

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

(19) World Intellectual Property Organization
International Bureau



(43) International Publication Date
4 December 2003 (04.12.2003)

PCT

(10) International Publication Number
WO 03/101079 A1

(51) International Patent Classification⁷: H04M 3/42

(21) International Application Number: PCT/US03/15807

(22) International Filing Date: 19 May 2003 (19.05.2003)

(25) Filing Language: English

(26) Publication Language: English

(30) Priority Data:
60/382,257 20 May 2002 (20.05.2002) US
10/439,601 16 May 2003 (16.05.2003) US

(71) Applicant: CALLWAVE, INC. [US/US]; 136 W. Canon Perdido St., Santa Barbara, CA 93101-8207 (US).

(72) Inventors: BRAHM, David; 137 Por La Mar Cr., Santa Barbara, CA 93101 (US). TRANDAL, David; 4007 Primavera Road, Santa Barbara, CA 93110 (US). SMITH, Robert; 2609 Elmira Street, Newbury Park, CA 91320 (US).

(74) Agent: ALTMAN, Daniel, E.; Knobbe, Martens, Olson & Bear, LLP, 2040 Main Street, 14th Floor, Irvine, CA 92614 (US).

(54) Title: SYSTEMS AND METHODS FOR CALL SCREENING

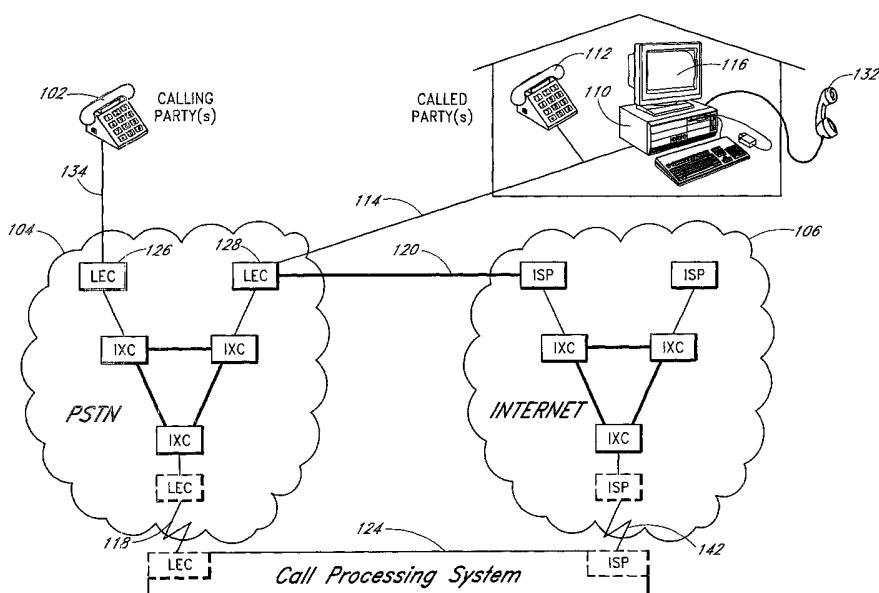
(81) Designated States (national): AE, AG, AL, AM, AT, AU, AZ, BA, BB, BG, BR, BY, BZ, CA, CH, CN, CO, CR, CU, CZ (utility model), CZ, DE (utility model), DE, DK (utility model), DK, DM, DZ, EC, EE (utility model), EE, ES, FI (utility model), FI, GB, GD, GE, GH, GM, HR, HU, ID, IL, IN, IS, JP, KE, KG, KP, KR, KZ, LC, LK, LR, LS, LT, LU, LV, MA, MD, MG, MK, MN, MW, MX, MZ, NI, NO, NZ, OM, PH, PL, PT, RO, RU, SC, SD, SE, SG, SK (utility model), SK, SL, TJ, TM, TN, TR, TT, TZ, UA, UG, UZ, VC, VN, YU, ZA, ZM, ZW.

(84) Designated States (regional): ARIPO patent (GH, GM, KE, LS, MW, MZ, SD, SL, SZ, TZ, UG, ZM, ZW), Eurasian patent (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European patent (AT, BE, BG, CH, CY, CZ, DE, DK, EE, ES, FI, FR, GB, GR, HU, IE, IT, LU, MC, NL, PT, RO, SE, SI, SK, TR), OAPI patent (BF, BJ, CF, CG, CI, CM, GA, GN, GQ, GW, ML, MR, NE, SN, TD, TG).

Published:

— with international search report

For two-letter codes and other abbreviations, refer to the "Guidance Notes on Codes and Abbreviations" appearing at the beginning of each regular issue of the PCT Gazette.



WO 03/101079 A1

(57) Abstract: The present invention provides flexible, user-definable call screening processes. The user (112) can optionally define to which telecommunication terminals a screened call is to be broadcast to and under what conditions. An incoming call is forwarded to a call management system (124) that asks the caller (102) to leave a voice message. The call management system (124) selectively couples the call to a POTS line or a VoIP-capable device (110) so that the user (112) can listen to the incoming message and thereby screen the incoming call. Based on the screening, the user (112) can instruct the call management system (124) to connect the caller (102) to the user (112).

SYSTEMS AND METHODS FOR CALL SCREENING

Background of the Invention

Field of the Invention

5 The present invention relates generally to telecommunications, and in particular to systems and methods for screening telephone calls.

Description of the Related Art

Conventional telephone systems often offer a Telephone Answering Service (TAS) that redirects incoming calls encountering a ring-no-answer condition, a busy condition, or a do-not-disturb condition, to a 10 network voice messaging system on which the caller can record a message for the called party. The called party is then provided with a Message-Waiting-Indicator (MWI). In many conventional systems, the MWI notification is in the form of a stutter dial tone or a flashing light on the called party's telephone. Upon detection of this indicator, the called party can dial into the voice messaging platform to retrieve the recorded message from his/her mailbox.

15 Alternatively, many residential telephone customers equip their homes with a Telephone Answering Machine (TAM) that automatically answers their phone and takes a message when an incoming call is not answered within the first three or four ring cycles. The TAM plays the caller's message over its speakers so that the call can be screened and if desired, picked up by the called party to initiate a two-way conversation. If the call is not picked up, the TAM provides a MWI notification, usually by illuminating a lamp on the TAM 20 device. Once again, the called party, upon detection of this indicator, can retrieve the recorded message from his/her mailbox.

The above described two classes of conventional automated telephone call answering solutions have distinct advantages and disadvantages. The TAS handles busy as well as unanswered calls but does not allow message screening. The TAM allows screening of unanswered calls but does not handle busy calls. 25 In addition, neither solution provides a timely notification of calls missed when the phone line is tied up while the called party is surfing the Internet on a dialup connection.

A more recent call answering service called the Internet Answering Machine (IAM), provided by CallWave, Inc., works with the "Call Forward On Busy" feature of the called party's phone line to answer calls while the called party is using the phone line to access the Internet via the called party's computer. Once 30 activated, callers no longer get annoying busy signals when the called party is online. Instead, callers hear a greeting after which they can leave a short message. The caller's phone number and message are transmitted in near real-time to the called party's computer so that the called party can screen the call and optionally choose to interact with the caller during the call. For example, the called party could choose to

answer the call, continue screening on an alternate telephone, or request that a telemarketer blocking message be played to the caller.

Some recent TAS systems provide call screening while recording a message from a caller. However, many of these conventional call screening methods disadvantageously only allow a call to be screened using a particular phone line of the called party, which may not be the most desirable phone line for performing the call screening operation. In addition, these conventional methods generally are not capable of providing the called party with the caller's Caller-ID. Further, these conventional methods may not allow calls to be screened while the particular phone is being used by the called party to access the Internet or the like via their computer. Lastly, these services typically require tight coupling with the called party's local telephone switch, thereby limiting the breadth of the serving territory that can be covered by a single system.

Summary of the Invention

Embodiments of the present invention are directed methods and systems for providing call screening in conjunction with a variety of network-based telephone call answering processes and services.

In one example embodiment, an Internet Answering Machine (IAM) system allows a called party to monitor a message being left by a caller even when the called party is connected to the Internet over a dial-up connection. Calls are forwarded from the called party's line to the IAM system using the called party's local phone company's fixed and variable call forward functions. These functions can include, by way of example, call forwarding on busy, call forwarding on ring-no-answer, and call-forward-all calls (sometimes called do-not-disturb). The IAM system answers the forwarded calls and plays a greeting to the caller. At the same time, a communication channel is opened with the called party over the public Internet and speech is "streamed" to the called party and played over the speakers of the called party's computer, which may be, by way of example, a personal computer or networked television. With streaming, a client application executing on the called party's computer can start playing the transmitted speech data in substantially or almost real-time, before an entire speech data file of a caller's message has been transmitted. In particular, the Internet channel is opened at the time a call arrives at the call answering system so the called party hears the caller's speech during the playing of the greeting.

If the called party wishes, the called party can instruct the IAM system to pickup the call by linking the called party to the caller. The IAM system interrupts the caller, who may be in the process of leaving a message, by playing a voice prompt, such as "please hold while we connect your call." The call management system causes the called party computer to be disconnected from the Internet, originates a new call from the IAM system to the called party's POTS (plain old telephone service - which refers to the standard telephone service that most homes use) phone, and bridges the two calls together.

In another embodiment, rather than opening a channel over the Internet to the called party's computer, a second call is selectively originated upon the arrival of the forwarded call to a second POTS Public Switched Telephone Network (PSTN) phone line or the called party's wireless/cellular phone. The IAM

system determines which of the POTS lines and cellular lines to call and which calls are to be forwarded based on a set of rules defined by the called party. These rules can include online and offline status (Internet presence), telephone presence (called party on the phone line/off the phone line), VIP Caller-ID filtering (calling number), called number, time of day, day of week, and other parameters. When the called party 5 answers, a brief greeting is played and the called party can monitor and interact with the caller as described above.

In still another embodiment, when the call is originated from the IAM system to the POTS phone line or wireless device, the call is originated, using by way of example the SS7 protocol, to the line with the calling party ID of the "original caller". Having the original calling party's number delivered with the outbound call and 10 then displayed on the POTS line or wireless phone display can help the called party decide how to handle the call.

In yet another embodiment, rather than the IAM system receiving only forwarded calls, the called party can selectively publish a unique phone number that terminates calls directly to the IAM system. The called party can monitor and selectively interact with their callers as described above.

15 In one embodiment, the call screening information is simultaneously multi-cast to multiple telephone and IP devices. Any one of the multi-cast destination devices can directly interact with the caller during the call.

