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(71) Applicant *(for all designated States except US)*: **TELECOMMUNICATION SYSTEMS, INC.** [US/US]; 275 West Street, Suite 400, Annapolis, MD 21401 (US).

(72) Inventor: **POREMBA, Todd**; 7053 24th Avenue NE, Seattle, WA 98115 (US).

(74) Agent: **BOLLMAN, William, H.**; Manelli Denison & Selter PLLC, 2000 M Street, NW, 7th Floor, Washington, DC 20036 (US).

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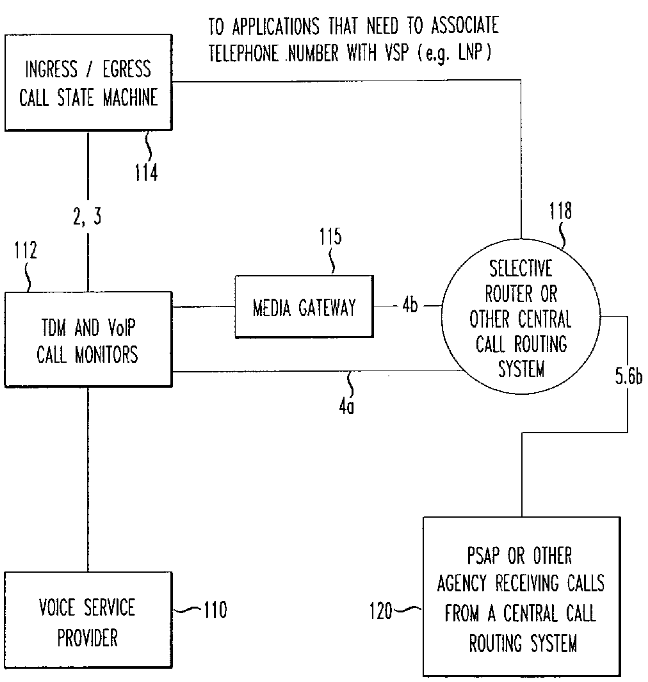

FIG. 1

(57) **Abstract**: A system monitors calls for a service provider. A number of simultaneous communications of a given technology type is monitored for either inbound communications and/or outbound communications associated with at least one service provider. A determination is made if the number of simultaneous communications from the service provider is in excess of a adjustable but set number of simultaneous communications of a given technology type that are permissible. A predetermined action is taken if the number of simultaneous calls or text messages (e.g., SMS, IM, email) is in excess of the set limit, e.g., the call may be terminated or other action taken. Both voice communications and/or non-voice communications (such as SMS, IM, Email, or MMS) can be monitored and throttled.

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INGRESS/EGRESS CALL MODULE

This application claims priority from U.S. Provisional Application 61/071,547 entitled "INGRESS/EGRESS CALL MODULE", filed May 5, 2008, and U.S. Provisional Application 61/129,006 entitled "INGRESS/EGRESS CALL MODULE", filed May 30, 2008, the entireties of which are expressly incorporated herein by reference.

BACKGROUND OF THE INVENTION

1. Field of the Invention

This invention relates generally to communications. More particularly, it relates to throttling on ingress to and egress from a central system for voice or text-based communications.

2. Background of the Related Art

9-1-1 is a phone number widely recognized in North America as an emergency phone number that is used to contact emergency dispatch personnel. Enhanced 9-1-1 (E9-1-1) is defined by an emergency call being selectively routed to an appropriate PSAP, based on a special identifier (a TN or "telephone number" that is either an ANI or "Automatic Number Identification" that identifies the caller's phone or the TN is a P-ANI, or "Pseudo Automatic Number Identifier", also referred to as "ESxK", that only identifies the PSAP the call should route to and the company routing the call), and includes the transmission of callback number and location information when 9-1-1 is used. E9-1-1 may be implemented for landline, cellular or VoIP networks. A Public Safety Answering Point (PSAP) is a dispatch office that receives 9-1-1 calls from the public. A PSAP may be a local, fire or police department, an ambulance service or a regional office covering all services. As used herein, the term "PSAP" refers to either a public safety answering point (PSAP), or to an Emergency Call Center (ECC), a VoIP term.

Regardless of the network type, a 9-1-1 service becomes E-9-1-1 when automatic number identification and automatic location information related to the call is provided to the 9-1-1 operator at the PSAP.

The current 911 infrastructure is designed to route a live voice call to a local public safety answering point (PSAP). This requires that voice circuits be available. The result of an E911 call is a direct circuit switched voice connection between an emergency service requestor and a suitable responder. 911 is further enhanced with the ability to deliver location over a data channel in parallel to the call. The location data is typically staged in a database that is queried by the PSAP to determine location information.

Fig. 6 shows a conventional landline public safety access point (PSAP) to automatic location identifier (ALI) connection.

In particular, Fig. 7 shows a PSAP **400** connected to one Automatic Location Identifier (ALI) database **401**. Upon receiving a 9-1-1 call, the PSAP **400** queries the ALI **401** for location data. The ALI database **401** accepts the query from the PSAP **400** for location. The query includes the telephone number of an emergency caller. The ALI database **401** relates the received telephone number to a physical street address and provides that street address (location information) back to the PSAP **400** in a manner that works for the customer premise equipment (CPE) display at the PSAP **400**.

An ALI is typically owned by a local exchange carrier (LEC) or a PSAP, and may be regional (i.e. connected to many PSAPs) or standalone (i.e. connected to only one PSAP).

Fig. 7 shows a context diagram for a conventional non-landline positioning center (e.g., an Internet based Voice over Internet Protocol (VoIP) positioning center).

In particular, the ALI database **401** includes a conventional emergency services key (ESQK or ESRK) in a location request sent to an appropriate positioning center **402** (XPC). The emergency services key (ESQK or ESRK) is used by the positioning center **402** as a key to look up the location and other call information associated with the emergency call.

In non-landline telephony, the PSAPs **400** query the ALI **401** for location information. However, the ALI **401** is not pre-provisioned with location data for non-landline calls (e.g. cellular, VoIP etc) and must communicate with other network entities to obtain and deliver location data to the PSAP **400**.

911 calls require voice circuits to be available to complete the voice call to a PSAP. For the most part, PSAPs are capable of receiving only voice calls. Connectivity with a PSAP, established either through the existing time division multiplexed (TDM)-based emergency services network (ESN), or directly over the public switched telephone network (PSTN) to the PSAP, is managed through dedicated telephone switches that cannot be directly dialed.

The present inventors have appreciated that during times of regional crises, such as during a hurricane, the local wireless infrastructure can become overloaded by call volume. This was experienced during the September 11, 2001, terrorist attacks during which voice telecommunications along the east coast was subjected to service failures.

Existing technology uses a limit on the number of voice circuits or network bandwidth available for voice calls or other calls based on other technologies. Existing technology is not dynamic and cannot be controlled from a central access point to a system. Moreover, existing technology cannot deliver location information at the time of call set up and cannot take advantage of the diversity, redundancy, and resiliency of IP networks.

