

EXHIBIT D

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Dolan et al.

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(54) **METHOD AND APPARATUS FOR PROVIDING EXPANDED TELECOMMUNICATIONS SERVICE**

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(73) Assignee: **Callwave, Inc.**, Santa Barbara, CA (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 619 days.

This patent is subject to a terminal disclaimer.

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(63) Continuation of application No. 10/255,567, filed on Sep. 26, 2002, now Pat. No. 6,898,275, which is a continuation of application No. 09/539,375, filed on Mar. 31, 2000, now Pat. No. 6,477,246.

(60) Provisional application No. 60/127,434, filed on Apr. 1, 1999.

(51) **Int. Cl.**
H04M 3/42 (2006.01)

(52) **U.S. Cl.** **379/211.02**; 379/88.17; 379/88.25; 379/93.33; 370/356

(58) **Field of Classification Search** 379/88.17, 379/88.25, 93.33, 211.02; 370/356
See application file for complete search history.

(56) **References Cited**

U.S. PATENT DOCUMENTS

3,936,613 A 2/1976 Nishigori et al.

(Continued)

FOREIGN PATENT DOCUMENTS

CA 1329852 5/1994

(Continued)

OTHER PUBLICATIONS

Johnson, Dave, "Now You're Talking-voice-response systems for home offices—Product Information"; *Home Office Computing*; http://www.findarticles.com; Feb. 1999.

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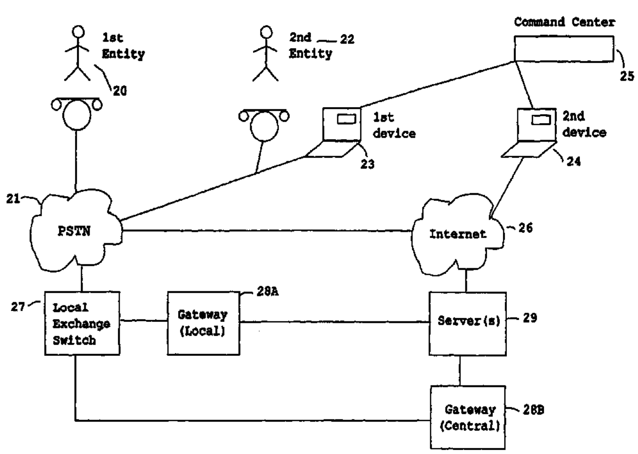
(74) *Attorney, Agent, or Firm*—Knobbe, Martens, Olson & Bear LLP

(57) **ABSTRACT**

System and method for putting control of incoming telephone calls in the hands of subscribers with the aid of computer software and the Internet. In effect, a very efficient multiplexer is provided that does not require a change in the behavior of the called party or the calling party. This system permits the called party to hear a spoken message by the calling party in real time, and the content of the spoken message permits the called party to decide how to handle the call. The system adaptively learns and captures the rules of the called party for handling calls, and learns which callers the called party always wishes to talk to. The system uses special control software on the called party's computer which is connected to the Internet. The system employs a central server in which all of the required intelligence is resident. Audio signals are exchanged via non-data channels provided by the telephone companies and by the Internet. Either a very simple Internet busy pick-up is provided, or a very complex messaging system is provided, as desired. A feature is the monitoring and screening of incoming calls before deciding how to handle them. Another feature is the capturing and storage of the decision making profile.

46 Claims, 14 Drawing Sheets



U.S. PATENT DOCUMENTS

3,956,595	A	5/1976	Sobanski
4,009,337	A	2/1977	Sakai et al.
4,022,983	A	5/1977	Braun et al.
4,485,470	A	11/1984	Reali
4,736,405	A	4/1988	Akiyama
4,809,321	A	2/1989	Morganstein et al.
4,893,336	A	1/1990	Wuthnow
4,994,926	A	2/1991	Gordon et al.
5,040,208	A	8/1991	Jolissaint
5,046,087	A	9/1991	Sakai
5,291,302	A	3/1994	Gordon et al.
5,404,537	A	4/1995	Olnowich et al.
5,434,908	A	7/1995	Klein
5,459,584	A	10/1995	Gordon et al.
5,467,388	A	11/1995	Redd, Jr. et al.
5,526,524	A	6/1996	Madduri
5,533,102	A	7/1996	Robinson et al.
5,577,111	A	11/1996	Iida et al.
5,583,918	A	12/1996	Nakagawa
5,619,557	A	4/1997	Van Berkum
5,640,677	A	6/1997	Karlsson
5,651,054	A	7/1997	Dunn et al.
5,751,795	A	5/1998	Hassler et al.
5,774,067	A	6/1998	Olnowich et al.
5,805,587	A	9/1998	Norris et al.
5,809,128	A	9/1998	McMullin
5,812,551	A	9/1998	Tsukazoe et al.
5,825,867	A	10/1998	Epler et al.
5,832,060	A	11/1998	Corlett et al.
5,894,504	A	4/1999	Alfred et al.
5,946,386	A	8/1999	Rogers et al.
5,960,064	A	9/1999	Foladare et al.
5,960,073	A	9/1999	Kikinis et al.
5,963,629	A	10/1999	Jung
5,995,594	A	11/1999	Shaffer et al.
5,995,603	A	11/1999	Anderson
6,014,436	A	1/2000	Florence et al.
6,032,051	A	2/2000	Hall et al.
6,034,956	A	3/2000	Olnowich et al.
6,035,031	A	3/2000	Silverman
6,044,059	A	3/2000	Olnowich
6,078,581	A	6/2000	Shtivelman et al.
6,104,800	A	8/2000	Benson
6,144,644	A	11/2000	Bajzath et al.
6,160,881	A	12/2000	Beyda et al.
6,167,127	A	12/2000	Smith et al.
6,169,796	B1	1/2001	Bauer et al.
6,178,183	B1	1/2001	Buskirk, Jr.
6,181,691	B1	1/2001	Markgraf et al.
6,208,638	B1	3/2001	Rieley et al.
6,212,261	B1	4/2001	Meubus et al.
6,230,009	B1	5/2001	Holmes et al.
6,243,378	B1	6/2001	Olnowich
6,253,249	B1	6/2001	Belzile
6,278,704	B1	8/2001	Creamer et al.
6,304,565	B1	10/2001	Ramamurthy
6,350,066	B1	2/2002	Bobo, II
6,353,663	B1	3/2002	Stevens et al.
6,405,035	B1	6/2002	Singh
6,411,601	B1	6/2002	Shaffer et al.
6,411,805	B1	6/2002	Becker et al.
6,438,216	B1	8/2002	Aktas
6,438,222	B1	8/2002	Burg
6,477,246	B1	11/2002	Dolan et al.
6,496,576	B2	12/2002	Tanaka et al.
6,501,750	B1	12/2002	Shaffer et al.
6,510,162	B1	1/2003	Fijolek et al.
6,512,930	B2	1/2003	Sandegren
6,519,258	B1	2/2003	Tsukazoe et al.
6,539,084	B1	3/2003	Long
6,549,612	B2	4/2003	Gifford et al.
6,564,321	B2	5/2003	Bobo, II
6,567,505	B1	5/2003	Omari et al.
6,643,034	B1	11/2003	Gordon et al.
6,721,408	B1	4/2004	Bain et al.
6,785,021	B1	8/2004	Gordon et al.
6,857,074	B2	2/2005	Bobo, II

FOREIGN PATENT DOCUMENTS

WO	WO 98/36551	8/1998
WO	WO 97/26749	7/1999
WO	WO 00/06084	10/2000

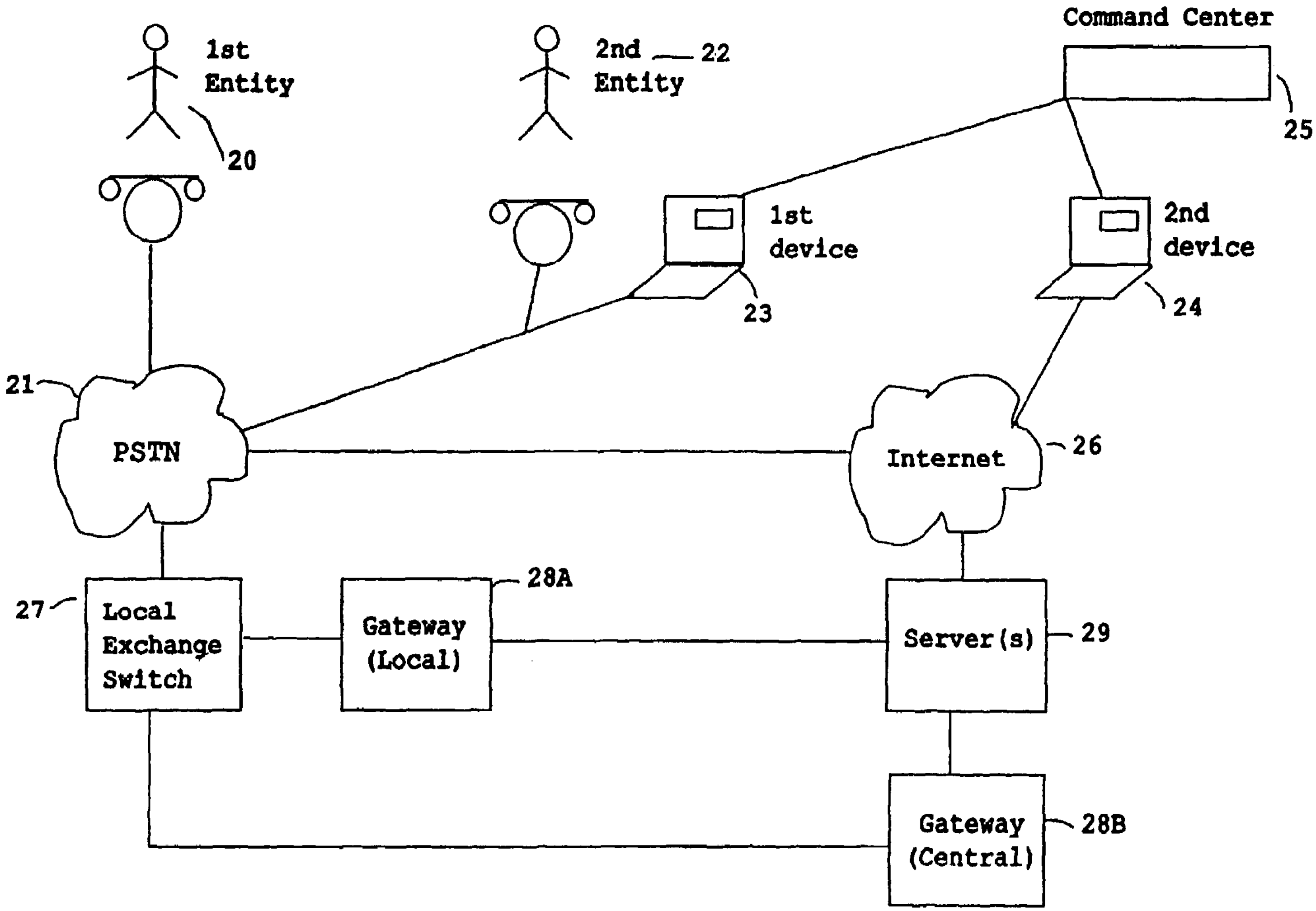
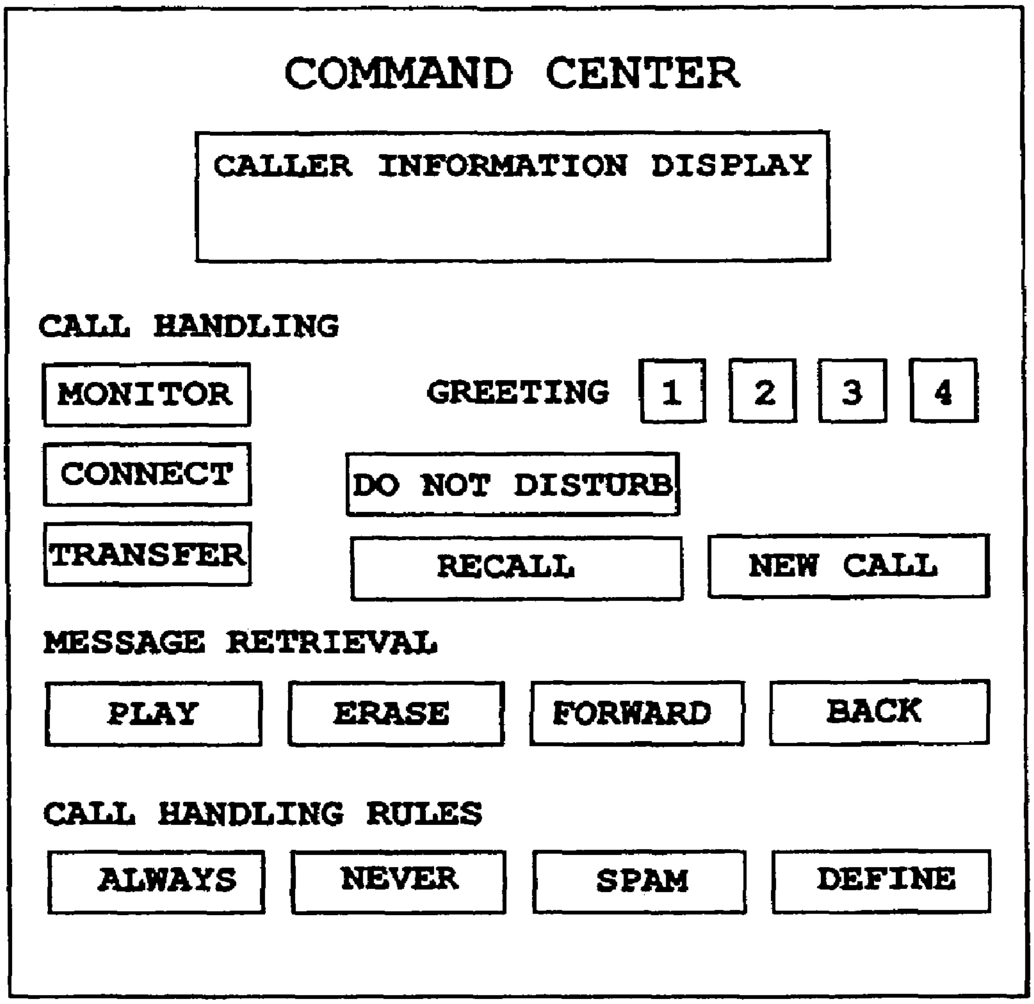

