(12) INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(19) World Intellectual Property Organization
International Bureau

(43) International Publication Date
12 September 2008 (12.09.2008)

PCT

(10) International Publication Number
WO 2008/107898 A2

(51) International Patent Classification:
H04Q 9/02 (2006.01)

(21) International Application Number:
PCT/IL2008/000307

(22) International Filing Date: 6 March 2008 (06.03.2008)

(25) Filing Language: English

(26) Publication Language: English

(30) Priority Data:
60/905,760 7 March 2007 (07.03.2007) US

(71) Applicant (*for all designated States except US*): OUT-SMART LTD. [IL/IL]; 1 Bat-Sheva Street, 71100 Lod (IL).

(72) Inventors; and
(75) Inventors/Applicants (*for US only*): GABBAY, Avi [IL/IL]; 32 Yad Shalom Street, 52354 Ramat Gan (IL). ARUCH, Ronny [IL/IL]; 39 Yitzhak Rabin Street, 55510 Kiryat Ono (IL).

(74) Agents: SANFORD T. COLB & CO. et al.; P.O. Box 2273, 76122 Rehovot (IL).

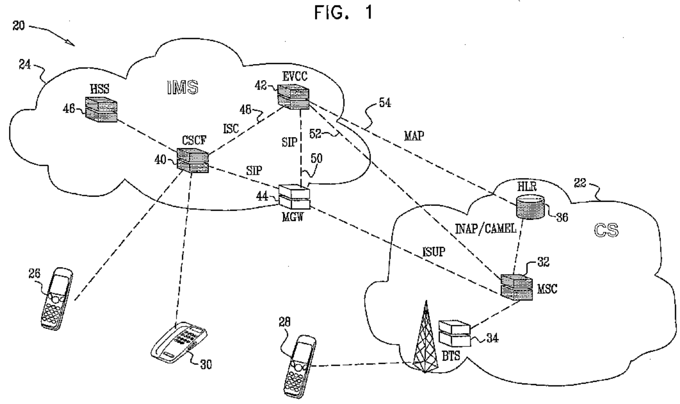
(81) Designated States (*unless otherwise indicated, for every kind of national protection available*): AE, AG, AL, AM, AO, AT, AU, AZ, BA, BB, BG, BH, BR, BW, BY, BZ, CA, CH, CN, CO, CR, CU, CZ, DE, DK, DM, DO, DZ, EC, EE, EG, ES, FI, GB, GD, GE, GH, GM, GT, HN, HR, HU, ID, IL, IN, IS, JP, KE, KG, KM, KN, KP, KR, KZ, LA, LC, LK, LR, LS, LT, LU, LY, MA, MD, ME, MG, MK, MN, MW, MX, MY, MZ, NA, NG, NI, NO, NZ, OM, PG, PH, PL, PT, RO, RS, RU, SC, SD, SE, SG, SK, SL, SM, SV, SY, TJ, TM, TN, TR, TT, TZ, UA, UG, US, UZ, VC, VN, ZA, ZM, ZW.

(84) Designated States (*unless otherwise indicated, for every kind of regional protection available*): ARIPO (BW, GH, GM, KE, LS, MW, MZ, NA, SD, SL, SZ, TZ, UG, ZM, ZW), Eurasian (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European (AT, BE, BG, CH, CY, CZ, DE, DK, EE, ES, FI, FR, GB, GR, HR, HU, IE, IS, IT, LT, LU, LV, MC, MT, NL, NO, PL, PT, RO, SE, SI, SK, TR), OAPI (BF, BJ, CF, CG, CI, CM, GA, GN, GQ, GW, ML, MR, NE, SN, TD, TG).

Published:
— *without international search report and to be republished upon receipt of that report*

(54) Title: ENHANCED VOICE CALL CONTINUITY SERVER

FIG. 1



(57) Abstract: A method for communication includes registering in a packet-switched (PS) network (24) a dual-mode handset (26), which is capable of voice communications over both the PS network and over a cellular circuit-switched (CS) network (22), and which is assigned a first telephone number in the cellular CS network. A second telephone number, different from the first telephone number, is assigned to the dual-mode handset for use in placing voice calls over the PS network. A call is established, using the second telephone number, between the dual-mode handset and a telephone (28) in the cellular CS network via a Voice Call Continuity (VCC) server (42), which is connected to communicate with both the PS and the CS networks. The VCC server may also support other types of handsets and other enhanced services.

WO 2008/107898 A2

ENHANCED VOICE CALL CONTINUITY SERVER

CROSS-REFERENCE TO RELATED APPLICATIONS

This application claims the benefit of U.S. Provisional Patent Application 60/905,760, filed March 7, 2007, whose disclosure is incorporated herein by reference.

FIELD OF THE INVENTION

The present invention relates generally to communication systems, and specifically to convergence of circuit-switched and packet network communication services.

BACKGROUND OF THE INVENTION

The IP Multimedia Subsystem (IMS) is an architectural framework for delivering Internet Protocol (IP) network services and content to mobile telephone users. It was originally designed by the 3rd Generation Partnership Project (3GPP). Technical aspects of IMS are defined in 3GPP Technical Specification 23.228, entitled "IP Multimedia Subsystem (IMS); Stage 2" (Release 8, 2007), which is incorporated herein by reference. This and other 3GPP technical specifications are available at the 3gpp.org Web site.

Users can connect to an IMS network using various types of IP access networks, including both wireless and fixed access networks. (Users in circuit-switched telephone networks, such as existing cellular and wireline networks, connect to IMS through appropriate gateways.) IMS defines a horizontal control layer, which separates the various access networks from the service layer. The core of this control layer is the Call Session Control Function (CSCF), which interacts with terminals, application servers, and other network components using the Session Initiation Protocol (SIP). A Home Subscriber Service (HSS) stores user profiles and performs authentication and authorization functions.

One type of application server (AS) that has been defined by 3GPP is the Voice Call Continuity (VCC) server, which permits mobile telephones to maintain a voice call without interruption as they move between circuit-switched (CS) and packet-switched (PS) domains. VCC is of particular interest for dual-mode telephone handsets, which support both cellular and broadband wireless packet technologies (such as WiFi and/or WiMAX). To provide VCC service, calls to and from a handset in the CS domain are also anchored in the PS domain, using IMS functionality, for example. As the handset becomes attached and detached from wireless access points during a call, the VCC server opens and tears down call legs in the CS and PS domains so that the call is transferred between the domains in a manner transparent to the end user. Further technical aspects of VCC are defined in 3GPP Technical Specification 23.206, entitled "Voice Call Continuity (VCC) between Circuit Switched (CS) and IP Multimedia Subsystem (IMS); Stage 2" (Release 7, 2007), which is incorporated herein by reference.

SUMMARY OF THE INVENTION

VCC, in principle, permits the mobile handset to take advantage of the network (CS or PS) that provides the best quality and lowest cost at any given time during a call, while moving seamlessly between networks. In practice, however, there are still limitations and gaps in the quality and types of services that are available to the VCC user. Furthermore, adoption of IMS by network operators has been slow, and there is a need for VCC functionality in non-IMS networks, as well.

Embodiments of the present invention that are described hereinbelow provide an enhanced VCC server and methods for providing VCC service that go beyond the conventional model of the 3GPP specifications. Although some of these embodiments are defined in the context of the IMS environment, the principles of the present invention may be applied more generally in the context of CS/PS network convergence, with or without IMS.

There is therefore provided, in accordance with an embodiment of the present invention, a method for communication, including:

registering in a packet-switched (PS) network a dual-mode handset, which is capable of voice communications over both the PS network and over a cellular circuit-switched (CS) network, and which is assigned a first telephone number in the cellular CS network;

assigning to the dual-mode handset a second telephone number, different from the first telephone number, for use in placing voice calls over the PS network; and

establishing a call, using the second telephone number, between the dual-mode handset and a telephone in the cellular CS network via a Voice Call Continuity (VCC) server, which is connected to communicate with both the PS and the CS networks.

In a disclosed embodiment, assigning the second telephone number includes defining a service zone within the PS network in which the second telephone number is permitted to be used, and establishing the call includes connecting a leg of the call to the dual-mode handset via an access point of the PS network that is within the service zone or connecting the leg of the call to the dual-mode handset via the cellular CS network using the first telephone number when the dual-mode handset is outside the service zone.

There is also provided, in accordance with an embodiment of the present invention, a method for communication, including:

establishing a voice call between first and second handsets using a Voice Call Continuity (VCC) server, which is connected to communicate with both a packet-switched (PS) network and a circuit-switched (CS) network, the voice call having one leg in the PS network and another leg in the CS network;

receiving at the VCC server a request from a third handset to take over the call from the first handset; and

responsively to the request, connecting a new leg to the call, using the VCC server, so as to enable the third handset to communicate with the second handset.

The first handset is typically disconnected from the call upon connecting the new leg. In a disclosed embodiment, connecting the new leg includes, at the VCC server, evaluating an identity associated with the third handset and determining, responsively to the identity that the third handset is authorized to take over the call from the first handset.

There is additionally provided, in accordance with an embodiment of the present invention, a method for communication, including:

establishing a voice call using a Voice Call Continuity (VCC) server, which is connected to communicate with both a packet-switched (PS) network and a cellular circuit-switched (CS) network, in which a dual-mode handset is connected via the cellular CS network to communicate with another handset;

detecting a disconnection between the dual-mode handset and the cellular CS network;

responsively to the disconnection, directing a request from the dual-mode handset to the VCC server to hand over the call for connection to the dual-mode handset via an access point of the PS network; and

responsively to the request and prior to tearing down the call, recovering the call by connecting the dual-mode handset to communicate with the other handset via the access point of the PS network.

In a disclosed embodiment, detecting the disconnection includes receiving a notification of the disconnection at the VCC server, and recovering the call includes connecting the dual-mode handset to continue the call when the VCC server receives the request within a predetermined time limit following the notification. Alternatively, the call is torn down if the request is not received within the predetermined time limit. Receiving the notification may include evaluating, responsively to the notification, a tear-down policy applicable to the dual-mode handset, and deciding whether to wait for the time limit before tearing down the call responsively to the tear-down policy.

