EXHIBIT C

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Brahm et al.

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(45) **Date of Patent: *Dec. 22, 2009**

(54) **SYSTEMS AND METHODS FOR CALL SCREENING**

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This patent is subject to a terminal disclaimer.

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(51) **Int. Cl.**
H04M 1/64 (2006.01)

(52) **U.S. Cl.** **379/88.23**; 379/211.02

(58) **Field of Classification Search** 379/67.1, 379/6, 142.04, 211.02, 88.23

See application file for complete search history.

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(57) **ABSTRACT**

The present invention provides flexible, user-definable call screening processes. The user 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 that asks the caller to leave a voice message. The call management system selectively couples the call to a POTS line or a VoIP-capable device so that the user can listen to the incoming message and thereby screen the incoming call. Based on the screening, the user can instruct the call management system to connect the caller to the user.

77 Claims, 7 Drawing Sheets

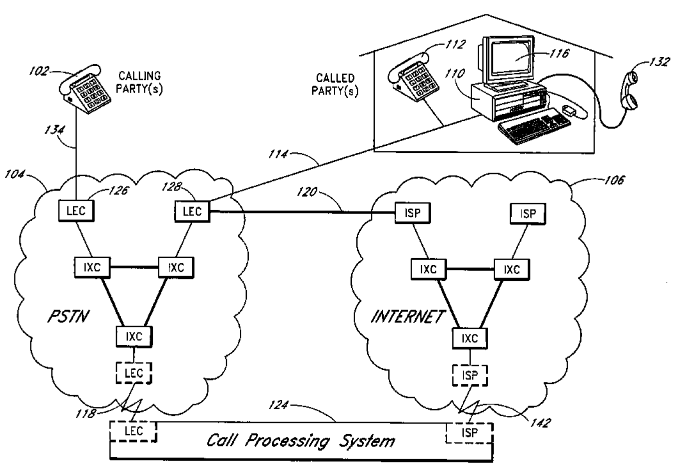

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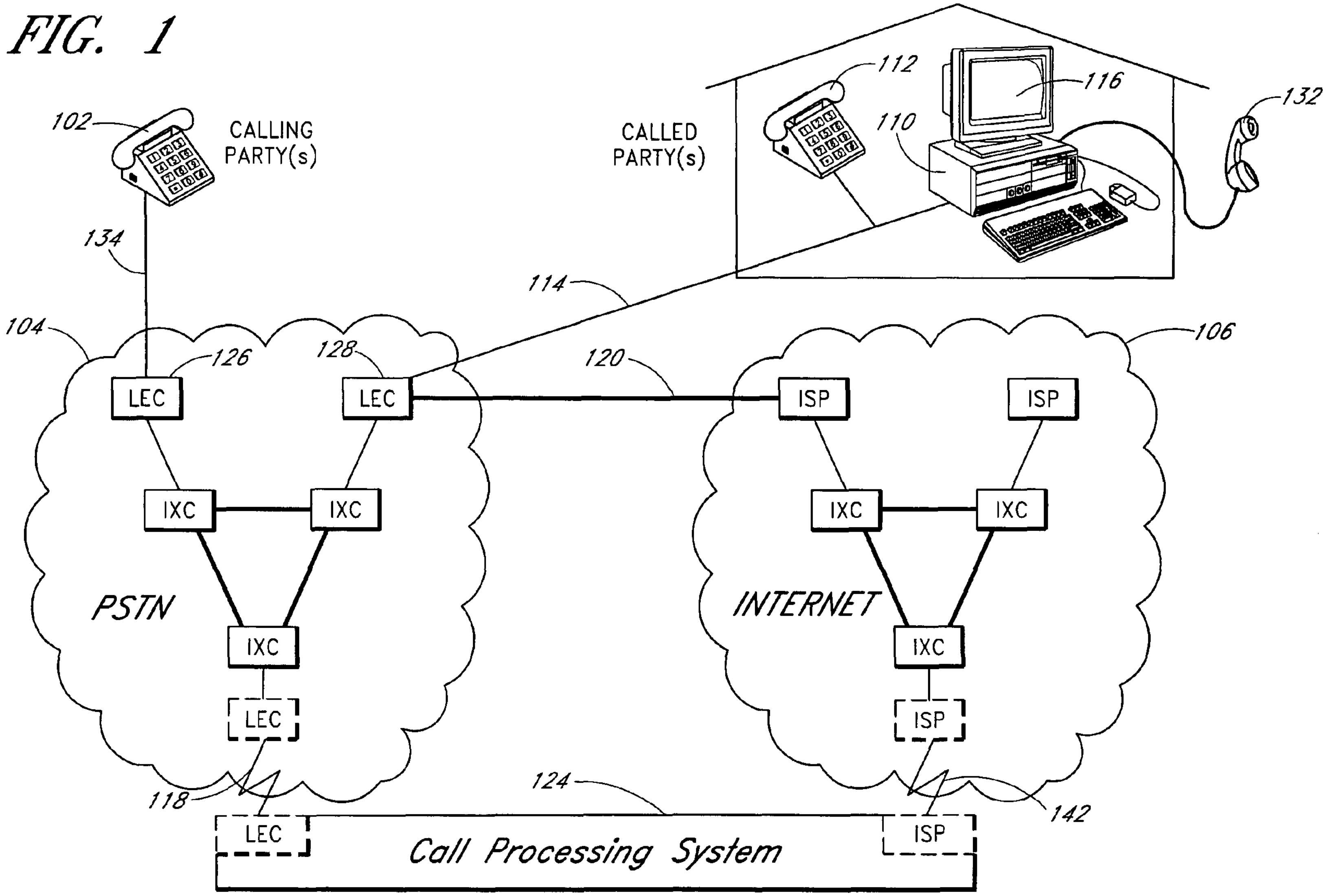
FIG. 1