1st Entity
20
2nd Entity
22
Command Center
25
1st device
23
2nd device
24
21
PSTN
Internet
26
27
Local Exchange Switch
Gateway (Local)
28A
Server(s)
29
Gateway (Central)
28B

Figure 1



Display presentation is visual and/or audible caller information.

Shown are action buttons for various call and message handling. Actions may also be taken by voice or touch tone commands.

Figure 2

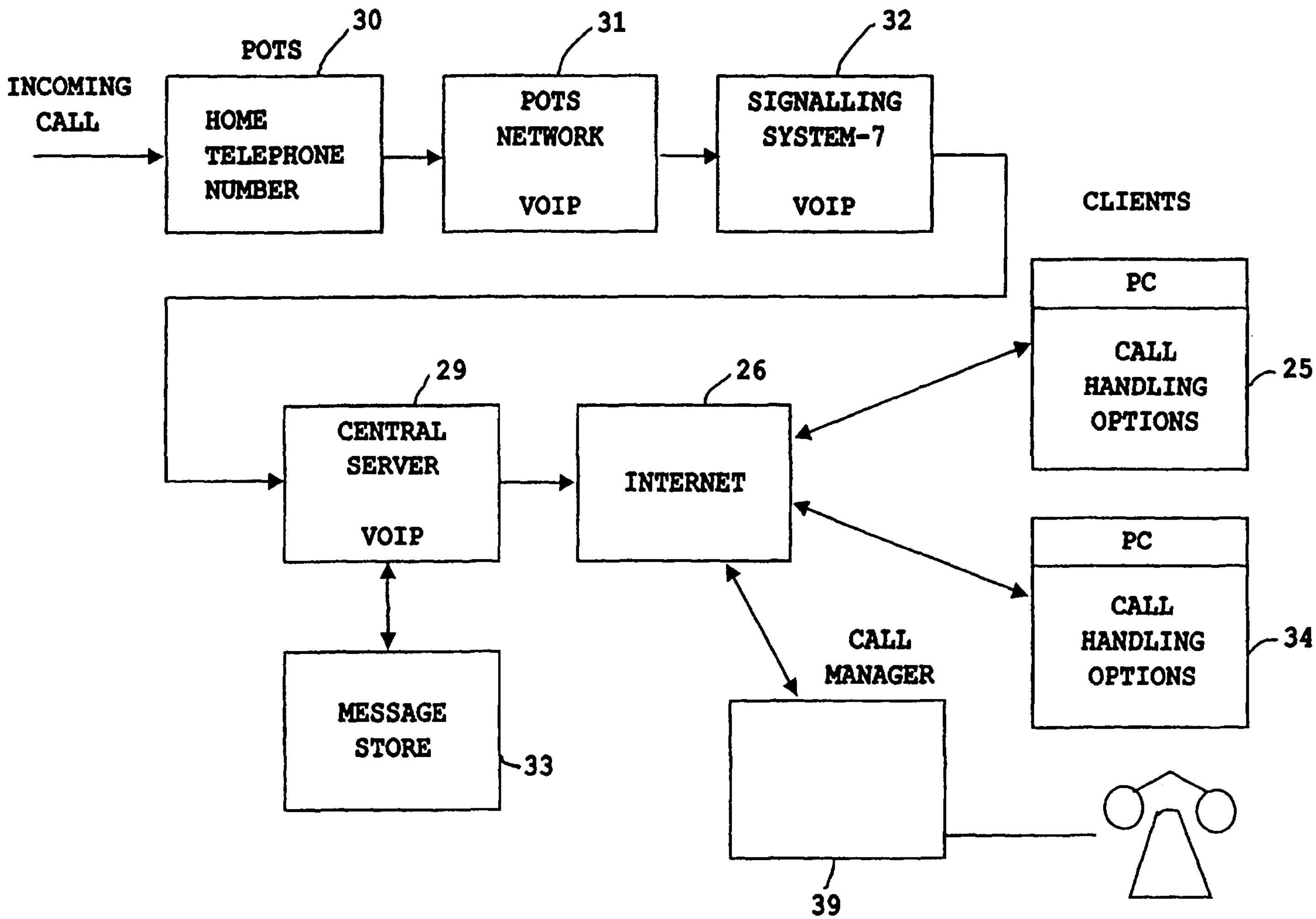

INCOMING CALL
POTS
30
HOME TELEPHONE NUMBER
31
POTS NETWORK
VOIP
32
SIGNALLING SYSTEM-7
VOIP
CLIENTS
PC
CALL HANDLING OPTIONS
25
29
CENTRAL SERVER
VOIP
26
INTERNET
PC
CALL HANDLING OPTIONS
34
CALL MANAGER
MESSAGE STORE
33
39

Figure 3

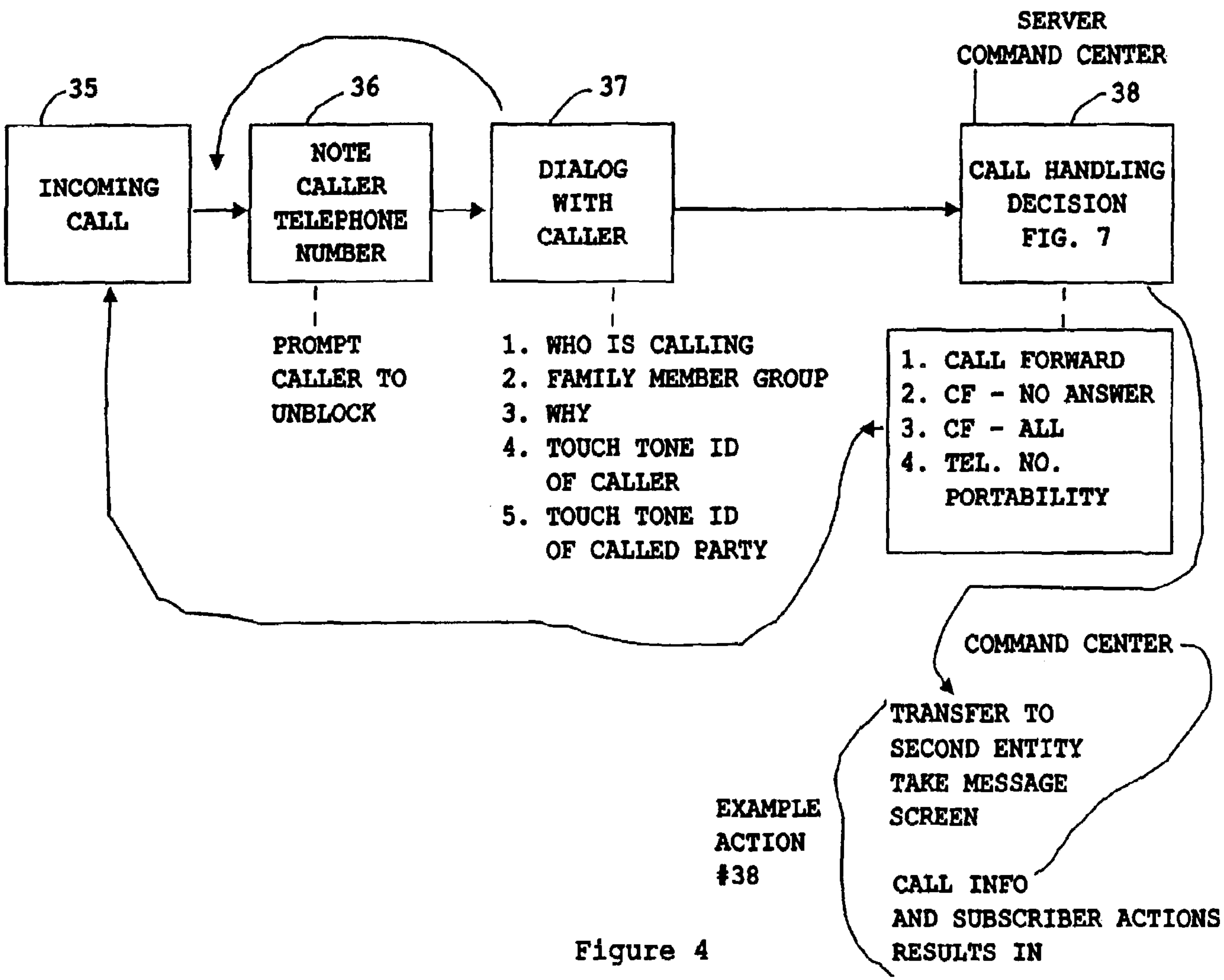

35
INCOMING CALL
36
NOTE CALLER TELEPHONE NUMBER
PROMPT CALLER TO UNBLOCK
37
DIALOG WITH CALLER
1. WHO IS CALLING
2. FAMILY MEMBER GROUP
3. WHY
4. TOUCH TONE ID OF CALLER
5. TOUCH TONE ID OF CALLED PARTY
SERVER COMMAND CENTER
38
CALL HANDLING DECISION FIG. 7
1. CALL FORWARD
2. CF - NO ANSWER
3. CF - ALL
4. TEL. NO. PORTABILITY
COMMAND CENTER
TRANSFER TO SECOND ENTITY
TAKE MESSAGE
SCREEN
CALL INFO AND SUBSCRIBER ACTIONS RESULTS IN
EXAMPLE ACTION #38