There is further provided, in accordance with an embodiment of the present invention, a method for communication, including:

storing a record on a Voice Call Continuity (VCC) server, which is connected to communicate with both a packet-switched (PS) network and a cellular circuit-switched (CS) network, associating first and second handsets with a subscriber identity of a given subscriber;

receiving at the VCC server a request to place a voice call to the subscriber, such that the voice call will have at least one leg in the PS network;

responsively to the request, sending signals from the VCC server to both of the first and second handsets so as to cause both of the handsets to ring; and

when the call is picked up by the subscriber using one of the first and second handsets, connecting the one of the handsets to complete the call.

Typically, the method includes, upon connecting the one of the handsets, disconnecting the other of the handsets from the call. Additionally or alternatively, sending the signals includes evaluating the record at the VCC server in order to ascertain that the given subscriber is subject to a dual-ringing policy, and to identify first and second network domains through which to send the signals to the first and second handsets, respectively.

There is moreover provided, in accordance with an embodiment of the present invention, a method for communication, including:

receiving at a Voice Call Continuity (VCC) server, which is connected to communicate with both a packet-switched (PS) network and a cellular circuit-switched (CS) network, a request to establish a call to a given telephone number;

evaluating the request at the VCC server so as to make a determination that the given telephone number is assigned to a private branch exchange (PBX) operated by a Centrex server on the PS network; and responsively to the determination, routing the call to the given telephone number via the Centrex server.

Typically, the method includes registering the given telephone number in a record maintained by the VCC server, wherein the record associates the given telephone number with the PBX, and wherein evaluating the request includes determining responsively to the record that the given telephone number is assigned to the PBX.

There is furthermore provided, in accordance with an embodiment of the present invention, apparatus for communication, including:

a service processor, which is configured to provide a Voice Call Continuity (VCC) service between a packet-switched (PS) network and a cellular circuit-switched (CS) network; and

interfaces for connecting the service processor to communicate with both the PS and the CS networks,

wherein the service processor is configured to carry out the methods described above.

The present invention will be more fully understood from the following detailed description of the embodiments thereof, taken together with the drawings in which:

BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 is a block diagram that schematically illustrates a communication system with an enhanced VCC server, in accordance with an embodiment of the present invention;

Fig. 2 is a block diagram showing functional components of an enhanced VCC server, in accordance with an embodiment of the present invention;

Figs. 3-8 are message flow diagrams that schematically illustrate functions performed by an enhanced VCC server, in accordance with embodiments of the present invention;

Fig. 9 is a block diagram that schematically illustrates a communication system with an enhanced VCC server, in accordance with an alternative embodiment of the present invention; and

Fig. 10 is a message flow diagram that schematically illustrates functions performed by an enhanced VCC server, in accordance with another embodiment of the present invention.

DETAILED DESCRIPTION OF EMBODIMENTS

SYSTEM OVERVIEW

Fig. 1 is a block diagram that schematically illustrates a communication system 20, in accordance with an embodiment of the present invention. System 20 comprises a circuit-switched (CS) telephone network 22 and an IP-based packet-switched (PS) network 24. In the present example, CS network 22 is assumed to be a cellular network, while PS network 24 is an IMS network. The principles of the present invention, however, may similarly be applied in providing Voice Call Continuity (VCC) services involving CS and PS networks of other kinds, such as a public switched telephone network (PSTN) and non-IMS PS networks.

Networks 22 and 24 may serve various kinds of handsets, including mobile handsets 26, 28 and a fixed IP phone handset 30. Typically, for purposes of VCC functionality, one or more of these handsets have dual-mode capability and are thus able to communicate through both CS network 22 and PS network 24. On the other hand, some of the enhanced services that are described hereinbelow may also be applied using single-mode handsets as at least one of the endpoints of the call. "Handsets" in this context may also include personal computers with soft phone capabilities, including proprietary soft-phone devices of the type described in PCT International Publication WO 2007/093989.

CS network 22 in this example is a conventional cellular telephone network, which is built around a mobile switching center (MSC) 32, as is known in the art. Mobile subscriber equipment, such as handset 28, communicates with network 22 via a base transceiver station (BTS) 34. A home location register (HLR) 36 stores subscriber information, including account status and user preferences.

PS network 24 is built around a CSCF 40, which may be embodied in one or more servers, as is known in the art. A HSS server 46 provides subscriber information in the conventional manner. An Enhanced Voice Call Continuity (EVCC) server 42 serves as a signaling gateway between networks 22 and 24 for purposes of VCC and other network convergence functions. A media gateway (MGW) 44 converts and transmits voice and other media traffic between the networks. EVCC server 42 has standard interfaces to the network components necessary for VCC functionality, including an IMS Service Control (ISC) interface 48 to CSCF 40; a SIP interface 50 to MGW 44; an application service interface 52 to MSC 32, such as an Intelligent Network Application Part (INAP) or Customized Applications for Mobile Enhanced Logic (CAMEL) interface; and a Mobile Application Part (MAP) interface 54 to HLR 36.

Fig. 2 is a block diagram that schematically shows functional details of EVCC server 42, in accordance with an embodiment of the present invention. Server 42 comprises a service processor 60, which is connected by appropriate interfaces to other components of system 20, as shown in Fig. 1 and detailed further in Fig. 2. Processor 60 typically comprises a general-purpose computer processor, which has the required high-speed input/output (I/O) connections and is programmed in software to carry out the functions that are described herein. Key functional blocks of this software are shown within the processor block in Fig. 2 and are described further hereinbelow. The software may be downloaded to processor 60 in electronic form, over a network, for example, or it may, alternatively or additionally, be provided on tangible storage media, such as optical, magnetic or electronic memory media. Further alternatively or additionally, at least some of the functions of processor 60 may be performed by dedicated or programmable hardware logic components.

For purposes of voice call continuity between CS network 22 and PS network 24, processor 60 performs a CS adaptation function 62, which interacts with the interrogating CSCF (I-CSCF) function 64 of CSCF 40 so as to serve as the proxy in the PS network for call legs in the CS network. A domain selection function (DSF) 66 and a domain transfer function (DTF) 68 are responsible for choosing the access network (CS or PS) with which a given handset is to communicate and for handing over the handset from one network to the other as needed. These functions operate in conjunction with HSS 46 and the Serving CSCF (S-CSCF) function 70 of CSCF 40. CS adaptation, domain selection and domain transfer are standard functions of a VCC server, as defined by the above-mentioned 3GPP specifications, and their basic implementations will be apparent to those skilled in the art. These functional components of server 42 exchange messages with CSCF 40 and HSS 46 via standard ISC, Ma and Sh interfaces, which are likewise defined by the 3GPP specifications. Embodiments of the present invention, however, add novel functionality to DSF 66 and DTF 68, beyond the standard, basic implementations, as is described in detail hereinbelow.

In addition to these IMS-related functions, processor 60 comprises a CS service component 72 for anchoring calls between the PS and CS networks. Component 72 comprises a service control function (SCF), which interacts with a service switching function (SSF) 74 of

MSC 32. The interface between component 72 and SSF 74 is typically a standard Intelligent Network (IN) service interface, such as a CAMEL interface, as defined for GSM networks. Alternatively or additionally, the interface may be based on INAP or any other suitable service protocol.

In the context of the enhanced VCC services provided by server 42, processor 60 performs a number of additional functions that are not part of the conventional VCC specifications. They include an identity management function (IMF) 78, which provides logic for handling multiple identities (i.e., multiple handsets or multiple telephone numbers) for the same subscriber. An enhanced call logic (ECL) function 80, implements call setup and tear-down in ways that provide network operators and subscribers with additional options and flexibility. An optional registrar function 76 permits server 42 to operate in non-IMS environments, as well. These functions are described with reference to the message flow diagrams in the figures that follow.

ADDITIONAL NUMBER

IMF 78 is capable of managing multiple telephone numbers for any given subscriber. Thus, the network operator may, for example, assign two numbers to a single dual-mode handset, one for use in accessing the PS network and the other for cellular CS networks. The IMF may then be programmed, for example, to restrict the subscriber's use of the PS number to a certain predefined service zone and to use whatever cellular network is available outside this zone.

Fig. 3 is a message flow diagram that schematically illustrates the operation of IMF 78 in call origination, in accordance with an embodiment of the present invention. In this scenario, an originating endpoint (EP) 90 places a call to a terminating endpoint 92 by sending a SIP INVITE message to CSCF 40. The message specifies an identifier (such as a telephone number) of the called party as the call destination. The CSCF passes an invite message to DTF 68 of EVCC server 42, which processes the call parameters at an initial session anchoring step 94.

DTF 68 sends the call origination and destination information, and possibly other call parameters, to IMF 78, which looks up the identities of the calling and called parties, at an identity determination step 96. If the calling party has two numbers, as in the example mentioned above, EVCC 42 may use this information in setting up call functions such as billing, caller ID and other applicable services. If the called party has two numbers, the IMF will instruct the DTF to refer the call to either the PS number or to the cellular number. (Call termination is described further hereinbelow with reference to Figs. 7 and 8.) In either case, the IMF conveys the appropriate instruction back to the DTF.

Based on the instruction from IMF 78, DTF 68 decides how the call should be routed to the terminating endpoint, at a final session anchoring step 98. In the example shown in Fig. 3, IMF 78 has determined that the terminating endpoint is located in the PS service zone, and the DTF therefore completes the call by sending an appropriate INVITE message directed to the PS number of the handset in question. Otherwise, the DTF passes the call through CS service component 72 to the cellular number of the handset, for completion through the cellular network in which the handset is located.

The call flow described above may apply to any call placed by or to a subscriber having access to both the CS and PS networks. A similar flow will pertain when a call is placed through CS network 22 (whereupon an IN trigger in MSC 32 will cause the call to be referred to the EVCC server), except that in this case, CS service component 72 will refer the call to DTF 68.

An additional number may be assigned to a handset, for example, by a fixed network operator in order to offer mobile PS services to subscribers without dependence on or cooperation from any cellular network operator. In this scenario, a subscriber having a dual-mode handset with a cellular telephone number in any cellular network may sign up to receive PS network service from the fixed network operator, who assigns an additional fixed telephone number to the subscriber. Then, whenever the subscriber is registered by CSCF 40 in the zone of service of the fixed network operator (via a WiFi or WiMAX access point authorized by the fixed network operator, for example), EVCC server 42 will enable calls to and from the subscriber to be routed via the PS network using the additional number. Otherwise, the calls will be routed through the cellular network to the original cellular number. PS calls in progress will likewise be handed over to the cellular number when PS network service is not available. Alternatively, more complex service rules may be defined, particularly if there is a cooperative relationship between the fixed and cellular network operators.