In another embodiment, a method of providing a called party the ability to screen calls comprises: receiving over a switched network at a call manager system a forwarded call from a calling party intended for 20 the called party, wherein signaling information associated with the forwarded call includes the calling party's phone number; playing a greeting to the calling party; originating a second call from the call manager system to the called party, wherein signaling information associated with the second call includes the calling party's phone number so that the second call appears to be originating from the calling party; and bridging the forwarded call with the second call.

25 In yet another embodiment, a method of processing calls comprises: receiving over a switched network at a call processing system a first call from a caller intended for a called party, wherein the first call includes signaling information having a phone number of the caller; and placing a new call from the call processing system over the switched network to a terminal associated with the called party, the new call including signaling information having at least a selected portion of the phone number of the caller so that the 30 called party can determine the identity of the caller of the first call and thereby screen the caller.

In still another embodiment, a method of providing a called party the ability to selectively accept phone calls comprises: receiving at a call processing system a first call from a calling party intended for the called party, wherein the first call includes at least a first portion of the calling party's phone number; and initiating a second call from the call processing system to the called party, the second call including at least

part of the first portion of the calling party's phone number to thereby provide the called party with information related to the identity of the calling party so that the called party can screen the first call.

In one embodiment, a call screening apparatus comprises: a first instruction stored in computer readable memory, the first instruction configured to cause a call from a calling party intended for a called party to be answered; a second instruction stored in computer readable memory, the second instruction configured to play a greeting to the calling party; a third instruction stored in computer readable memory, the third instruction configured to maintain a communication channel over the Internet with a networked computer associated with the called party while the greeting is being played; a fourth instruction stored in computer readable memory, the fourth instruction configured to receive and stream speech from the calling party over the Internet communication channel to the networked computer, wherein the streamed speech is intended to be screened by the called party by the networked computer; a fifth instruction stored in computer readable memory, the fifth instruction configured to receive a command, via the Internet communication channel, from the called party to connect the calling party to the called party; a sixth instruction stored in computer readable memory, the sixth instruction configured to cause the called party's networked computer to go offline; and a seventh instruction stored in computer readable memory, the seventh instruction configured to originate a second call from the call manager system to the called party, and to bridge the calling party's call with the second call.

In another embodiment, a method of providing a called party the ability to screen calls comprises: receiving at a first call processing apparatus a call from a first user for a second user; receiving a voice communication from the first user at the first call processing apparatus; and multi-casting at least a portion of the voice communication to a plurality of client devices at substantially the same time so that the first user's call can be screened.

In still another embodiment, a method of providing a called party the ability to screen calls comprises: receiving at a first call processing apparatus a call from a first user for a second user; and multi-casting a call alert to a plurality of client devices at substantially the same time so that the first user's call can be screened.

Brief Description of the Drawings

Embodiments of the present invention will now be described with reference to the drawings summarized below. These drawings and the associated description are provided to illustrate example embodiments of the invention, and not to limit the scope of the invention.

Figure 1 illustrates an example telecommunications system that can be used in accordance with one embodiment of the present invention.

Figure 2 further details the subsystems that comprise the IAM system depicted in Figure 1 described above.

Figure 3 displays an example menu of call screening/handling options available to the called party during the processing of the inbound call.

Figures 4A-4B illustrate a first example call screening process in accordance with the present invention.

5 Figures 5A-5B illustrate a second example call screening process in accordance with the present invention.

Throughout the drawings, like reference numbers are used to refer to items that are identical or functionally similar.

10 Detailed Description of Preferred Embodiments

The present invention provides systems and methods for call screening. As will be described in greater detail below, in one embodiment, an IAM system allows a called party to monitor a message being left by a caller even when the called party is using a phone line to access a computer network, such as the Internet.

15 Throughout the following description, the term "Web site" is used to refer to a user-accessible network site that implements the basic World Wide Web standards for the coding and transmission of hypertextual documents. These standards currently include HTML (the Hypertext Markup Language) and HTTP (the Hypertext Transfer Protocol). It should be understood that the term "site" is not intended to imply a single geographic location, as a Web or other network site can, for example, include multiple geographically 20 distributed computer systems that are appropriately linked together. Furthermore, while the following description relates to an embodiment utilizing the Internet and related protocols, other networks, such as networked interactive televisions, and other protocols may be used as well. In addition, unless otherwise indicated, the functions described herein are preferably performed by executable code and instructions running on one or more general-purpose computers. However, the present invention can also be 25 implemented using special purpose computers, state machines, and/or hardwired electronic circuits. In addition, a communications line is referred to as "busy" when the communication line is being utilized in such a way that a conventional incoming call will not be connected to the communications line. Thus, for example, if a user is utilizing a conventional line capable of only conducting one of a conventional voice session and a data session, but not both at the same time, for a data session, the line will be busy.

30 Figure 1 illustrates an example telecommunications system that can be used in accordance with the present invention. As illustrated, the telecommunications system includes:

- a plurality of user telephone stations 102, 112.
- a plurality of user computer terminals 110.
- a call processing system 124 that acts as an Internet Answering Machine (IAM) system.

These devices are linked together using various line and trunk circuits to a Public Switched Network (PSTN) 104 and to a common data network, such as the Internet 106.

Figure 2 further decomposes the IAM system 124 into its functional components:

- a Call Management (CM) subsystem 108, which serves as the interface to the PSTN 104 to manage inbound and outbound telephone calls.
- a Router subsystem 140, which serves as the interface to the Internet 106 to manage communications between online IP client devices and the various IAM servers.
- an online presence detection Internet Session Management (SM) subsystem 122, which monitors the status of subscriber data terminals to determine availability for call handling services.
- a shared Media Storage (MS) subsystem 138, which persistently archives the callers voice messages and the called party/subscriber's personal greeting(s).
- an IAM Database (DB) subsystem 136 in which called party/subscriber IAM service parameters are stored.

Theses various subsystems are interconnected via a Local Area Network (LAN) and/or via a Wide Area Network (WAN). Other embodiments of the IAM system 124 are described in U.S. Patent Application 09/539,375, filed March 31, 2000, now U.S. Patent No. 6,477,246, the contents of which are incorporated herein in their entirety by reference.

CallWave, Inc. operates one such IAM system. As is well known in the field of Internet telecommunications, an IAM service works with the "Call Forward On Busy" feature of a standard phone line to answer calls while the subscriber is online and is using the phone line to access the Internet. Once activated, callers no longer get annoying busy signals when the subscriber is online. Instead, callers hear a brief greeting after which they can leave a short message. The recording can be streamed in substantially real-time or sent to the subscriber over the Internet within seconds after the recording has completed. Just like a home telephone answering machine, the subscriber can elect to interact with the caller while they are still on the line or can call them back at a later time.

Referring back to Figure 1, the user telephone stations 102, 112 are respectively connected to local exchange switches 126, 128 via telephone lines 134, 114. The stations 102, 112 can optionally be conventional POTS (Plain Old Telephone Service) telephones or local extensions behind a corporate Private Branch Exchange (PBX).

The telephone stations 102, 112 can be coupled to the same switch or different switches. If the telephone stations 102, 112 are coupled to the same switch, the switch will be local to both the calling and called parties, such as for intra-LATA or local calls. If telephone stations 102, 112 are coupled to different switches, each switch may be local only to one of the parties, as is the case for non-local calls such as inter-LATA (long-distance) calls.

In the illustrated embodiment, the CM subsystem 108 is coupled into the PSTN 104 through voice trunk circuits 118 directly interfacing with the Inter Exchange Carrier's (IXC) circuit switched or packet switched telephony network. Thus, advantageously the IAM system 124 does not have to be directly serviced by the same Local Exchange Carrier's (LEC) switch or PBX as the calling or called terminals 102 and 112.

5 Indeed, the IAM system 124 or its individual subsystem components can be located in a different country than the called and calling parties. In this instance, the IAM system 124 is optionally configured as, or to appear as, a telephone end office and can interface with the PSTN 104 as a Class 5 switch. In other embodiments, the IAM system 124 is locally attached to a LEC switch with a physical line or local trunk interface circuit. This switch may or may not be serving telephone stations 102 and/or 112.

10 The IAM voice trunk circuits 118 are not limited to a particular signaling convention. For example, the present invention can be utilized with a Common Channel Signaling system, such as Signaling System 7 (SS7), having separate voice/user data and signaling channels. In addition, the present invention can be used with other signaling methods, such as the following trunk-side signaling interfaces: ISDN-PRI; Advanced Intelligent Network; and/or Service Node architectures. Preferably, the selected signaling system provides a

15 suite of call presentation information to the IAM system 124, including one or more of:

- ANI – Automatic Number Identification: phone number and privacy indicator of the calling party ("Caller-ID").
- DNIS – Dialed Number Identification: phone number of the IAM system's voice trunks 118 that the call was forwarded to.
- OCN – Original Called Number Identification: phone number of the original called party (subscriber to the IAM service).
- Call Type – Forwarded call due to a BCF, RNA, or DND/CFA condition. In addition, directly dialed inbound calls can be handled as well. In this instance, the caller will be required to implement a second stage of dialing to enter the subscriber's phone number or the subscriber could be assigned a unique personal number that is directly dialed by their callers.

20 The telephone lines 134, 114 may be shared with one or more computer terminals. For example, telephone terminal 112 shares the telephone line 114 with a computer terminal 110. While in the illustrated example the computer terminal 110 is a personal computer, the computer terminal 110 can be an interactive television, a networked-enabled personal digital assistant (PDA), other IP (Internet Protocol) device, or the like. Alternatively, the computer terminal 110 can be a personal computer having a monitor, keyboard, a mouse, a disk drive, sound card or similar sound reproduction circuitry such as a codec, streaming media playback software, such as the Media Player program available from Microsoft, speakers, and a modem, such as a standard V.90 56K dial-up modem. The modem can optionally be configured to dial-up a number under

control of an application, such as a contact manager application or telecommunications client application phone dialer, stored and executing on the computer terminal 110.

The telephone line 114, can be used to establish a dial-up connection for computer terminals, such as terminal 110 via the computer modem, to an Internet Service Provider (ISP) offering dial-in remote access 5 service connections from the PSTN 104 via trunk interface circuits 120. The computer terminal 110 can also be connected to the Internet 106 via a broadband connection, such as a DSL line, a television cable line, or a T1 line.