Figure 4

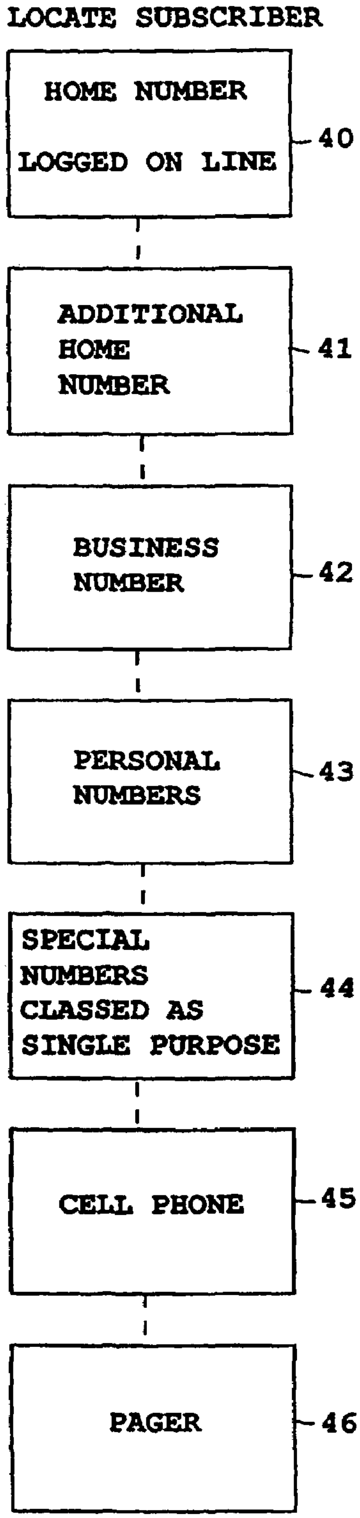


Figure 5

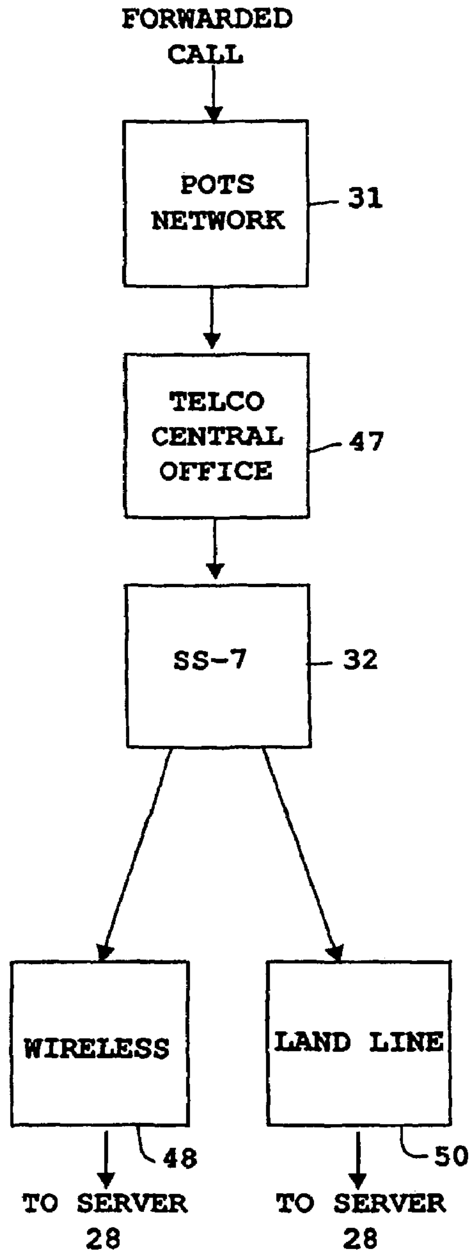


Figure 6

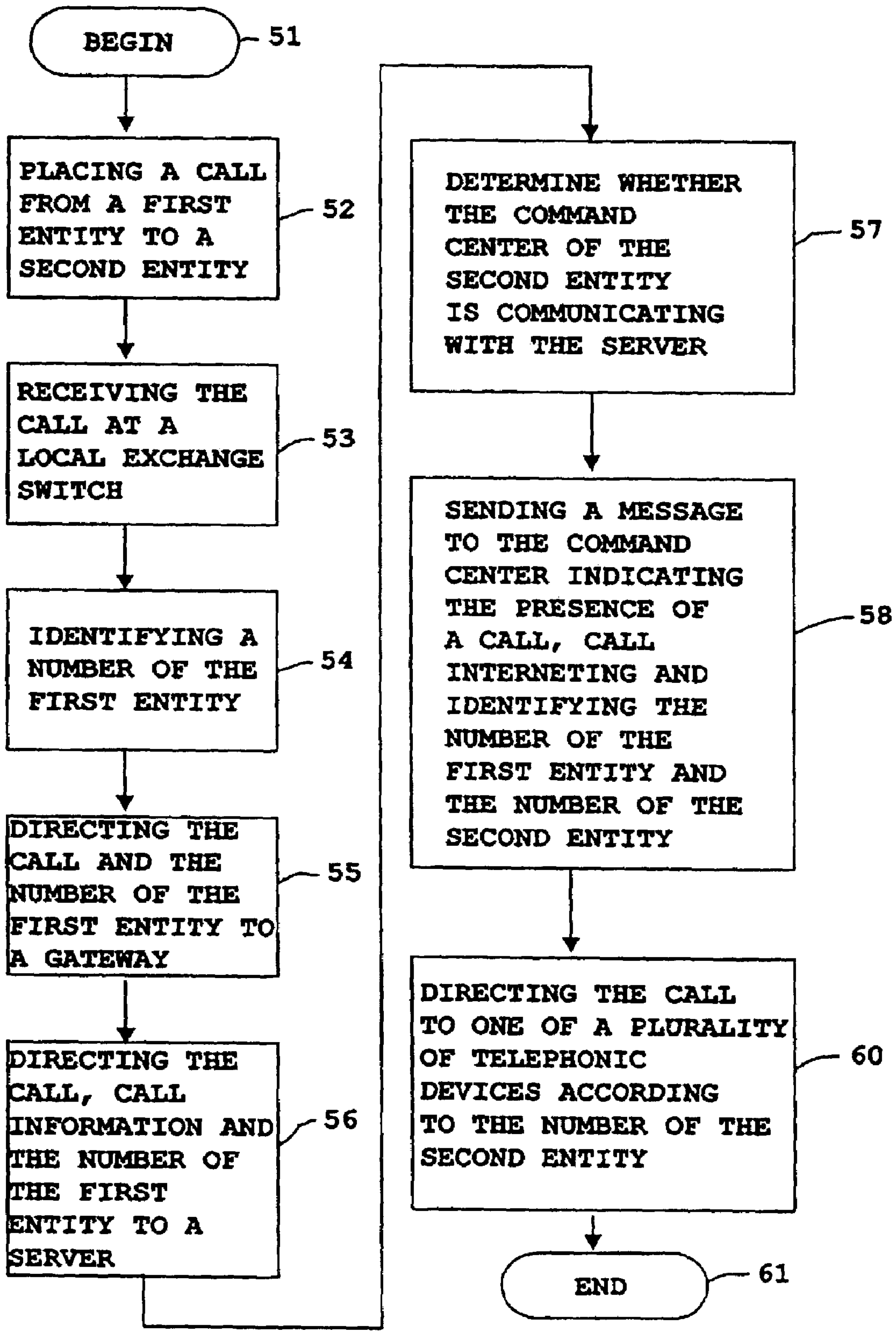

BEGIN
51
PLACING A CALL FROM A FIRST ENTITY TO A SECOND ENTITY
52
RECEIVING THE CALL AT A LOCAL EXCHANGE SWITCH
53
IDENTIFYING A NUMBER OF THE FIRST ENTITY
54
DIRECTING THE CALL AND THE NUMBER OF THE FIRST ENTITY TO A GATEWAY
55
DIRECTING THE CALL, CALL INFORMATION AND THE NUMBER OF THE FIRST ENTITY TO A SERVER
56
DETERMINE WHETHER THE COMMAND CENTER OF THE SECOND ENTITY IS COMMUNICATING WITH THE SERVER
57
SENDING A MESSAGE TO THE COMMAND CENTER INDICATING THE PRESENCE OF A CALL, CALL INTERNETING AND IDENTIFYING THE NUMBER OF THE FIRST ENTITY AND THE NUMBER OF THE SECOND ENTITY
58
DIRECTING THE CALL TO ONE OF A PLURALITY OF TELEPHONIC DEVICES ACCORDING TO THE NUMBER OF THE SECOND ENTITY
60
END
61

Figure 7

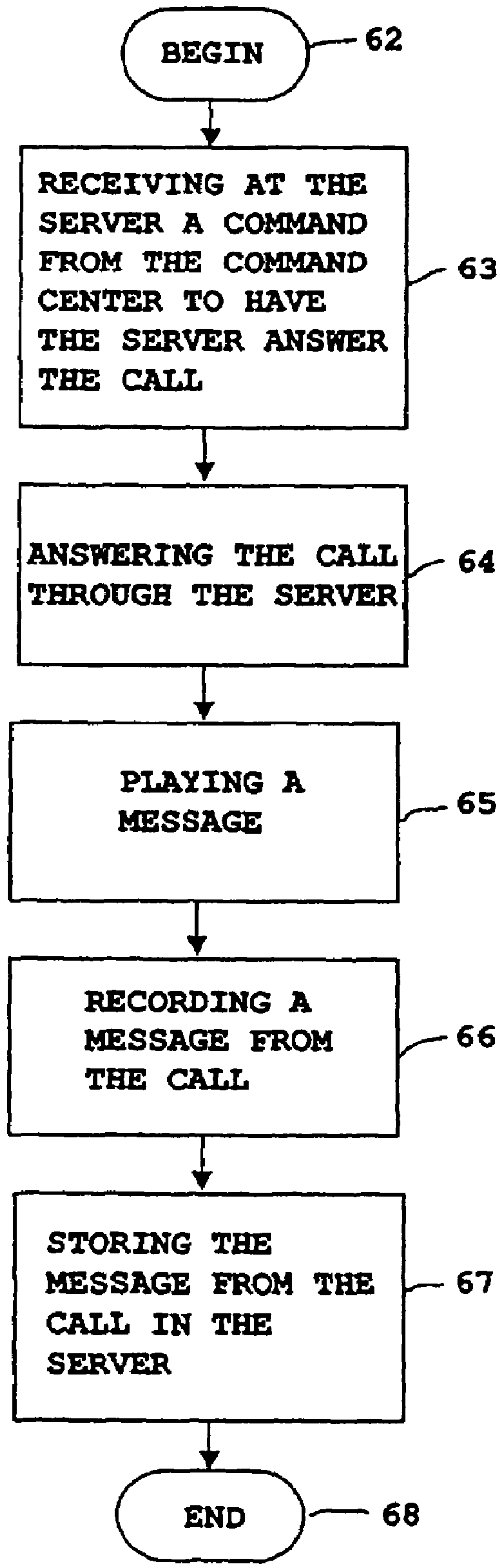

BEGIN
62
RECEIVING AT THE SERVER A COMMAND FROM THE COMMAND CENTER TO HAVE THE SERVER ANSWER THE CALL
63
ANSWERING THE CALL THROUGH THE SERVER
64
PLAYING A MESSAGE
65
RECORDING A MESSAGE FROM THE CALL
66
STORING THE MESSAGE FROM THE CALL IN THE SERVER
67
END
68