CALL TAKEOVER

Mobile telephone users frequently have multiple handsets, such as a desktop handset or automobile handset, in addition to their mobile (dual-mode) handset. At times a user in the midst of a call on PS network 24 may wish to have another one of these other handsets take over the call. For example, the user arriving at his or her desk may wish to transfer the call to the desktop handset; or the user in the midst of a call while leaving the office may wish to transfer the call from the desktop handset to a mobile handset or from the mobile handset to the automobile handset. In an embodiment of the present invention, IMF 78 of EVCC server 42 is programmed to receive and transfer a call from a first handset to a second handset, in response to a takeover request from the second handset.

Fig. 4 is a message flow diagram that schematically illustrates this aspect of the operation of EVCC server 42, in accordance with an embodiment of the present invention. Initially, a call is set up between an originating endpoint 100 and terminating endpoint 92. (This call is set up in the conventional manner and is represented in Fig. 4 by a bar 104.) The user of endpoint 100 decides during the call that the call should be taken over by another endpoint 102. For this purpose, the user inputs a special code via endpoint 102, which causes this endpoint to send a new SIP INVITE message to CSCF 40. The CSCF passes the INVITE message to DTF 68 in the normal manner, just as it would pass an ordinary INVITE message to initiate a new call.

DTF 68 begins processing of the new INVITE message, at an initial handover detection step 106. As in the embodiment of Fig. 3, the DTF passes the message parameters, including the identity of endpoint 102, to IMF 78. The IMF recognizes that the message is actually a takeover request and evaluates the identity record of endpoint 102, at a takeover identity management step 108. At this step, in the present example, the IMF determines that endpoint 102 is paired with endpoint 100 and is authorized to take over calls from endpoint 100.

IMF 78 returns control to DTF 68 with instructions to hand over the call from endpoint 100 to endpoint 102. The DTF completes preparations for call takeover, at a final takeover execution step 110. It then sends a REINVITE message via the CSCF to terminating endpoint 92, which causes the terminating endpoint to continue communicating with endpoint 102 in place of endpoint 100.

A similar procedure may be invoked by the user of the terminating endpoint in order to transfer the terminating leg of the call to another handset.

AUTOMATIC CALL RECOVERY

Reference is now made to Figs. 5 and 6, which are message flow diagrams that schematically illustrate a method for automatic call recovery, in accordance with an embodiment of the present invention. This method may be used to handle cases of sudden disconnection of a call, such as may occur when the cellular signal is lost on a leg of a call carried through CS network 22. It is assumed in the example shown in Figs. 5 and 6 that the signal is lost by originating endpoint 90, which is a dual-mode handset. A similar procedure may be used by a suitably-configured terminating endpoint.

When originating endpoint 90 detects that it has been disconnected from the present call on the cellular network, it initiates teardown of the call anchor on CSCF 40, as well, by sending a SIP BYE message. The CSCF passes the BYE message to DTF 68 of EVCC server 42, which receives and processes the message, at a call release detection step 120. The DTF refers the teardown request to ECL 80 in order to determine the appropriate teardown policy for the subscriber in question, at a teardown policy determination step 122. If the teardown policy indicates that this subscriber has automatic call recovery capability, the DTF initiates a delayed call release, at a release delay step 124. The meaning of the delayed call release is that rather than completing teardown of the call immediately, EVCC server 42 will give the user a certain predetermined time in which to hand over the call from CS network 22 to PS network 24.

In the meanwhile, upon detecting that originating endpoint 90 has been disconnected from the cellular network, a client program running on the originating endpoint checks whether the endpoint has a current connection to an access point of PS network 24. If so, endpoint 90 sends a SIP INVITE message over this connection, as illustrated in Fig. 5, asking for the call to be handed over to the PS network. DTF 68 detects the INVITE message, at an initial handover detection step 126. If the DTF receives this message before the timer that was set at step 124 expires, it passes the message parameters to ECL 80, which undertakes a call recovery step 128. Based on the policy applicable to this subscriber, the ECL may instruct the DTF to carry out a handover of the call to PS network 24, at a handover recovery step 130. In this case, DTF 68 passes a REINVITE message via CSCF 40 to terminating endpoint 92, which then continues the call with the originating endpoint via the PS network.

On the other hand, as shown in Fig. 6, if DTF 68 does not receive the handover request from endpoint 90 before the timer expires, the DTF tears down the call, at a call release step 132. Call release proceeds in the conventional way, by transmission of a SIP BYE message via CSCF 40 to terminating endpoint 92.

CALL TERMINATION WITH DUAL RINGING OPTION

Reference is now made to Figs. 7 and 8, which are message flow diagrams that schematically illustrate methods for call termination, in accordance with an embodiment of the present invention. In particular, these figures illustrate the operation of ECL 80 and IMF 78 in EVCC server 42 in determining how calls should be routed to the appropriate terminating endpoint or endpoints.

In Fig. 7, originating endpoint 90 initiates a call to a subscriber of the services provided by EVCC server 42 by sending the appropriate SIP INVITE message to CSCF 40, which then passes the message to DTF 68. The DTF begins processing of the call, at an initial session termination anchoring step 140, and passes the relevant call parameters, including the call destination, to IMF 78. The IMF looks up the identity of the subscriber who is to receive the call, at a managed identity determination step 142. The service properties in the identity record may indicate, for example, that the subscriber has an additional fixed number (as explained above in reference to Fig. 3), or may be subject to enhanced call logic in setting up the call, or has other service preferences that are applicable to the call.

IMF 78 passes the relevant service properties to DTF 68, which applies the properties in a final session termination anchoring step 144. In this example, based on these properties, DTF 68 passes control of the call to ECL 80, which determines the applicable ringing policy, at a policy determination step 145. If the destination subscriber has a dual-mode handset or other dual-mode capability, ECL 80 passes control to DSF 66, at a domain selection step 146. The DSF checks the current access connections (CS and/or PS) of the receiving handset and thus decides whether to route the call to terminating endpoint 92 through network 22 or through network 24. In the example shown in Fig. 7, the call is routed through PS network 24 by sending a SIP INVITE message via CSCF 40 to the terminating endpoint.

Fig. 8 shows a scenario similar to that of Fig. 7, except that in this case, the subscriber to whom the call is directed has two handsets, identified in the figure as terminating endpoints 150 and 152, and the identity record accessed by IMF 78 at step 142 indicates that the call is subject to enhanced call logic. Therefore, at step 144, DTF 68 refers the call to ECL 80 to determine how the call should be routed to the terminating endpoints, at a policy determination step 154. ECL 80 instructs DSF 66 to open two terminating legs for the call, one for each terminating endpoint. In response, the DSF chooses a first domain (PS or CS) for placing the call to endpoint 150, at a first domain selection step 156, and a second domain for placing the call to endpoint 152, at a second domain selection step.

DSF 66 sends SIP INVITE messages via CSCF 40 to both endpoints. If one (or both) of endpoints 150 and 152 is in the CS domain, the DSF may initiate a call via CS network 22 to one (or both) of the terminating endpoints. In any case, when the user receiving the call picks up one of the handsets, the call is completed, and the other, unused handset is typically disconnected from the call.

NON-IMS SYSTEM CONFIGURATIONS

Fig. 9 is a block diagram that schematically illustrates a communication system 160, in accordance with an alternative embodiment of the present invention. In this embodiment, EVCC server 42 is configured to operate as a gateway between CS network 22 and a PS network 161, which operates in accordance with the Internet Protocol but does not have IMS capabilities. In this scenario, there is no CSCF available to handle registration of handsets in the PS domain and to intervene between transport and application layers. Therefore, registrar 76 (Fig. 2) of the EVCC server handles handset registration, and the functional components of the EVCC server interact directly with the relevant application servers in network 161. Otherwise, the internal architecture of the EVCC server is similar to that shown above in Fig. 2.

One type of application server with which EVCC server 42 may interact in network 161 is an IP Centrex server 162. This type of sever can be used, for example, to provide private branch exchange (PBX) functionality to users in an organization (such as a business enterprise) that extends over both IP and CS telephones, including mobile users of cellular and dual-mode handsets. Server 162 may be deployed by such an organization or by a network service provider, who may then offer Centrex services to organizations and other customer groups. One server of this sort is the BroadWorks® Mobile PBX, which is sold by BroadSoft® Inc. (Gaithersburg, Maryland). Another is described in PCT International Publication WO 2005/084128.

Fig. 10 is a message flow diagram that schematically illustrates a method for interaction between EVCC server 42 and IP Centrex server 162, in accordance with an embodiment of the present invention. In this scenario, an originating endpoint 170 places a call to the telephone number of a terminating endpoint 172, which is part of the PBX served by server 162. The originating endpoint sends a SIP INVITE message directly to EVCC server 42, and DTF 68 extracts the relevant message parameters at an initial session anchoring step 174. The DTF passes these parameters to IMF 78, which looks up the call destination parameters and determines that the identity of the call destination belongs to a PBX, at an identity determination step 176.

DTF 68 passes the identity properties to DTF 68, for completion of call establishment at a final session anchoring step 178. To find out how the call should be routed to the terminating endpoint, the DTF transfers control of the call to ECL 80, which determines that the call should be routed through server 162, at an application server identification step 180. ECL 80 passes this information back to DTF 178, which then routes the call accordingly by sending a SIP INVITE message to server 162. The server uses its own logic to pass the call to terminating endpoint 172 via the appropriate network.

Although the operation of EVCC server 42 in a non-IMS network is described hereinabove only with reference to one specific type of application server, a similar procedure may be used in providing voice call continuity for other types of application servers with which the EVCC server may interact. Furthermore, the novel EVCC features that are described above for the IMS environment with reference to Figs. 3-8 may similarly be provided by the EVCC server, *mutatis mutandis*, in non-IMS environments, such as in system 160 (Fig. 9).