SUMMARY OF THE INVENTION

In accordance with the principles of the present invention, a method of monitoring a state for calls for a service provider comprises monitoring, from a central call routing system, a number of communications for at least one of inbound communications and outbound communications associated with at least one service provider. A determination is made from a state machine if the number of communications from the at least one service provider is in excess of a predetermined number of communications that are permissible. The state machine performs a predetermined action if the number of calls associated with the at least one service provider is in excess of the predetermined number of communications.

In accordance with another aspect of the invention, a system for throttling calls and data messaging being handled by a given service provider comprises a central call routing system to monitor a number of simultaneous communications of a given technology type associated with a given service provider. A state machine dynamically determines a number of simultaneous communications associated with the given service provider, and initiates a predetermined action against any simultaneous communication in excess of a predetermined limited number of simultaneous communications.

BRIEF DESCRIPTION OF THE DRAWINGS

Features and advantages of the present invention will become apparent to those skilled in the art from the following description with reference to the drawings, in which:

Figs. 1 depicts exemplary message flows for an exemplary communications system, in accordance with the principles of the present invention.

Figs. 2 depicts exemplary message flows for another exemplary communications system, in accordance with the principles of the present invention.

Fig. 3 shows an exemplary ingress/egress system and message flow for another exemplary communications system, in accordance with the principles of the present invention.

Fig. 4 shows exemplary message flow for yet another exemplary communications system, in accordance with the principles of the present invention.

Fig. 5 shows an exemplary message flow for another exemplary communications system, in accordance with the principles of the present invention.

Fig. 6 shows a conventional landline public safety access point (PSAP) to automatic location identifier (ALI) connection.

Fig. 7 shows a context diagram for a conventional non-landline positioning center (e.g., an Internet based voice over Internet Protocol (VoIP) positioning center).

DETAILED DESCRIPTION OF ILLUSTRATIVE EMBODIMENTS

The present invention prevents any one voice service provider or one communication technology from monopolizing a central system's resources.

In accordance with the principles disclosed herein, a module keeps state for all voice calls or non-voice-based requests (SMS, IM, Email) inbound to a central system and outbound from a central system. Any request attempt (ingress) from a service provider in excess of their voice, SMS, IM, or Email allowance are immediately terminated or some other default or chosen action. Additionally or alternately, any request, regardless of originating service provider, in excess of a set limit for a given communication technology (e.g. only 5 concurrent SMS requests are allowed to the central system), are immediately terminated or some other default or chosen action.

Similarly or alternatively, preferably any request that would result in an excess of the allowed number of attempted completions (egress) regardless of service provider or technology is immediately terminated or some other default or chosen action.

The present invention provides benefits to multiple voice service providers (VSP) such that they can use the same circuit(s) as long as the monitoring device and/or IECSM can determine which VSPs calls are which. Moreover, local number portability is no longer an issue because the IECSM can inform systems downstream which carrier is using a specific TN for the current active call.

The invention allows multiple VSPs to use the same circuits or trunks. Moreover, it provides a solution that does not require the locking and unlocking of ported telephone numbers (LNP).

The present invention has applicability with virtually any voice service provider that operates a central call routing system. In accordance with the invention, the central call routing system throttles calls, SMSs, emails, IMs, etc. based on both incoming calls and data messages from other VSPs as well as outgoing calls and data messages to the agency/service that the relevant central call routing system serves.

The present invention provides a Next Generation 9-1-1 messaging center that allows a selective router to receive TDM calls and make routing decision for those calls. The Next Generation 9-1-1 messaging center disclosed herein comprises a media gateway, a back-to-back user agent (B2BUA) that allows SIP body rewrites, and software that integrates with the B2BUS and allows it to query various databases.

The Next Generation 9-1-1 messaging center disclosed herein converts a time division multiplex (TDM) call to voice over Internet protocol (VoIP), and inserts the appropriate location information into the SIP Invite body in the form of PIDF-LO as recommended by NENA i3 standards. The Next Generation 9-1-1 messaging center brings voice, video, and text from any originating device to Internet Protocol (IP) capable PSAPs.

The Next Generation 9-1-1 messaging center also intercepts selective transfer requests from Internet Protocol (IP) public service access points (PSAPs) and determines if the responder to receive that transfer is IP capable or TDM capable. If the responder is IP capable, the Next Generation 9-1-1 Messaging Center forwards the call over the IP network. If the responder is TDM capable, the Next Generation 9-1-1 messaging center determines the dialing pattern necessary for the selective router and sends calls to the selective router for completion.

The Next Generation 9-1-1 infrastructure disclosed herein replaces or augments selective routers and automatic location identification (ALI) databases. The Next Generation 9-1-1 systems disclosed herein eliminate the need for MPCs, GMLCs, and VPCs. ALI databases continue to provide traditional data for Local Exchange Carrier (LEC) subscribers. Other customer operated databases may be expanded to provide further personal information.

The invention provides a module that keeps state information for all voice calls inbound to a system and outbound from a system. Any call set up attempt (ingress) from a voice service provider in excess of the relevant allowance is immediately terminated (or other default of chosen action). Similarly, any call attempt that would otherwise result in an excess of the allowed number of attempted completions (egress) would instead be immediately terminated (or other default or chosen action).

Those having particular use of the invention include any voice service provider that operates a central call routing system. The invention provides the ability to throttle calls or data messages based on both incoming calls from other VSPs as well as outgoing calls to the agency/service that the relevant central call routing system serves.

Fig. 1 shows exemplary message flow for an exemplary communications system, in accordance with the principles of the present invention.

In particular, as shown in Fig. 1, various communications are illustrated between an "Ingress/Egress Call State Machine" (IECSM) **114**, a central call routing system **112**, a voice service provider **110**, a media gateway **115**, a selective router or other central call routing system (hereafter referred to as a selective router **118**), and a PSAP or other agency receiving calls from a central call routing system (hereafter referred to as a receiving agency **120**).

In step 1 of Fig. 1, a voice service provider (VSP) **110** initiates a call to the central call routing system **112**.

In step 2, monitoring devices (not shown) within the central call routing system **112** detect all calls and report the state to an IECSM **114**.

In step 3, an IECSM **114** receives notification of a call attempt from the monitoring devices within the central call routing system **112**. For example, if a VSP **110** is allowed, e.g., no more than three simultaneous calls, and the call in question would be the fourth call, the IESCM **114** sends notice to the monitoring device within the central call routing system **112** to perform some desired action, such as to immediately terminate the fourth call. Similarly, for example, if a set limit is that only ten calls are allowed to simultaneously egress the central call routing system **112**, and ten calls are already currently active, then on the next (i.e., 11th) call attempt from a VSP **110**, the IECSM **114** performs some desired action on the call, such as termination, even if the call would be the VSPs **110** only active call.

The IECSM **114** is preferably capable of informing other systems regarding which VSP **110** is using a given telephone number (TN), thus changing most processes involving Local Number Portability.

In step 4, a call attempt is passed to a central call routing system **112**.

In step 4a, if the central call routing system **112** is IP-based, the TDM calls must be converted to IP. On the other hand, as depicted in step 4b, if the central call routing system **112** is TDM based, the VoIP calls must be converted to TDM.

In step 5, a receiving agency **120** receives the call.