Figure 8

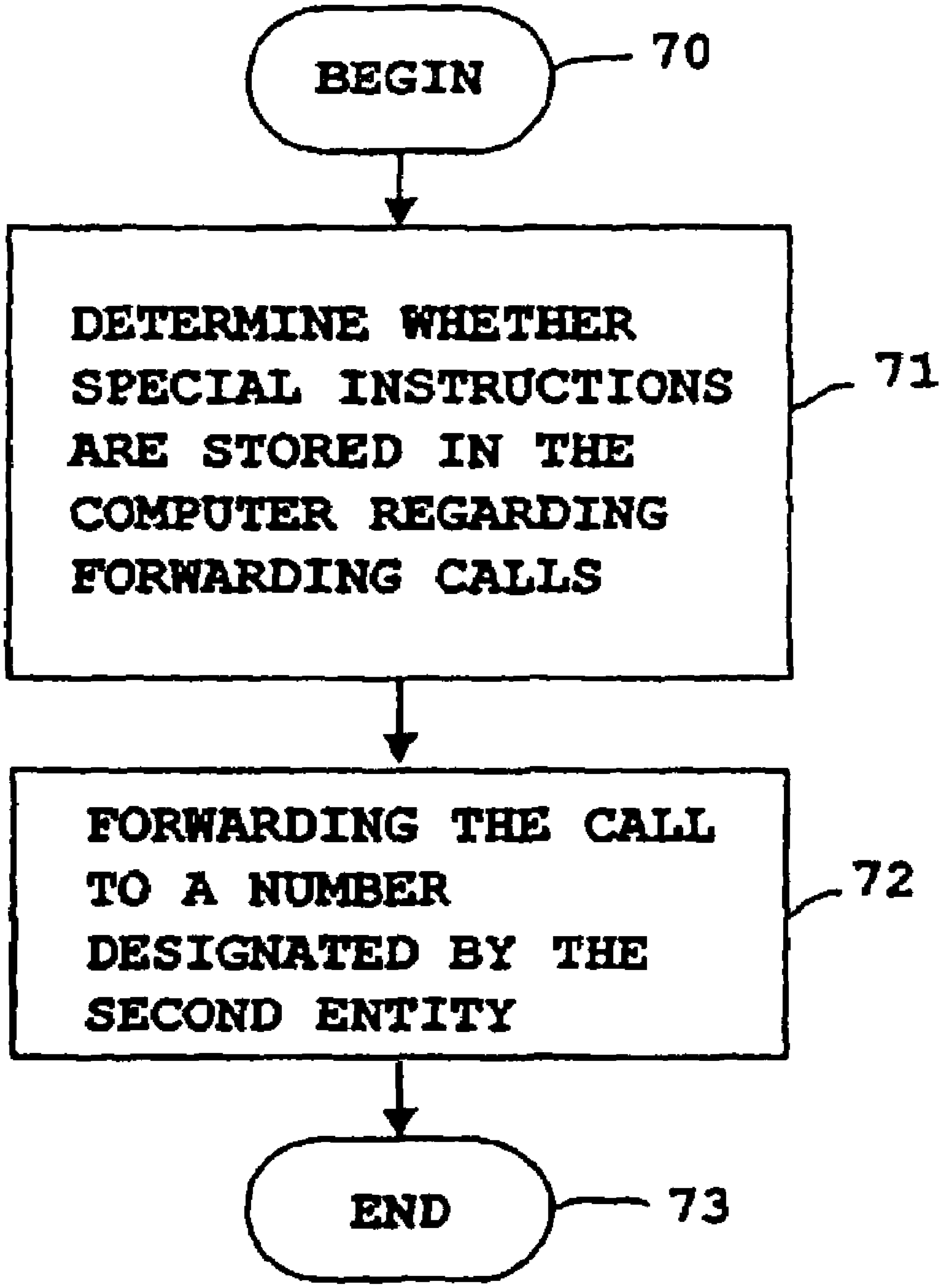

BEGIN
70
DETERMINE WHETHER SPECIAL INSTRUCTIONS ARE STORED IN THE COMPUTER REGARDING FORWARDING CALLS
71
FORWARDING THE CALL TO A NUMBER DESIGNATED BY THE SECOND ENTITY
72
END
73

Figure 9

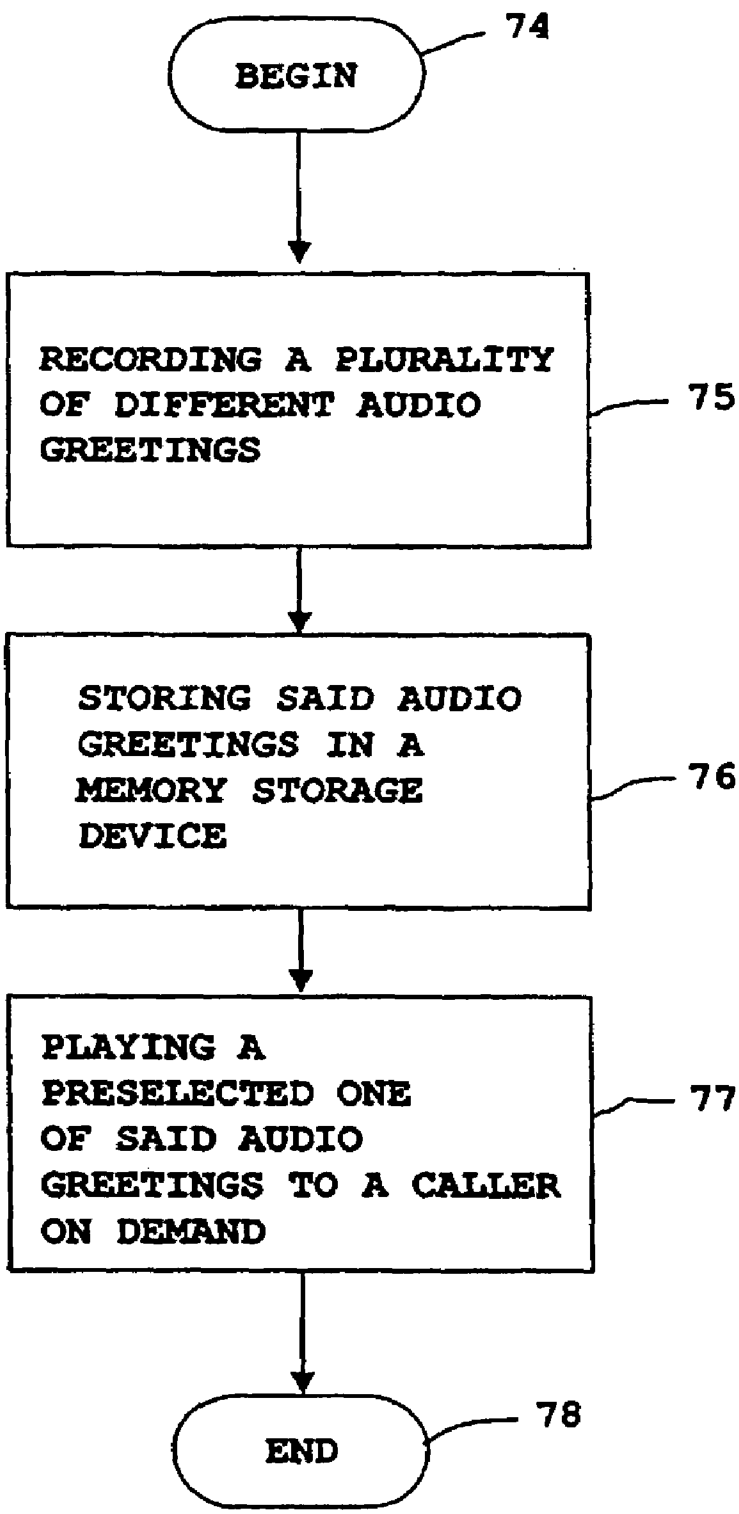

BEGIN
74
RECORDING A PLURALITY OF DIFFERENT AUDIO GREETINGS
75
STORING SAID AUDIO GREETINGS IN A MEMORY STORAGE DEVICE
76
PLAYING A PRESELECTED ONE OF SAID AUDIO GREETINGS TO A CALLER ON DEMAND
77
END
78

