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(54) VOICE OVER IP ENDPOINT CALL ADMISSION

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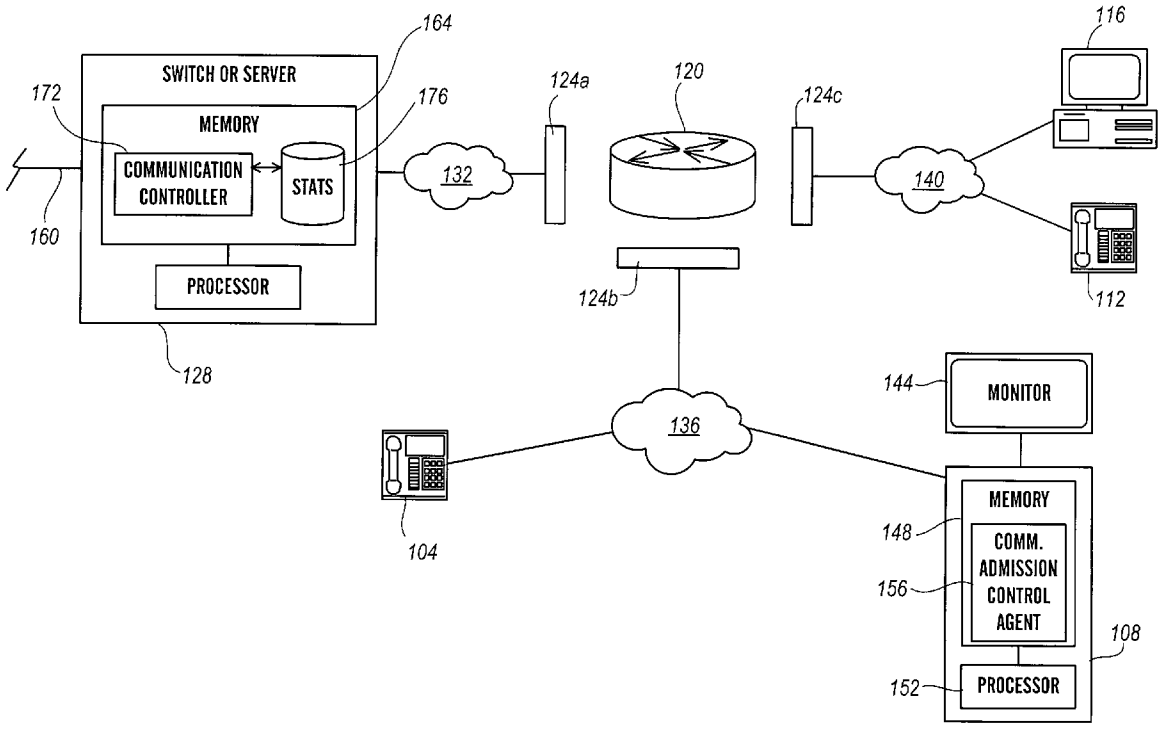
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(57) ABSTRACT

The present invention is directed generally to an intelligent endpoint or communication device that can collect available bandwidth-related information metrics and/or perform call admission control functions. The present invention is further directed to an architecture comprising a switch or media server in communication with a plurality of subscriber communication devices in which the subscriber communication devices act as network nodes to collect available bandwidth-related information.



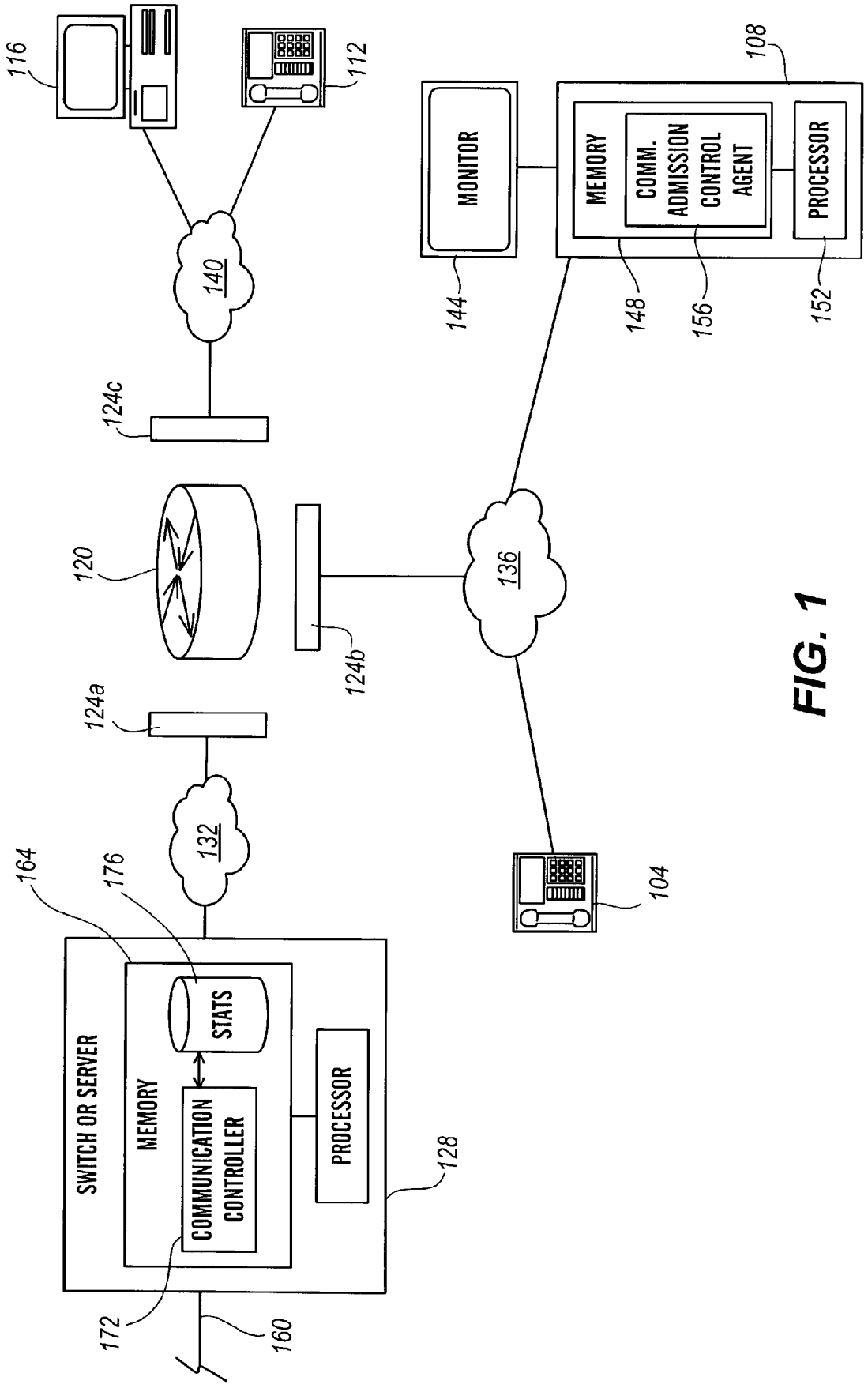

116
112
140
124c
120
124a
124b
136
132
164
176
SWITCH OR SERVER
MEMORY
STATS
COMMUNICATION CONTROLLER
PROCESSOR
128
172
160
104
144
MONITOR
108
MEMORY
COMM. ADMISSION CONTROL AGENT
PROCESSOR
148
156
152