In addition, the computer terminal 110 can be equipped with a Voice over Internet Protocol (VoIP) 10 software module and a headset or a handset 132, including a microphone and speaker, allowing voice communications to be conducted over a computer network, such as the Internet 106. VoIP communicates information via packet switching, which opens a connection just long enough to send a small packet of data. Each packet includes a destination address informing the network where to send the packet along with the 15 actual voice data payload. If the receiving station is also a VoIP terminal, then when the receiving terminal receives the packets, VoIP software executing on the receiving terminal reassembles the packets into the original data stream. The data stream is then converted to a voice signal. If the receiving station is a conventional telephone, then a VoIP gateway converts the packets into a voice signal that is then connected to the PSTN 104.

In one embodiment, the VoIP process is performed using the H.323 standardized protocol established by the International Telecommunications Union (ITU).¹ Advantageously, H.323 provides 20 specifications for real-time, interactive videoconferencing, data sharing and audio applications such as IP telephony. Alternatively, the Session Initiation Protocol (SIP), established by the Internet Engineering Task Force (IETF), can be used. SIP is generally more efficient than the H.323 protocol as SIP is specifically intended for IP telephony. Alternatively, proprietary protocols could be deployed where multi-vendor interoperability is not required.

25 Optionally residing and executing on the computer terminal 110 is a communications management Client application 116. The Client application 116 is used to provide enhanced communication services, as discussed in greater detail below. The Client application 116 is connected to and communicates with the IAM system 124 via the Internet 106, other public wide area computer networks, or the like.

The IAM system 124 optionally hosts a Web site used by subscribers of the IAM service to setup and 30 manage their accounts, to view information about incoming calls, and to instruct the IAM system 124 on how to route incoming calls to one or more destination stations. Many of these same functions can be implemented by the Client application 116 as well.

The CM subsystem 108 manages communications with the Client application 116 and with forwarded calls. The CM subsystem 108 can interact with callers and called parties through voice prompts, 35 voice commands, and/or DTMF touch-tone entries. The CM subsystem 108 is optionally configured to

perform additional functions, such as acting as a telephone answering system that answers calls, playing outgoing greetings and announcements, recording incoming messages, and bridging calls. In addition, as will be described in greater detail below, the CM subsystem 108 further provides a call screening process.

The SM subsystem 122 monitors the Internet for online IP devices registered to IAM subscribers to 5 determine their availability for handling inbound call screening and call handling services. When a user or subscriber connects to the Internet using, for example, a dial-up ISP, the Client application 116 executing on the subscriber's computer terminal 110 makes the subscriber's online presence known to the IAM system 124. Presence detection can be performed by the SM subsystem 122 polling or pinging the computer terminal 110 via the telecommunications Client application 116, or by the telecommunications Client 10 application 116 transmitting a "Login/I'm alive" message and subsequent periodic "keep alive" messages to the SM subsystem 122. Just prior to the normal termination of the online Internet session, the Client application 116 sends a "Logout" message to the SM subsystem 122. Abnormal Internet session termination conditions are detected by the SM subsystem 122 timing out the expected Client "Keep alive" message.

If, rather than using a dial-up connection, the user or subscriber is using a broadband, always on- 15 connection, such as via a DSL line or cable modem, the Client application 116 becomes active when the computer 110 is turned on or powered up and stays on until the user manually shuts down the Client application 116, or the computer 110 is turned off or powered down.

Figures 4A-4B illustrate one example embodiment of the present invention, including an abstraction 20 of the previously described telecommunications system and an example call flow diagram. For clarity, the detailed breakout of the network elements and individual subsystems of the IAM system 124 illustrated in Figures 1 and 2 are not shown in Figure 4A. In this example, the calling party is associated with telephone terminal 102 and the called party is associated with terminal 112. In this embodiment, the called party is subscribed to an Internet call answering service that forwards calls to the remote IAM system 124 upon the occurrence of selected conditions, wherein the IAM system 124 transmits a notification to the called party 25 regarding the call.

With reference to Figure 1, the called party's station 102 has been configured with the local switching system 128 to forward calls on busy (BCF), ring-no-answer (RNA), or do-not-disturb (DND) to the voice trunk 30 circuits 118 connecting the CM subsystem 108 to the PSTN 104. The calling party initiates a call using the calling party telephone station 102 by dialing the number of a called party's phone line 114. The PSTN 104 routes this call to the called party's local switching system 128 causing the called party's telephone terminal 112 to either ring or to forward the call immediately if the line 114 is busy or set to do-not-disturb. If, for example, the called party does not answer within a certain amount of time or after a certain amount of rings, the associated switching system 128 detects a no-answer condition and invokes a switch operation command termed "call forwarding on RNA". The call is then forwarded to a phone number of the CM subsystem 108.

Based at least in part on the OCN of the forwarded call (i.e. the original called party's phone number), the CM subsystem 108 queries the SM subsystem 122 to determine whether the called party is a registered subscriber, is online or offline, and what the subscriber's call handling preferences are. If the called party's computer 110 is online, the CM subsystem 108 opens a communication channel over the public Internet 106 to the Client application 116 running on the called party's computer terminal 110. The Caller-ID of the calling party, if available, and if not designated as private, is transmitted to the Client application 116 and is displayed to the subscriber along with an optional sound notification. The sound notification can be in the form of ringing produced using the called party's computer terminal 110 speakers.

The CM subsystem 108 proceeds to play a greeting to the calling party. The greeting can be a "canned" greeting or a personalized greeting previously recorded by the subscriber and stored in the MS subsystem 138. The CM subsystem 108 records and stores the caller's message in the MS subsystem 138, while simultaneously "streaming" the message speech through the opened Internet channel to the Client application 116 on the called party's computer terminal 110. The Client application 116 uses the computer terminal's codec to play the streamed speech through the speakers on the called party's computer terminal 110, thereby allowing the called party to listen to and screen the call. Optionally, to prevent the calling party from hearing any sounds made by the called party during the screening process, the audio return path over the Internet channel to the CM subsystem 108 is muted.

While monitoring the Caller-ID of the incoming call, via the Incoming Call field illustrated in Figure 3 for example, and listening to the corresponding streaming message, the called party is presented with one or more of the following options (see Figure 3 which depicts an example Client application popup dialog menu):

1. do nothing.
2. pickup (answer) the call to talk to the caller using a software telephone running on the "home PC" (the computer terminal 110).
3. pickup (answer) the call to talk to the caller using the "home phone" on the phone line used to connect to the Internet (the user telephone station 112).
4. pickup (answer) the call to talk to the caller after transferring the call to an alternate phone or to an alternate PC.
5. continue screening the call after transferring it to an alternate phone or to an alternate PC.
6. terminate the call substantially immediately – with a do not disturb message.
7. do not answer the call.

The called party may choose to ignore the incoming call. For example, the call may not have been urgent enough to interrupt what they are doing or the call may have been intended for another member of the household. Under option (1), the called party can close the call handling options dialog box illustrated in Figure 3 using the "CLOSE" option, thereby informing the IAM system 124 that no further instructions for caller interaction will be forthcoming. Alternatively, the called party, having screened the Caller-ID of the

incoming call and/or the associated caller's message, can simply continue doing what they were doing before the call arrived. After the caller has left a complete message, as indicated by the caller terminating the call or after a predetermined recording time period, the IAM system 124 downloads the recorded message to the subscriber's computer terminal 110 and updates the Client application's call log, which lists the calls handled by the IAM system 124 for the called party. The message is archived in the MS subsystem 138 and is also available locally on the computer terminal 110 for playback at the called party's convenience.

Under option (2), the called party may decide to pickup the call in progress to talk to the calling party using the computer terminal 110. Having screened the call, the called party can signal the IAM system 124 to indicate a desire to talk to the calling party using VoIP. For example, the called party can activate the "HOME PC (VOIP)" option displayed in Figure 3. After the called party has selected option (2), the Client application 116 sends an instruction by way of an Internet-based client/server control message to the IAM system 124. Upon receiving the instruction, the IAM system 124 interrupts the recording and streaming process and plays a canned audio prompt to the calling party. The audio prompt can be, for example, "please hold while your call is being connected," followed by audible ringing. The IAM system 124 then bridges, in full duplex mode, the inbound call from the calling party to the CM subsystem 108 with the outbound VoIP call from the CM subsystem 108 to the called party computer 110.

The IAM system 124 will stay bridged between the calling party and called party for the duration of the call and may respond to internal events or called party actions. For example, the IAM system 124 can selectively interrupt the bridged call if a time limit is exceeded and play an announcement to notify the calling party and/or the called party that the call will be terminated shortly. The IAM system 124 can also initiate or transmit a warning message directly to the Client application 116 that then displays a visual notice regarding call termination or the like on the called party's computer terminal 110.

Figures 4A-4B illustrate an example call process workflow that can be used when a called party is online and can answer screened calls via a VoIP session. In this example, after screening the call, the called party agrees to talk directly to the caller. Of course, after screening the call the called party could have elected to decline the call. With reference to Figures 4A-4B, at state 401, the calling party phone 102 (hereinafter, referred to as the "calling party") calls the called party phone line 114 connected to the telephone 112 and computer 110. In this example, the computer 110 is using the phone line 114 to access the Internet; i.e. the computer is online and hence the phone line is busy.

At state 402, the PSTN 104 detects that the called party phone line 114 is busy. At state 403, in accordance with a call forwarding service, the PSTN 104 forwards the call on busy to the IAM system 124 via the voice trunk circuits 118. At state 404, the IAM system 124 transmits an incoming call alert to the computer 110 that is displayed to the called party by the Client application 116. At the same time or shortly thereafter, at state 405 the IAM system 124 answers the forwarded incoming call. At state 406, the PSTN 104 establishes a full duplex, 2-way talk path with the calling party. At state 407, the IAM system 124 plays a

greeting to the calling party. At state 408, the calling party optionally begins leaving a voice message that is recorded by the IAM system 124. Alternatively, similar to a telephone answering machine, the calling party can begin speaking to the called party even while the IAM system 124 is playing the greeting. At state 409 the IAM system 124 begins streaming the message being left by the calling party in substantially real-time to the Client application 116 or other media player executing on the computer 110, that then plays the message to the called party.

At state 410 of Figures 4A-4B, the IAM system 124 generates a tone or other audio signal to indicate to the calling party that the calling party should begin recording a message. At state 411, the calling party begins leaving a voice message. At state 412, the IAM system 124 begins streaming the message being left by the calling party in substantially real-time to the Client application 116 or other media player executing on the computer 110, which plays the message to the called party.