In step 6a, the VSP **110** terminates the call, or as depicted in step 6b, a receiving agency **120** terminates the call. In either scenario, the IECSM **114** changes the "ingress" state for that VSP **110** as well as changes the "egress" state for the agency **120** receiving calls.

Fig. 2 shows exemplary message flow for another exemplary communications system, in accordance with the principles of the present invention.

In particular, as shown in Fig. 2, various communications are illustrated between a voice service provider **110**, a media gateway (Note: In this and other uses, the term media gateway should be considered synonymous with the media gateway and the media gateway controller that may be used in conjunction with it.) **115**, a selective router or other central call routing system (hereafter referred to as a selective router **118**), and a PSAP or other agency receiving calls from a central call routing system (hereafter referred to as a receiving agency **120**), an Internet Protocol (IP) PSAP **122**, a police responder **130**, a Next Generation Messaging Center (NGMC) **124**, a Location Information Server (LIS) **126**, and a Automatic Location Identification (ALI) Database **128**.

The LIS **126** can be a database service that provides locations of endpoints. In practice, "LIS Steering" may be required to determine which of a possible plurality of LISs to query.

As shown in step 1 of Fig. 2, a Voice Service Provider **110** sends a call out on a 9-1-1 trunk (not shown).

In step 2, the call reaches the selective router **118** and is routed using a telephone number to a trunk group.

In step 2a, a trunk group terminates at a media gateway **115** because the PSAP is now an IP PSAP **122**.

In step 2b, a trunk terminates at a TDM based PSAP **120**.

In step 3, the media gateway **115** messages the NGMC **124**.

In step 4, the NGMC **124** queries either an LIS **126** or ALI **128** (which one is optional but one must be used) using a telephone number to retrieve the location information.

In step 5, the NGMC **124** sends a new message with Presence Information Data Format-Location Object (PIDF-LO) to the media gateway **115**.

In step 6, the IP PSAP **122** receives location information at the time of call set up.

In step 6a, the call goes to an IP based PSAP, e.g., IP PSAP **122**, based on a trunk decision made by the selective router **118**.

In step 6b, the call goes to a TDM based PSAP, e.g., TDM PSAP **120**, based on a trunk decision made by the selective router **118**.

In step 7, the IP PSAP **122** may initiate a selective transfer using, in this case, a SIP Invite to SOS.Police.

In step 8, the media gateway **115** again queries the NGMC **124**.

In step 9, the NGMC **124** determines who the responder is and then determines how to reach that responder using a database of responders: e.g., using either another Session Initiation Protocol (SIP) Uniform Resource

Identifier (URI), or using a dialing pattern that the selective router can interpret, such as "*1".

In step 10, the NGMC **124** sends what it has determined to the media gateway **115**.

In step 11, the media gateway **115** either sends the SIP universal resource indicator (URI) to another IP node, or, as in the example in Fig. 2, will send the dialing pattern to the selective router **118**.

In step 12, the selective router **118** interprets the dialing pattern and sends the call out to the trunk group of the appropriate police or other emergency responder **130**.

Fig. 3 shows an exemplary ingress/egress system and message flow for another exemplary communications system, in accordance with the principles of the present invention.

In particular, as shown in Fig. 3, call flow is illustrated between an IECSM **114**, a central call routing system **112**, a Communications Service Provider (CSP) **111**, a media gateway **115**, a selective router or other central call routing system (hereafter referred to as a selective router **118**), and a PSAP or other agency receiving calls from a central call routing system (hereafter referred to as a receiving agency **120**).

In step 1 of Fig. 3, a CSP **111** initiates a request to the central call routing system **112**. It is noted that in the example shown, it is assumed that Short Message Service (SMS), Instant Messaging (IM), Email requests, etc. will use another system to initiate an IP call request. This request may look to the receiving system like another VoIP call request.

In step 2, monitoring devices within the central call routing system **112** detect all requests and report the state to the IECSM **114**.

In step 3, an IECSM **114** receives notification of a request from the monitoring devices (not shown) within the central call routing system **112**. For example, if a CSP **111** is subject to a set limit of no more than three simultaneous requests, and a given request in question would be the fourth simultaneous request, the IECSM **114** sends notice to the monitoring device within the central call routing system **112** to perform some desired action, such as terminate the call.

Similarly, as another example, if only ten simultaneous requests are allowed to egress the central call routing system **112**, and ten requests are currently active, then on the next request from a CSP **111**, the IECSM **114** performs some desired action on the call, such as termination, even if the call would be the CSP's **111** only active call.

Additionally, the IECSM **114** can be configured to allow only a given number of requests (e.g., three) originated from a particular technology (e.g., from an SMS system). In this case, upon receipt of a 4th request made over SMS, the IECSM **114** performs some desired action on the request, such as termination.

The IECSM **114** can also inform other systems regarding which CSP **111** is using a given telephone number, thus changing most processes involving Local Number Portability.

In step 4, the request is passed to a central call routing system **112**.

In step 4a, if the central call routing system **112** is IP-based, the TDM calls must be converted to IP.

In step 4b, if the central call routing system is TDM based, the VoIP calls must be converted to TDM. In the case where the request is not a call (such as SMS, IM, or Email), the request is passed forward by another system as if it were a VoIP call so that services can be provided by the central call routing system **112**.

In step 5, an end agency, e.g., PSAP **120**, receives the request and performs the appropriate action, i.e., either carries out a voice conversation or provides information to communicate to an SMS, IM, or Email user.

In step 6a, a CSP **111** terminates the call, or as depicted in step 6b, the receiving agency, e.g., PSAP **120**, terminates the call. In either scenario, the IECSM **114** changes the "ingress" state for that CSP **111**, i.e., changes the "ingress" state for the technology in question, as well as changes the "egress" state for the agency receiving calls.

Fig. 4 shows an exemplary message flow for another exemplary communications system, in accordance with the principles of the present invention.

In particular, as shown in Fig. 4, various communications are illustrated between a Voice Service Provider **110**, a selective router or other central call routing system (hereafter referred to as a first selective router **118a** and a second selective router **118b**), a first selective router **118a**, a PSAP or other agency receiving calls from a central call routing system (hereafter referred to as a first receiving agency **120a** and a second receiving agency **120b**), a first PSAP **132**, a national media gateway network **134**, a National Transfer VPC **136**, a first ALI **128a**, and a second ALI **128b**.

In step 1 of Fig. 4, the Voice Service Provider **110** sends a call into the local E9-1-1 system.

In step 2, the first selective router **118a** sends the call to the first PSAP **120a**.

In step 3, the first PSAP **120a** determines that the call must be transferred to a PSAP not connected to their selective router, e.g., the second PSAP **120b**. The first PSAP **120a** dials a 10-digit number that corresponds to an IVR system (not shown) in the National Transfer VPC **136**. The IVR system retrieves the call back number and the destination PSAP, e.g., second PSAP **120b**, or destination city/state.

In step 4, the National Transfer VPC **136** retrieves location information from the local ALI, e.g., ALI **128a**, querying as if it is a PSAP. The National Transfer VPC **136** assigns the call an ESRN and ESQK appropriate to the location provided to the IVR.

In step 5, the call is sent out on the national media gateway network **134**.