FIG. 1

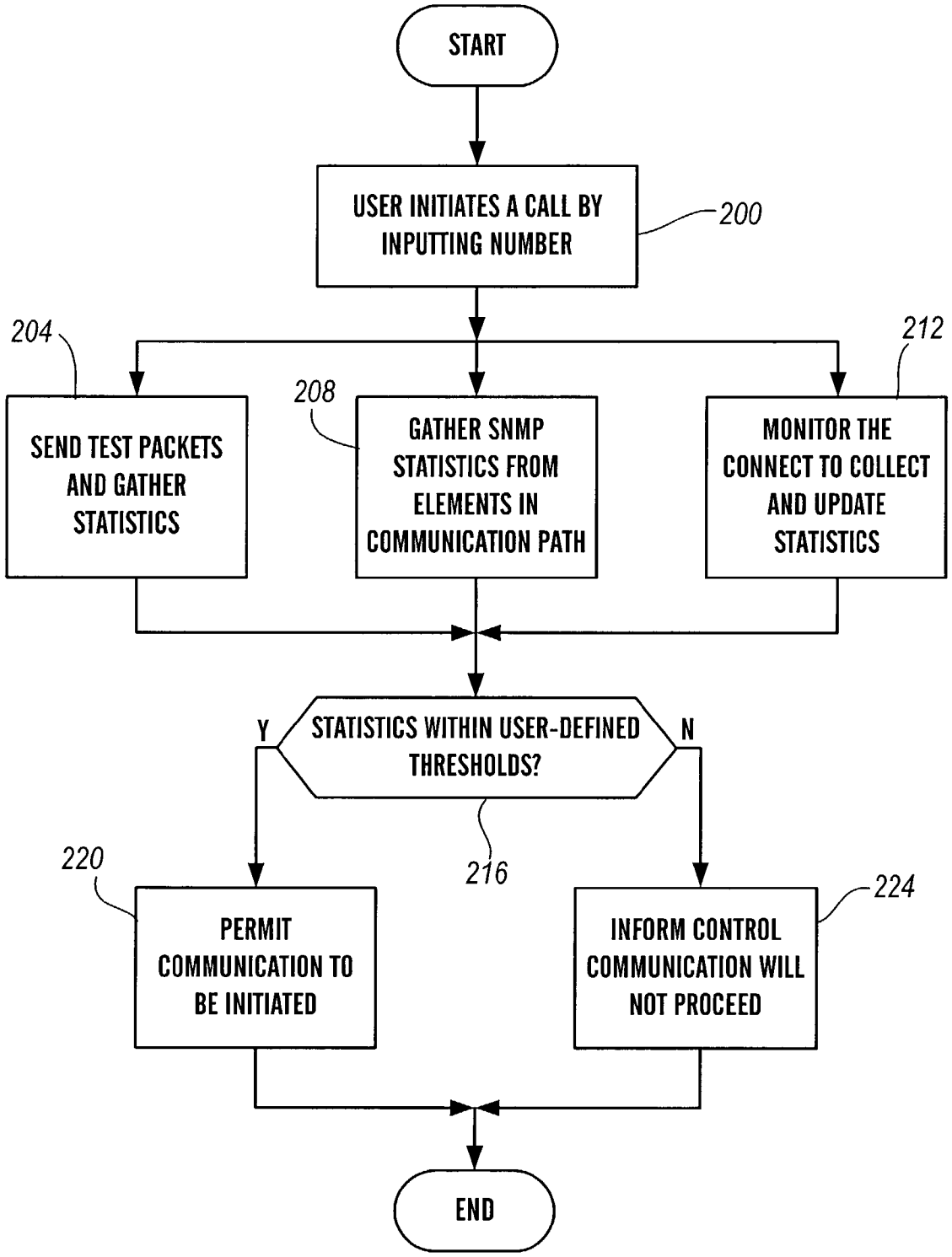

START
USER INITIATES A CALL BY INPUTTING NUMBER
200
204
208
212
SEND TEST PACKETS AND GATHER STATISTICS
GATHER SNMP STATISTICS FROM ELEMENTS IN COMMUNICATION PATH
MONITOR THE CONNECT TO COLLECT AND UPDATE STATISTICS
Y
STATISTICS WITHIN USER-DEFINED THRESHOLDS?
N
216
220
224
PERMIT COMMUNICATION TO BE INITIATED
INFORM CONTROL COMMUNICATION WILL NOT PROCEED
END

FIG. 2

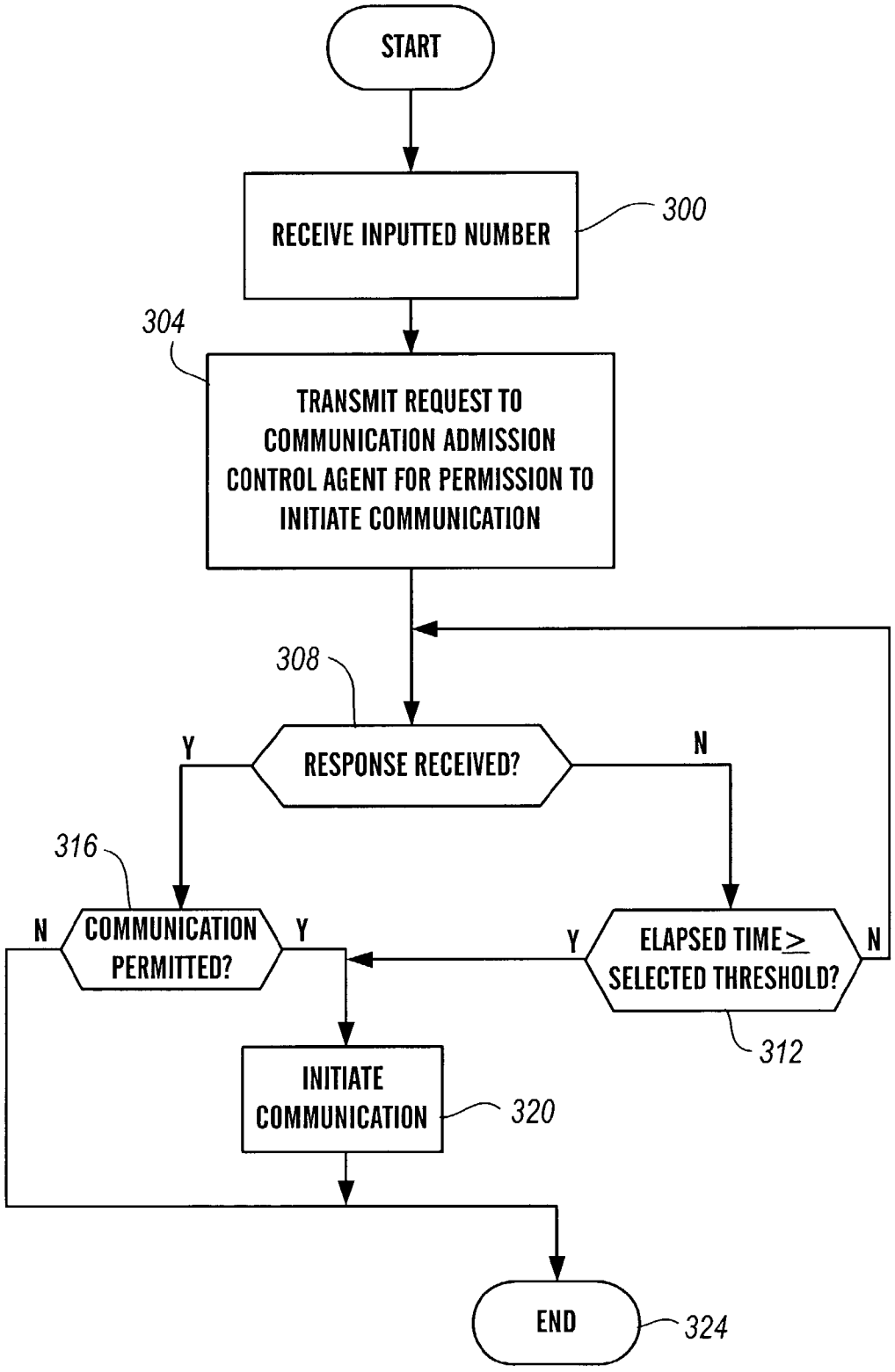

START
RECEIVE INPUTTED NUMBER
300
304
TRANSMIT REQUEST TO COMMUNICATION ADMISSION CONTROL AGENT FOR PERMISSION TO INITIATE COMMUNICATION
308
RESPONSE RECEIVED?
Y
N
316
COMMUNICATION PERMITTED?
N
Y
ELAPSED TIME≥ SELECTED THRESHOLD?
Y
N
312
INITIATE COMMUNICATION
320
END
324