Figure 10

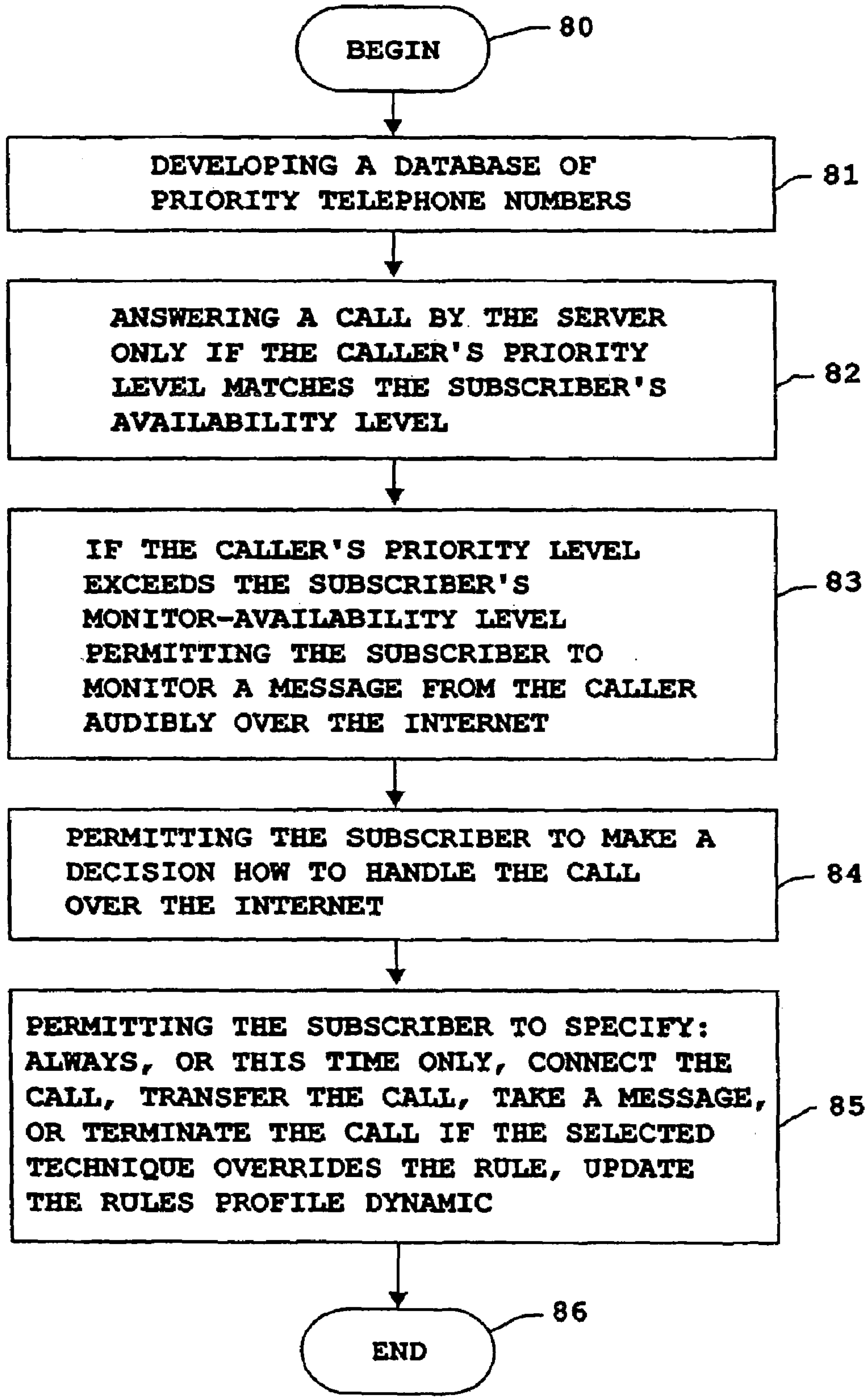

BEGIN
80
DEVELOPING A DATABASE OF PRIORITY TELEPHONE NUMBERS
81
ANSWERING A CALL BY THE SERVER ONLY IF THE CALLER'S PRIORITY LEVEL MATCHES THE SUBSCRIBER'S AVAILABILITY LEVEL
82
IF THE CALLER'S PRIORITY LEVEL EXCEEDS THE SUBSCRIBER'S MONITOR-AVAILABILITY LEVEL PERMITTING THE SUBSCRIBER TO MONITOR A MESSAGE FROM THE CALLER AUDIBLY OVER THE INTERNET
83
PERMITTING THE SUBSCRIBER TO MAKE A DECISION HOW TO HANDLE THE CALL OVER THE INTERNET
84
PERMITTING THE SUBSCRIBER TO SPECIFY: ALWAYS, OR THIS TIME ONLY, CONNECT THE CALL, TRANSFER THE CALL, TAKE A MESSAGE, OR TERMINATE THE CALL IF THE SELECTED TECHNIQUE OVERRIDES THE RULE, UPDATE THE RULES PROFILE DYNAMIC
85
END
86

Figure 11

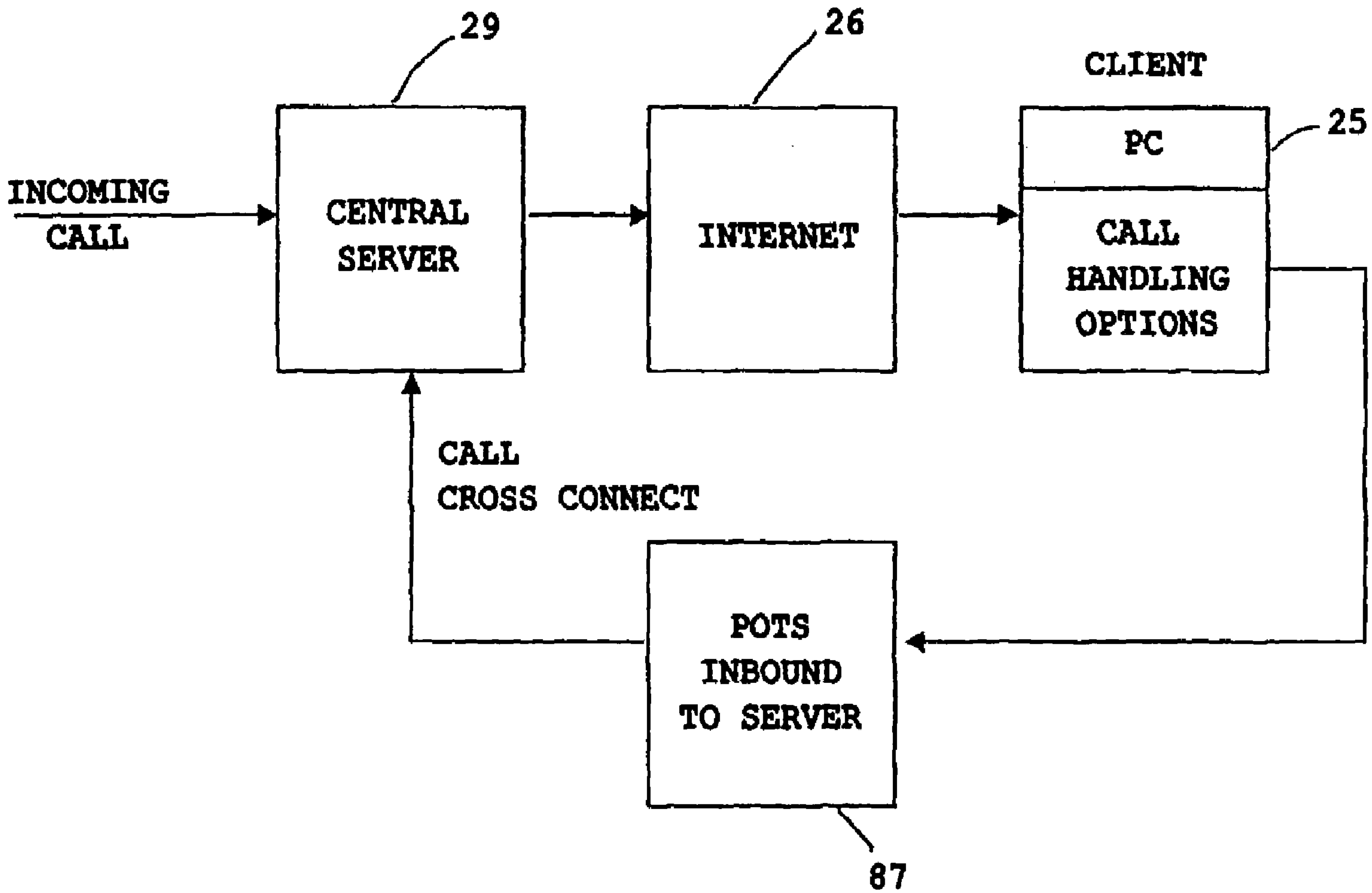

29
26
CLIENT
25
PC
INCOMING
CALL
CENTRAL
SERVER
INTERNET
CALL
HANDLING
OPTIONS
CALL
CROSS CONNECT
POTS
INBOUND
TO SERVER
87

Figure 12

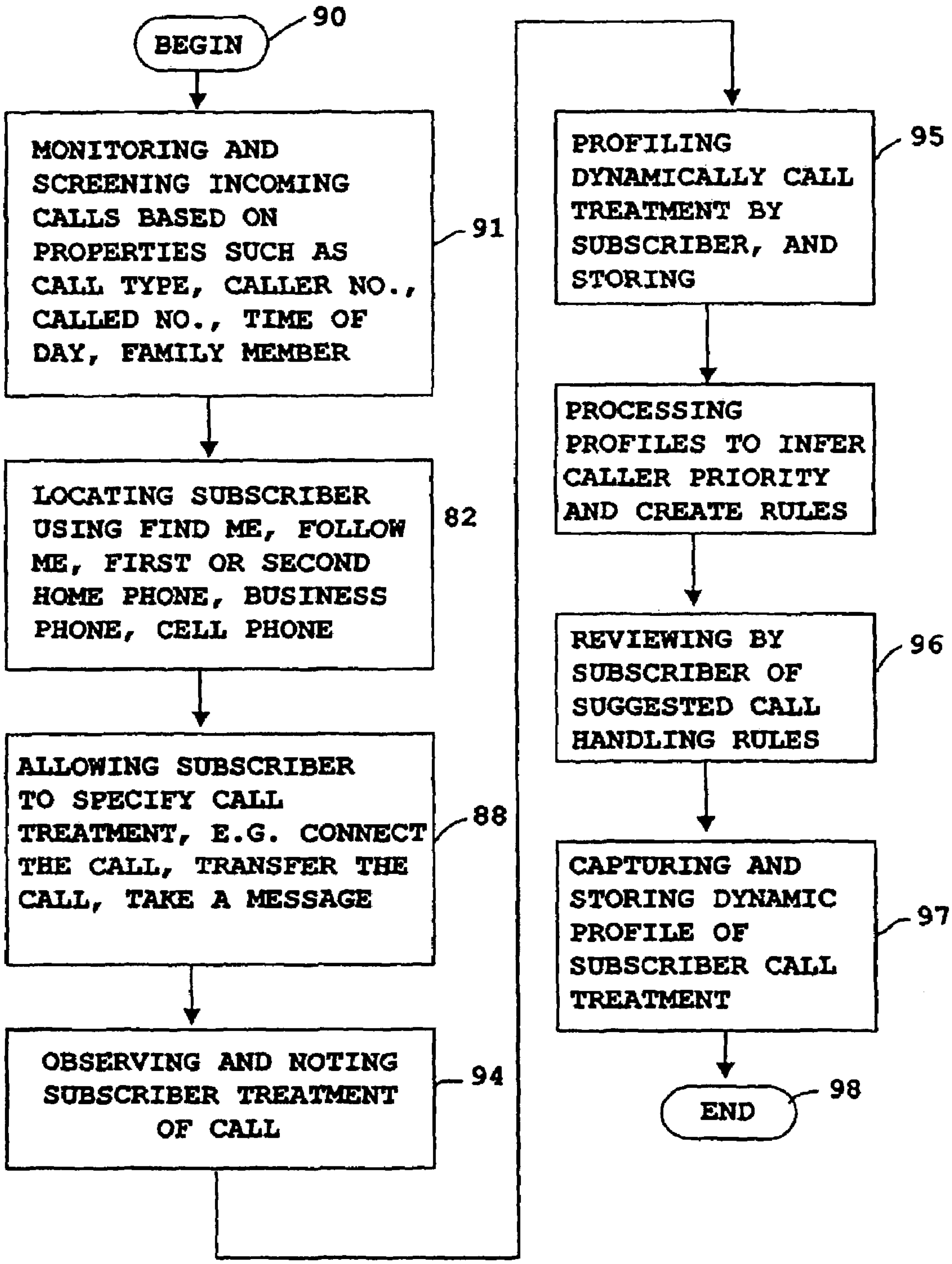

BEGIN
90
MONITORING AND SCREENING INCOMING CALLS BASED ON PROPERTIES SUCH AS CALL TYPE, CALLER NO., CALLED NO., TIME OF DAY, FAMILY MEMBER
91
LOCATING SUBSCRIBER USING FIND ME, FOLLOW ME, FIRST OR SECOND HOME PHONE, BUSINESS PHONE, CELL PHONE
82
ALLOWING SUBSCRIBER TO SPECIFY CALL TREATMENT, E.G. CONNECT THE CALL, TRANSFER THE CALL, TAKE A MESSAGE
88
OBSERVING AND NOTING SUBSCRIBER TREATMENT OF CALL
94
PROFILING DYNAMICALLY CALL TREATMENT BY SUBSCRIBER, AND STORING
95
PROCESSING PROFILES TO INFER CALLER PRIORITY AND CREATE RULES
REVIEWING BY SUBSCRIBER OF SUGGESTED CALL HANDLING RULES
96
CAPTURING AND STORING DYNAMIC PROFILE OF SUBSCRIBER CALL TREATMENT
97
END
98