In step 6, the national media gateway network **134** forwards the call to the second selective router **118b**.

In step 7, the second selective router **118b** forwards the call to the second PSAP **120b** provided to the National Transfer VPC's **136** IVR.

In step 8, the second PSAP **120b** queries the local ALI **128b** for an ESQK.

In step 9, the national transfer VPC **136** provides the caller's callback number and location information.

In step 10, the local ALI **128b** delivers the caller's information.

Fig. 5 shows an exemplary message flow for another exemplary communications system, in accordance with the principles of the present invention.

In particular, as shown in Fig. 5, various communications are illustrated between a first service provider **110a**, a second voice service provider **110b**, a third voice service provider **110c**, an IP selective router **140**, a first media gateway **115a**, a second media gateway **115b**, a selective router **118**, and a PSAP or other agency receiving calls from a central call routing system (hereafter referred to as a receiving agency **120**). The IP selective router **140** includes a Policy Routing Function (PRF) **142**, a VPC/Emergency Services Routing Proxy (ESRP) **144**, and a conference bridge **146**.

The ESRP **144** preferably is a SIP proxy server that selects the next hop routing within an ESInet based on location and policy. A "PSAP Proxy" is used in some implementations to facilitate completion of calls to Legacy PSAPs.

The Policy Routing Function (PRF) **142** is preferably an entity that defines attributes such as hours of operation, default routing, and overflow routing.

In step 1a of Fig. 5, all VoIP Service Providers **110a**, **110b**, and **110c**, send calls to the IP Selective Router **140** within the system **150** that provides parallel service to a TDM-based selective router **118**.

In step 2b, the IP Selective Router **140** routes a call to a media gateway **115b** local to the correct receiving agency **120**.

In step 3, the media gateway **115b** converts a call back to TDM and passes the call to the receiving agency's **120** TDM-based PBX over local cables, just as if they were trunks from a selective router **118**.

The IP Selective Router **140** interprets the receiving agency's **120** equipment signaling for a selective transfer and sets up the call to the responder from the receiving agency's **120** local media gateway **115b** using a PRI.

In step 1b, the voice service provider **110b** sends calls to the media gateway **115b** local to the TDM based selective router **118**.

In step 2a, the media gateway **115a** local to the TDM based selective router **118** signals to the IP selective router **140**.

In step 3, the IP selective router **140** routes the call, through the media gateway **115a** local to the TDM selective router **118**, and to a media gateway **115** local to the correct receiving agency **120**.

In step 4, the media gateway **115b** local to the receiving agency **120** converts the call back to TDM and passes the call to the receiving agency's **120** TDM PBX over local cables, just as if they were trunks from a selective router **118**.

The IP selective router **140** interprets the receiving agency **120** equipment signaling for a selective transfer and sets up the call to the responder from the receiving agency's **120** local media gateway **115b** using a PRI.

One of ordinary skill in the art will appreciate that the present invention can be used with an Emergency Call Routing Function (ECRF) that receives location information (either civic address or geo-coordinates) as input. The ECRF uses the information to provide a URI that routes an emergency call toward the appropriate PSAP for the caller's location.

One of ordinary skill in the art will appreciate that the present invention can also be used with a Location Validation Function (LVF) that validates location objects against the next generation address data.

Advantageously, the invention may be implemented without the need for immediate PSAP upgrades. It enables transfers from any selective routers disclosed herein to any other selective router disclosed herein with only minor configuration changes. No interoperability issues are presented or additional software loads required.

Implementation of the IP selective router disclosed herein provides for a slow migration to IP call routing while at the same time extends the life of existing selective routers. The invention allows for dynamic addition of call answering stations at a PSAP by simple activation of additional DS0s from the local media gateway into a PBX. It establishes a diverse, redundant IP network for use by voice service providers to PSAPs. Additional services will also be able to make use of the networks disclosed herein.

The Next Generation Messaging Center disclosed herein makes selective routers and ALI databases next generation capable. The Next Generation Messaging Center disclosed herein establishes a diverse, redundant IP network for use by voice service providers to PSAPs. Additional services are able to utilize this network. The invention affects only PSAPs migrating to IP, with no disruption to other E9-1-1 customers.

While the invention has been described with reference to the exemplary embodiments thereof, those skilled in the art will be able to make various modifications to the described embodiments of the invention without departing from the true spirit and scope of the invention.

CLAIMS

What is claimed is:

1. A system for throttling calls and data messaging being handled by a given service provider, comprising:

a central call routing system to monitor a number of communications of a given technology type associated with a given service provider; and

a state machine to dynamically determine a number of simultaneous communications associated with said given service provider, and to initiate a predetermined action against any simultaneous communication in excess of a predetermined limited number of simultaneous communications.

2. The system for throttling calls and data messaging being handled by a given service provider according to claim 1, wherein:

said predetermined action is a termination of said any simultaneous communications in excess of said predetermined limited number.

3. The system for throttling calls and data messaging being handled by a given service provider according to claim 1, wherein:

said communications are voice calls.

4. The system for throttling calls and data messaging being handled by a given service provider according to claim 1, wherein:

said communications are text-based communications.

5. The system for throttling calls and data messaging being handled by a given service provider according to claim 4, wherein:

said text-based communications are Short Messaging System (SMS) communications.

6. The system for throttling calls and data messaging being handled by a given service provider according to claim 4, wherein:

said text-based calls are Instant Messaging (IM) communications.

7. The system for throttling calls and data messaging being handled by a given service provider according to claim 4, wherein:

said text-based calls are Email communications.

8. A method of monitoring a state for calls for a service provider, comprising:

monitoring, from a central call routing system, a number of communications for at least one of inbound communications and outbound communications associated with at least one service provider; and

determining, from a state machine, if said number of communications associated with said at least one service provider is in excess of a predetermined number of communications that are permissible;

wherein said state machine performs a predetermined action if said number of communications from said at least one service provider is in excess of said predetermined number of communications.

9. The method of monitoring a state for calls for a service provider according to claim 8, wherein:

said predetermined action is a termination of communications in excess of said predetermined number of communications.

10. The method of monitoring a state for calls for a service provider according to claim 8, wherein:

said communications are voice calls.

11. The method of monitoring a state for calls for a service provider according to claim 8, wherein:

said communications are non-voice based communications.

12. The method of monitoring a state for calls for a service provider according to claim 11, wherein:

said non-voice based communications are Short Messaging System (SMS) communications.