It will thus be appreciated that the embodiments described above are cited by way of example, and that the present invention is not limited to what has been particularly shown and described hereinabove. Rather, the scope of the present invention includes both combinations and subcombinations of the various features described hereinabove, as well as variations and modifications thereof which would occur to persons skilled in the art upon reading the foregoing description and which are not disclosed in the prior art.

CLAIMS

1. A method for communication, comprising:

registering in a packet-switched (PS) network a dual-mode handset, which is capable of voice communications over both the PS network and over a cellular circuit-switched (CS) network, and which is assigned a first telephone number in the cellular CS network;

assigning to the dual-mode handset a second telephone number, different from the first telephone number, for use in placing voice calls over the PS network; and

establishing a call, using the second telephone number, between the dual-mode handset and a telephone in the cellular CS network via a Voice Call Continuity (VCC) server, which is connected to communicate with both the PS and the CS networks.

2. The method according to claim 1, wherein assigning the second telephone number comprises defining a service zone within the PS network in which the second telephone number is permitted to be used, and wherein establishing the call comprises connecting a leg of the call to the dual-mode handset via an access point of the PS network that is within the service zone.

3. The method according to claim 2, wherein establishing the call comprises connecting the leg of the call to the dual-mode handset via the cellular CS network using the first telephone number when the dual-mode handset is outside the service zone.

4. A method for communication, comprising:

establishing a voice call between first and second handsets using a Voice Call Continuity (VCC) server, which is connected to communicate with both a packet-switched (PS) network and a circuit-switched (CS) network, the voice call having one leg in the PS network and another leg in the CS network;

receiving at the VCC server a request from a third handset to take over the call from the first handset; and

responsively to the request, connecting a new leg to the call, using the VCC server, so as to enable the third handset to communicate with the second handset.

5. The method according to claim 4, and comprising disconnecting the first handset from the call upon connecting the new leg.

6. The method according to claim 4, wherein connecting the new leg comprises, at the VCC server, evaluating an identity associated with the third handset and determining, responsively to the identity that the third handset is authorized to take over the call from the first handset.

7. A method for communication, comprising:

establishing a voice call using a Voice Call Continuity (VCC) server, which is connected to communicate with both a packet-switched (PS) network and a cellular circuit-switched (CS) network, in which a dual-mode handset is connected via the cellular CS network to communicate with another handset;

detecting a disconnection between the dual-mode handset and the cellular CS network;

responsively to the disconnection, directing a request from the dual-mode handset to the VCC server to hand over the call for connection to the dual-mode handset via an access point of the PS network; and

responsively to the request and prior to tearing down the call, recovering the call by connecting the dual-mode handset to communicate with the other handset via the access point of the PS network.

8. The method according to claim 7, wherein detecting the disconnection comprises receiving a notification of the disconnection at the VCC server, and wherein recovering the call comprises connecting the dual-mode handset to continue the call when the VCC server receives the request within a predetermined time limit following the notification.

9. The method according to claim 8, and comprising tearing down the call if the request is not received within the predetermined time limit.

10. The method according to claim 8, wherein receiving the notification comprises evaluating, responsively to the notification, a tear-down policy applicable to the dual-mode handset, and deciding whether to wait for the time limit before tearing down the call responsively to the tear-down policy.

11. A method for communication, comprising:

storing a record on a Voice Call Continuity (VCC) server, which is connected to communicate with both a packet-switched (PS) network and a cellular circuit-switched (CS) network, associating first and second handsets with a subscriber identity of a given subscriber;

receiving at the VCC server a request to place a voice call to the subscriber, such that the voice call will have at least one leg in the PS network;

responsively to the request, sending signals from the VCC server to both of the first and second handsets so as to cause both of the handsets to ring; and

when the call is picked up by the subscriber using one of the first and second handsets, connecting the one of the handsets to complete the call.

12. The method according to claim 11, and comprising, upon connecting the one of the handsets, disconnecting the other of the handsets from the call.

13. The method according to claim 11, wherein sending the signals comprises evaluating the record at the VCC server in order to ascertain that the given subscriber is subject to a dual-ringing policy, and to identify first and second network domains through which to send the signals to the first and second handsets, respectively.

14. A method for communication, comprising:

receiving at a Voice Call Continuity (VCC) server, which is connected to communicate with both a packet-switched (PS) network and a cellular circuit-switched (CS) network, a request to establish a call to a given telephone number;

evaluating the request at the VCC server so as to make a determination that the given telephone number is assigned to a private branch exchange (PBX) operated by a Centrex server on the PS network; and

responsively to the determination, routing the call to the given telephone number via the Centrex server.

15. The method according to claim 14, and comprising registering the given telephone number in a record maintained by the VCC server, wherein the record associates the given telephone number with the PBX, and wherein evaluating the request comprises determining responsively to the record that the given telephone number is assigned to the PBX.

16. Apparatus for communication, comprising:

a service processor, which is configured to provide a Voice Call Continuity (VCC) service between a packet-switched (PS) network and a cellular circuit-switched (CS) network; and interfaces for connecting the service processor to communicate with both the PS and the CS networks,

wherein the service processor is configured to record an assignment to a dual-mode handset, which is capable of voice communications over both the PS network and the CS network, of a first telephone number in the cellular CS network and of a second telephone number, different from the first telephone number, for use in placing voice calls over the PS network, and to establish a call, using the second telephone number, between the dual-mode handset and a telephone in the cellular CS network.

17. The apparatus according to claim 16, wherein the service processor is configured to accept a definition of a service zone within the PS network in which the second telephone number is permitted to be used, and to cause a leg of the call to the dual-mode handset to be established via an access point of the PS network that is within the service zone.

18. The apparatus according to claim 17, wherein the service processor is configured to cause the leg of the call to the dual-mode handset to be established via the cellular CS network using the first telephone number when the dual-mode handset is outside the service zone.

19. Apparatus for communication, comprising:

a service processor, which is configured to provide a Voice Call Continuity (VCC) service between a packet-switched (PS) network and a cellular circuit-switched (CS) network; and

interfaces for connecting the service processor to communicate with both the PS and the CS networks,

wherein the service processor is configured to establish a voice call between first and second handsets, the voice call having one leg in the PS network and another leg in the CS network, and to receive a request from a third handset to take over the call from the first handset, and to cause a new leg to be connected to the call, responsively to the request, so as to enable the third handset to communicate with the second handset.

20. The apparatus according to claim 19, wherein the service processor is configured to cause the first handset to be disconnected from the call upon connecting the new leg.

21. The apparatus according to claim 19, wherein the service processor is configured, upon receiving the request, to evaluate an identity associated with the third handset so as to

determine, responsively to the identity, that the third handset is authorized to take over the call from the first handset before causing the new leg to be connected.

22. Apparatus for communication, comprising:

a service processor, which is configured to provide a Voice Call Continuity (VCC) service between a packet-switched (PS) network and a cellular circuit-switched (CS) network; and

interfaces for connecting the service processor to communicate with both the PS and the CS networks,

wherein the service processor is configured to detect, in a voice call in which a dual-mode handset is connected via the cellular CS network to communicate with another handset, a disconnection between the dual-mode handset and the cellular CS network, and to accept, responsively to the disconnection, a request from the dual-mode handset to the VCC server to hand over the call for connection to the dual-mode handset via an access point of the PS network, and to recover the call, responsively to the request and prior to tearing down the call, by connecting the dual-mode handset to communicate with the other handset via the access point of the PS network.

23. The apparatus according to claim 22, wherein the service processor is configured to receive a notification of the disconnection, and to connect the dual-mode handset to continue the call when the request is received at the service processor within a predetermined time limit following the notification.

24. The apparatus according to claim 23, wherein the service processor is configured to tear down the call if the request is not received within the predetermined time limit.

25. The apparatus according to claim 23, wherein the service processor is configured to evaluate, responsively to the notification, a tear-down policy applicable to the dual-mode handset, and to decide whether to wait for the time limit before tearing down the call responsively to the tear-down policy.

26. Apparatus for communication, comprising:

a service processor, which is configured to provide a Voice Call Continuity (VCC) service between a packet-switched (PS) network and a cellular circuit-switched (CS) network; and interfaces for connecting the service processor to communicate with both the PS and the CS networks,

wherein the service processor is configured to store a record associating first and second handsets with a subscriber identity of a given subscriber, and upon receiving a request to place a voice call to the subscriber, such that the voice call will have at least one leg in the PS network, to send signals to both of the first and second handsets so as to cause both of the handsets to ring, and to connect one of the handsets to complete the call when the call is picked up by the subscriber using the one of the handsets.

27. The apparatus according to claim 26, wherein the service processor is configured to disconnect the other of the handsets from the call upon connecting the one of the handsets.

28. The apparatus according to claim 26, wherein the service processor is configured to evaluate the record in order to ascertain that the given subscriber is subject to a dual-ringing policy, and to identify first and second network domains through which to send the signals to the first and second handsets, respectively.

29. Apparatus for communication, comprising:

a service processor, which is configured to provide a Voice Call Continuity (VCC) service between a packet-switched (PS) network and a cellular circuit-switched (CS) network; and

interfaces for connecting the service processor to communicate with both the PS and the CS networks,

wherein the service processor is configured to receive a request to establish a call to a given telephone number, to evaluate the request so as to make a determination that the given telephone number is assigned to a private branch exchange (PBX) operated by a Centrex server on the PS network, and responsively to the determination, to route the call to the given telephone number via the Centrex server.

30. The apparatus according to claim 29, wherein the service processor is configured to register the given telephone number in a record maintained by the service processor, wherein the record associates the given telephone number with the PBX, and to determine responsively to the record that the given telephone number is assigned to the PBX.