At state 413, the called party notifies the IAM system 124 that the called party wants to take the call. At state 414, the IAM system 124 interrupts the calling party, via a tone or voice notification. At state 415, the IAM system 124 requests that the calling party hold or wait while the IAM system 124 connects the calling party to the called party. At state 416, the IAM system 124 bridges the calling party with the called party computer 110, via the VoIP software module 130, by establishing a VoIP session. This entails bridging the two calls together through the IAM system so that the caller and the called party can converse (state 417):

- the inbound call from the calling party 102 connected into the CM subsystem 108 through the PSTN 104; is bridged with
- the outbound call from the CM subsystem 108 connected through the Router subsystem 140 and the Internet 106 to the VoIP session running on the subscriber's computer terminal 110.

Either party can terminate the call at state 418 by hanging up their telephone (calling party station set 102, or the called party can terminate the VoIP session on the computer terminal 110). At state 419, the IAM system 124 releases the bridging resources and signals completion of the call by sending a corresponding control message to the called party which is either displayed or played to the called party via the Client application 116.

In another embodiment, the SM subsystem 122 detects the presence of the called party on different IP devices, such as other computers or web-enabled cellular phones, at other locations. For example, the session manager SM subsystem 122 optionally interfaces with other instant messaging services, such as:

- AOL®'s Instant Messenger™,
- MSN®'s Instant Messenger™,
- Yahoo!® Messenger,
- ICQ

where presence of the called party can be detected on other IP networks and at other geographic locations. The same call/session dialog described above is similarly performed in this embodiment.

Under option (3), the called party may decide to pickup the call in progress to talk to the calling party via a POTS telephone, such as the telephone terminal 112. Having screened the call, the called party can 5 signal the IAM system 124 to indicate a desire to talk to the calling party. If the called party activates, by way of example, the "TALK @ HOME" key illustrated in Figure 3 with the Home Phone radio button depressed, the Client application 116 sends an instruction to the IAM system 124 and then substantially immediately terminates the called party's dial-up Internet session in order to make available the called party's phone line 114. Upon receiving the instruction from the Client application 116, the IAM system 124 interrupts the 10 recording and streaming process and plays a canned voice prompt, such as "please hold while your call is being connected," followed by audible ringing. The IAM system 124 then proceeds to originate a new call on a free outbound voice trunk 118 from the IAM system 124 to the called party's phone line 114. The call from the IAM system 124 to the called party can be a local, intra-state, inter-state, or International PSTN call, as needed. Optionally, the call originated by the IAM system 124 is to be jurisdictionally interstate so as to be 15 rated and billed or charged as an interstate call. For example, in one embodiment, a six digit Information Element in the SS7 call setup message may be configured with the geographic area code and prefix of the Call Processing IAM System 124 so as to cause the rating of the outgoing call to be Inter-state rather than Intra-state.

When the called party's phone line 114 is answered a brief announcement is played to the called 20 party and the IAM system 124 then bridges, in full duplex mode, the inbound call between the calling party and IAM system 124 with the outbound call between the IAM system 124 and called party's line 114.

In addition, the user can specify call handling rules that determine, at least in part, the call treatment for an incoming call based on one or more conditions. A rule can specify, for example, that if one or more conditions are met for a call, the call will be processed in accordance with a corresponding specified 25 treatment. For example, the following conditions and automatic treatments can be defined:

Conditions:

- Time-of-Day (can include a range of times), Day of Week (can include a range of days), Day of Year (holiday)
- Calling Party Number (Caller ID, non-local area code, phone type, caller 30 name)
- Called Party Number
- Subscriber presence (IP device)
- Telephony presence (phone)

35 Treatments:

- Take a voice message (using selective greeting(s))
- Take the call on home PC

5

- Take the call on home phone
- Take the call on work phone
- Take the call on another phone
- Remote screen on another phone(s) or other device(s)
- Block call (for example, using an audio message, a SIT tone or the like)
- Do not answer call
- Multiparty conference

By way of example, a subscriber can specify that if a call from a specified calling party number is received at a specified time of day (8:00-5:00), during the work week (Monday-Friday), the call should be forwarded to a specified phone, which can be the subscriber's work phone. By way of another example, a subscriber can specify that if a call to a specified phone number associated with the subscriber is received, on a holiday, remote screening should be performed using a different one of the subscriber's phone numbers.

If the call treatment specifies that the caller is to be connected to the subscriber using a given device, the subscriber can optionally still be provided with the ability to manually specify further call treatment, such as similarly described above. For example, the called party can be presented with one or more of the following options: take a voice message (using selective greeting(s)); take the call on home PC; take the call on home phone; take the call on office phone; take the call on another phone; remote screen on another phone(s) or other device(s); block call; do not answer call. Depending on the device the subscriber is currently using, the options can be provided via a visual menu, a voice menu, or the like.

Figures 5A-5B illustrate an example call process workflow that can be used when a called party requests to interrupt the caller message and to talk to the caller using the home telephone. With reference to Figures 5A-5B, at state 501, the calling party calls the called party phone line 114 connected to the telephone 112 and computer 110. In this example, the computer 110 is using the phone line 114 to access the Internet or other computer network, and so is online. At state 502, the PSTN 104 detects that the called party phone line 114 is busy. At state 503, in accordance with a call forwarding service, the PSTN 104 forwards the call on busy to the IAM system 124 via the voice trunk circuits 118. At state 504, the IAM system 124 transmits an incoming call alert to the computer 110 that is displayed to the called party by the Client application 116. At the same time or shortly thereafter, at state 505 the IAM system 124 answers the forwarded incoming call. At state 506, the PSTN 104 establishes a full duplex, 2-way talk path with the calling party. At state 507, the IAM system 124 plays a greeting to the calling party. At state 508, the calling party optionally begins leaving a voice message that is recorded by the IAM system 124. Once again, the calling party can begin speaking to the called party even while the IAM system 124 is playing the greeting. At state 509 the IAM system 124 begins streaming the message being left by the calling party in substantially real-time to the Client application 116 or other media player executing on the computer 110, that then plays the message to the called party.

At state 510 of Figures 4A-4B, the IAM system 124 generates a tone or audible signal to indicate to the calling party that the calling party should begin recording a message. At state 511, the calling party

begins leaving a voice message. At state 512, the IAM system 124 begins streaming the message being left by the calling party in substantially real-time to the Client application 116 or other media player executing on the computer 110, which plays the message to the called party.

At state 513, the called party notifies the IAM system 124 that the called party wants to take the call 5 via the home telephone 112, and the IAM system 124 instructs the Client application 116 to terminate the online session of the computer 110 by disconnecting from the Internet 106. At state 514, the IAM system 124 interrupts the calling party, via a tone or voice prompt. At state 515, the IAM system 124 requests that the calling party hold or wait while the IAM system 124 connects the calling party to the called party. At state 516, the Client application 116 terminates the online session of the computer 110 by disconnecting from the 10 Internet 106 thereby idling the called party's telephone line 114. At state 517, the IAM system 124 initiates a new call to the called party phone 112. At state 518, the called party answers the new incoming call from the IAM system 124. At state 519, the IAM system 124 generates a call announcement to the called and/or calling party. At state 520, the IAM system 124 bridges the call between the calling party phone 102 and the called party phone 112. The calling and called parties can now conduct a normal telephone conversation at 15 state 521. Again, either party can terminate the call by simply hanging up their telephone (state 522). At state 523, the IAM system 124 then releases the bridging resources and terminates the remaining call by optionally notifying the calling/called party that the other party has hung up and then disconnecting the call.

Under option (4), the called party may decide to pickup the call in progress to talk to the calling party using a communications device other than the telephone terminal 112 or the computer 110. Having screened 20 the call, the called party signals the IAM system 124 by, for example, activating the "TALK REMOTELY" button option illustrated in Figure 3, to indicate a desire to talk to the calling party. As similarly discussed above with respect to option (3), based on the called party selecting option (4), the Client application 116 sends a corresponding instruction to the IAM system 124 along with a specification of the desired destination station phone number. The destination number specification can be an index into the subscriber's electronic 25 phone book or may literally be the desired destination phone number. For example, the called party can select via the "Would you like to TALK to this caller" option that the called party wants to talk to the calling party using the called party's cell phone, office phone, other phone, or at a phone associated with a phone number entered by the called party in the "ENTER PHONE #" field.

Upon receiving the instruction from the Client application 116, the IAM system 124 interrupts the 30 recording and streaming process and plays a voice prompt to the caller. The IAM system 124 then proceeds to originate a new call on a free outbound voice trunk circuit 118. In contrast to option (3) described above, the Client application 116 does not terminate the online Internet session of the subscriber's computer terminal 110. In fact, the Client application 116 may continue online call monitoring operation while the above described transferred call is in progress. Multiple subsequent inbound calls could be simultaneously handled 35 in this manner.

By way of example and not limitation, the destination station of the outbound call from the IAM system 124 can include:

- 5 a wireless or cellular phone or device;
- a called party's phone line and/or extension at work;
- another POTS line of the called party (e.g. a second home phone number); or
- 10 a neighbor's or friend's phone line.

In addition, the called party can optionally specify the destination station by manually entering a phone number while the call is being screened. This new entry could be automatically journaled in an electric phone book available to the Client application 116 or an extension to the options menu could popup to query 15 the subscriber to determine if this is desired. Alternatively, the called party selecting an existing entry from the phone book can optionally dynamically assign the destination station. The phone book entries can be persistently stored locally on the computer terminal 110, in a centralized medium like the IAM DB subsystem 136, or in both.

Under option (5), the called party may decide to allow the call in progress to be remotely screened 15 using a communications device other than the telephone terminal 112 or the computer 110. This can optionally be in addition to continued screening of this call on the subscriber's computer 110 or instead of continued screening of this call on the subscriber's computer 110. The called party signals the IAM system 124, using the "SCREEN REMOTELY" option illustrated in Figure 3, to indicate a desire to remotely screen 20 the incoming call. Once again, as similarly discussed with respect to options (3) and (4) above, based on the called party selecting option (5), the Client application 116 sends a corresponding instruction to the IAM system 124 along with a specification of the desired destination phone number. The destination number 25 selection and specification is identical to that utilized in option (4) above, however the Screen Remotely options are used, rather than the Talk Remotely options. For example, the called party can select via the "Would you like to SCREEN to this caller" option that the called party wants to screen the calling party using the called party's cell phone, office phone, other phone, or at a phone associated with a phone number entered by the called party in the "ENTER PHONE #" field.

Upon receipt of this instruction, the IAM system 124 initiates an outbound call from the CM subsystem to the specified destination phone number. The call setup signaling information for this outbound call is modified by the IAM system 124 to deliver the calling party number from the inbound call in the 30 outbound call's ANI field. This allows the forwarded destination station to display the "original Caller-ID" to use as a first level filter for remotely screening the call. Should the remote called party decide to ignore this call, they simply do not answer it and the IAM system 124 will abort the transferred call attempt after a programmable time interval or a programmable number of ring cycles.