13. The method of monitoring a state for calls for a service provider according to claim 11, wherein:

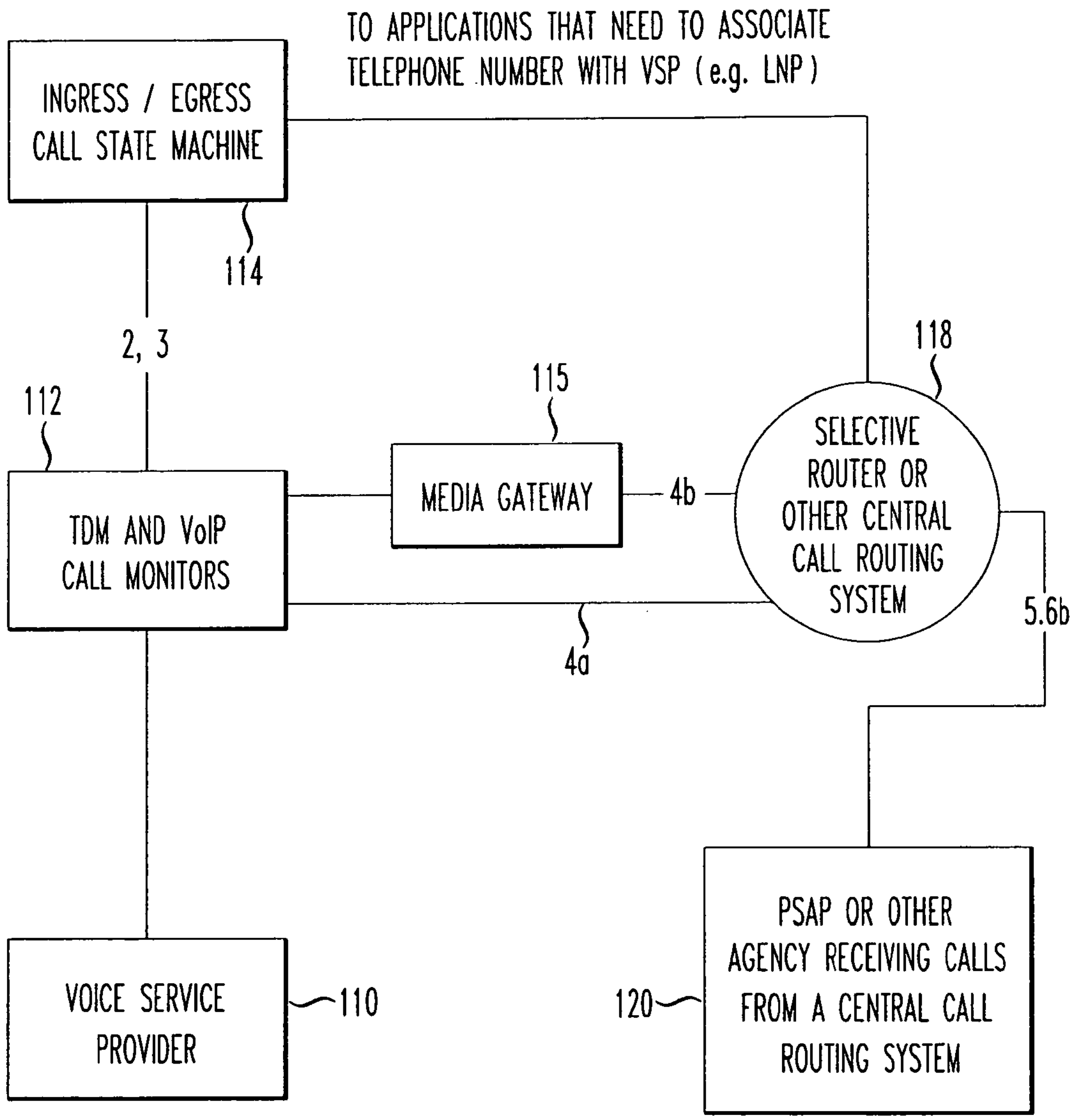
said non-voice based calls are Instant Messaging (IM) communications.

14. The method of monitoring a state for calls for a service provider according to claim 11, wherein:

said non-voice based calls are Email communications.

15. The method of monitoring a state for calls for a service provider according to claim 8, wherein:

said predetermined action is a default action.



TO APPLICATIONS THAT NEED TO ASSOCIATE TELEPHONE NUMBER WITH VSP (e.g. LNP)
INGRESS / EGRESS CALL STATE MACHINE
114
2, 3
112
TDM AND VoIP CALL MONITORS
115
MEDIA GATEWAY
4b
118
SELECTIVE ROUTER OR OTHER CENTRAL CALL ROUTING SYSTEM
4a
5.6b
VOICE SERVICE PROVIDER
110
120
PSAP OR OTHER AGENCY RECEIVING CALLS FROM A CENTRAL CALL ROUTING SYSTEM