FIG. 3

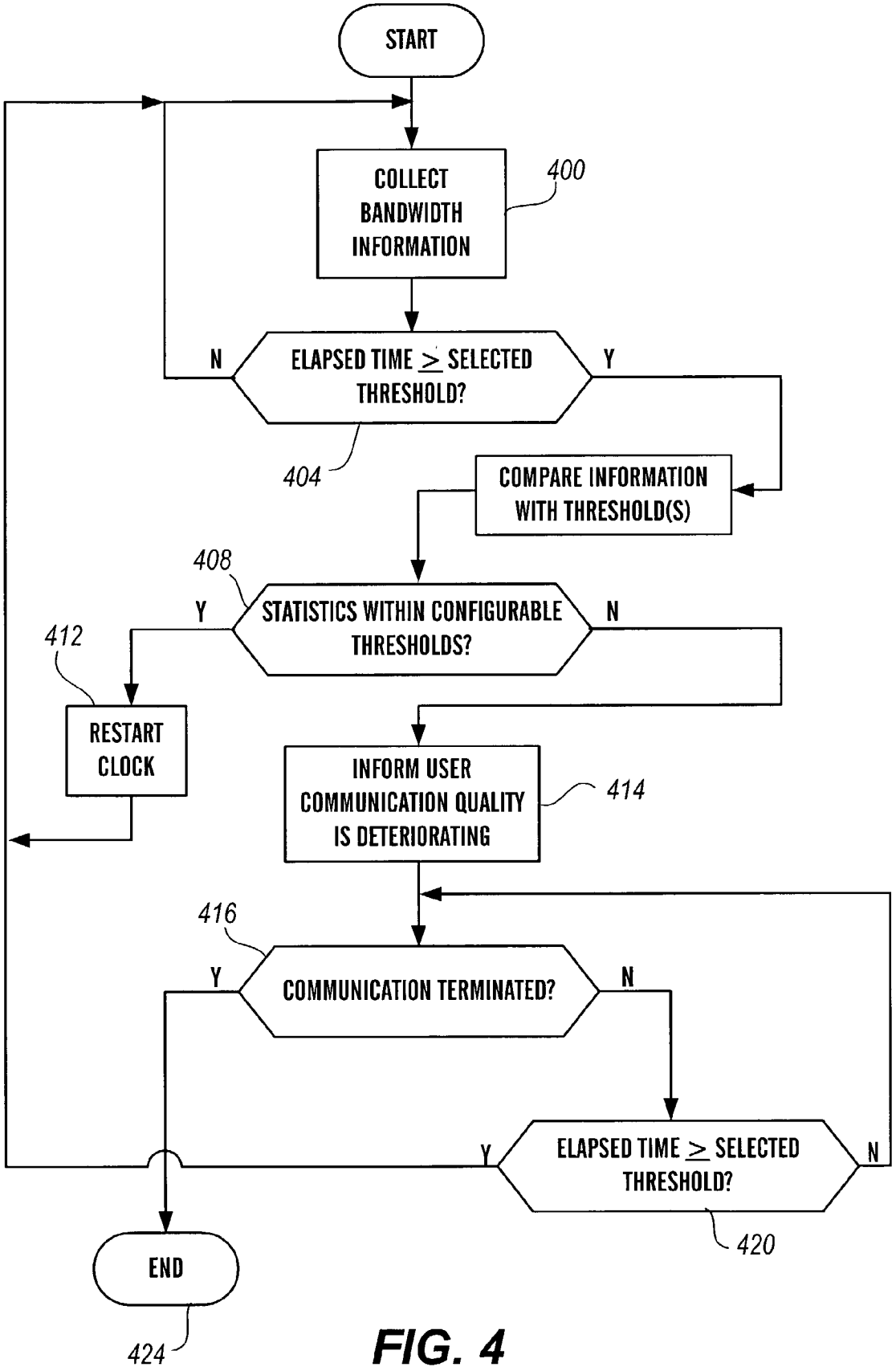

START
COLLECT BANDWIDTH INFORMATION
400
ELAPSED TIME ≥ SELECTED THRESHOLD?
N
Y
404
COMPARE INFORMATION WITH THRESHOLD(S)
408
STATISTICS WITHIN CONFIGURABLE THRESHOLDS?
Y
N
412
RESTART CLOCK
INFORM USER COMMUNICATION QUALITY IS DETERIORATING
414
416
COMMUNICATION TERMINATED?
Y
N
ELAPSED TIME ≥ SELECTED THRESHOLD?
Y
N
420
END
424