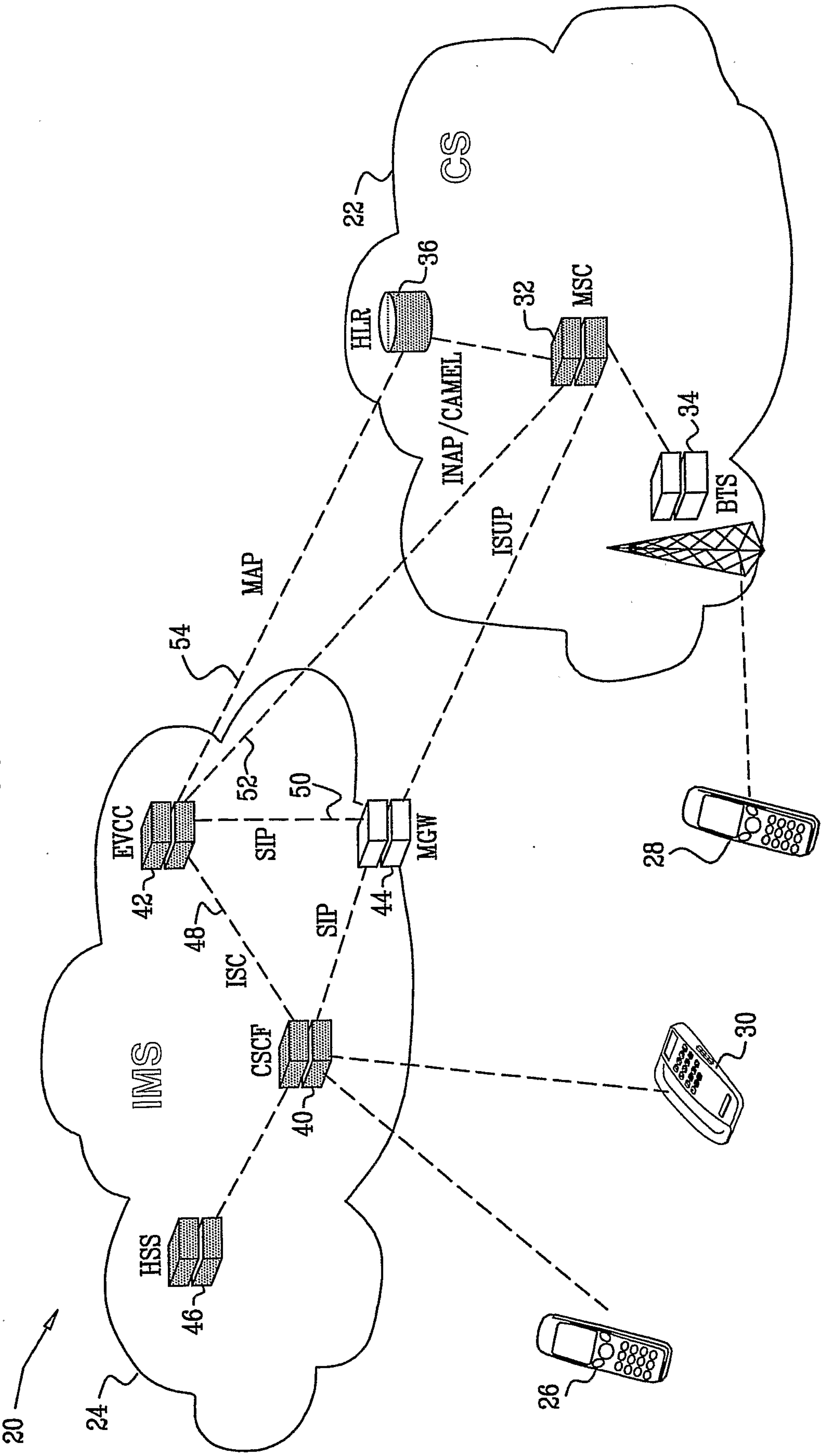

CS
22
HLR
36
MSC
32
INAP/CAMEL
34
BTS
MAP
ISUP
54
52
50
EVCC
SIP
MGW
42
44
48
ISC
SIP
IMS
CSCF
40
30
28
HSS
46
26
24
20

FIG. 1

FIG. 2

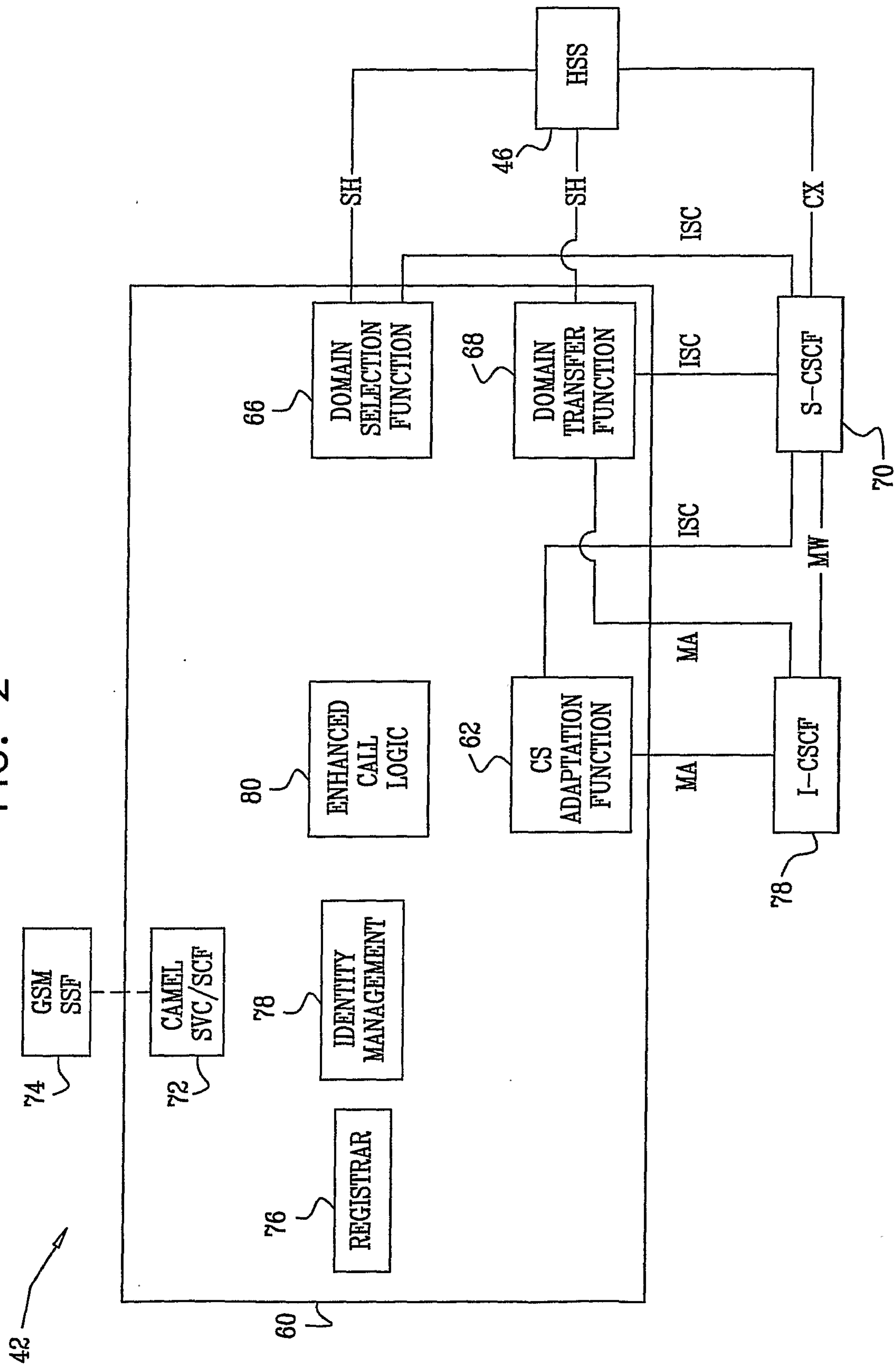
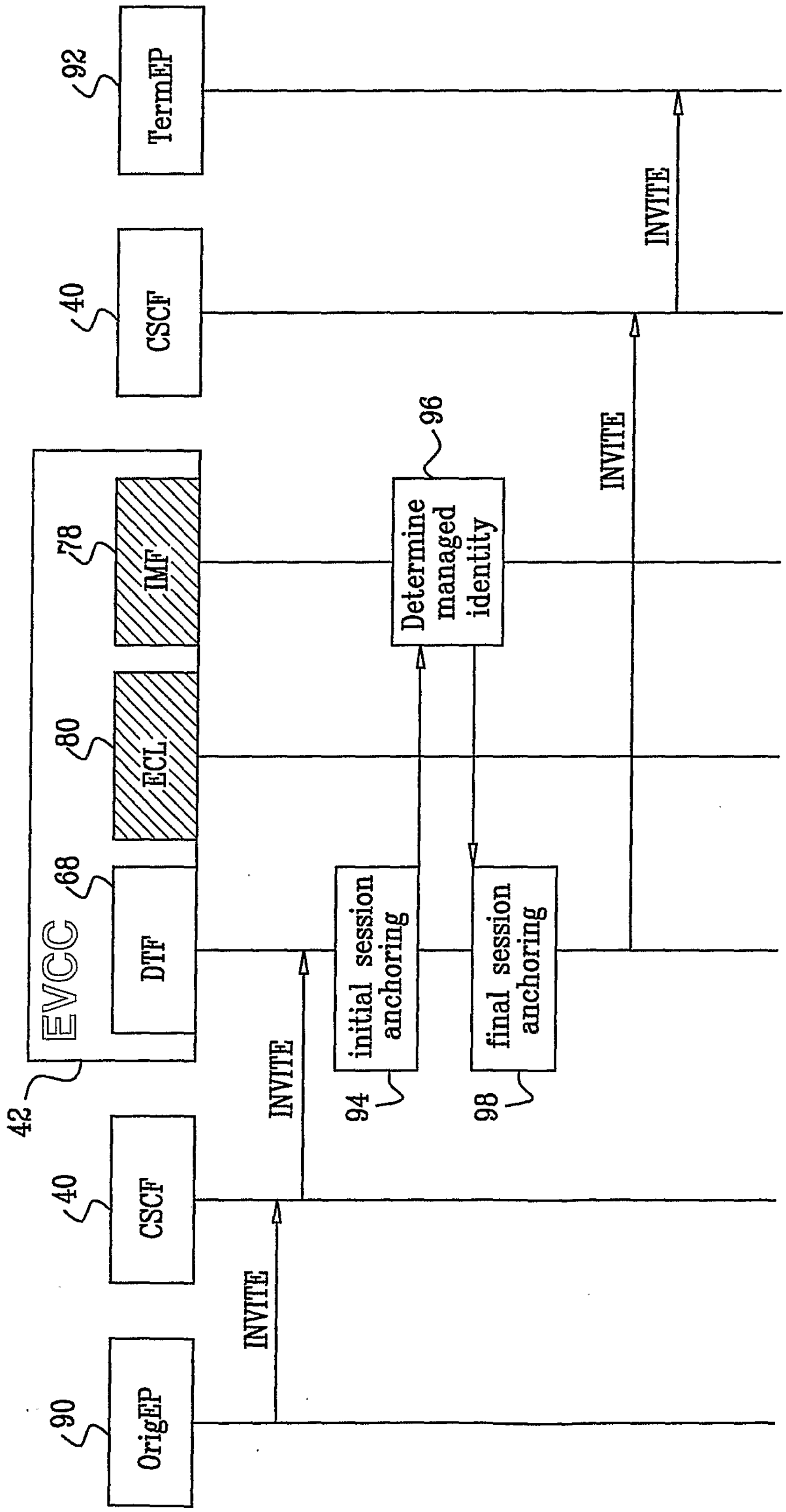

