500. Experimental results indicate that the elimination of the DC values in an SNR enhancement layer (S0-S2) does not adversely impact picture quality, possibly due to the already fine quantization performed at the base layer. A benefit of this design is that the exactly same encoding/decoding hardware or software can be used both for the base and SNR enhancement layers. In a similar fashion – spatial scalability (at any ratio) may be introduced by applying the H.264 base layer coding to a downsampled image and upsampling the reconstructed image before calculating the residual. Further, standards other than H.264 can be used for compressing both layers.

In the codecs of the present invention, in order to decouple the SNR and temporal scalabilities, all motion prediction within a temporal layer and across temporal layers may be performed using the base layer streams only. This feature is shown in FIG. 4 by the open arrowheads 415 indicating temporal prediction in the base layer block (L) rather than in the combination of L and S blocks. For this all layers may be coded at CIF resolution resolution pictures be derived by decoding the base layer stream having a certain temporal resolution, and downsampling in each spatial dimension by a dyadic factor (2), using appropriate low-pass filtering. In this manner, SNR scalability can be used to also provide spatial scalability. It will be understood that CIF/QCIF resolutions are referred to only for purposes of illustration. Other resolutions (e.g., VGA/QVGA) can be supported by the inventive codecs without any change in codec design. The

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codecs may also include traditional spatial scalability features in the same or similar manner as described above for the inclusion of the SNR scalability feature. Techniques provided by MPEG-2 or H.264 Annex F may be used for including traditional spatial scalability features.

The architecture of codecs designed to decouple the SNR and temporal scalabilities described above, allows frame rates in ratios of 1:4 (L0 only), 1:2 (L0 and L1), or 1:1 (all three layers). A 100% bitrate increase is assumed for doubling the frame rate (base is 50% of total), and a 150% increase for adding the S layer at its scalability point (base is 40% of total). In a preferred implementation, the total stream may, for example, operate at 500 Kbps, with the base layer operating at 200 Kbps. A rate load of 200/4=50 Kbps per frame may be assumed for the base layer, and (500-200)/4=75 Kbps for each frame. It will be understood that the aforementioned target bitrates and layer bitrate ratio values are exemplary and have been specified only for purposes of illustrating the features of the present invention, and that the inventive codecs can be easily adapted to other target bitrates, or layer bitrate ratios.

Theoretically, up to 1:10 scalability (total vs. base) is available when the total stream and the base layer operate at 500 Kbps and 200 Kbps, respectively. TABLE I shows examples of the different scalability options available when SNR scalability is used to provide spatial scalability.

TABLE I

Scalability Options

Temporal (fps)	**QCIF* (Kbps) L only**	**CIF (Kbps) L to L+S**
7.5 (L0)	50	50-125
15 (L0+L1)	100	100-250
30 (L0+L1+L2)	200	200-500

* Although no QCIF component is present in the bitstreams, it can be provided by scaling down the CIF image by a factor of 2. In this example, the lower resolution of QCIF mably allows this operation to be perforn CIF layer without eable effect on quality.

FIG. 6 shows alternate SNR scalable encoder 600, which is based on a single encoding loop scheme. The structure and operation of SNR scalable encoder 600 is based on that of encoder 300 (FIG. 3). Additionally in encoder 600, DCT coefficients that are quantized by Q0 are inverse-quantized and subtracted from the

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original unquantized coefficients to obtain the residual quantization error (QDIFF 610) of the DCT coefficients. The residual quantization error information (QDIFF 610) is further quantized with a finer quantizer Q1 (Block 620), entropy coded (VLC/BAC), and output as the SNR enhancement layer S. It is noted that there is a single coding loop in operation, i.e., the one operating at the base layer.