FIG. 4

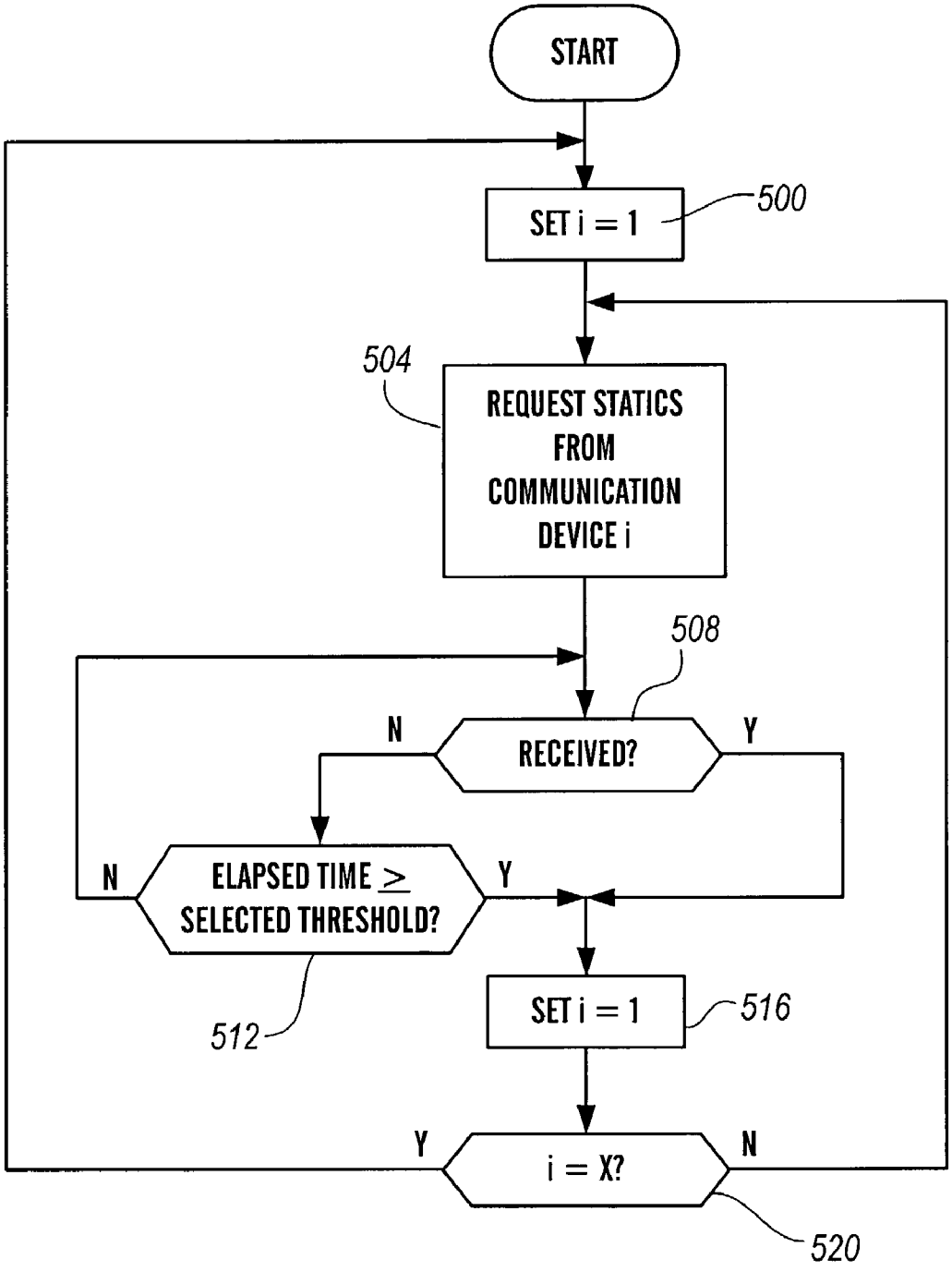

START
SET i = 1
500
504
REQUEST STATICS FROM COMMUNICATION DEVICE i
508
N
RECEIVED?
Y
N
ELAPSED TIME ≥ SELECTED THRESHOLD?
Y
512
SET i = 1
516
Y
i = X?
N
520

FIG. 5

VOICE OVER IP ENDPOINT CALL ADMISSION

CROSS REFERENCE TO RELATED APPLICATIONS

[0001] Cross-reference is made to copending U.S. application Ser. No. 10/028,874, filed Oct. 22, 2001, to Rankine et al., entitled "Real Time Control Protocol Session Matching"; Ser. No. 10/109,784, filed Mar. 29, 2002, to Chavez et al., entitled "Emergency Bandwidth Allocation with an RSVP-Like Protocol"; Ser. No. 10/165,719, filed Jun. 7, 2002, to Krumm-Heller et al., entitled "Apparatus and Method for Automatically and Dynamically Reconfiguring Network Provisioning", and Ser. No. 10/_____, filed concurrently herewith, to Minhazuddin et al., entitled "Instantaneous User Initiation Voice Quality Feedback", each of which is incorporated herein fully by reference.

FIELD OF THE INVENTION

[0002] The present invention relates generally to communications over networks and specifically to voice communications over data networks.

BACKGROUND OF THE INVENTION

[0003] Distributed processing networks are being increasingly used for live voice communications between network nodes using Voice over IP or VoIP technology. In VOIP technology, after the speech is digitized, the digitized speech is divided into packets. Each packet includes a header and a data payload of one to several frames of encoded speech. Distributed processing networks for delivering the packets to desired endpoints are typically designed to provide a Best Effort or BE single service model that does not discriminate in packet delivery between services and does not control service access or quality. Quality of Service or QoS architectures have been developed for BE environments to provide guaranteed transmission characteristics end-to-end such as available bandwidth, maximum end-to-end delay, maximum end-to-end delay variation (jitter), and packet/cell loss levels to provide continuous data streams suitable for real-time phone calls and video conferencing. Such QoS architectures include protocols such as the Resource ReSerVation Protocol or RSVP and the Real-Time Transfer Protocol or RTP.

[0004] RSVP is a signaling protocol that guarantees receivers a requested end-to-end QoS. RSVP serves as an Internet signaling protocol through the transmission of QoS parameters. Under RSVP, an end point negotiates with the network to allocate or reserve protected resources for traffic that the end point will generate or receive. The two messages that perform the reservation request and installation are the Path and Resv messages. Robustness is achieved through maintaining a soft state network by transmitting periodic refresh messages to maintain a reservation and path state along the reservation path. If the intermediate nodes do not receive the refresh message, the reservation will time out and be deleted.

[0005] RTP is a voice bearer channel transfer protocol. RTP neither guarantees a QoS nor provides for resource reservations. RTP runs on the transport layer of the Open Systems Interconnection or OSI model and defines a session by two components, namely its profile and payload format where the payload is the data being transmitted. The payload format specifies the format of the data within the RTP packet such as encoding and compression schemes. RTP functions include loss detection for quality estimation and rate adaptation, sequencing of data, intra- and intermedia synchronization, session identification using a session id, source identification using a synchronization source id or SSRC, and basic membership information.

[0006] The Real-Time Control Protocol or RTCP, a companion protocol to RTP, is used by applications to monitor the delivery of RTP streams. Media packets are transmitted between endpoints during a session according to RTP while additional performance information governing the communication link (e.g., key statistics about the media packets being sent and received by each end point such as jitter, packet loss, round-trip time, etc.) are collected by the end points and transmitted to a session monitor according to RTCP. The network monitor can be, for example, VoIP Monitoring Manager™ or VMon™ by Avaya, Inc.

[0007] Under either the RSVP or RTP protocols, VoIP introduces a whole new range of QoS problems which were not previously significant or, in some cases, even encountered in circuit-switched networks. Voice telephony depends upon reliable, low latency, real-time delivery of audio data. In VoIP, values for latency, packet loss, and jitter can increase substantially, particularly during periods of heavy network traffic, causing a user to experience a much poorer quality of communication (e.g., audio or video distortion, unacceptable levels of asynchronization between audio and video streams, etc.) than would be experienced if the call were made by a traditional circuit-switched telephony network. This is particularly true when the network allows any and all calls are allowed to occur, regardless of available bandwidth and the concomitant low quality of the call to be placed and the detrimental impact on the quality of other calls.

[0008] To provide a higher QoS, call admission control (CAC) functionality has been employed to control bandwidth usage. In one approach, CAC-type functionality is built into the communication protocol. For example, in RSVP calls are disallowed when a reservation fails. In another approach, CAC functionality is built into the switch or media server (e.g., the Private Branch Exchange or PBX). Under the H.323 standard for example, an endpoint in the H.323 zone of a call admission controller or gatekeeper must receive permission (using a bandwidth request message) from the call admission controller before making a call. Based on a restriction on the number of concurrent IP calls that can be placed generally and/or on a critical measure, such as not being able to reserve the bandwidth required, the call admission controller responds with a bandwidth confirm message permitting the call to be placed or a bandwidth reject message refusing to make the necessary connection for the call. The remaining bandwidth is reserved for electronic mail, file transfers, and other local-area network or LAN protocols.