HSS
46
SH
SH
CX
ISC
ISC
ISC
MW
MA
MA
DOMAIN SELECTION FUNCTION
66
DOMAIN TRANSFER FUNCTION
68
S-CSCF
70
ENHANCED CALL LOGIC
80
CS ADAPTATION FUNCTION
62
I-CSCF
78
GSM SSF
74
CAMEL SVC/SCF
72
IDENTITY MANAGEMENT
78
REGISTRAR
76
60
42

FIG. 3


OrigEP
90
CSCF
40
EVCC
42
DTF
68
ECL
80
IMF
78
CSCF
40
TermEP
92
INVITE
INVITE
initial session anchoring
94
Determine managed identity
96
final session anchoring
98
INVITE
INVITE

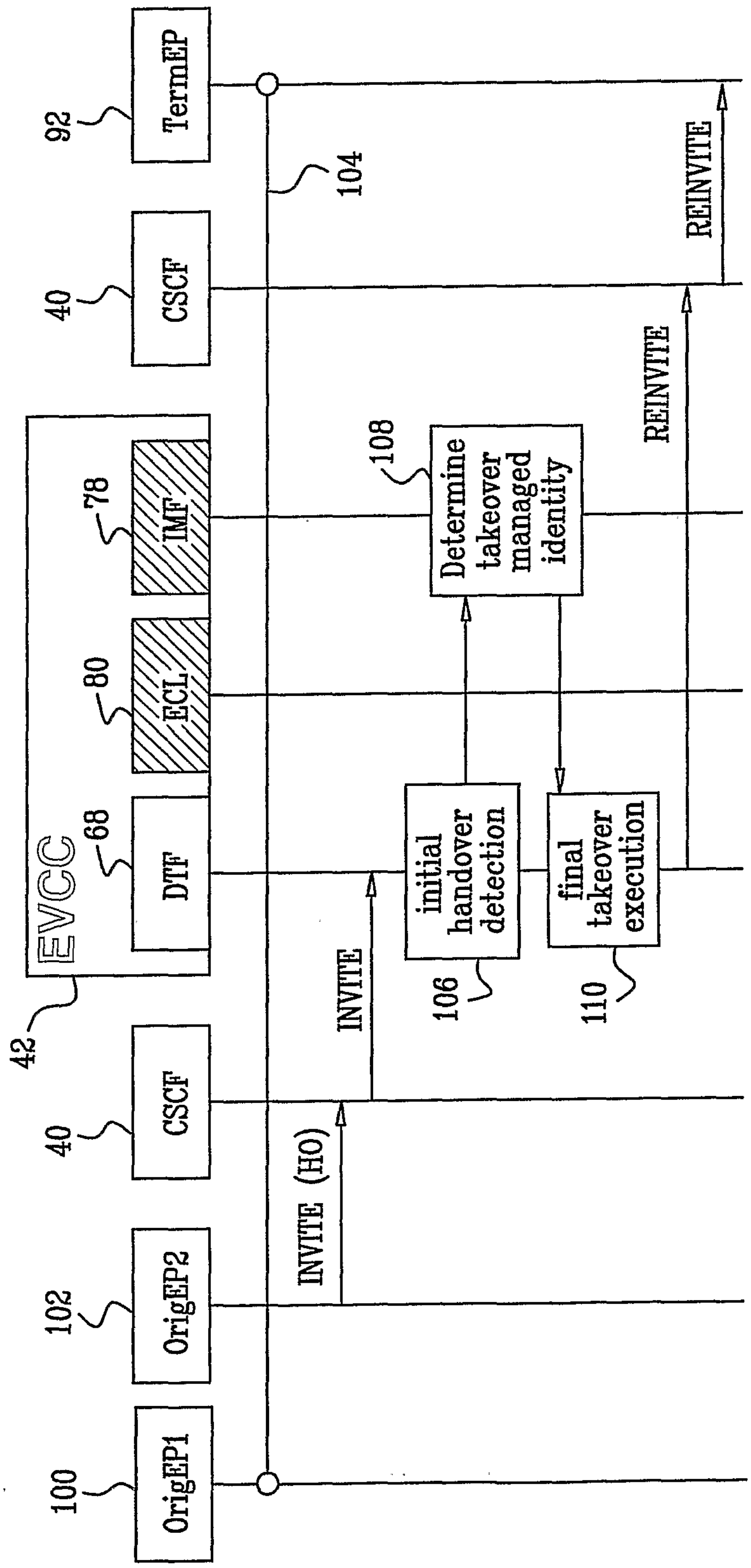

OrigEP1
100
OrigEP2
102
CSCF
40
EVCC
42
DTF
68
ECL
80
IMF
78
CSCF
40
TermEP
92
104
INVITE (HO)
INVITE
initial handover detection
106
Determine takeover managed identity
108
final takeover execution
110
REINVITE
REINVITE

FIG. 4

FIG. 5

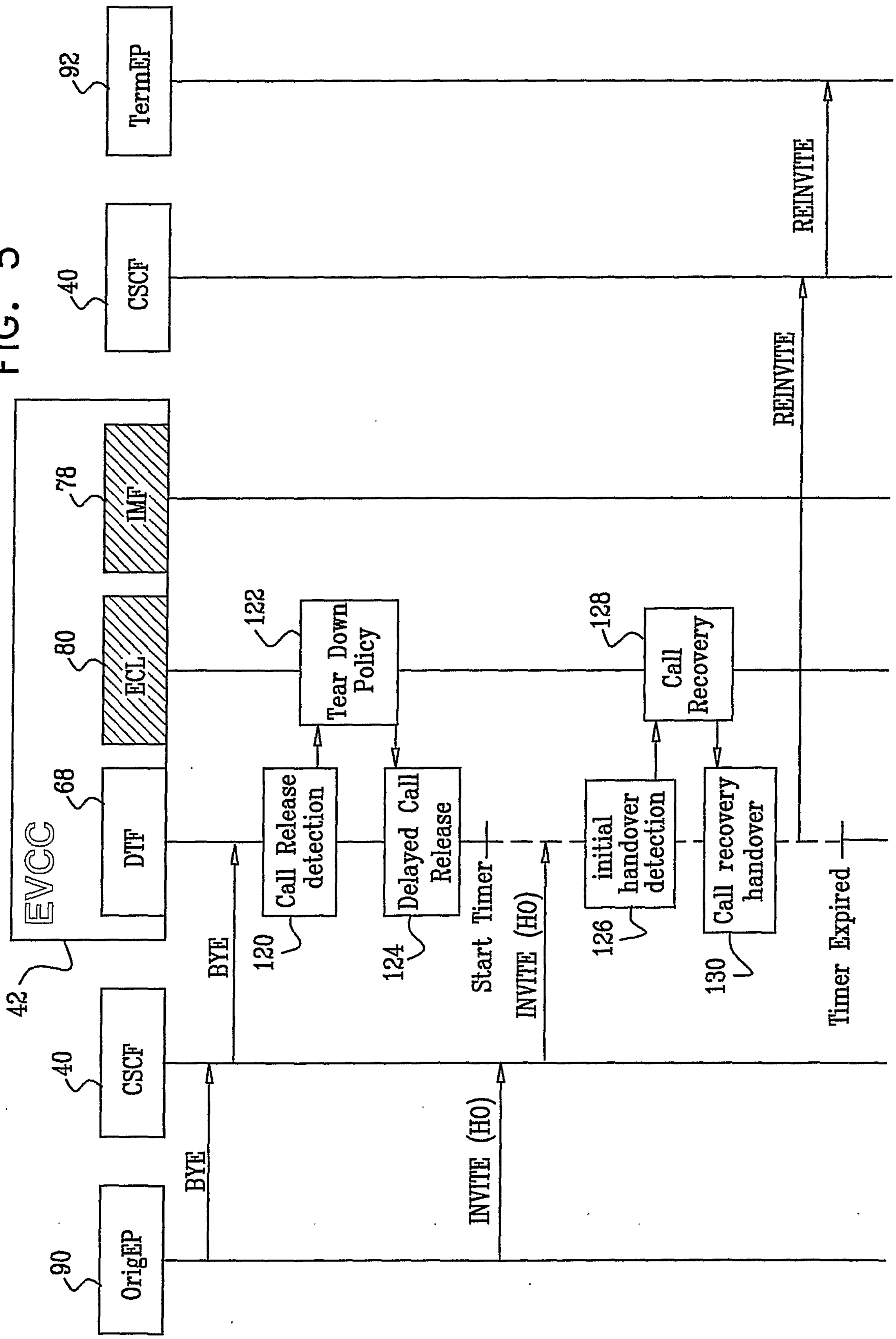

90
OrigEP
40
CSCF
42
EVCC
68
DTF
80
ECL
78
IMF
40
CSCF
92
TermEP
BYE
BYE
120
Call Release detection
122
Tear Down Policy
124
Delayed Call Release
Start Timer
INVITE (HO)
INVITE (HO)
126
initial handover detection
128
Call Recovery
130
Call recovery handover
REINVITE
REINVITE
Timer Expired