Terminal 140/video 230 encoders may be configured to provide spatial scalability enhancement layers in addition to or instead of the SNR quality enhancement layers. For encoding spatial scalability enhancement layers, the input to the encoder is the difference between the original high-resolution picture and the upsampled reconstructed coded picture as created at the encoder. The encoder operates on a downsampled version of the input signal. FIG. 7 shows exemplary encoder 700 for encoding the base layer for spatial scalability. Encoder 700 includes a downsampler 710 at the input of low-resolution base layer encoder 720. For a full resolution input signal at CIF resolution, base layer encoder 720 may with suitable downsampling operate at QCIF, HCIF (half CIF), or any other resolution lower than CIF. In an exemplary mode, base layer encoder 720 may operate at HCIF. HCIF-mode operation requires downsampling of a CIF resolution input signal by about a factor of $\sqrt{2}$ in each dimension, which reduces the total number of pixels in a picture by about one-half of the original input. It is noted that in a video conferencing application, if a QCIF resolution is desired for display purposes, then the decoded base layer will have to be further downsampled from HCIF to QCIF.

It is recognized that an inherent difficulty in optimizing the scalable video encoding process for video conferencing applications is that there are two or more resolutions of the video signal being transmitted. Improving the quality of one of the resolutions may result in corresponding degradation of the quality of the other resolution(s). This difficulty is particularly pronounced for spatially scalable coding, and in current art video conferencing systems in which the coded resolution and the :esolutions are identical. The inventive t oupling the coded l resolution from the intended display resolution provides yet another tool in a codec designer's arsenal to achieve a better balance between the quality and bitrates associated with each of the resolutions. According to the present invention, the choice of coded resolution for a particular codec may be obtained by considering the rate-distortion (R-D) performance of the codec across different spatial resolutions, taking into account the total bandwidth available, the desired bandwidth partition

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across the different resolutions, and the desired quality difference differential that each additional layer should provide.

Under such a scheme, a signal may be coded at CIF and one-third CIF (1/3CIF) resolutions. Both CIF and HCIF resolution signals may be derived for display from the CIF-coded signal. Further, both 1/3CIF and QCIF resolution signals may similarly be derived for display from the 1/3CIF-coded signal. The CIF and 1/3CIF resolution signals are available directly from the decoded signals, whereas the latter HCIF and QCIF resolution signals may be obtained upon appropriate downsampling of the decoded signals. Similar schemes may also be applied in the case of other target resolutions (e.g., VGA and one-third VGA, from which half VGA and quarter VGA can be derived).

The schemes of decoupling the coded signal resolution from the intended display resolution, together with the schemes for threading video signal layers (FIG. 4, and FIGS. 15 and 16), provide additional possibilities for obtaining target spatial resolutions with different bitrates, in accordance with the present invention. For example, in a video signal coding scheme, spatial scalability may be used to encode the source signal at CIF and 1/3CIF resolutions. SNR and temporal scalabilities may be applied to the video signal as shown in FIG. 4. Further, the SNR encoding used may be a single loop or a double loop encoder (e.g., encoder 600 FIG. 6 or encoder 500 FIG. 5), or may be obtained by data partitioning (DP). The double loop or DP encoding schemes will likely introduce drift whenever data is lost or removed. However, the use of the layering structure will limit the propagation of the drift error until the next L0 picture, as long as the lost or removed data belongs to the L1, L2, S1, or S2 layers. Further taking into account the fact that the perception of errors is reduced when the spatial resolution of the displayed video signal is reduced, it is possible to obtain a low bandwidth signal by eliminating or removing data from the L1, L2, S1, and S2 layers, decoding the 1/3CIF resolution, and displaying it npled at a QCIF resolution. The loss of d lownsampling will e errors in the corresponding L1/S1 and L2/S2 pictures, and will also propagate errors to future pictures (until the next L0 picture), but the fact that the display resolution is reduced makes the quality degradation less visible to a human observer. Similar schemes may be applied to the CIF signal, for display at HCIF, 2/3 CIF or at any other desired resolution. These schemes advantageously allow the use of quality scalability to effect spatial scalability at various resolutions, and at various bitrates.