FIG. 1

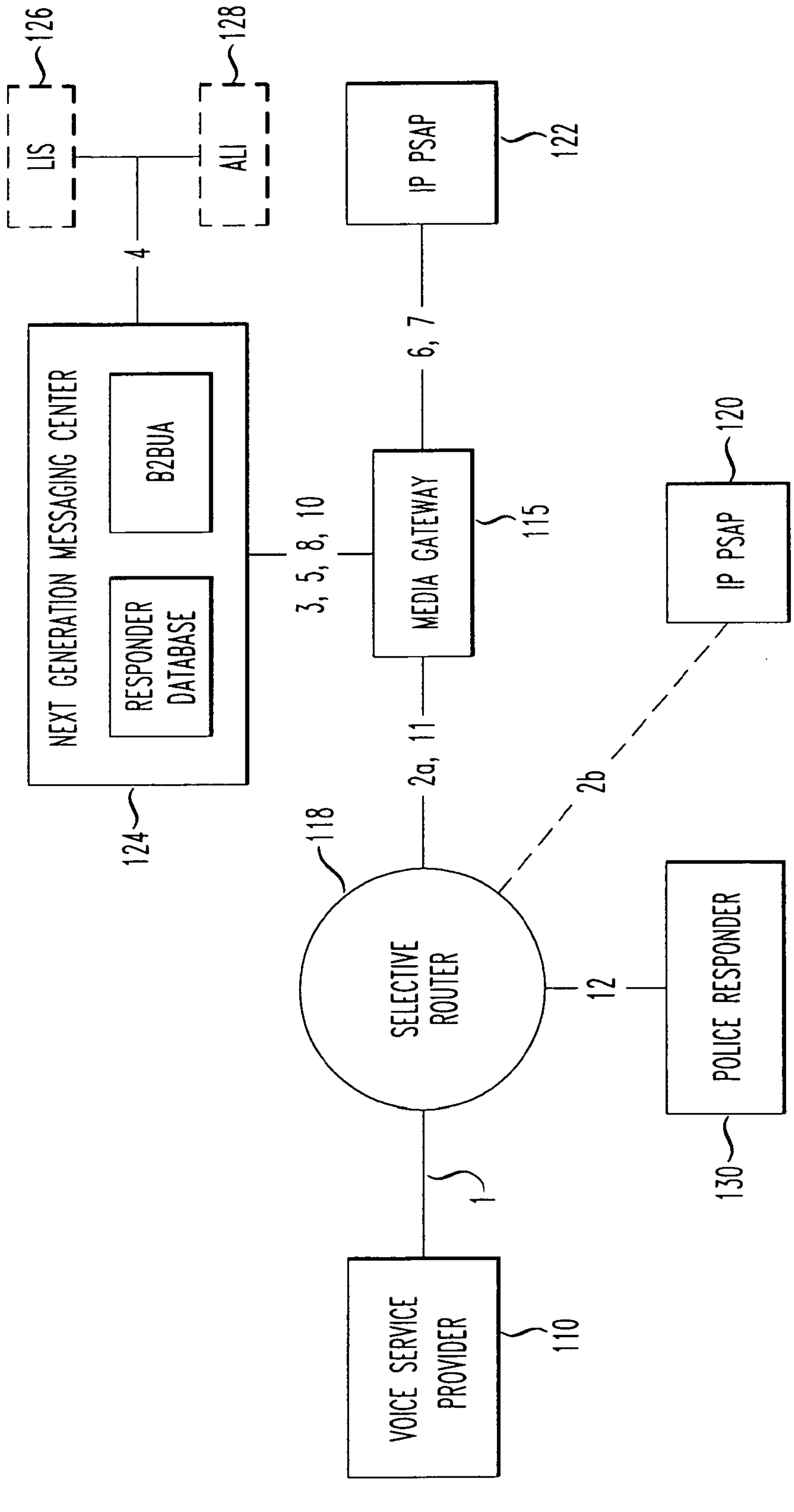

LIS
126
ALI
128
4
IP PSAP
122
NEXT GENERATION MESSAGING CENTER
B2BUA
RESPONDER DATABASE
124
3, 5, 8, 10
6, 7
MEDIA GATEWAY
115
IP PSAP
120
2a, 11
2b
118
SELECTIVE ROUTER
12
POLICE RESPONDER
130
1
VOICE SERVICE PROVIDER
110

FIG. 2

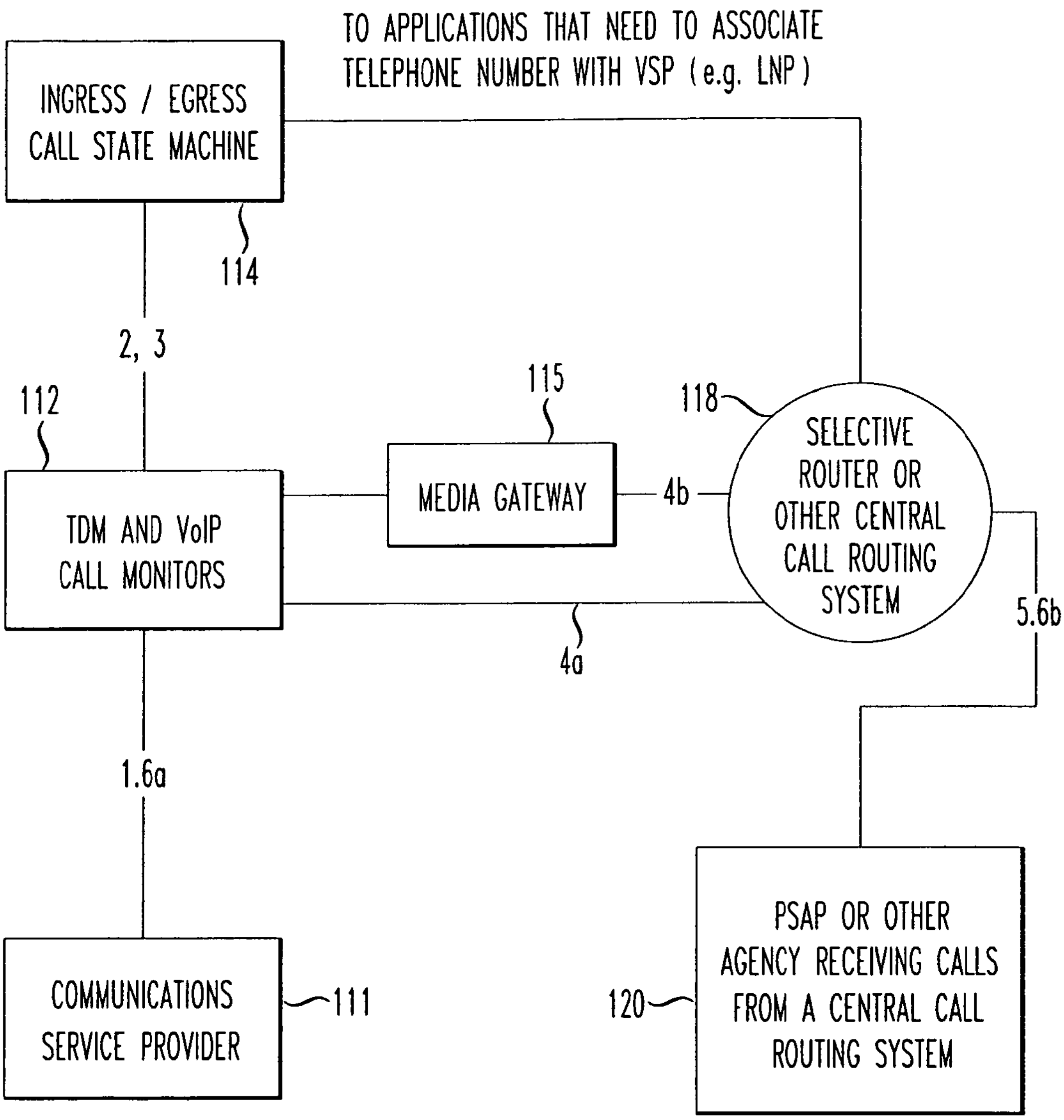

TO APPLICATIONS THAT NEED TO ASSOCIATE TELEPHONE NUMBER WITH VSP (e.g. LNP)
INGRESS / EGRESS CALL STATE MACHINE
114
2, 3
112
115
118
TDM AND VoIP CALL MONITORS
MEDIA GATEWAY
4b
SELECTIVE ROUTER OR OTHER CENTRAL CALL ROUTING SYSTEM
4a
5.6b
1.6a
COMMUNICATIONS SERVICE PROVIDER
111
120
PSAP OR OTHER AGENCY RECEIVING CALLS FROM A CENTRAL CALL ROUTING SYSTEM