[0009] Although a higher QoS can be realized using CAC functionality, there are drawbacks. First, there is no reason provided to the user regarding the failure to place the call. This can lead to user frustration and ultimately to a lack of utilization of IP telephony. Although some telephones, such as the Cisco IP Phone 7960™, do permit a user to press a button and view the current values for latency, packet loss,

and jitter, this feature is only enabled after a call is placed. Second, it can be very difficult for a call admission controller to determine accurately whether or not to place a call. For example, if the controller is in a first subnet, a first endpoint is in a second subnet, and a second endpoint is in a third subnet, the controller is often unable to determine whether or not there is sufficient bandwidth available between the second and third subnets for the first and second endpoints to conduct a communication of acceptable quality. Although the controller can base the decision on pings to the second and third subnets, this is generally not an accurate indicator of available bandwidth. Third, CAC techniques provide the user with no option about placing the call if predefined criteria are not satisfied. There are situations when a user may want to place a call even though the call may be of poor quality. For example, a user may want to place a call in an emergency situation.

SUMMARY OF THE INVENTION

[0010] These and other needs are addressed by the various embodiments and configurations of the present invention. The present invention is directed generally to an intelligent endpoint or communication device that can collect bandwidth related metrics and/or perform call admission control functions.

[0011] In one embodiment, a method for controlling contact admission is provided. The method includes the steps of:

[0012] (a) receiving from a first user at a first endpoint a signal (e.g., a telephone number, an address on a data network, etc.) associated with initiation of a voice communication with a second endpoint;

[0013] (b) collecting bandwidth-related metrics or bandwidth information associated with an expected quality of the voice communication by performing one or more of the following:

[0014] (i) sending one or more test packets to the second endpoint to obtain a first subset of bandwidth information;

[0015] (ii) requesting, from one or more intermediate nodes or network entities in the communication path between the first and second endpoints, a second subset of bandwidth information; and

[0016] (iii) requesting, from a switch or media server in communication with the first and second endpoints, a third subset of bandwidth information (the third subset of bandwidth information is typically from communications between endpoints other than the first and second endpoints);

[0017] (c) comparing some or all of the collected bandwidth information with one or more configurable thresholds;

[0018] (d) when the collected bandwidth information fails to satisfy the thresholds, notifying the first user of a likelihood of a low quality of the communication and/or not permitting initiation of the communication between the first and at least a second endpoints; and

[0019] (e) when the collected bandwidth information satisfies the threshold(s), permitting initiation of the communication between the first and second endpoints.

[0020] The first and second endpoints can be any suitable communication device, such as an IP hard phone, an IP softphone, a telephone other than an IP hard phone and softphone, and a personal digital assistant.

[0021] The bandwidth information can include one or more of: received RTP packets, received RTP octets, round trip time, jitter buffer delay, jitter, packet loss burst size, a number of out-of-order packets, an out-of-order distance, Reservation Protocol status, call state, sender channel state, IP Differential Service Code Point, available bandwidth, router buffer size, a number of dropped packets by a router, router bandwidth utilization, and router processor utilization.

[0022] This embodiment decentralizes the call admission control functionality from the switch/media server to the endpoints serviced by the switch/media server. The endpoint and not the switch/media server is best able to evaluate available bandwidth for the requested communication. For example, this embodiment is particularly beneficial for endpoints in virtual private networks where there is a higher likelihood of call failure due to network constraints which are frequently unknown to the switch/server (e.g., the use of a low speed modem link). This embodiment is also particularly beneficial in evaluating available bandwidth between endpoints in different subnets from the switch/media server. The embodiment can reduce significantly network congestion relative to conventional systems by reducing the numbers of calls placed, particularly during periods of high usage.

[0023] Other forms of call admission control assume some kind of administrative load on the routers and the switch/server. This embodiment is extremely lightweight in terms of cost of implementation, e.g., it only affects the endpoints themselves rather than all network entities in a selected path. It also requires minimal administration other than the user enabling the functionality on the endpoint.

[0024] The embodiment provides more options to the user to help improve quality for VoIP calls. In a non-mission-critical application, a system that disallows and/or discourages low quality calls may be able to stop overload of a system when the user would exit the call of a certain quality anyway. The thresholds can be configured by network administration and/or by the user himself or herself. User configurable thresholds are particularly attractive as the personal preferences of each user, which vary widely from user to user, can be taken into account by each endpoint. The embodiment can be configured to permit the user to place a call even when the quality will be poor and/or provide detailed reasons for not placing the call, thereby reducing user frustration relative to conventional CAC systems.

[0025] In another embodiment, an endpoint or communication device collects bandwidth information during a voice communication and, when the collected bandwidth information fails to satisfy one or more voice quality threshold(s), informs a user that the voice quality is below a selected level.

[0026] In current phones, such as the Cisco I-Button™ the user is able to view performance metrics, such as latency and packet loss, by pressing a button on the phone. The user, however, must continually press the button to refresh the metrics and determine when call quality is deteriorating,

which can be very detracting during the conversation. In contrast, the communication device of this embodiment itself determines automatically when call quality is deteriorating and warns the user accordingly. This permits the user to wrap up the conversation before voice quality deteriorates to a level that is unacceptable.

[0027] In yet another embodiment, method for collecting bandwidth information is provided. In the method, a switch or a media server in communication with a plurality of subscriber communication devices uses the subscriber communication devices as network nodes to collect bandwidth information. In one configuration, the switch or media server progressively collects bandwidth information from each subscriber communication device in the network, e.g., LAN or enterprise network, served by the switch or media server. Any subscriber communication device involved in a communication when requested for bandwidth information is typically skipped.

[0028] This embodiment effectively treats the subscriber communication devices as network probes to provide a real-time, complete, and detailed picture of the bandwidth utilization levels across selected parts of the network. The use of existing devices to perform bandwidth information collection is much less expensive and far simpler than installing a myriad of dedicated network probes at various points in the network.

[0029] These and other advantages will be apparent from the disclosure of the invention(s) contained herein.

[0030] The above-described embodiments and configurations are neither complete nor exhaustive. As will be appreciated, other embodiments of the invention are possible utilizing, alone or in combination, one or more of the features set forth above or described in detail below.

BRIEF DESCRIPTION OF THE DRAWINGS

[0031] **FIG. 1** is a block diagram showing an implementation of a first embodiment of the present invention;

[0032] **FIG. 2** is a flowchart showing an embodiment of the operation of the communication admission control agent of **FIG. 1**;

[0033] **FIG. 3** is a flowchart showing an embodiment of the operation of the communication controller of **FIG. 1**;

[0034] **FIG. 4** is a flowchart showing yet another embodiment of the operation of the communication admission control agent; and

[0035] **FIG. 5** is a flowchart showing yet another embodiment of the operation of the communication controller of **FIG. 1**.

DETAILED DESCRIPTION

[0036] **FIG. 1** depicts a VoIP architecture **100** according to an embodiment of the present invention. The architecture includes a first, second, third, and fourth end points **104**, **108**, **112**, and **116**, a router **120** having associated interfaces **124***a-c*, and switch or server **128**. The various components are interconnected by first, second, and third subnets **132**, **136**, and **140**, respectively, which collectively form a local area network. For illustration purposes, the first and third endpoints **104** and **112** are configured as IP hard-phones, while the second and third endpoints **108** and **116** are configured as IP soft-phones. As will be appreciated, the various end points can be any communication device, such as a plain old telephone, a soft phone, an IP hard phone, a personal or laptop computer, a PDA, a pager, and a wireless phone. It is further to be understood that the architecture can include any number of subnetworks and routers, depending on the application, and that the subnets can form a network topology other than a LAN, such as a wide-area network, an enterprise network, and a metropolitan-area network.