FIG. 6

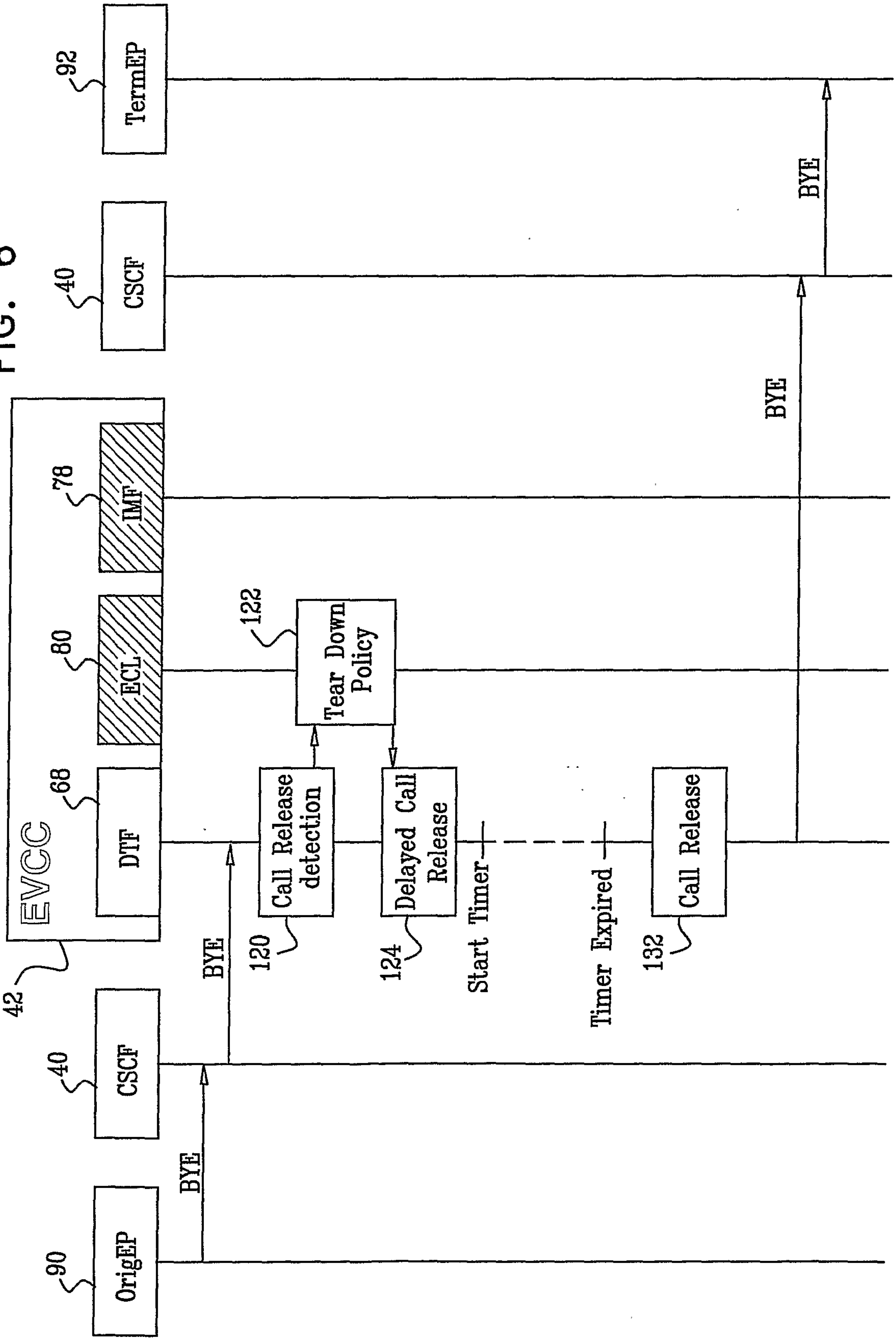
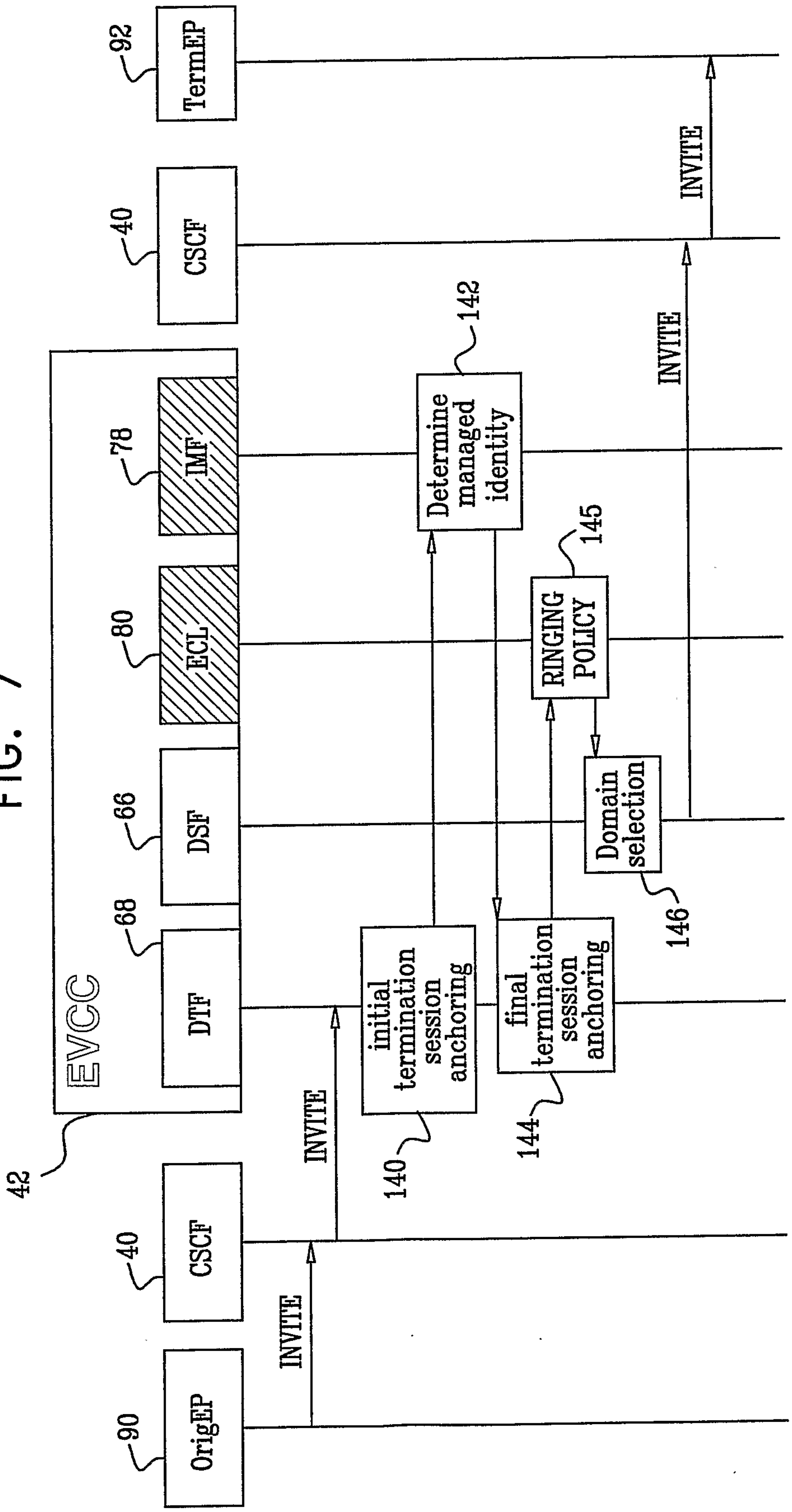

OrigEP
90
CSCF
40
EVCC
42
DTF
68
ECL
80
IMF
78
CSCF
40
TermEP
92
BYE
BYE
Call Release detection
120
Tear Down Policy
122
Delayed Call Release
124
Start Timer
Timer Expired
Call Release
132
BYE
BYE

FIG. 7


90
OrigEP
40
CSCF
42
EVCC
68
DTF
66
DSF
80
ECL
78
IMF
40
CSCF
92
TermEP
INVITE
INVITE
140
initial termination session anchoring
142
Determine managed identity
144
final termination session anchoring
145
RINGING POLICY
146
Domain selection
INVITE
INVITE