FIG. 3

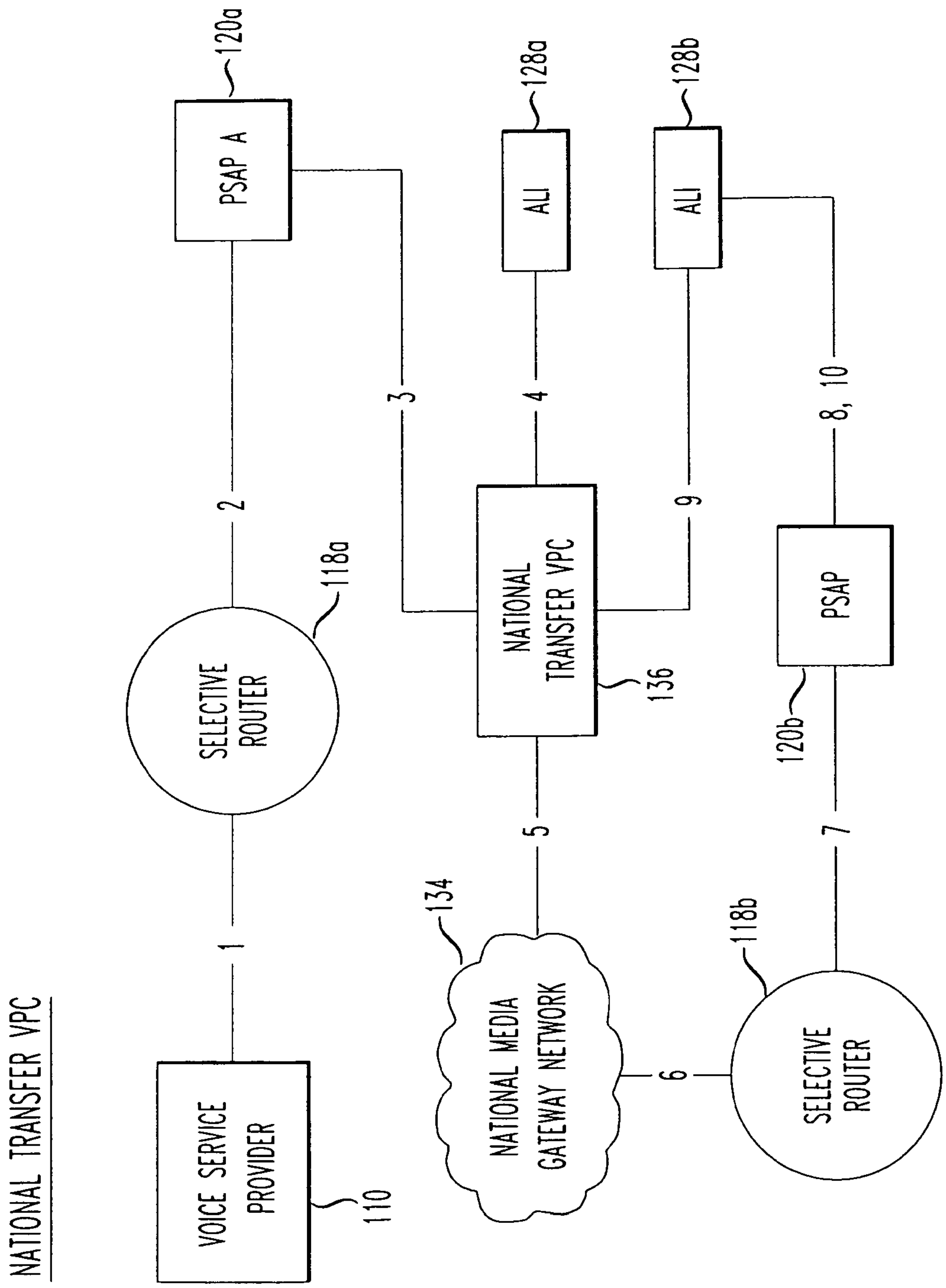

NATIONAL TRANSFER VPC
VOICE SERVICE PROVIDER
110
1
SELECTIVE ROUTER
118a
2
PSAP A
120a
3
NATIONAL TRANSFER VPC
136
4
ALI
128a
5
NATIONAL MEDIA GATEWAY NETWORK
134
6
SELECTIVE ROUTER
118b
7
PSAP
120b
8, 10
9
ALI
128b

FIG. 4

5/7

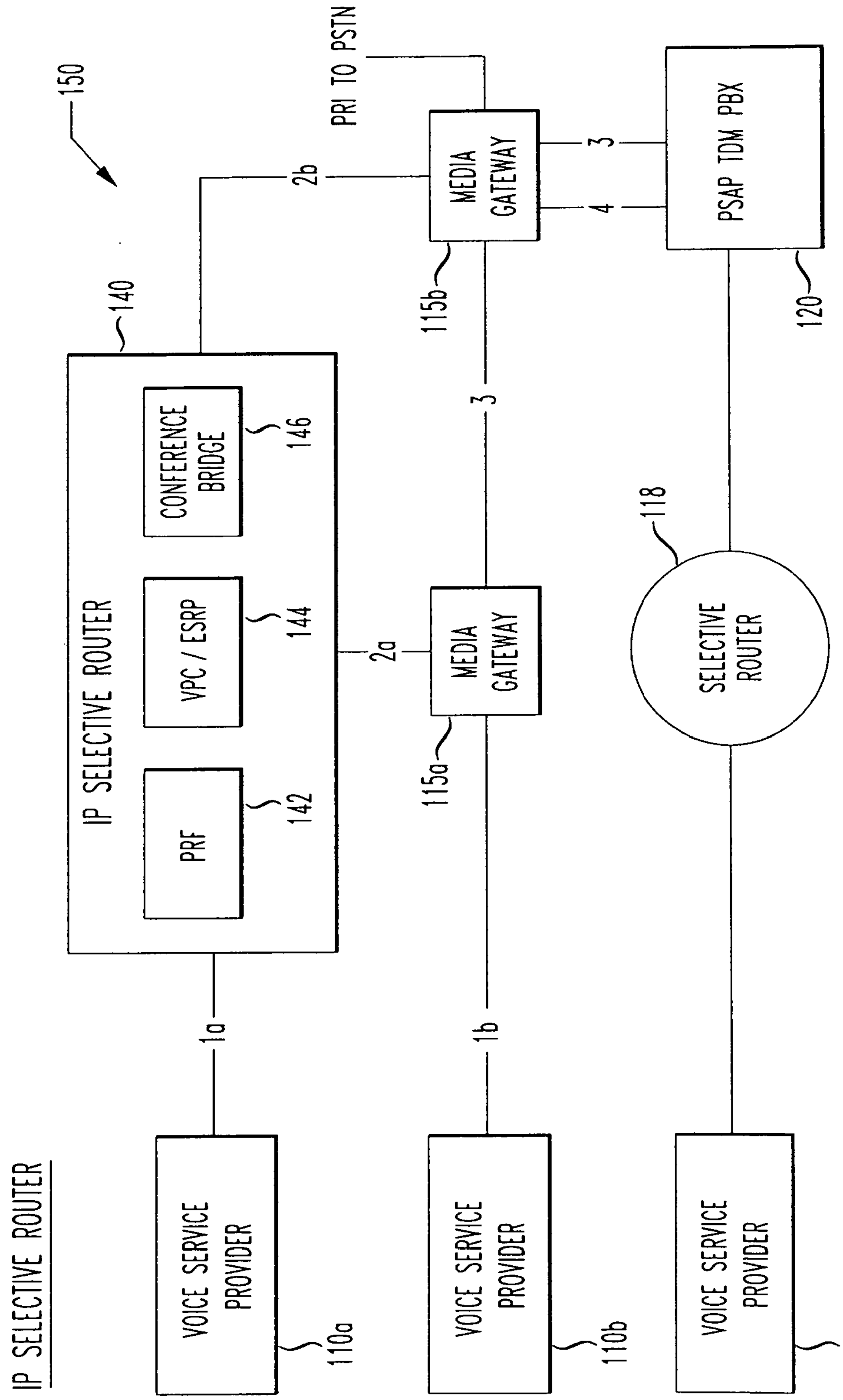

IP SELECTIVE ROUTER
150
VOICE SERVICE PROVIDER
110a
VOICE SERVICE PROVIDER
110b
VOICE SERVICE PROVIDER
110c
1a
1b
IP SELECTIVE ROUTER
140
PRF
142
VPC / ESRP
144
CONFERENCE BRIDGE
146
2a
2b
MEDIA GATEWAY
115a
MEDIA GATEWAY
115b
3
PRI TO PSTN
3
4
PSAP TDM PBX
120
SELECTIVE ROUTER
118