If the remote called party answers the transferred call, the IAM system 124 plays a brief greeting 35 prompt to the remote party to announce the remote screening call in progress. The caller message streaming

can start at the beginning of the recording or cut over to live recording in real time. The output talk path from the remote party back to the IAM system 124 is active but is muted with respect to the original calling party call. This allows the remote party to monitor the inbound call without the original calling party knowing that they are doing so. If the remote party decides to pickup the call in progress to talk to the calling party, they 5 instruct the IAM system 124 to bridge the two calls together by depressing a DTMF key or by uttering a voice command. Upon receiving this instruction, the IAM system 124 interrupts the recording and streaming process and plays a canned audio prompt to the calling party. Once again, the audio prompt can be, for example, "please hold while your call is being connected," followed by audible ringing. The IAM system 124 then bridges, in full duplex mode, the inbound call from the calling party to the CM subsystem 108 with the 10 outbound call from the CM subsystem 108 to the remote called party station.

Once again, the IAM system 124 will stay bridged between the calling party and remote called party for the duration of the call and may respond to internal events or called party actions. For example, the IAM system 124 can selectively interrupt the bridged call if a time limit is exceeded and play an announcement to notify the calling party and/or the remote called party that the call will be terminated shortly. The IAM system 15 124 can also transmit a warning message (such as a short text message) over the Internet 106 to the remote station set that then displays a visual notice regarding call termination or the like.

Under option (6), the called party may decide to not accept calls from the calling party. Having screened the call, the called party can signal the IAM system 124 to abort the caller's message recording and to communicate the request to not be called in the future. For example, the called party can activate the 20 "BLOCK CALL" option illustrated in Figure 3. The called party can either select a do not answer option or a "Tell them to TAKE ME OFF their list" option. As in the cases above, based on a user action the Client application 116 sends a corresponding instruction to the IAM system 124. Upon receiving the "Tell them to TAKE ME OFF their list" instruction from the Client application 116, the IAM system 124 interrupts the recording and streaming process, plays a voice prompt to the caller, such as: "The person you have called 25 does not accept solicitations, please remove this phone number from your calling list. Thank you and goodbye." The IAM system 124 then disconnects the call. The options dialog box could pop an additional query to determine if the subscriber would like the IAM system 124 to always apply this treatment to future calls from this Calling phone number. This automatic call handling rule would be stored in a table of subscriber preference settings local to the IAM system 124 (for example in a simple extension table of the 30 subscriber's phone book). If directed to do so, the IAM system 124 would automatically screen-out future calls from this caller and not "bother" the subscriber with needing to handle them.

Option (7) is a variant of option (6). The called party can monitor the Caller-ID of the incoming call and decide to not accept calls from this calling party. The IAM system 124 could be optionally configured to delay answering the incoming call for a fixed time interval or for a specific number of ring cycles in order to 35 allow the subscriber time to review the Caller-ID. If the Client application 116 instructs the IAM system 124 to

block the call in this manner before the incoming call has been answered, the IAM system 124 will ignore the call (i.e. let it ring). If the Client application 116 instruction comes after the incoming call has been answered, the IAM system 124 will apply the call treatment described above for option (6). Alternatively, the IAM system 124 could be configured to instead default to a standard Internet answering call when the Client application

5 116 instruction comes after the incoming call has already been answered. Once again, the options dialog box could pop an additional query to determine if the subscriber would like the IAM system 124 to always apply this treatment to future calls from this Calling phone number. Again, these automatic call handling rules would be stored in the IAM system 124 and, when directed to do so, the IAM system 124 would automatically screen-out future calls from this caller and not "bother" the subscriber with needing to handle them.

10 The above scenarios describe situations in which the called party's computer 110 is on-line and serves as the initial IAM call screening device. Alternatively, the IAM system 124 could be configured to automatically forward the call notification announcement and streamed caller message to an alternate device such as a POTS or wireless telephone or another online IP device. This alternative call screening device selection could be configured to vary based on the availability of the online presence of the called party's 15 computer 110 or alternate IP devices, on the Caller-ID of the calling party 102, on the dialed number for the Called party 112, on the reason that the call was directed to the IAM system 124 (for example, call forwarding on busy, ring-no-answer, or do-not-disturb conditions), on time of day, day of week, etc. Configuration rules governing the automatic call handling treatment can be stored in the IAM DB subsystem 136.

20 The following process describes a typical IAM call screening scenario when the called party's computer 110 is offline. When the called number forwards on busy, ring-no-answer, or do-not-disturb, and arrives on one of the IAM voice trunks 118 along with the signaling information, the CM subsystem queries the SM subsystem 122 and/or the IAM DB subsystem 136 using the incoming call's OCN (the original called party number) to determine that the call is for a registered subscriber, to determine the subscriber's online/offline presence status, and to retrieve that subscriber's call handling preference rules. Assuming that 25 the subscriber has previously configured the account to handle diverted offline calls, the IAM system carries out the specified call handling treatment. This could be simply to answer the call and take a message. Alternatively, it could include one of the seven call management options previously described. For example, the subscriber may have specified that automatic remote call screening on their cell phone was desired when their home computer 110 was not online. In this case, the CM subsystem 108 originates another call to the 30 destination device, based on the previously described configuration rules. Additionally, the CM subsystem 108 may optionally delay answering the incoming calling party's call for a predetermined amount of time or number of rings. This gives the called party additional time to answer the call originated from the CM subsystem 108.

35 Normally, when the IAM system 124 originates a call, the calling party ID passed in the SS7 and/or ISDN-PRI trunk signaling is the calling party ID of the trunks originating the call. In one embodiment, the CM

subsystem 108 modifies the network signaling to replace the calling party ID of the trunks to be that of the phone number of the original calling party. Thus, a Caller-ID device will advantageously display the phone number of the original calling party. The call is processed in an analogous above described fashion for handling a remote screening call forwarded by the subscriber from the online computer 110 to a wireless station. If the called party answers the call, the IAM system 124 plays a brief announcement of the call to the subscriber and the inbound call is answered by the IAM system 124 (if not already answered due to timeout reasons). For example, the CM subsystem 108 might announce the call as "This is an Internet Answering Machine call for John Doe". The IAM system 124 then bridges the inbound calling party call with the outbound called party call. The IAM system 124 optionally mutes the return talk path to prevent sound traveling back to the calling party so that the calling party is unaware that their call is being screened. Preferably, though not required, the called party is bridged onto the call as the called party is either listening to a personal/system greeting or, as the calling party is beginning to leave a message for the called party. In this manner, the called party can further screen the call.

Once the called party begins to screen the call, the called party may decide not to connect to the calling party. The called party, having screened the caller who is in the process of leaving a message, can hang up, thereby terminating the bridged call without the calling party being aware that the screening process took place. The calling party can continue to leave a message for the called party after the bridged call is terminated. Alternatively, the called party may decide to pickup the incoming call to talk to the calling party. Having screened the calling party, the called party can signal the CM subsystem 108, by entering a touch-tone signal or providing a voice command, to indicate a desire to talk to the calling party. The IAM system 124 will then bridge in full duplex mode the call between the calling party and CM subsystem 108 with the call between the CM subsystem 108 and called party.

The IAM system 124 will stay bridged between the calling party and called party. The IAM system 124 can selectively interrupt the bridged call if a predetermined time limit is reached and play an announcement to the calling and/or the called party that the call will be terminated shortly. Optionally, this announcement will only be played to the called party. In another case, the IAM system 124 can selectively interrupt the bridged calls to announce to the called party that the call will be terminating unless the called party authorizes billing by entering a touch-tone command or by providing a verbal authorization to charge the called party's telephone number or a credit card.

In another embodiment, an external bridging system (hardware and/or software), including a connecting switch, is used to bridge calls. The CM subsystem 108 can instruct the connecting switch located within the PSTN 104 equipped with the call bridging system to create a 3-party conference call between the calling party, the called party, and the IAM system. This process advantageously reduces the number of voice ports needed on the IAM system 124. In this scenario, the called party can optionally signal the switch with touch-tone or voice commands to cause the IAM system 124 to connect back into the bridged call.

The example embodiments described above referred to calls forwarded from a called party's line. Another embodiment uses a personal number uniquely assigned to each subscriber by which calls to that number can be screened. The personal number can be, for example a telephone number that has been acquired through governmental telephone number administration bodies, provisioned in the PSTN network, 5 assigned to the IAM system 124 and registered to an individual subscriber.

The personal number call screening process will now be described. A calling party 102 at a phone dials a phone number published by a subscriber to the IAM system 124. The call routes through the PSTN 104 and terminates on the IAM voice trunk 118 along with its associated call signaling information. Thus, for example, rather than using an existing wireless or POTS phone number, a subscriber can publish a private 10 phone number, wherein all calls to the private phone number undergo an automatic screening process, as previously described. This technique enables the subscriber to better manage their incoming call costs.

Using the called party personal phone number or normal phone number as a search key or index, the IAM system 124 extracts or retrieves call treatment actions and conditions stored in association with the called party personal number or normal phone number. The call treatment conditions and actions can include 15 some or all of those described above. For example, the conditions can include one or more of:

- Time-of-Day (can include a range of times), Day of Week (can include a range of days), Day of Year (holiday)
- Calling Party Number (Caller ID, non-local area code, phone type, caller name)
- Called Party Number
- Subscriber presence (IP device)
- Telephony presence (phone)

The call treatments can include one or more of:

- Take a voice message (using selective greeting(s))
- Take the call on computer
- Take the call on POTS phone (specify home, work, other POTS phone)
- Take the call on a wireless phone
- Remote screen on another phone(s) or other device(s)
- Block call (for example, using an audio message, a SIT tone or the like)
- Do not answer call
- Multiparty conference

The following is a more detailed description of example call treatment actions that can be executed 35 or orchestrated by the IAM system 124:

Do not answer action: the incoming calling party phone number is screened against a list of phone numbers or subscriber-specified other criteria (such as no caller id available), wherein if the calling phone number matches the listed phone numbers and/or the other criteria, the incoming call is not answered;

Take call on computer (online/offline status action): if the called party is online when a call is received, initiate a VoIP session with the Client application 116 running on the subscriber's IP device 110 and screen the call as described above;

Take the call on POTS phone action: originate a call to a specified POTS line and bridge the POTS call with remote screening as described above;

5 Take the call on POTS phone action: originate a call to a wireless phone action and bridge the wireless call with remote screening as described above; or

multi-party conference action: in which the call screening session described above is broadcast to multiple phones and IP Clients substantially at the same time.