102
CALLING PARTY(s)
134
104
126
128
LEC
LEC
IXC
IXC
PSTN
IXC
LEC
118
114
CALLED PARTY(s)
112
116
110
132
120
106
ISP
ISP
IXC
IXC
INTERNET
IXC
ISP
142
124
LEC
Call Processing System
ISP

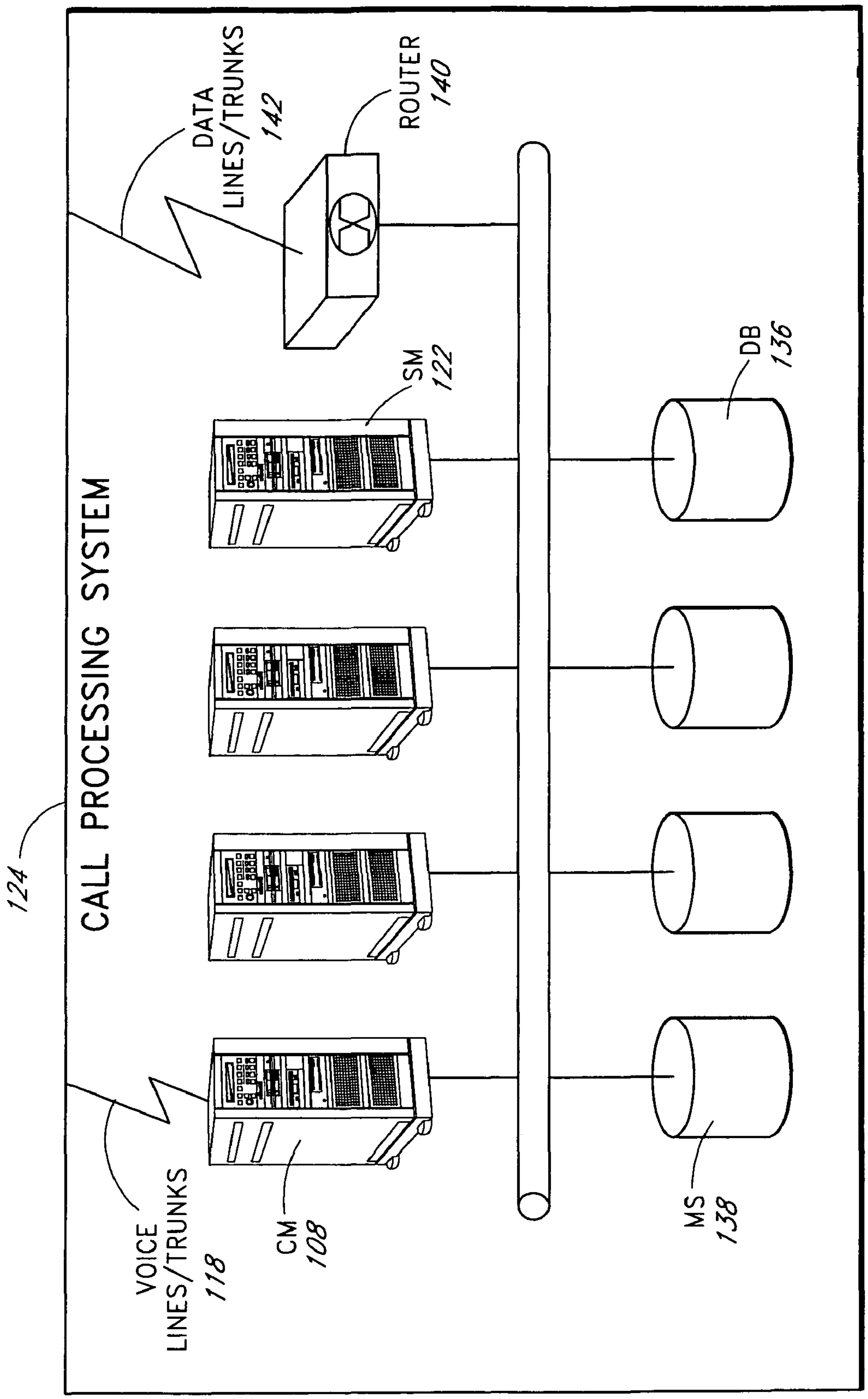

124
CALL PROCESSING SYSTEM
DATA LINES/TRUNKS 142
ROUTER 140
SM 122
DB 136