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FIG. 8 shows the structure of an exemplary spatially scalable enhancement layer encoder 800, which, like encoder 500, uses the same H.264 encoder structure for encoding the coding error of the previous layers but includes an upsampler block 810 on the reference (REF) signal. Since non-negative input is assumed for such an encoder, the input values are offset (e.g., by offset 340) prior to coding. Values that still remain negative are clipped to zero. The offset is removed after decoding and prior to the addition of the enhancement layer to the upsampled base layer.

For the spatial enhancement layer encoding, like for the SNR layer encoding (FIG. 6), it may be advantageous to use frequency weighting in the quantizers (Q) of the DCT coefficients. Specifically, coarser quantization can be used for the DC and its surrounding AC coefficients. For example, a doubling of the quantizer step size for the DC coefficient may be very effective.

FIG. 9 shows the exemplary structure of another spatially scalable video encoder 900. In encoder 900, unlike in encoder 800, the upsampled reconstructed base layer picture (REF) is not subtracted from the input, but instead serves as an additional possible reference picture in the motion estimation and mode selection blocks 330 of the enhancement layer encoder. Encoder 900 can accordingly be configured to predict the current full resolution picture either from a previous coded full resolution picture (or future picture, for B pictures), or an upsampled version of the same picture coded at the lower spatial resolution (inter-layer prediction). It should be noted that, whereas encoder 800 can be implemented using the same codec for the base and enhancement layers with only the addition of downsampler 710, upsampler 810, and offset 340 blocks, encoder 900 requires that the enhancement layer encoder's motion estimation (ME) block 330* is modified. It is also noted that enhancement layer encoder 900 operates on the regular pixel domain, rather than a differential domain.

It is also possible to combine the n a previous high ution picture and the upsampled base layer picture by using the B picture prediction logic of a standard single-layer encoder, such as an H.264 encoder. This can be accomplished by modifying the B picture prediction reference for the high resolution signal so that the first picture is the regular or standard prior high resolution picture, and the second picture is the upsampled version of the base layer picture. The encoder then performs prediction as if the second picture is a regular B picture, thus

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utilizing all the high-efficiency motion vector prediction and coding modes (e.g., spatial and temporal direct modes) of the encoder. Note that in H.264, "B" picture coding stands for 'bi-predictive' rather than 'bi-directional', in the sense that the two reference pictures could both be past or future pictures of the picture being coded, whereas in traditional 'bi-directional' B picture coding (e.g., MPEG-2) one of the two reference pictures is a past picture and the other is a future picture. This embodiment allows the use of a standard encoder design, with minimal changes that are limited to the picture reference control logic and the upsampling module.

In an implementation of the present invention, the SNR and spatial scalability encoding modes may be combined in one encoder. For such an implementation, video-threading structures (e.g., shown in two dimensions in FIG. 4) may be expanded in a third dimension, corresponding to the additional third scalability layer (SNR or spatial). An implementation in which SNR scalability is added on the full resolution signal of a spatially scalable codec may be attractive in terms of range of available qualities and bitrates.

FIGS. 10-14 show exemplary architectures for a base layer decoder 1000, SNR enhancement layer decoder 1100, a single-loop SNR enhancement layer decoder 1200, a spatially scalable enhancement layer decoder 1300 and spatially scalable enhancement layer decoder 1400 with interlayer motion prediction, respectively. These decoders complement encoders 300, 500, 600, 700, 800 and 900. Decoders 1000, 1100, 1200, 1300, and 1400 may be included in terminal 140 decoders 230A as appropriate or needed.

The scale video coding/decoding configurations of terminal 140 present a number of options for transmitting the resultant layers over the HRC and LRC in system 100. For example, (L0 and S0) layers or (L0, S0 and L1) layers may be transmitted over HRC. Alternate combinations also may be used as desired, upon due consideration of network conditions, and the bandwidths of high and low y channels. For example, depending o litions, it may be able to code S0 intra-mode but not to transmit S0 in a protected HRC. In such case, the frequency of intra-mode coding, which does not involve prediction, may depend on network conditions or may be determined in response to losses reported by a receiving endpoint. The S0 prediction chain may be refreshed in this manner (i.e. if there was an error at the S0 level, any drift is eliminated).

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