10 In one embodiment, the multi-party conference action can include the following states. A first call processing apparatus, such as the IAM system 124, receives a call from a first user for a second user. A voice communication from the first user is received at the first call processing apparatus. At least a portion of the voice communication is multicasted to a plurality of client devices, which can include for example POTS, wireless, cellular and/or VoIP phone devices, at substantially the same time so that the first user's call can be

15 screened. An instruction is received via a first of the plurality of client devices to connect the first user to a first of the plurality of client devices. The second user is then bridged to the first of the plurality of client devices.

In another embodiment, the multi-party conference action can include the following states. A first call processing apparatus, such as the IAM system 124, receives a call from a first user for a second user. A call alert is then multicasted to a plurality of client devices, which can include for example POTS, wireless, cellular and/or VoIP phone devices, at substantially the same time so that the first user's call can be screened. In addition, a voice communication received from the first user can be multicasted to the plurality of client devices at substantially the same time. The call alert can include at least a portion of Caller ID information associated with the first call. An instruction can then be received via a first of the plurality of client devices to

20 connect the first user to a first of the plurality of client devices. The second user's call is then bridged to the first of the plurality of client devices.

25 Thus, as described above, embodiments of the present invention provide flexible, user definable call screening processes that can advantageously optionally be used even when the user is online. Further embodiments advantageously enable the user to define to which telecommunication terminals the screened call is to be broadcast to and under what conditions.

30 It should be understood that certain variations and modifications of this invention would suggest themselves to one of ordinary skill in the art. The scope of the present invention is not to be limited by the illustrations or the foregoing descriptions thereof.

WHAT IS CLAIMED IS:

1. A method of providing a called party the ability to screen calls, the method comprising:
 - 5 receiving over a switched network at a call manager system a forwarded call from a calling party intended for the called party, wherein signaling information associated with the forwarded call includes the calling party's phone number;
 - 10 playing a greeting to the calling party;
 - originating a second call from the call manager system to the called party, wherein signaling information associated with the second call includes the calling party's phone number so that the second call appears to be originating from the calling party; and
 - 15 bridging the forwarded call with the second call.
2. The method as defined in Claim 1, further comprising causing a computer associated with the called party to cease accessing the Internet before originating the second call.
3. The method as defined in Claim 1, further comprising:
 - 15 receiving at the call manager system a second forwarded call from a second calling party intended for the called party, wherein signaling information associated with the second forwarded call includes the second calling party's phone number;
 - receiving a voice communication from the second calling party;
 - 20 originating a third call from the call manager system to a cellular phone associated with the called party, wherein signaling information associated with the third call includes the second calling party's phone number so that the third call appears to be originating from the second calling party;
 - 25 receiving an instruction via the cellular phone from the called party to transmit the voice communication from the second calling party to the cellular phone so that the called party can screen the second caller; and
 - after at least beginning to transmit the voice communication from the second calling party,
- 25 receiving an instruction from the called party to connect the second calling party to the cellular phone.
4. The method as defined in Claim 1, wherein the calling party is separated from the call manager system by at least a first local carrier switch.
5. The method as defined in Claim 1, further comprising terminating the bridge between the 30 forwarded call and the second call after a predetermined amount of time.
6. The method as defined in Claim 1, further comprising:
 - receiving a third call from a second calling party;
 - streaming a voice communication from the second calling party to a networked computer associated with the called party;

receiving a second instruction from the called party while the voice communication is being streamed; and

playing a predetermined message to the second calling party indicating that the third call is not being accepted at least partly in response to the second instruction.

5 7. A method of processing calls, comprising:

receiving over a switched network at a call processing system a first call from a caller intended for a called party, wherein the first call includes signaling information having a phone number of the caller; and

10 placing a new call from the call processing system over the switched network to a terminal associated with the called party, the new call including signaling information having at least a selected portion of the phone number of the caller so that the called party can determine the identity of the caller of the first call and thereby screen the caller.

15 8. The method as defined in Claim 7, wherein at least a portion of an SS7 jurisdictional information data field on the new call from the call processing system is set to a geographic area code of the call processing system.

9. The method as defined in Claim 7, further comprising bridging the first call and the new call in response to an instruction from the called party.

20 10. The method as defined in Claim 7, wherein the signaling information includes an identifier to be displayed to the called party, wherein the identifier indicates that the new call is being placed by the call processing system.

11. The method as defined in Claim 7, wherein the signaling information includes a number associated with the call manager system in an NPA field, and at least a part of the selected portion of the caller phone number in an NXX-YYYY field.

25 12. The method as defined in Claim 7, further comprising transmitting an audible representation of at least a portion of the caller phone number.

13. The method as defined in Claim 7, further comprising providing the calling party's area code to the called party using an audio signal.

14. The method as defined in Claim 7, further comprising receiving an instruction from the called party to forward the caller to a second terminal.

30 15. The method as defined in Claim 7, further comprising providing the called party with a visual indication that the new call is being placed by the call processing system.

16. The method as defined in Claim 7, further comprising providing the called party with a second portion of the caller phone number after the called party agrees to speak with the caller.

35 17. The method as defined in Claim 7, further comprising providing the called party with a second portion of the caller phone number using digitized audio data.

18. The method as defined in Claim 7, further comprising:
receiving an instruction from the called party to forward the caller to a second terminal; and
placing a third call from the call processing system over the switched network to the second terminal, the third call including signaling information having the phone number of the caller so that a user at the second terminal can determine the identity of the caller and thereby screen the caller.
- 5 19. The method as defined in Claim 18, wherein the second terminal is a cellular phone.
20. The method as defined in Claim 7, wherein the terminal is a POTs phone.
21. The method as defined in Claim 7, wherein the first call is to a phone number assigned to the call processing system.
- 10 22. A method of providing a called party the ability to selectively accept phone calls, the method comprising:
receiving at a call processing system a first call from a calling party intended for the called party, wherein the first call includes at least a first portion of the calling party's phone number; and
initiating a second call from the call processing system to the called party, the second call including at least part of the first portion of the calling party's phone number to thereby provide the called party with information related to the identity of the calling party so that the called party can screen the first call.
- 15 23. The method as defined in Claim 22, wherein the information related to the calling party's phone number includes Caller ID information.
- 20 24. The method as defined in Claim 22, wherein, based at least in part on an instruction from the called party received after initiating the second call, selectively causing at least one of the following actions are to be taken:
refuse the calling party's call;
establish a voice over Internet (VoIP) call session with an IP device of the called party and transmitting a caller voice communication to the IP device so that the called party can screen the calling party; and
bridge the first call with a POTs line associated with the called party.
- 25 25. The method as defined in Claim 22, further comprising determining whether the first call is to be bridged with a wireless phone based at least in part on the calling party's phone number.
- 30 26. The method as defined in Claim 22, further comprising streaming an incoming message in substantially real-time to a plurality of terminals associated with the called party at substantially the same time.
27. A call screening apparatus, comprising:
a first instruction stored in computer readable memory, the first instruction configured to cause a call from a calling party intended for a called party to be answered;

a second instruction stored in computer readable memory, the second instruction configured to play a greeting to the calling party;

5 a third instruction stored in computer readable memory, the third instruction configured to maintain a communication channel over the Internet with a networked computer associated with the called party while the greeting is being played;

a fourth instruction stored in computer readable memory, the fourth instruction configured to receive and stream speech from the calling party over the Internet communication channel to the networked computer, wherein the streamed speech is intended to be screened by the called party by the networked computer;

10 a fifth instruction stored in computer readable memory, the fifth instruction configured to receive a command, via the Internet communication channel, from the called party to connect the calling party to the called party;

a sixth instruction stored in computer readable memory, the sixth instruction configured to cause the called party's networked computer to go offline; and

15 a seventh instruction stored in computer readable memory, the seventh instruction configured to originate a second call from the call manager system to the called party, and to bridge the calling party's call with the second call.

28. The call screening apparatus as defined in Claim 27, further comprising an eighth instruction configured to provide a caller-ID associated with the calling party to the called party.

20 29. The call screening apparatus as defined in Claim 27, further comprising a client application configured to execute on the called party's computer, wherein the client application provides an online indication to a server when the called party's computer is connected to the Internet.

30. 30. A method of providing a called party the ability to screen calls, the method comprising:
receiving at a first call processing apparatus a call from a first user for a second user;
25 receiving a voice communication from the first user at the first call processing apparatus;
and
multi-casting at least a portion of the voice communication to a plurality of client devices at substantially the same time so that the first user's call can be screened.

31. The method as defined in Claim 30, further comprising receiving an instruction via a first of the plurality of client devices to connect the first user to a first of the plurality of client devices.

30 32. The method as defined in Claim 30, further comprising receiving an instruction via a first of the plurality of client devices to connect the first user to a first of the plurality of client devices and causing the second user to be bridged to the first of the plurality of client devices.

33. The method as defined in Claim 30, where at least a first of the plurality of client devices is 35 capable of communicating using Internet protocol.

34. The method as defined in Claim 30, where at least a first of the plurality of client devices is a telephone.

35. The method as defined in Claim 30, where at least a first of the plurality of client devices is a cellular telephone.

5 36. A method of providing a called party the ability to screen calls, the method comprising:
receiving at a first call processing apparatus a call from a first user for a second user; and
multi-casting a call alert to a plurality of client devices at substantially the same time so that
the first user's call can be screened.