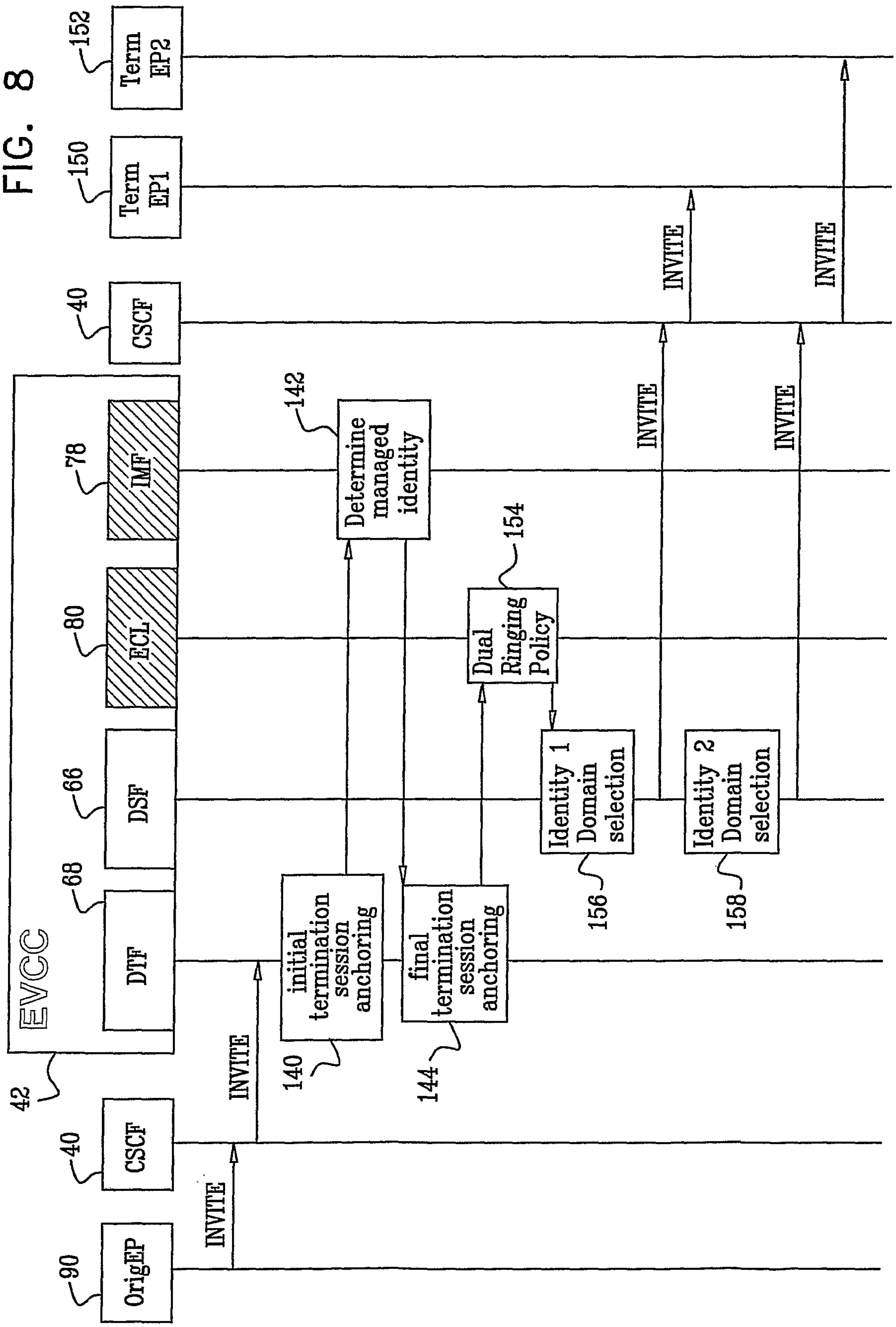

90
OrigEP
40
CSCF
42
EVCC
68
DTF
66
DSF
80
ECL
78
IMF
40
CSCF
150
Term EP1
152
Term EP2
INVITE
INVITE
initial termination session anchoring
140
Determine managed identity
142
final termination session anchoring
144
Dual Ringing Policy
154
Identity 1 Domain selection
156
INVITE
INVITE
Identity 2 Domain selection
158
INVITE
INVITE

FIG. 8

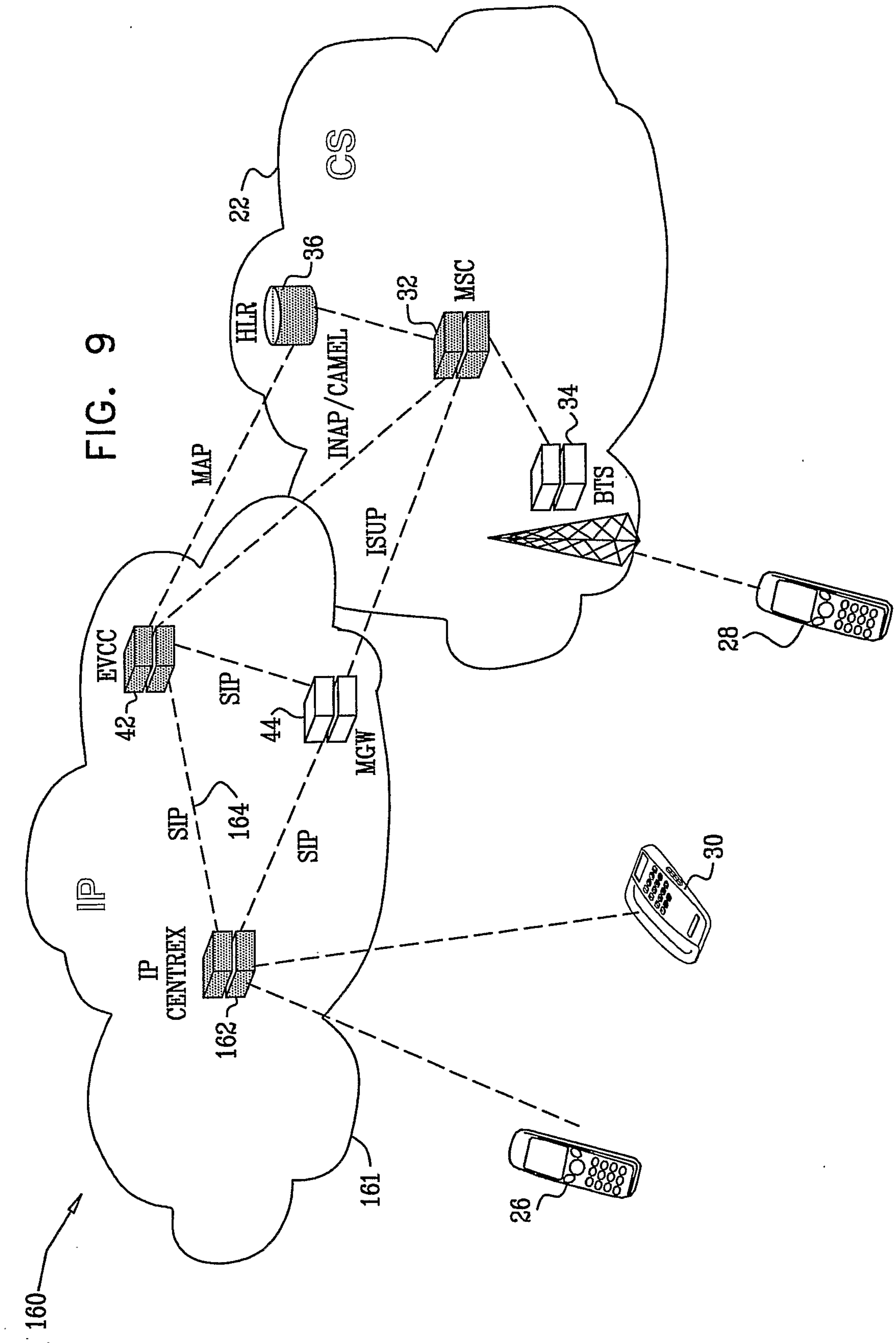
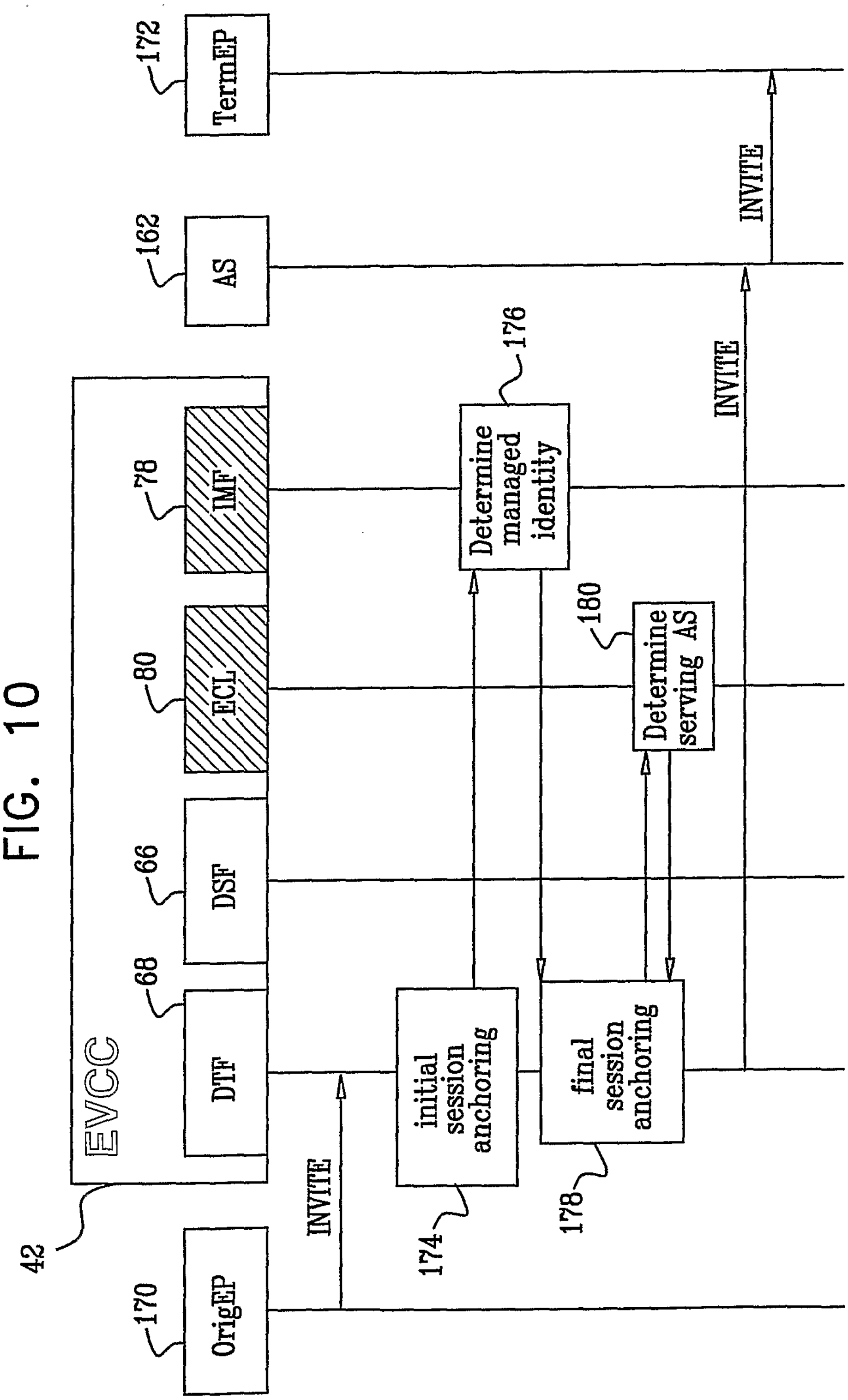

CS
22
HLR
36
32
MSC
MAP
INAP/CAMEL
ISUP
34
BTS
28
EVCC
42
SIP
44
MGW
SIP
164
SIP
IP
IP
CENTREX
162
30
161
26
160

FIG. 9

FIG. 10


42
EVCC
OrigEP
170
DTF
68
DSF
66
ECL
80
IMF
78
AS
162
TermEP
172
INVITE
initial session anchoring
174
Determine managed identity
176
final session anchoring
178
Determine serving AS
180
INVITE
INVITE