Figure 13

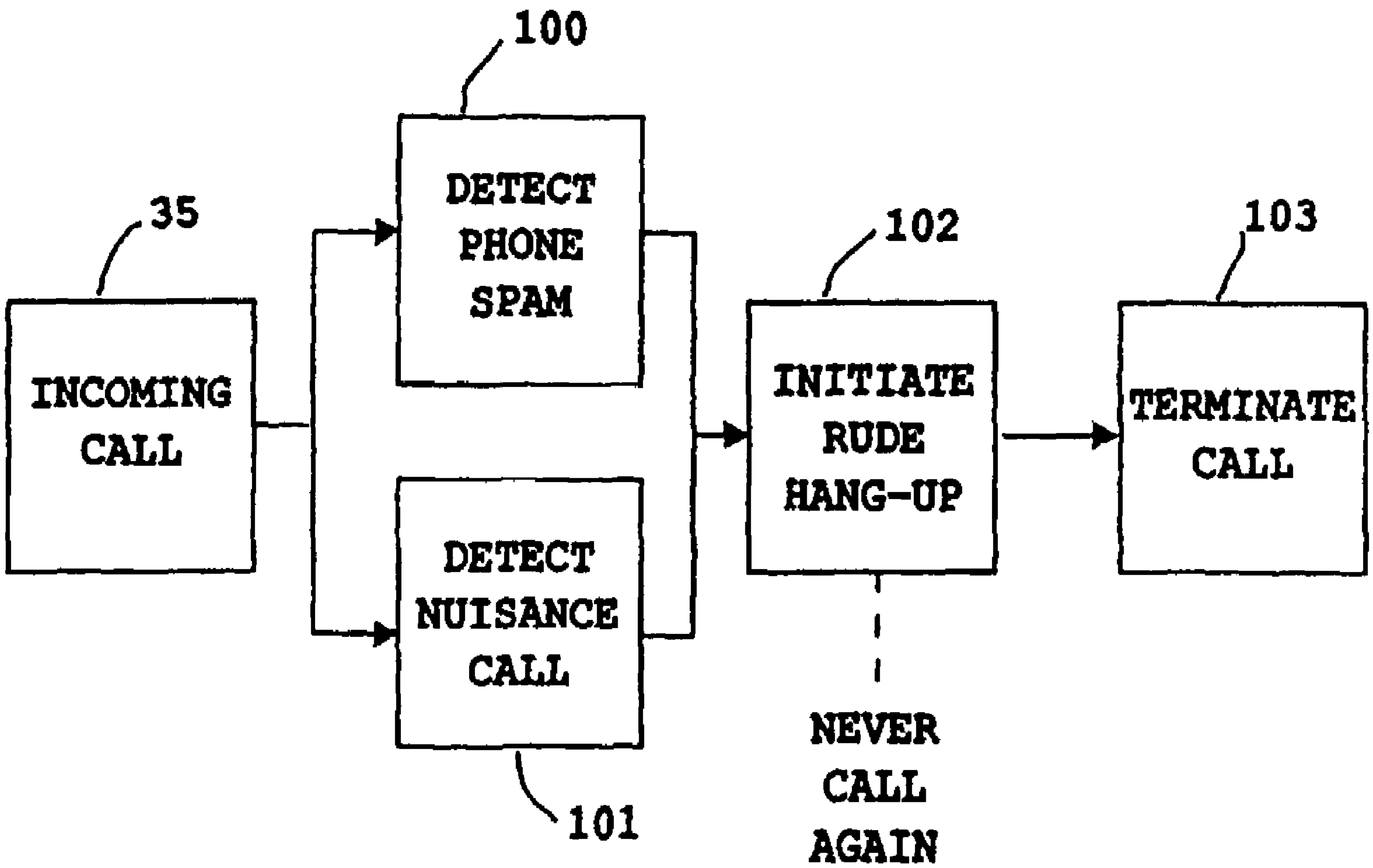

INCOMING CALL
35
DETECT PHONE SPAM
100
DETECT NUISANCE CALL
101
INITIATE RUDE HANG-UP
102
NEVER CALL AGAIN
TERMINATE CALL
103

Figure 14

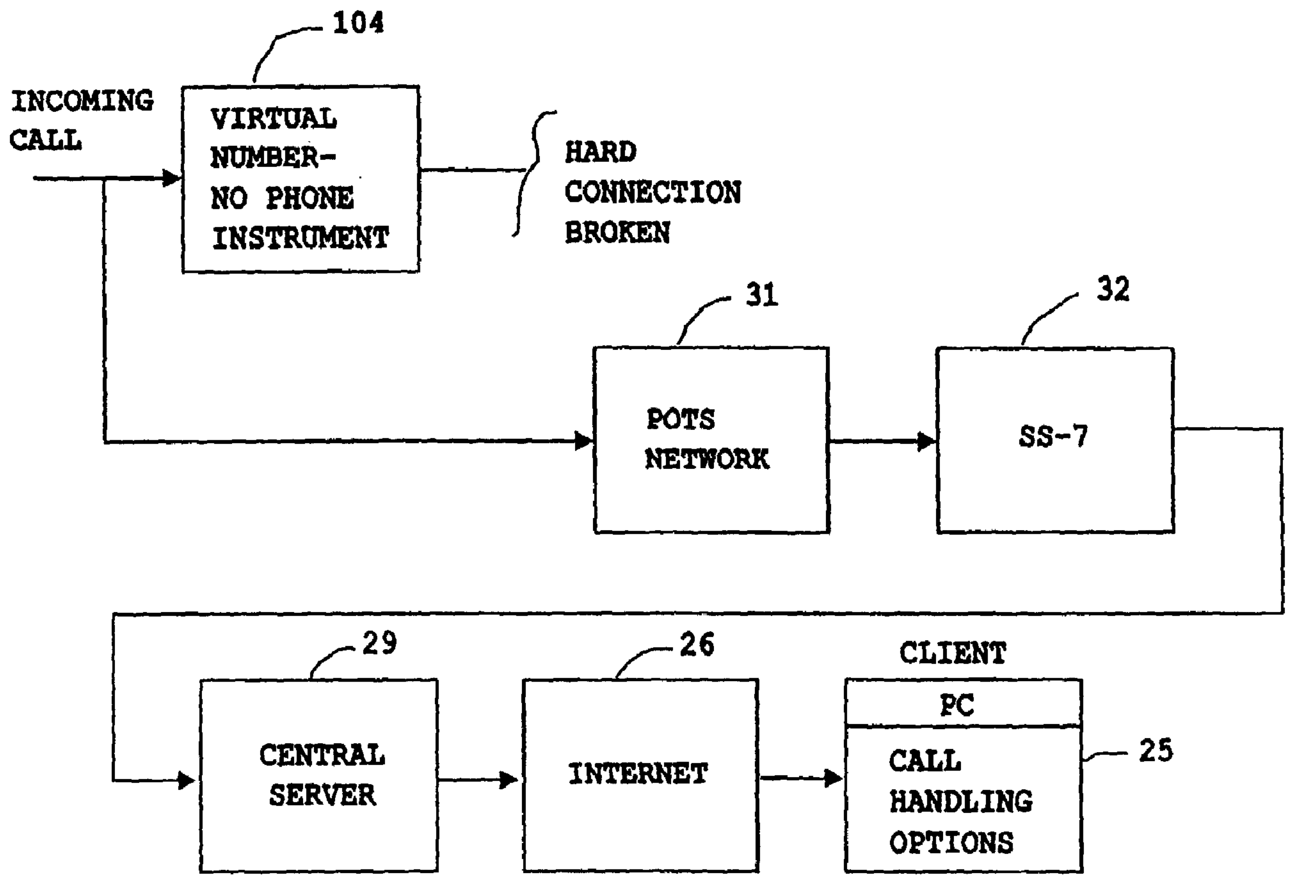

104
INCOMING
CALL
VIRTUAL
NUMBER-
NO PHONE
INSTRUMENT
HARD
CONNECTION
BROKEN
31
32
POTS
NETWORK
SS-7
29
26
CLIENT
PC
CENTRAL
SERVER
INTERNET
CALL
HANDLING
OPTIONS
25

Figure 15

METHOD AND APPARATUS FOR PROVIDING EXPANDED TELECOMMUNICATIONS SERVICE

CROSS REFERENCE TO RELATED APPLICATIONS

This application is a continuation of U.S. patent application Ser. No. 10/255,567 (filed 26 Sep. 2002) now U.S. Pat. No. 6,898,275, which is a continuation of U.S. patent application Ser. No. 09/539,375 (filed 31 Mar. 2000; now U.S. Pat. No. 6,477,246), which claims the benefit of U.S. Provisional Patent Application 60/127,434 (filed 1 Apr. 1999). The entire disclosure of all of these priority applications is hereby incorporated by reference herein.

TECHNICAL FIELD

The present invention relates generally to a telecommunications system, and more particularly to the control of telephone calls in a telecommunication system by use of personal computer software via the Internet.

BACKGROUND OF THE INVENTION

Many telephone subscribers have a personal computer on their desk and frequently the personal computer is logged in to the same telephone line that would normally be used by the telephone. This is for use of the computer on the Internet. Frequently, incoming telephone calls receive a busy signal because the computer is logged on to the Internet. Thus there are many lost calls. Many individuals and small businesses are searching for ways to simplify and control their telecommunications systems. Many of them are reluctant to acquire additional telephone lines at current prices.

Most telecommunications systems today have limited intelligence. It is estimated that 75% of business calls end in voice mail, an often unsatisfactory conclusion. Calls not completed may result in irritated customers and lost sales. The present invention addresses this waste of human and business resources by using the Internet and Internet telephony to deliver control of a customers telecommunication for the individual or small business.

BRIEF SUMMARY OF THE INVENTION

The present invention ranges from an internet busy pick-up through a very complex messaging system. All that is required at the subscribers location is special software for use on a personal computer in connection with the internet. The system of the present invention operates a central server which receives incoming telephone calls when a user is connected to the Internet. The company's central server delivers the calling and called number information to the user's desktop computer for all calls. The user may elect to pick up that call in which case a direct Internet connection is made between the users desktop computer and the telephone system. If the user does not answer the call or the user is not logged on line to the Internet, the company's central server takes the message or optionally forwards the call to a traditional phone line or a cell phone. The information about the call is then spoken to the customer who can again make a decision whether to take that call.

It is presently contemplated that the system of the present invention takes the call only if the subscriber's line is busy and the calling parties number is recorded in memory storage at the central server. It is also contemplated at present that the caller can provide a spoken message which can be played via the Internet to the subscriber at which time the subscriber can make a decision as to how to handle the call. These features are offered without the necessity for a behavior change by either the calling party or the called party. However, other messaging features can be offered, if desired. This makes the system of the present invention as complex as is desired. It can be extremely simple for the unsophisticated customer and yet can offer many advanced features for those that desire them.

Other features of the present invention are disclosed or apparent in the section entitled: "BEST MODE FOR CARRYING OUT THE INVENTION."

BRIEF DESCRIPTION OF DRAWINGS

For a fuller understanding of the present invention, reference is made to the accompanying drawing in the following detailed description of the Best Mode of Carrying Out the Present Invention. In the drawing, the same reference characters are used to refer to the same elements of the present invention throughout the several figures of the drawing.

FIG. **1** is a schematic representation of one embodiment of the present invention.

FIG. **2** is an example of a typical screen for a control panel.

FIG. **3** is a more detailed version of the communication path of the system of the present invention.

FIG. **4** is a schematic diagram of the call handling process.

FIG. **5** is the process of locating the subscriber.

FIG. **6** is an expansion of FIG. **3** showing how a Telco central office can be connected by way of wireless connections or by way of land line connections to the server.