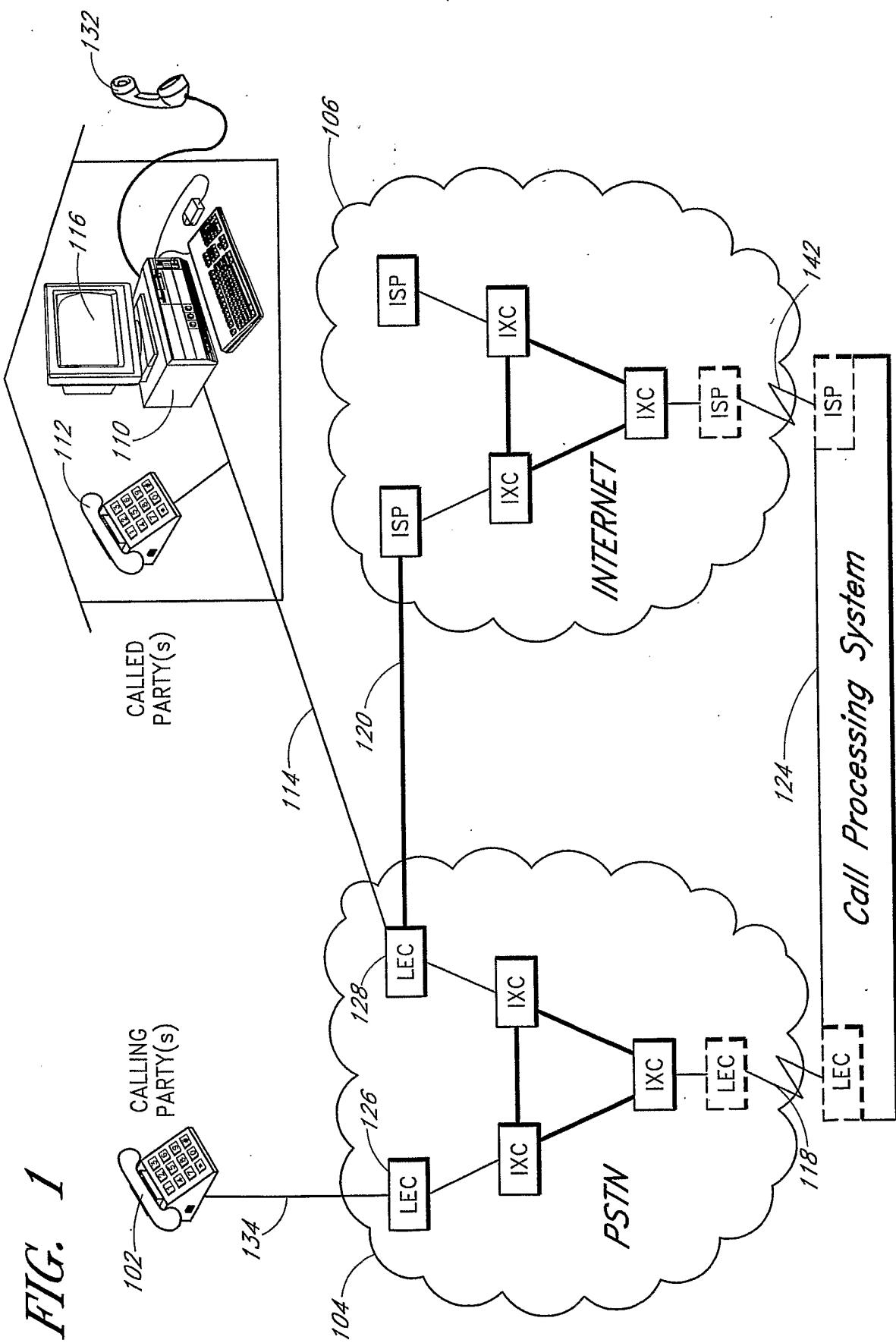
10 37. The method as defined in Claim 36, further comprising multi-casting at least a portion of a
voice communication received from the first user to the plurality of client devices at substantially the same
time.

38. The method as defined in Claim 36, wherein the call alert includes at least a portion of
Caller ID information associated with the first call.

15 39. The method as defined in Claim 36, further comprising receiving an instruction via a first of
the plurality of client devices to connect the first user to a first of the plurality of client devices.

40. The method as defined in Claim 36, further comprising receiving an instruction via a first of
the plurality of client devices to connect the first user to a first of the plurality of client devices and causing the
second user to be bridged to the first of the plurality of client devices.

1/7



2/7

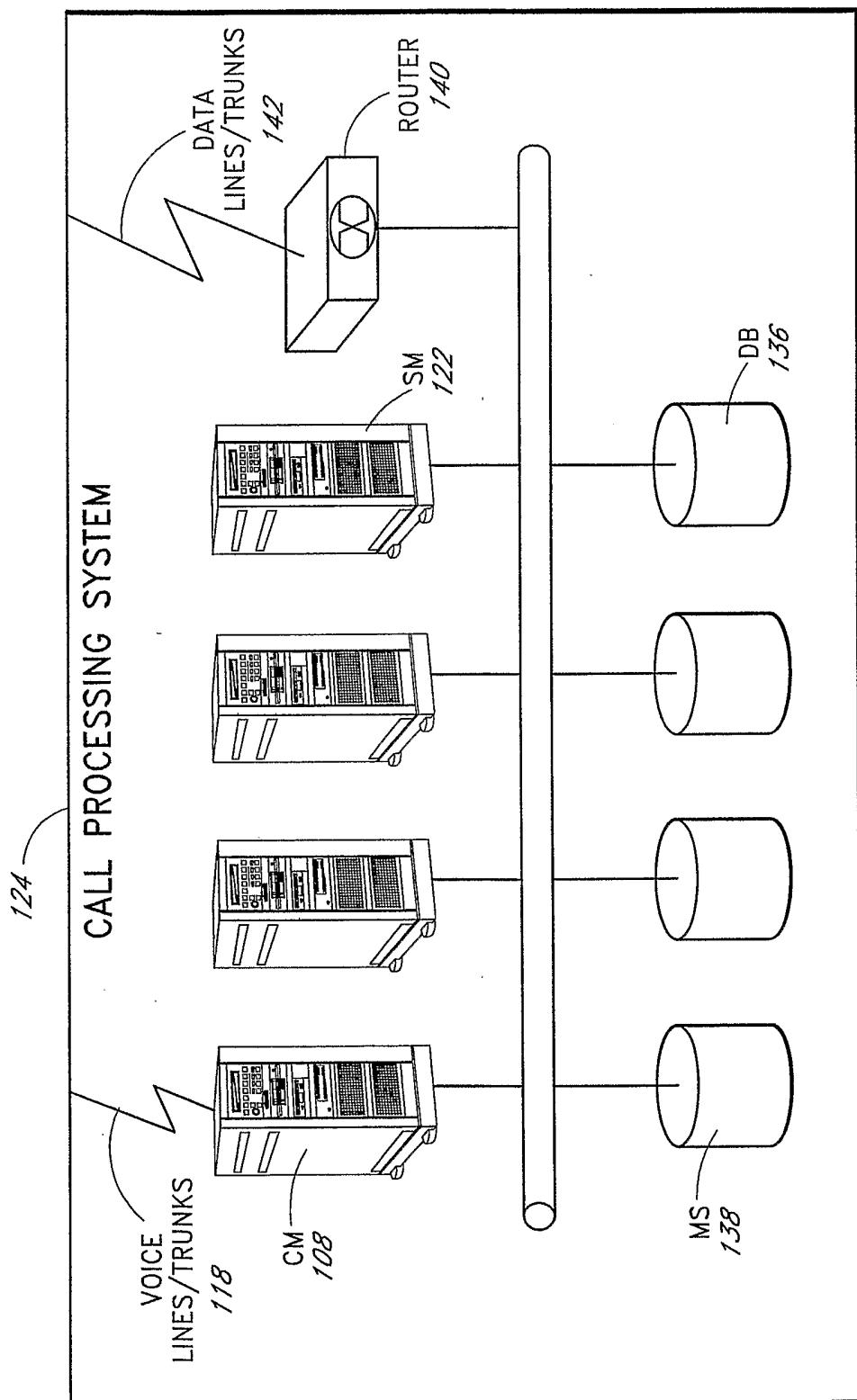


FIG. 2

3/7

 Incoming Call
775-965-0678

Would you like to TALK to this caller on your:

HOME PC (VoIP)
 HOME Phone
(This will end your dialup Internet session)

CLOSE

TALK @ HOME

Would you like to TALK to this caller on your:

CELL Phone 775-455-4230
 OFFICE Phone 775-690-4125
 OTHER Phone
 ENTER PHONE #

Open PHONE BOOK

Would you like to SCREEN this call on your:

CELL Phone 775-455-4230
 OFFICE Phone 775-690-4125
 OTHER Phone
 ENTER PHONE #

Open PHONE BOOK

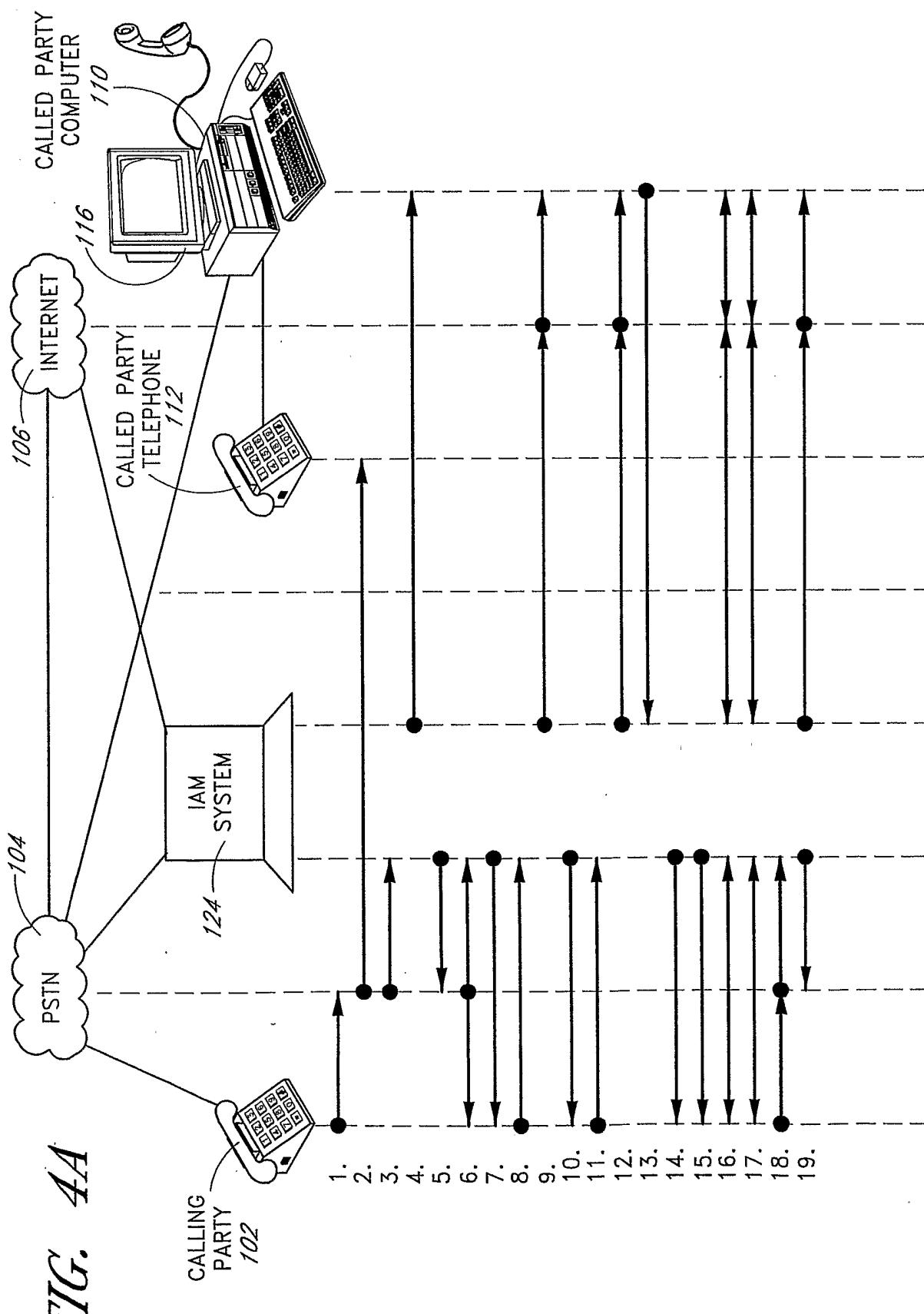
Would you like to BLOCK this call:

Tell them to TAKE ME OFF their list
 Do NOT ANSWER this Call

BLOCK CALL

FIG. 3

4/7



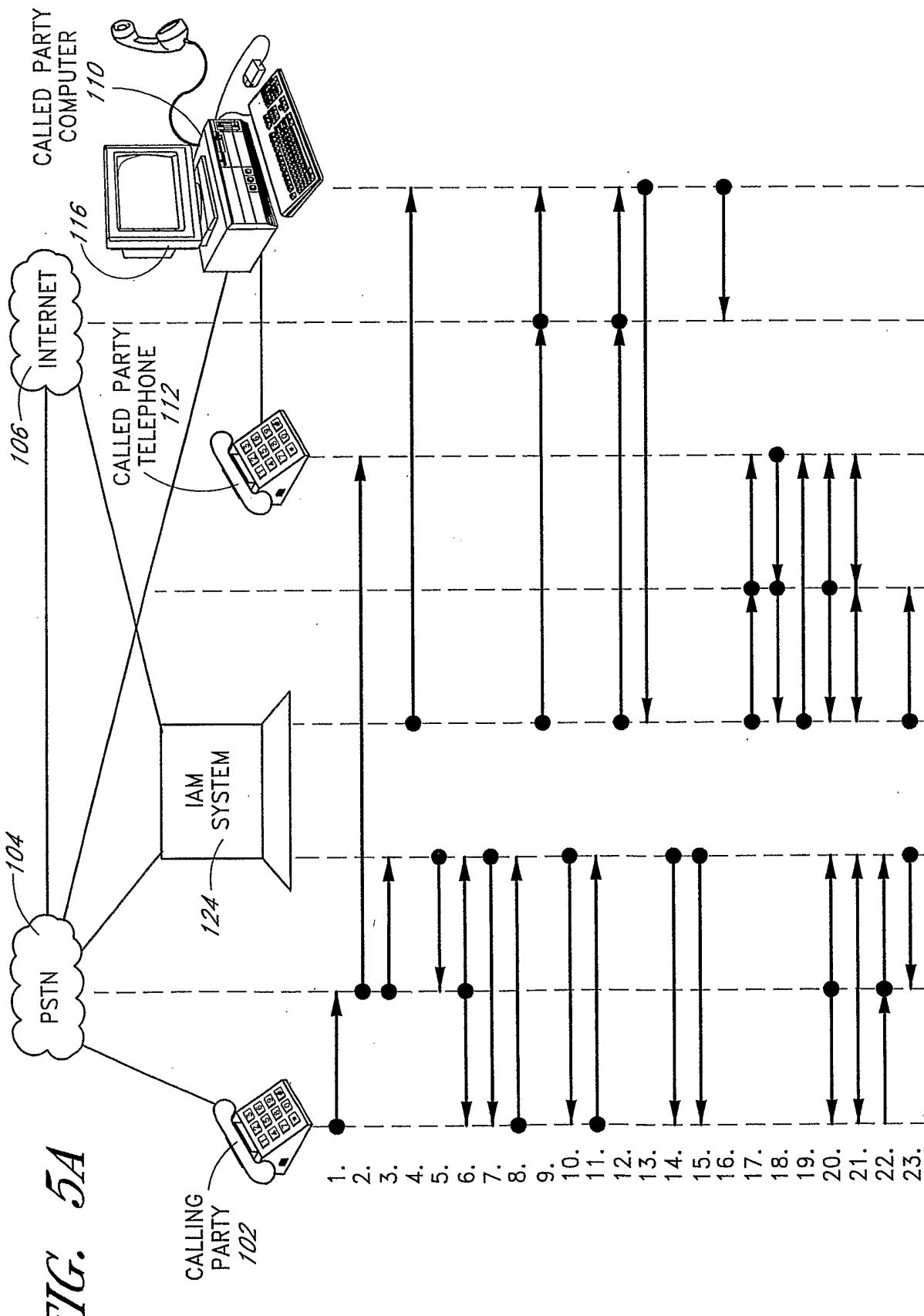
5/7

PROCESS FLOW STEPS

401. Calling Party 102 originates call to Called Party Phone Line 114
402. Called Party LEC switch 128 detects Called Party Telephone Line 114 busy
403. Called Party LEC switch 128 forwards the call to the IAM Line/Trunk 118
404. IAM system 124 notifies online Computer 110 of incoming call via Internet 104
405. IAM system 124 answers incoming call from Calling Party 102
406. PSTN 104 establishes 2-way talk path between IAM system 124 and Calling Party 102
407. IAM system 124 plays greeting to Calling Party 124
408. Calling Party 102 can talk over the greeting to Called Party 112
409. IAM system 124 streams Calling Party's 102 audio comments to the online Computer 110
410. IAM system 124 generates tone to signal Calling Party 102 to begin message recording
411. Calling Party 102 begins recording voice message
412. IAM system 124 streams Calling Party's voice message to the online Computer 110
413. Called Party 112 signals IAM system 124 to pickup the call to TALK to the Calling Party using their HOME PC
414. IAM system 124 interrupts Calling Party 102 message recording by generating tone
415. IAM system 124 requests that Calling Party 102 hold while bridging resources are allocated
416. IAM system 124 bridges the Calling Party 102 call with the Called Party 112 call
417. (Normal 2-way conversation between Calling Party telephone 102 and Called Party Computer 110)
418. Calling Party 102 or Called Party 112 terminates call by hanging up
419. IAM system 124 releases bridging resources and signals call completion to second Party
 - Internet 104 control message sent to Called Party computer for display
 - Call release signal sent to PSTN 104 to alert Calling Party Telephone 102

FIG. 4B

6/7



7/7

PROCESS FLOW STEPS

501. Calling Party 102 originates call to Called Party Phone Line 114
502. Called Party LEC switch 128 detects Called Party Telephone Line 114 busy
503. Called Party LEC switch 128 forwards the call to the IAM Line/Trunk 118
504. IAM system 124 notifies online Computer 110 of incoming call via Internet 104
505. IAM system 124 answers incoming call from Calling Party 102
506. PSTN 104 establishes 2-way talk path between IAM system 124 and Calling Party 102
507. IAM system 124 plays greeting to Calling Party 124
508. Calling Party 102 can talk over the greeting, to Called Party 112
509. IAM system 124 streams Calling Party's audio comments to the online Computer 110
510. IAM system 124 generates tone to signal Calling Party 102 to begin message recording
511. Calling Party 102 begins recording voice message
512. IAM system 124 streams Calling Party's voice message to the online Computer 110
513. Called Party 112 signals IAM system 124 to pickup the call to TALK to the caller using their HOME PHONE
514. IAM system 124 interrupts Calling Party 102 message recording by generating tone
515. IAM system 124 requests that Calling Party 112 hold while bridging resources are allocated
516. Client application 116 running on Called Party's Computer 110 terminates Internet session (releasing phone Line 114)
517. IAM system 124 originates new call to Called Party telephone 112
518. Called Party 112 answers Incoming call from IAM system 124
519. IAM system 124 announces call to Called Party 112
520. IAM system 124 bridges the Calling Party 102 call with the Called Party 112 call
521. (Normal 2-way conversation between Calling Party Telephone 102 and Called Party Telephone 112)
522. Calling Party 102 or Called Party 112 terminates call by hanging up
523. IAM system 124 releases bridging resources and signals call completion to second Party

FIG. 5B

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US03/15807

A. CLASSIFICATION OF SUBJECT MATTER

IPC(7) : H04M 3/42

US CL : 379/211.01

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

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

U.S. : 379/211.01, 88.18, 88.19, 88.2, 88.21, 142.08, 207.13, 207.15, 211.02, 215.01

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched
NONE

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

Please See Extra Sheet.

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	US 5,832,060 A (CORLETT et al) 03 November 1998, Abstract, column 1, line 38 - column 6, line 22.	1-40
A	US 5,960,064 A (FOLADARE et al) 28 September 1999, Abstract, column 2, lines 2 - 63.	1-40
A	US 6,014,436 A (FLORENCE et al) 11 January 2000, Abstract, column 2, line 43 - column 3, line 49.	1-40
A	US 6,144,644 A (BAJZATH et al) 07 November 2000, Abstract, column 1, line 66 - column 2, line 29.	1-40

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

*	Special categories of cited documents:	
"A"	document defining the general state of the art which is not considered to be of particular relevance	"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
"E"	earlier document published on or after the international filing date	"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
"L"	document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)	"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art
"O"	document referring to an oral disclosure, use, exhibition or other means	"&" document member of the same patent family
"P"	document published prior to the international filing date but later than the priority date claimed	

Date of the actual completion of the international search

13 JULY 2003

Date of mailing of the international search report

31 AUG 2003

Name and mailing address of the ISA/US
Commissioner of Patents and Trademarks
Box PCT
Washington, D.C. 20231

Facsimile No. (703) 305-3230

Authorized officer

BING BUI

Telephone No.

(703) 308-5858

INTERNATIONAL SEARCH REPORT

International application No. PCT/US03/15807

B. FIELDS SEARCHED

Electronic data bases consulted (Name of data base and where practicable terms used):

EAST

SEARCH TERMS: FORWARDING, IDENTIFICATION, IDENTITY, ANI, CALL WAITING, INTERNET, CELLULAR, MOBILE, SCREENING, VOICE MAIL, PLAY, PROMPT