[0037] As depicted, the second end point **108** is a personal computer including a computer display monitor **144** and a computer comprising memory **148** and a processor **152**. The memory **148** includes a communication admission control agent **156** to collect session-related information, such as latency, packet loss, jitter, available bandwidth, and jitter buffer delay to be used in determining whether or not a communication or call should be placed. As will be appreciated, the other end points preferably include a communication admission control agent as well. Although the various endpoints are shown as subscribers to the switch or server, it is to be understood that one or more of the endpoints can be nonsubscribers.

[0038] The switch or media server preferably performs automatic communication routing functions to the various endpoints. The switch or media server **128** is connected to one or more communication lines **160** (e.g., telephone lines or trunks) over which it receives incoming contacts on the public switched telephone network or IP network. As will be appreciated, a "contact" can be any form, mode, or type of single media or multimedia communication, such as a wired or wireless live voice communication (whether circuit-switched or packet-switched), electronic mail, and video conferencing.

[0039] The switch or media server **128** can be any architecture for routing contacts to one or more endpoints. Illustratively, the switch or server of **FIG. 1** can be the subscriber-premises equipment disclosed in U.S. Pat. Nos. 6,192,122; 6,173,053; 6,163,607; 5,982,873; 5,905,793; 5,828,747; and 5,206,903, all of which are incorporated herein by this reference; Avaya Inc.'s Definity™ private-branch exchange (PBX)-based ACD system; or Nortel Networks' IPConnect™. The switch or media server typically is a stored-program-controlled system that conventionally includes interfaces to external communication links, a communications switching fabric, service circuits (e.g., tone generators, announcement circuits, etc.), memory **164** for storing control programs and data, and a processor **168** (i.e., a computer) for executing the stored control programs to control the interfaces and the fabric and to provide automatic contact-distribution functionality. Included in the memory **164** is a communication or call controller **172** to supervise the placement of contacts among subscribers, such as the first, second, third, and fourth endpoints, and between a subscriber and a nonsubscriber and a database **176** of performance metrics or bandwidth information collected during communications by subscribers.

[0040] The bandwidth information in the memory **164** can vary depending on the application. For each current or historical communication, the database **164** can include one or more of the following: received RTP packets (an integer that is cumulative for the session and is reset to zero at the

start of each new RTP session), received RTP octets (an integer that is cumulative for the session and is reset to zero at the start of each new RTP session), round trip time (an integer expressed in units of milliseconds that is reset to zero at the start of each new session), jitter buffer delay (an integer representing the delay imparted on the bearer channel by the jitter buffer at an endpoint and is expressed in milliseconds), jitter (an integer indicating a level of distortion of the interpacket arrival times compared to the interpacket times of the original transmission), packet loss burst size (an integer indicating the maximum number of consecutive packets lost in the last RTCP reporting interval), the number of out-of-order packets (an integer representing the number of packets received out-of-order in the last RTCP reporting interval), the out-of-order distance (an integer representing the number of packets after an out-of-order packet was received from when it was expected to be received), RSVP status (the RSVP status field reports the state of RSVP protection of the receiver end point's RTP session only (e.g., states can include receiver RSVP not in use, receiver RSVP disabled, receiver RSVP installation pending, receiver RSVP reservation failed, and receiver RSVP reservation installed), call state, sender channel state, DCSP (an integer that is the value of the IP Differentiated Service Code Point or DSCP field of the incoming RTP packets), available bandwidth, router buffer size (one or more integers equal to the number of packets enqueued or stacked for processing at the queried interface of the router or the stacking maximum capacity of the buffer at the queried router interface), dropped packets (the dropped packet field(s) comprise at least the following information, whether or not the queried interface of the router is dropping packets and, if so, how many and why (e.g., excessive delay, duplication and/or fragmentation)), router bandwidth utilization (an integer equal to the percent bandwidth utilization at a queried router interface at a selected point in time), and router processor utilization (an integer equal to the percent utilization of the router processor at a selected point in time).

[0041] **FIGS. 2 and 3** depict an embodiment of the operations of the communication admission control agent **156** in the endpoint seeking to place a call and the communication controller **172**, respectively. As can be seen from these figures, the call admission control functionality is decentralized to the endpoints compared to the conventional approach of centralizing call admission control functions in the call admission controller of the switch or media server.

[0042] Referring to **FIG. 2**, the agent **156** in step **200** detects that a user is attempting to initiate a contact. For example, this can be detected by detecting an off-hook state of the telephone, receiving an inputted series of signals such as would be associated with a telephone number or IP address.

[0043] The agent **156** in response to the detection can perform one or more of the statistic collection steps **204**, **208**, and **212**, depending on the implementation.

[0044] In step **204**, the agent **156** in the contact-initiating endpoint sends test packets to the switch or server and/or to one or more of the destination endpoints to assess available bandwidth. For example, the initiating endpoint could send a test Resv message to the destination endpoint to attempt to set up a dummy or test reservation. If the reservation cannot be set up, the agent **156** would conclude that insufficient bandwidth is available for the contact to be placed. The contents of the Resv message would be based on the parameters required by the anticipated contact. Alternatively, test RTP/RTCP packets can be sent between the two endpoints to measure one or more of the bandwidth information noted above, such as jitter, packet delay, and packet loss. The packets would have a dummy payload and the packet headers would include information such as time stamps. The format of the test packets is set forth in RFC 1889. In either of the two previous examples, a marker bit or flag would be included in the exchanged packets to notify the receiving endpoint that the packet is associated with an available bandwidth test. The details to implement either of these examples will be readily appreciated by one of ordinary skill in the art who is associated with the RSVP and/or RTP/RTCP protocols. The two examples can be performed simultaneously using the same set of test packets. Other protocols may also be used for statistic collection. As will be appreciated, a proprietary protocol can also be used to perform statistic collection by transmitting packets between two or more endpoints. These techniques can measure currently difficult-to-measure parameters such as echo.

[0045] In step **208**, the agent **156** in the contact-initiating endpoint collects bandwidth information from intermediate functional elements, entities or components (e.g., routers and one or more of their associated interfaces) in the communication path. This statistic collection can be done using any suitable protocol. For example, the collection can be done using pinging as defined in the Internet Control Message Protocol or ICMP. In another approach, the bandwidth information can be collected using the Simple Network Management Protocol or SNMP. To substantially minimize bandwidth utilization, the statistic collection is generally stopped when enough information is collected to assess available bandwidth for the anticipated contact. While some bandwidth will be utilized by numerous endpoints simultaneously or near simultaneously collecting bandwidth information, the utilized bandwidth is considered to be insignificant when compared to the amount of bandwidth consumed by unrestricted or uncontrolled call placement.

[0046] In step **212**, the agent **156** in the contact-initiating endpoint queries polls the switch or server **128** (which is the nearest gateway for the agent **156**) or one or more other switches or servers acting as a monitor or otherwise located in a different network for relevant bandwidth information stored in their respective database **176**. As will be appreciated, some or all of the bandwidth information gathered in steps **204** and **208** by the agents **156** in the various endpoints are forwarded to the switch or server for updating the database **176**. Depending on the database rules or policies, this bandwidth information is retained in the database for a selected period of time. In one configuration, agents continue to collect bandwidth information after a contact is effected by the corresponding endpoint(s) until the contact is terminated and forwards the collected bandwidth information to the switch or server for database updating. In this manner, the endpoints act as network nodes, and any contact initiating endpoint may be able to obtain, nonintrusively, current information about bandwidth utilization on all or part of the communication path to the destination endpoint. In one configuration, steps **204** and **208** are not performed to avoid unnecessary use of bandwidth if relevant bandwidth

information, as defined by the user or network administrators, is already contained in the database **176**.