FIG. **7** is a sequence of steps on a flow chart indicating the handling of a call.

FIG. **8** is a flow chart showing a series of steps in the handling of incoming telephone calls.

FIG. **9** is a flow chart illustrating further steps in the call handling flow chart.

FIG. **10** is a sequence of steps in a flow chart.

FIG. **11** is a flow chart for steps in deciding how the subscriber answers a call.

FIG. **12** illustrates calling back into the central server while an incoming call is being held at the server.

FIG. **13** is a sequence of steps in a method flow chart illustrating capturing the dynamic profile of a subscriber.

FIG. **14** is a schematic diagram indicating how the system can handle undesired incoming calls.

FIG. **15** is a schematic diagram illustrating a second embodiment of the call control system of the present invention.

BEST MODE FOR CARRYING OUT THE INVENTION

FIG. **1** is a schematic representation of the organization of one embodiment of the present invention. FIG. **1** shows a first entity **20** connected to the Public Switched Telephone Network (PSTN) **21**. A second entity **22** is also connected to the PSTN **21**. The second entity **22** is illustrated as having a first device **23** indirectly connected to the Internet **26** through a PSTN **21** provided dial-up connection shared with the telephone of the second entity **22**. The second entity **22** is also shown to have a second device **24** directly connected to the Internet **26**. Both devices host a software based Command Center **25**. The devices may or may not be logged onto the Internet **26**. FIG. **1** also shows a local exchange switch **27** connected to the PSTN **21**. The system directs the call and the call information of the first entity **20** through the PSTN **21** to the local exchange switch **27** and then to a gateway **28**; said gateway can be implemented as a local gateway **28**A near the switching system **27** or a centralized gateway **28**B near the server **29**. The gateway forwards the call information to the Server (or array of Servers) **29** which then communicates to the Command Center **25** over an Internet Protocol connection that is by a plurality of means, including a single phone line dial up connection (e.g. as shown in the case of the first device **23**) an always on landline home connection (for example as shown in the case of the second device **24**) an always on landline office connection or an always on wireless connection. The Server **29** and the Command Center **25** then interact to coordinate the dialog with the first entity **20** over the PSTN **21** and the second entity **22** over the appropriate device **23** or **24**.

Referring now to FIG. **2** of the drawings, an example of a typical screen for command center **25** a control panel is shown. It should be understood that the control panel may have messages on it other than the ones shown in the figure. The control panel of FIG. **2** would typically show up in a small area of the monitor screen of the subscriber's device desktop (e.g. computer) **23** & **24**. For example, the control panel of FIG. **2** may occupy only a 1 inch by 2 inch corner of the screen. The control panel of FIG. **2** displays status of the incoming call and allows the second entity **22** to coordinate interactions with the server **29**. The command center interface supports a plurality of call control inputs including a) no input b) transfer call over circuit switching network c) take call over Internet Protocol d) reoriginate the call over circuit switched or Internet Protocol network e) have the server interact with the caller to provide information or record the audible signal.

Referring now to FIG. **3** of the drawings, there is shown a more detailed version of the communication path of the system of the present invention. An incoming call is illustrated as being directed to a home telephone number **30** in the Plain Old Telephone System (POTS) which is passed on to the POTS Network **31**. The POTS Network **31** is shown connected to a Signaling System 7 (SS7) **32**. The telephone system employs what is known as Common Channel Signaling (CCS). This is a signaling system used in telephone networks that separates signaling information from user data. A specified channel is exclusively designated to carry signaling information for all other channels in the system. The SS7 **32** is one of the standard CCS systems used by the telephone company. The SS7 **32** is connected to the central server **28**. The SS7 **32** normally connects between central offices. Because the present invention uses the SS7 **32**, it appears to be a central office to the telephone companies. The system of the present invention is actually a class 5 telephone office. A message store memory **33** is located at the central server **29** for storing messages. The central server **29** communicates through the internet **26** to the personal desktop computer **25** of the client. The central server **29** is shown as also communicating to a second client having a personal desktop computer **34**. A call merger **39** connects from the Internet **26** to a telephone instrument.

When an incoming call comes in to the home telephone number **30**, the central server **29** interacts with the caller. It makes a record of the caller's telephone number or prompts the caller to unblock the telephone so as to give the caller's telephone number. The central server **29** may give the caller options that can be answered by a touch tone response, or it may ask for a voice message to be passed on to the subscriber. If an audio message is given to the central server **29**, that message is passed along using Voice Over IP (VOIP) which is used in the telephone system and over the internet. The letters IP stand for Internet Protocol.

Teleconferencing over the internet is done using a standard developed by the International Telecommunications Union (ITU). This standard is known as ITU-T H.323. This provides for audio and video in a teleconferencing context. From the standpoint of VOIP, the video component of the teleconferencing signal is ignored and only the audio is used. This permits audio to be transferred from the incoming call at the home telephone number **30** to the client's personal desktop computer **25**. The client can listen to the message from the incoming call before making a decision as to the handling of the call. The server communicates to a command center running on a multiplicity of platforms and providing a control interface to the second entity. The command center is a software and device solution that can be hosted alternatively on a personal computer, a handheld computing device, a wireless telephone, a television, a web interface appliance, or a command center server using voice and DTMF tone interaction with a telephone device.

Referring now to FIG. **4**, there is shown a schematic diagram of the call handling process. Box **35** indicates the arrival of an incoming call. The options may include forward the call, call forward no-answer, forward the call always, and it may provide for switching the call to a different telephone number, for example that of a cell phone or other device. The caller may be presented with voice mail type options. For example, if you wish to talk with Mr. Jones, press 1, if you wish to talk with Mrs. Jones press 2, if you wish to talk with Susie Jones press 3. The caller may be required to give a touch tone ID, or to provide a touch tone ID of the called party. The dialog process is for the purpose of obtaining as much information as possible as to who is calling, which family member is being called, and why. The caller may be asked to speak a message into the telephone as would be done with an answering machine. This message is recorded and passed on to the subscriber so that he can listen to it to aid him in making the call handling decision. As shown at box **37**, the caller's telephone number is noted by the central server **29**, or the caller is prompted to unblock the telephone number. Box **36** indicates that a dialog is conducted by the central server **29** with the caller. Box **38** shows the call handling decision. The system identifies the first entity by the following methods: detecting caller ID and or called number information from the call information received from the switch, by means of voice prompts from the system and tone response from the first entity by which the first entity identifies their number, or the person whom they are calling, or by means of capturing an audible signal from the first entity.

The central server **29** may go through a process of locating the subscriber. This is illustrated in FIG. **5**. This service is sometimes referred to as find me/follow me. As indicated in FIG. **5**, the subscriber may have his home number logged on line for the Internet as indicated in block **40**. However, the subscriber may have an additional home number as shown in block **41** or the subscriber may have a business number as shown in block **42**. The subscriber may have a personal number as in block **43**, or a special number classed as a single purpose number as in block **44**. In addition the subscriber may have a cell phone **45** or a pager **46**.

It should be understood that as the central server **29** goes through the processes of locating the subscriber, the caller is not aware of any of the procedures that the central server **28** is going through. The caller is unaware of any of the special numbers that the subscriber may have, or equipment such as pagers or cell phones.

It is not necessary for the equipment used by the system such as the central server **29** to be located close to the subscriber or close to the called number. For example, as shown in FIG. **6**, the POTS Network **31** can be connected through a Telco central office **47** via the SS7 **32** by way of wireless connections **48** or by way of land line connections **50** to the server **29** at a remote location.

It should be understood that the message store memory **33** shown in FIG. **3** may include a list of numbers that the subscriber wishes to speak to always. These may be family members, or business partners, or the like. These may be added to at the subscriber's discretion. Other instructions for call handling may be recorded in the message store memory **33**. One of the first things done by the central server **29** is to check to see if there are special instructions for handling of a given incoming call. FIG. **7** shows a sequence of steps on a flow chart indicating the handling of a call. The first box **51** is "begin". Box **52** is "placing a call from a first entity to a second entity". Box **53** is "receiving the call at a local exchange switch". The next step is shown in box **54** as "identifying a number of the first entity". Box **55** is "directing the call and the number of the first entity to a gateway". Box **56** is "directing the call, call information, and the number of the first entity to a server". Box **57** is "determining whether the Command Center of the second entity is communicating with the server". Box **58** is "sending a message to the Command Center indicating the presence of a call and identifying the number of the first entity and the number of the second entity". Box **60** is "directing the call to one of a plurality of telephonic devices according to the number of the second entity" and the last block on FIG. **7** is box **61** "end".

FIG. **8** is a flow chart showing a series of steps in the handling of incoming telephone calls. The first block **62** is "begin". Block **63** is "receiving at the server a command from the command center to have the server answer the call". Block **64** is "answering the call through the server". Block **65** is "playing a message". Block **66** is "recording a message from the call". Block **67** is "storing the message from the call in the server". The last block is **68** "end".

FIG. **9** is a flow chart illustrating further steps in the call handling flow chart. The first block **70** is "begin". Block **71** is "determining whether special instructions are stored in the computer regarding forwarding calls". Block **72** is "forwarding the call to a number designated by the second entity". Block **73** is "end".

As has been indicated hereinbefore, an audible message from the caller may be played to the subscriber to aid him in making a decision for handling the call. However, the subscriber may also record messages, and these messages may be played to the caller as well. There may be a number of different messages depending upon the circumstances, and these may be selectively played as desired. Referring now to FIG. **10**, there is shown a sequence of steps in a flow chart. Block **74** is "begin". Block **75** is "recording a plurality of different audio greetings". Block **76** is "storing said audio greetings in a memory storage device". Block **77** is "playing a preselected one of said audio greetings to a caller on demand". Block **78** is "end". Referring now to FIG. **11**, this figure shows a flow chart for steps in deciding how the subscriber answers a call. The first block **80** is "begin". Block **81** is "developing a data base of important telephone numbers". Block **82** is "answering a call by the server only if the line is busy and the caller number is in the data base". Block **83** is "permitting the subscriber to monitor a message from the caller audibly over the Internet". Block **84** is "permitting the subscriber to make a decision how to handle the call over the Internet". Block **85** is "permitting the subscriber to specify: always, or this time only, connect the call, transfer the call, take a message, or terminate the call". Block **86** is "end".

The client or subscriber has many ways to deal with an incoming call. He can elect not to answer and to take a message, or he can have the call played through the personal desktop computer **25** and talk to the caller via the Internet **26**, or he can have the call transferred to a different instrument such as a second telephone **24** (FIG. **1**) or a cell phone **45** (FIG. **5**). There is another way for the subscriber to handle a call. While the central server **29** is holding the call, the client can pick up a cell phone or a regular POTS phone and call in to the central server **29** and have a call cross-connect right there. This is illustrated in FIG. **12**. This figure illustrates an incoming call arriving at the central server **29**. It is connected to the Internet **26** and to the client's personal desktop computer **25**. Then, the client may elect to make a POTS inbound call to the server **87**. This means, for example, picking up a cell phone or a regular POTS phone and dialing the number to the central server **29**. At the central server **29**, a call cross-connect is made, and the client can talk to the caller making the incoming call.