FIG. 5

FIG. 6
(PRIOR ART)

LANDLINE PSAP ALI INTERACTION

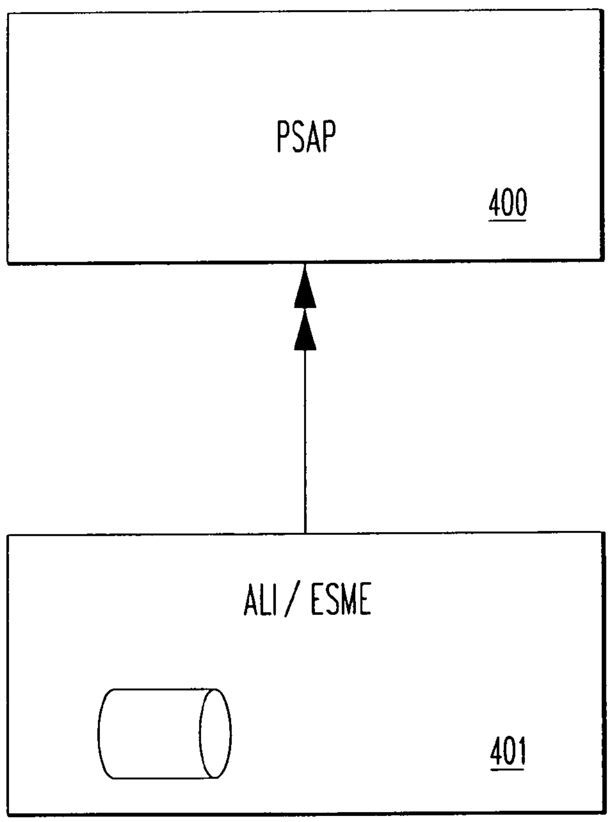
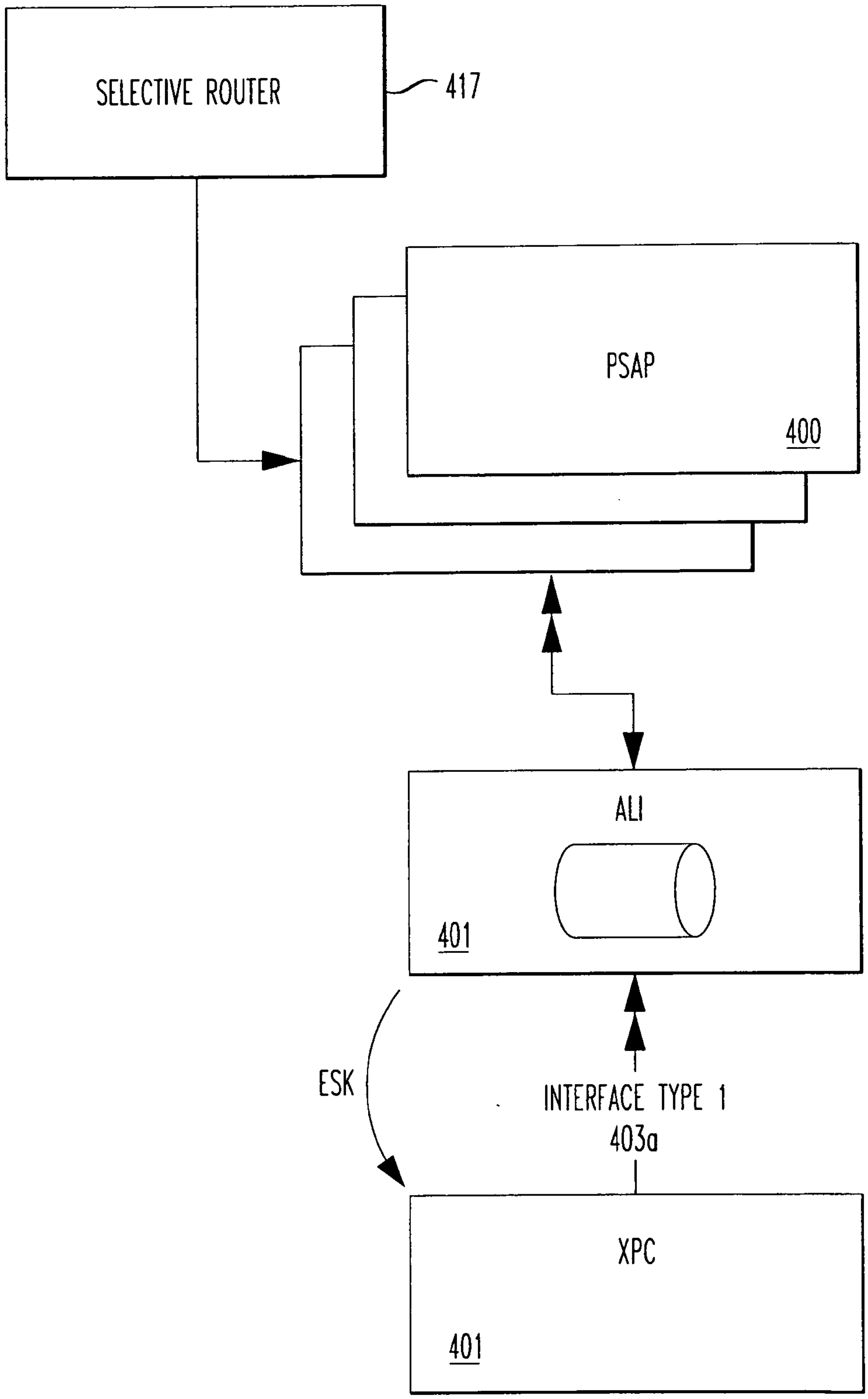


FIG. 7
(PRIOR ART)



INTERNATIONAL SEARCH REPORT

International application No.
PCT/US2009/002752

A. CLASSIFICATION OF SUBJECT MATTER
IPC(8) - H04L 12/26 (2009.01)
USPC - 370/230.1
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)
IPC(8) - H04L 12/26 (2009.01)
USPC - 370/230.1

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

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)
MicroPatent, Patbase, Google Patent, IEEEXPLORE, Google Scholar

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	US 2005/0271051 A1 (HOLLOWAY et al) 08 December 2005 (08.12.2005) entire document	1-15
Y	US 6,330,313 B1 (HUNT) 11 December 2001 (11.12.2001) entire document	1-15
Y	US 2005/0148351 A1 (REDING et al) 07 July 2005 (07.07.2005) entire document	4-7, 11-14
Y	US 6,898,274 B1 (GALT et al) 24 May 2005 (24.05.2005) entire document	15

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* Special categories of cited documents:
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"P" document published prior to the international filing date but later than the priority date claimed
"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
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"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 obvious to a person skilled in the 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
24 June 2009	06 JUL 2009

Name and mailing address of the ISA/US	Authorized officer:
Mail Stop PCT, Attn: ISA/US, Commissioner for Patents P.O. Box 1450, Alexandria, Virginia 22313-1450 Facsimile No. 571-273-3201	Blaine R. Copenheaver PCT Helpdesk: 571-272-4300 PCT OSP: 571-272-7774