[**0047**] After the bandwidth information is collected by performing one or more of steps **204**, **208**, and **212**, the agent **156** in the contact-initiating endpoint determines in step **216** whether the collected bandwidth information is within user-defined thresholds used to define a contact of acceptable quality. For example, the user can configure a packet loss to 5%, the packet round trip time to 150 milliseconds, and the jitter to 80 milliseconds. When the packet loss, round trip time, and/or jitter buffer delay exceeds the corresponding threshold, the bandwidth information is not within the user defined thresholds. The user can modify these and other thresholds to suit his or her personal taste. As will be appreciated, some users are more tolerable of lower call quality than others.

[**0048**] When the collected bandwidth information is within user-defined thresholds, the agent **156** in step **220** forwards a message to the controller **172** indicating that the contact can be connected. When the collected bandwidth information is not within user-defined thresholds, the agent **156** in step **224** notifies the user that the contact will not be connected and the reason(s) for the nonconnection. The user can then reconfigure the thresholds and attempt to make the contact or wait until a later time to again attempt the contact. In one configuration, the user can override the agent and proceed with the contact such as by inputting an authorization code. This is particularly important in an emergency situation.

[**0049**] Alternatively, the agent **156** can warn the user of the probable low quality for the contact if connected and permit the user to decide whether or not he or she wishes to continue with making the contact.

[**0050**] Referring now to **FIG. 3**, the controller **172** in step **300** receives an inputted series of signals such as would be associated with a telephone number or IP address. In step **304**, the controller sends a request to the agent **156** of the contact-initiating endpoint for permission to initiate the communication. In steps **308** and **312**, the controller **172** waits for a response. If no response is received within a selected time, the controller **172** proceeds to step **320** discussed below. If a response is received timely, the controller **172** in step **316** determines whether or not the endpoint granted permission for the contact to be effected. If not, the controller **172** proceeds to step **324** and terminates operation with respect to the current transaction. If so, the controller **172** in step **320** initiates the contact and then proceeds to step **324**. In one configuration when the agent **156** grants permission for the contact, the controller **172** effects the contact only when a second tier of criterion selected by network administration are satisfied. Such criterion can be one or more of the bandwidth information noted above and/or one of the prior art criterion set forth previously.

[**0051**] A number of variations and modifications of the invention can be used. It would be possible to provide for some features of the invention without providing others.

[**0052**] For example in one alternative embodiment shown in **FIG. 4**, the agent **156** in an endpoint continues to collect bandwidth information during a contact or session and provides a warning to the user when the quality of the communication deteriorates below user-defined thresholds. This embodiment can be used with the network architecture of the present invention or a conventional network architecture, such as any of the architectures described previously. Referring to **FIG. 4**, after a contact is initiated the agent **156** in step **400** collects bandwidth information using any of the techniques noted previously. The agent **156** may forward these bandwidth information periodically to the switch or server for database updating. In step **404**, the agent **156** determines if the elapsed time since clock reset has greater than or equal to a selected threshold. If not, the agent **156** repeats steps **400** and **404**. If so, the agent **156**, in step **408**, compares at least some of the collected bandwidth information with the user configured thresholds. If the collected bandwidth information is within the thresholds, the clock is reset to zero in step **412** and the agent repeats steps **400** and **404**. If the collected bandwidth information is not within user thresholds, the agent **156** informs the user in step **414** that the quality of the communication is deteriorating. In step **416**, the agent **156** determines if the user has terminated the communication. If the communication was not terminated by the user in response to the warning, the agent **156** in step **420** determines if the elapsed time since the performance of step **412** equals or exceeds a selected threshold. If so, the agent **156** returns to and repeats steps **400** and **404**. If not, the agent **156** repeats step **416**. If the communication was terminated by the user in response to the warning, the agent **156** terminates operation in step **424**.

[**0053**] In another alternative embodiment, the switch or server **128** uses one or more of the endpoints (whether or not involved in a session) as network nodes to collect periodically any of the bandwidth information noted above. This embodiment can be used with the network architecture of the present invention or a conventional network architecture, such as any of the architectures described previously. Referring to **FIG. 5, a** counter i is set to 1 in step **500**. In step **504**, bandwidth information is requested from communication device or endpoints. A plurality of the subscriber communication devices or endpoints in architecture **100** are assigned identifiers to permit the controller **172** to sequentially collect bandwidth information from each of the identified endpoints. In response to the request, the endpoint collects the bandwidth information using any of the techniques noted above. If the endpoint is involved in a communication, the controller **172** typically skips that endpoint and proceeds to the next endpoint as the bandwidth information from that communication is already being directed to the controller **172**. In steps **508** and **512**, the controller **172** determines if the requested bandwidth information has been received within a selected time threshold. If not, the controller **172** repeats steps **500** and **504**. If so, the controller updates the database **176** and in step **516** increments the counter i by one. In step **520**, the controller determines if i is equal to X, the highest identifier in the LAN. If so, the controller returns to step **500** and starts the above process over. If not, the controller returns to step **504**.

[**0054**] In a further embodiment, agent **156** and/or controller **172** is/are configured as logic circuit(s) such as an Application Specific Integrated Circuit.

[**0055**] In yet another alternative embodiment, the division of responsibilities between the agent **156** and controller **172** is different from that discussed above.

[0056] The present invention, in various embodiments, includes components, methods, processes, systems and/or apparatus substantially as depicted and described herein, including various embodiments, subcombinations, and subsets thereof. Those of skill in the art will understand how to make and use the present invention after understanding the present disclosure. The present invention, in various embodiments, includes providing devices and processes in the absence of items not depicted and/or described herein or in various embodiments hereof, including in the absence of such items as may have been used in previous devices or processes, e.g., for improving performance, achieving ease and/or reducing cost of implementation.

[0057] The foregoing discussion of the invention has been presented for purposes of illustration and description. The foregoing is not intended to limit the invention to the form or forms disclosed herein. Although the description of the invention has included description of one or more embodiments and certain variations and modifications, other variations and modifications are within the scope of the invention, e.g., as may be within the skill and knowledge of those in the art, after understanding the present disclosure. It is intended to obtain rights which include alternative embodiments to the extent permitted, including alternate, interchangeable and/or equivalent structures, functions, ranges or steps to those claimed, whether or not such alternate, interchangeable and/or equivalent structures, functions, ranges or steps are disclosed herein, and without intending to publicly dedicate any patentable subject matter.

What is claimed is:

1. A method for controlling contact admission, comprising:

(a) receiving from a first user at a first endpoint at least one signal associated with initiation of a voice communication with at least a second endpoint;

(b) collecting bandwidth information associated with an expected quality of the voice communication by performing at least one of the following substeps:

(i) sending at least one test packet to the at least a second endpoint to obtain a first subset of bandwidth information;

(ii) requesting, from at least one intermediate entity in at least one communication path between the first and second endpoints, a second subset of bandwidth information; and

(iii) requesting, from a switch or media server in communication with the first endpoint and at least a second endpoint, a third subset of bandwidth information, the third subset of bandwidth information being associated with communications between endpoints other than the first and at least a second endpoints;

(c) comparing at least some of the collected bandwidth information with at least one threshold; and

(d) when the at least some of the collected bandwidth information fails to satisify the at least one threshold, at least one of (i) notifying the first user of a likelihood of a low quality of the communication and (ii) not permitting initiation of the communication between the first and at least a second endpoints.

2. The method of claim 1, wherein the at least one signal is a telephone number or an address on a data network and further comprising:

when the at least some of the collected bandwidth information satisfies the at least one threshold, permitting initiation of the communication between the first and at least a second endpoints.