The central server **29** is able to capture and store the dynamic profile of the subscriber's rules for treatment of incoming calls. This is illustrated in FIG. **13**. FIG. **13** is a sequence of steps in a method flow chart. Block **90** is "begin". Block **91** is "monitoring and screening incoming calls based on properties such as call type, caller number, called number, time of day, family member called". Block **92** is "locating subscriber using find me, follow me, first or second home phone, business phone, cell phone and various Internet devices". Block **93** is "allowing subscriber to specify call treatment, e.g., connect the call, transfer the call, take a message". Block **94** is "observing and noting subscriber treatment of call". Block **95** is "profiling dynamically call treatment by subscriber". Block **96** is "reviewing by subscriber of suggested call handling rules". The subscriber is not asked to enter his rules for handling calls but rather the subscriber is presented with rules determined implicitly by the central server **28**. The subscriber than has an opportunity to approve or to reject the proposed rules. Block **97** is "capturing and storing dynamic profile of subscriber call treatment". This dynamic profile may be stored in the message store memory **33** of the central server **28**. In addition an address book may be compiled at that same location so that if the subscriber wishes to place a call, or to return a call following a telephone message, it is easy to initiate using the numbers logged in the address book in the message store memory **33** of the central server **28**. In FIG. **13**, the last block **98** is "end".

The central server **28** of the system of the present invention gradually becomes more useful to the subscriber as it learns the subscriber's profile. This profiling also adds another dimension to the system of the present invention. This is an advertising or media dimension. The profiling creates a community of users. People that call each other on the telephone have a community of interests. Thus the profiling information that is stored becomes, in effect, a collaborative filter based on telephone numbers. Recording of calls made to other numbers indicates similar patterns of behavior. This is based on shared interests. Thus, this gradual interactive development of a profile based on decisions made by the client may be used for advertising purposes, if desired.

FIG. **14** is a schematic diagram indicating how the system can handle undesired incoming calls. At the left of FIG. **14** there is illustrated an incoming call **35**. The call is coupled to a detector for detecting phone "spam" **100**. The incoming call **35** is also connected to a detector for detecting nuisance calls **101**. Both of these types of telephone calls are unsolicited, unwanted, and undesirable types of calls. When these types of calls are detected, they are forwarded to block **102** which initiates a "rude" hang-up. This type of hang-up would have an audible message to the caller saying something such as "never call again". Then, the signal is passed on to block **103** which terminates the call.

Typically, the system only answers calls for the client when the client's telephone line is busy and he is logged on the Internet. However it can be arranged for the system to answer all of the client's incoming calls. This is illustrated in FIG. **15**. FIG. **15** is a schematic diagram illustrating a second embodiment of the call control system of the present invention. In this embodiment of the invention, the hard connection from the POTS to the subscriber is broken. This is done by the central server system providing a virtual phone number to the client without providing a telephone instrument. This is illustrated in block **104**. The virtual telephone number is given to the client who uses it in his advertising so that all calls will be made to that number but in fact those calls will be intercepted by the system of the present invention.

The present invention has been particularly shown and described with respect to certain preferred embodiments and features thereof. However, it should be readily apparent to those of ordinary skill in the art that various changes and modifications in form and detail may be made without departing from the spirit and scope of the inventions as set forth in the appended claims. The inventions illustratively disclosed herein may be practiced without any element which is not specifically disclosed herein.

What is claimed:

1. A method of providing a called entity the ability to selectively accept phone calls, the method comprising:

receiving over a PSTN, at a call processing apparatus, a call from a first entity directed to a second entity's virtual telephone number;

opening a communication channel over the Internet with a networked computer associated with the second entity;

recording speech from the first entity and transmitting the speech over the Internet communication channel to the networked computer to be played by the networked computer to the second entity; and

receiving an instruction, over the Internet communication channel from the second entity via the networked computer, and at least partly in response to the received instruction, causing the first entity call to be selectively connected to one of a plurality of potential second entity terminal destinations.

2. The method as defined in claim **1**, wherein the first entity is connected to the PSTN via a local switch.

3. The method as defined in claim **1**, wherein the call processing apparatus is connected to the PSTN via a local switch.

4. The method as defined in claim **1**, wherein the second entity is connected to the Internet via a dial-up connection.

5. The method as defined in claim **1**, wherein the second entity is connected to the Internet via an always on connection.

6. The method as defined in claim **1**, wherein the second entity is connected to the Internet via a wireless connection.

7. A method of using Voice over Internet Protocol (VoIP) to provide a called party the ability to audibly screen calls via a networked computer, the method comprising:

receiving, at a first system, a call from a calling patty intended forte called party, wherein signaling information associated with the call includes the calling party's phone number;

playing a greeting to the calling party;

opening a communication channel over the Internet with a networked computer associated with the called party;

transmitting in substantially real time at least a portion of the signaling information for display to the called party on the networked computer; and

transmitting in substantially real time a VoIP voice communication from the calling party over the communication channel to the networked computer so that the called party can audibly listen to the voice communication in substantially real time and screen the calling part's call.

8. The method as defined in claim **7**, further comprising providing a user interface to the called party, wherein the user interface is configured to be displayed on only a portion of a display associated with the networked computer, and the display includes a control which ambles the called party to indicate that the called patty wants to take the call.

9. The method as defined in claim **7**, further comprising providing a user interface to the called party, wherein the user interface is configured to be displayed on a display associated with the networked computer, and wherein the user interface includes,

a first control which enables the called party to indicate that the called party wants to take the call using the Internet Protocol, and

a second control which enables the called party to indicate that the called party wants to transfer the call over a circuit switching network.

10. The method as defined in claim **7**, wherein the networked computer is connected to the called party's telephone line.

11. The method as defined in claim **7**, further comprising determining, via an application hosted on the networked computer, that the networked computer is connected to the called party's telephone line prior to transmitting the VoIP voice communication to the networked computer.

12. The method as defined in claim **7**, wherein the call is received at the first system via a Public Switched Telephone Network (PSTN).

13. The method as defined in claim **7**, wherein the call was forwarded to the first system by a call directing method selected from the group including busy call forward and no answer call forward.

14. The method as defined in claim **7**, wherein the call is received at the first system via a PSTN gateway.

15. The method as defined in claim **7**, wherein the networked computer shares a PSTN dial-up line with a POTs telephone.

16. A method of providing a called party the ability to screen calls using a wireless phone via an audible message from the calling party, the method comprising:

receiving, at a first system, a call from a calling party intended for the called party; playing a greeting via the first system to the calling party;

opening a communication channel with a wireless phone associated with the called party; and

transmitting in substantially real time via the first system an audible voice communication from the calling party over the communication channel to the wireless phone so that the called party can listen to and screen the audible voice communication in substantially real time.

17. The method as defined in claim **16**, wherein the wireless phone is configured to provide a first control enabling the acceptance of the call at the wireless phone.

18. The method as defined in claim **16**, wherein the wireless phone hosts a command center application configured to provide a first control enabling the transfer of the cell.

19. The method as defined in claim **16**, wherein the call was forwarded to the first system by a call directing method selected from the group including busy call forward, no answer call forward, call forward all, and telephone number portability.

20. The method as defined in claim **16**, further comprising sending a message to a user interface over an Internet Protocol connection indicating the presence of a call.

21. The method as defined in claim **16**, further comprising receiving via a called party user interface a call handling instruction selected from a group including:

transfer call over a circuit switching network, and

take call over an Internet Protocol network.

22. The method as defined in claim **16**, further comprising recording the voice communication as the tint system for later retrieval by the called party.

23. The method as defined in claim **16**, wherein the wireless phone is a cellular phone.

24. A method of enabling a called party to audibly screen calls via a device having a call management user interface displayed on a screen, the method comprising:

receiving, at a first system, a call from a calling party intended for the called party;

playing a greeting to the calling party;

opening a communication channel to a communication device having a display, wherein the display displays a user interface including,

a first control that, when selected by the called party, causes the first system to enable the called party to talk to the caller;

a second control that, when selected by the called party, causes the first system to transfer the call to another communication device;

transmitting in substantially real time an audible voice communication from the calling party over the communication channel to the communication device so that the called party can listen to the audible voice communication in substantially real time and screen the calling party's call; and

receiving an indication at the first system that the called party selected the first control or the second control while the voice communication from the calling party is being transmitted in substantially real time.

25. The method as defined in claim **24**, wherein the communication device is a wireless telephone.

26. The method as defined in claim **24**, wherein the voice communication is transmitted in substantially real time using a voice over Internet protocol.

27. The method as defined in claim **24**, wherein selection of the second control causes the call to be transfered over the Internet to another communication device.

28. The method as defined in claim **24**, wherein selection of the second control causes the call to be transferred over public switched telephone network to another communication device.

29. The method as defined in claim **24**, wherein the call was forwarded to the first system by a call directing method selected from a group including busy call forward and no answer call forward.

30. The method as defined in claim **24**, wherein the user interface providing the first control and the second control is no greater than two square inches in size.

31. A method of locating a called party and providing the called party the ability to audibly screen calls, the method comprising;

receiving, at a first system, a call from a calling party intended for the called party;

playing a greeting to the culling party;

retrieving from computer readable memory associated with the first system stored information corresponding to a plurality of terminals associated with the called party;

determining at which of the plurality of terminals the called party is currently located;

opening a communication channel with the terminal the called party is currently located at; and

transmitting in substantially real time a voice communication from the calling party over the communication channel to the terminal at which the called party is currently located so that the called party can listen to the voice communication in substantially real time and screen the calling party's call.

32. The method as defined in claim **31**, wherein the information corresponding to a plurality of terminals includes a home number, a business number, and a cell phone number.

33. The method as defined in claim **31**, wherein the called party is not aware of the process of determining at which of the plurality of terminals the called party is currently located and opening a communication channel with the terminal the called party is currently located at.

34. The method as defined in claim **31**, wherein the called party is not aware of the phone number associated with the terminal the called party is currently located at.

35. The method as defined in claim **31**, wherein the terminal the called party is currently located at is a cell phone.

36. The method as defined in claim **31**, wherein the voice communication is transmitted in substantially real time using VoIP.

37. The method as defined in claim **31**, wherein the voice communication is transmitted in substantially real time over a wireless network.

38. The method as defined in claim **31**, wherein the voice communication is transmitted in substantially real time over the public switched telephone network.

39. The method as defined in claim **31**, wherein a user interface is displayed on the terminal the called party is currently located at, the user interface providing a control via which the user can indicate that the call is to be forwarded to another terminal.

40. The method as defined in claim **31**, wherein a user interface no greater than two inches square is displayed on the terminal the called party is currently located at, the user interface providing a control via which the user can indicate that the call is to be forwarded to another terminal.

41. A method of providing a called party the ability to screen calls that are forwarded to a call processing system, the method comprising:

receiving at a call processing system a call from a first entity intended for a second entity that was forwarded via a PSTN, wherein the call was forwarded by a call directing method selected from the group including busy call forward and no answer call forward; playing a greeting to the first entity;

opening a communication channel with a first communication device associated with the second entity; and

transmitting in substantially real time via the first system a voice communication from the first entity over the communication channel to the communication device so that the second entity can listen to and screen the voice communication in substantially real time.

42. The method as defined in claim **41**, wherein the voice communication is transmitted in substantially real time using Voice over Internet protocol.

43. The method as defined in claim **41**, wherein the first communicating on device is a wireless phone.

44. The method as defined in claim **41**, wherein the system includes a database that stores contact information for a plurality of telecommunication devices associated with the called party.

45. The method as defined in claim **41**, wherein a user intertface is displayed on the communication device the called party is currently located at the user interface providing a control via which the user can indicate that the call is to be forwarded to another communication device.

46. The method as defined in claim **41**, wherein a user interface no greater than two inches square is displayed on the communication device the second entity is currently located at, the user interface providing a control via which the user can indicate that the call is to be forwarded to another communication device.

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