3. The method of claim 1, wherein the first and second endpoints are one of an IP hard phone, an IP softphone, a telephone other than an IP hard phone and softphone, and a personal digital assistant.

4. The method of claim 1, wherein the bandwidth information comprises one or more of the following: received RTP packets, received RTP octets, round trip time, jitter buffer delay, jitter, packet loss burst size, a number of out-of-order packets, an out-of-order distance, Reservation Protocol status, call state, sender channel state, IP Differentiated Service Code Point, available bandwidth, router buffer size, a number of dropped packets by a router, router bandwidth utilization, and router processor utilization.

5. The method of claim 1, wherein substep (i) of the collecting step is performed.

6. The method of claim 5, wherein the first subset of bandwidth information comprises at least one of the following: received RTP packets, received RTP octets, round trip time, jitter buffer delay, jitter, packet loss burst size, a number of out-of-order packets, an out-of-order distance, and Reservation Protocol status.

7. The method of claim 1, wherein substep (ii) of the collecting step is performed and wherein the at least one intermediate entity is a router and/or an associated interface of the router.

8. The method of claim 7, wherein the second subset of bandwidth information comprises at least one of the following: router buffer size, a number of dropped packets by a router, router bandwidth utilization, router processor utilization.

9. The method of claim 1, wherein substep (iii) of the collecting step is performed.

10. The method of claim 9, wherein the third subset of bandwidth information comprises at least one of the following: received RTP packets, received RTP octets, round trip time, jitter buffer delay, jitter, packet loss burst size, a number of out-of-order packets, an out-of-order distance, and Reservation Protocol status.

11. The method of claim 2, wherein the switch or media server performs the following steps:

receiving the at least one of a telephone number and address on a data network; and

receiving from the first endpoint a communication initiation message, wherein:

(i) when the communication initiation message indicates insufficient bandwidth is available for the communication, not connecting the first and at least a second endpoints; and

(ii) when the communication initiation message indicates sufficient bandwidth is available for the communication, connecting the first and at least a second endpoints.

12. A communication device comprising, comprising:

an input operable to receive, from a first user, at least one signal associated with initiation of a voice communication with at least a second communication device;

a communication admission control agent operable to (a) collect bandwidth information associated with an expected quality of the voice communication by performing at least one of (i) sending at least one test packet to the at least a second communication device to obtain a first subset of bandwidth information; (ii) requesting, from at least one intermediate entity in at least one communication path between the first and second communication devices, a second subset of bandwidth information; and (iii) requesting, from a switch or media server in communication with the first and at least a second communication devices, a third subset of bandwidth information, the third subset of bandwidth information being associated with communications between communication devices other than the first and at least a second communication devices; (b) compare at least some of the collected bandwidth information with at least one threshold; and (c) when the at least some of the collected bandwidth information fails to satisfy the at least one threshold, at least one of (i) notifying the first user of a likelihood of a low quality of the communication and (ii) not permitting initiation of the communication between the first and at least a second communication devices.

13. The device of claim 12, wherein the at least one signal is a telephone number or an address on a data network and wherein, when the at least some of the collected bandwidth information satisfies the at least one threshold, the agent is operable to initiate the communication between the first and at least a second communication devices.

14. The device of claim 12, wherein the first and second communication devices are one of an IP hard phone, an IP softphone, a telephone other than an IP hard phone and softphone, and a personal digital assistant.

15. The device of claim 12, wherein the bandwidth information comprises one or more of the following: received RTP packets, received RTP octets, round trip time, jitter buffer delay, jitter, packet loss burst size, a number of out-of-order packets, an out-of-order distance, Reservation Protocol status, call state, sender channel state, IP Differentiated Service Code Point, available bandwidth, router buffer size, a number of dropped packets by a router, router bandwidth utilization, and router processor utilization.

16. The device of claim 12, wherein the agent performs (i).

17. The device of claim 16, wherein the first subset of bandwidth information comprises at least one of the following: received RTP packets, received RTP octets, round trip time, jitter buffer delay, jitter, packet loss burst size, a number of out-of-order packets, an out-of-order distance, and Reservation Protocol status.

18. The device of claim 12, wherein the agent performs (ii) and the at least one intermediate entity comprises a router and/or an associated interface of the router.

19. The device of claim 18, wherein the second subset of bandwidth information comprises at least one of the following: router buffer size, a number of dropped packets by a router, router bandwidth utilization, router processor utilization.

20. The device of claim 12, wherein the agent performs (iii).

21. The device of claim 20, wherein the third subset of bandwidth information comprises at least one of the following: received RTP packets, received RTP octets, round trip time, jitter buffer delay, jitter, packet loss burst size, a number of out-of-order packets, an out-of-order distance, and Reservation Protocol status.

22. The device of claim 13, wherein the switch or media server is operable to receive the at least one of a telephone number and address on a data network; and receive from the first endpoint a communication initiation message, and wherein:

(i) when the communication initiation message indicates insufficient bandwidth is available for the communication, the switch or media server does not connect the first and at least a second communication devices; and (ii) when the communication initiation message indicates sufficient bandwidth is available for the communication, the switch or media server connects the first and at least a second communication devices.

23. The device of claim 12, wherein the first communication device is a subscriber of the switch or media server.

24. A method for controlling contact admission, comprising:

(a) during a voice communication first and second communication devices, collecting bandwidth information associated with a quality of the voice communication;

(c) comparing at least some of the collected bandwidth information with at least one voice quality threshold; and

(d) when the at least some of the collected bandwidth information fails to satisfy the at least one voice quality threshold, informing at least one of a first and second user associated, respectively, of the first and second communication devices that the voice quality is below a selected level.

25. The method of claim 24, wherein the voice communication is a live voice conversation between the first and second users.

26. The method of claim 24, wherein the first and second communication devices are one of an IP hard phone, an IP softphone, a telephone other than an IP hard phone and softphone, and a personal digital assistant.

27. The method of claim 24, wherein the collected bandwidth information comprises one or more of the following: received RTP packets, received RTP octets, round trip time, jitter buffer delay, jitter, packet loss burst size, a number of out-of-order packets, an out-of-order distance, Reservation Protocol status, call state, sender channel state, IP Differentiated Service Code Point, available bandwidth, router buffer size, a number of dropped packets by a router, router bandwidth utilization, and router processor utilization.

28. A method for collecting bandwidth information, comprising:

(a) providing a switch or a media server in communication with a plurality of subscriber communication devices;

(b) selecting a first subscriber communication device;

(c) requesting bandwidth information from the first subscriber communication device;

(d) performing with the first subscriber communication device at least one of the following substeps:

(i) sending at least one test packet to another subscriber communication device to obtain a first subset of bandwidth information; and

(ii) requesting, from at least one intermediate node in at least one communication path between the first communication device and another communication device, a second subset of bandwidth information; and

(e) selecting a second subscriber communication device; and

(f) repeating steps (c) and (d) with respect to the second subscriber communication device.

29. The method of claim 28, further comprising:

(g) selecting a third subscriber communication device; and

(h) requesting bandwidth information from the third subscriber communication device, wherein the third subscriber communication device does not perform step (d) with respect to the third communication device, when the third subscriber communication device is in use by a user.

30. The method of claim 28, wherein the bandwidth information comprises a plurality of the following: received RTP packets, received RTP octets, round trip time, jitter buffer delay, jitter, packet loss burst size, a number of out-of-order packets, an out-of-order distance, Reservation Protocol status, IP Differentiated Service Code Point, available bandwidth, router buffer size, a number of dropped packets by a router, router bandwidth utilization, and router processor utilization.

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