VOICE LINES/TRUNKS 118
CM 108
MS 138

FIG. 2

FIG. 3

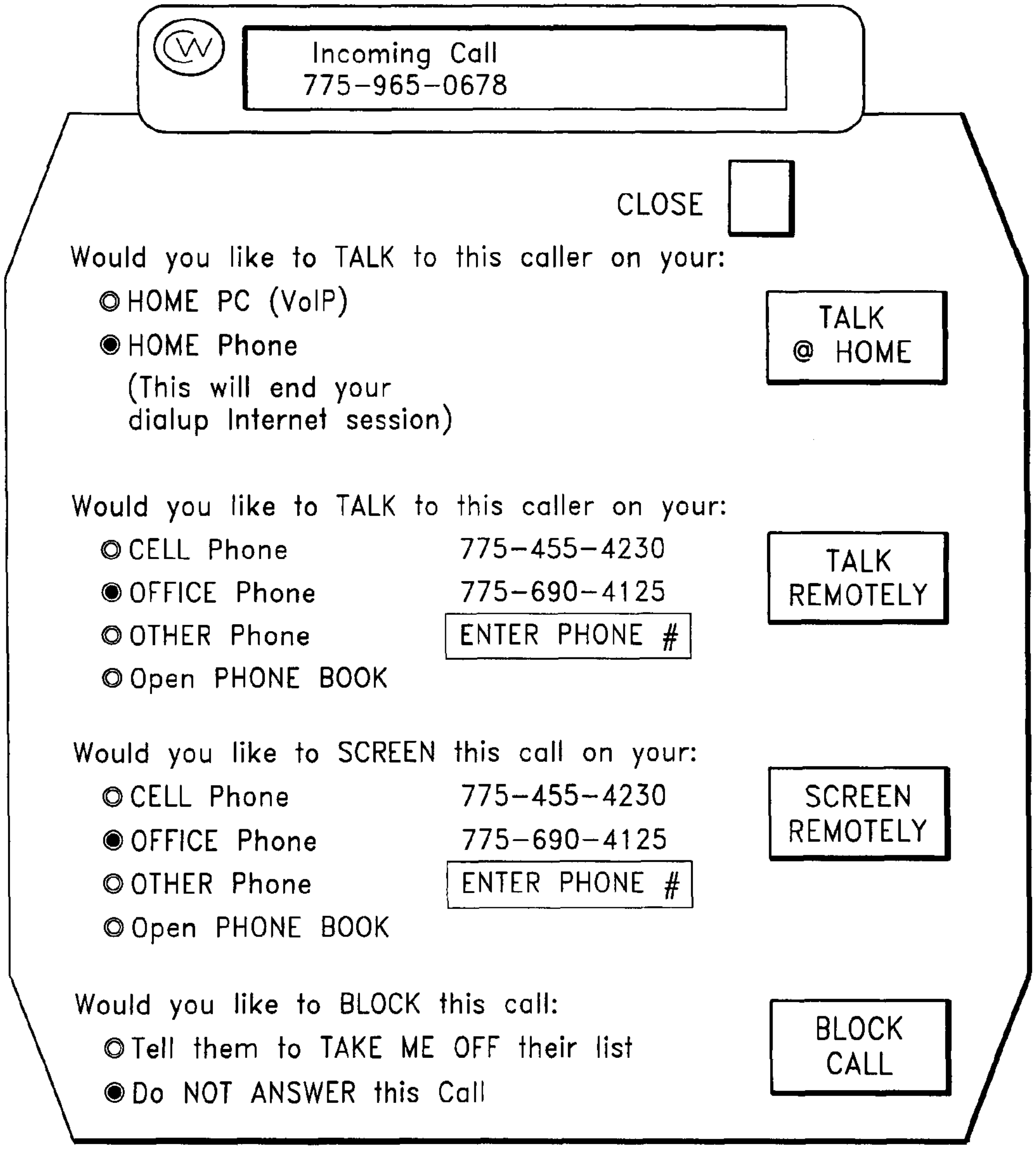
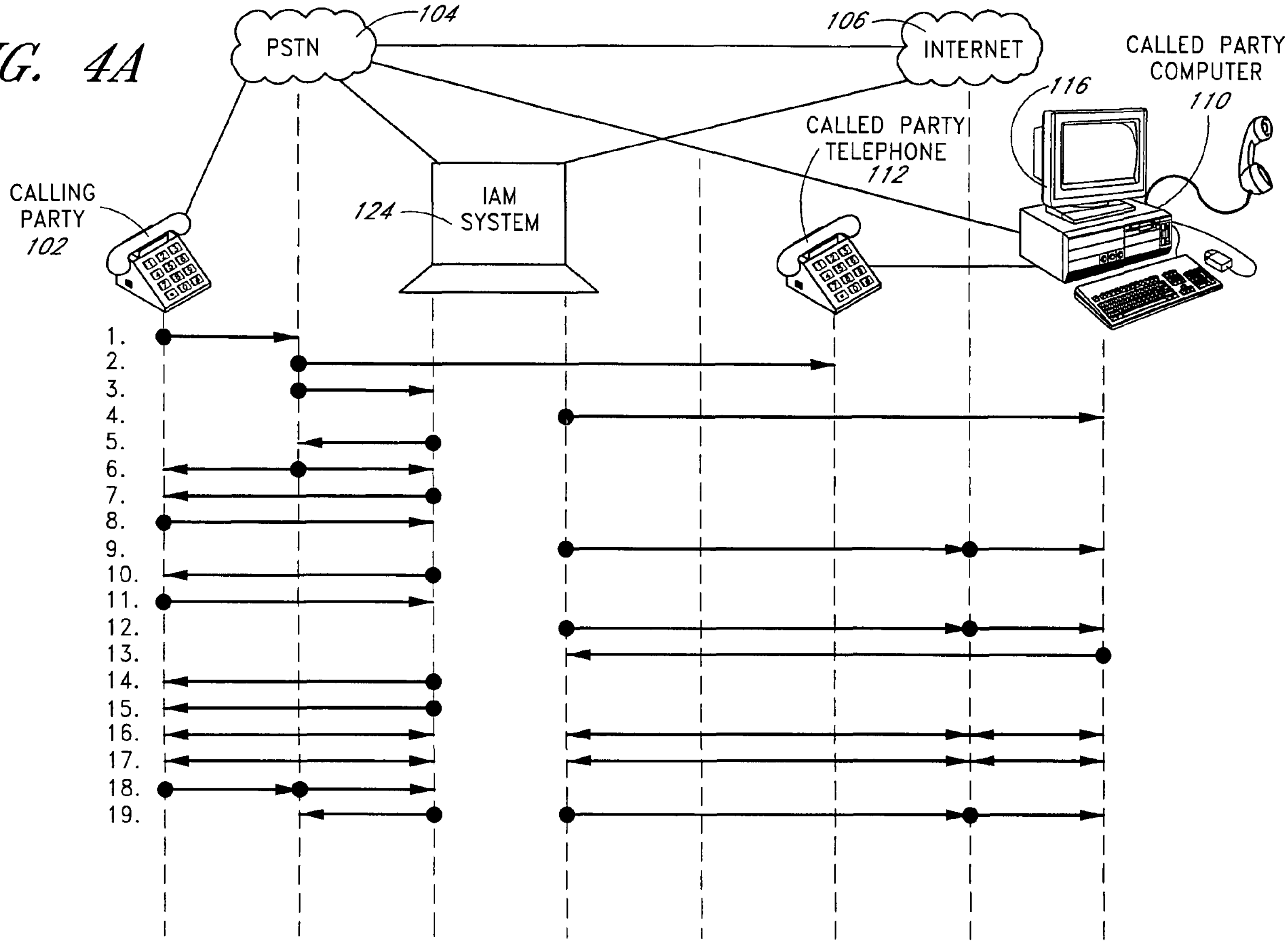

Incoming Call
775-965-0678
CLOSE
Would you like to TALK to this caller on your:
HOME PC (VoIP)
HOME Phone
(This will end your dialup Internet session)
TALK @ HOME
Would you like to TALK to this caller on your:
CELL Phone 775-455-4230
OFFICE Phone 775-690-4125
OTHER Phone ENTER PHONE #
Open PHONE BOOK
TALK REMOTELY
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
SCREEN REMOTELY
Would you like to BLOCK this call:
Tell them to TAKE ME OFF their list
Do NOT ANSWER this Call
BLOCK CALL

FIG. 4A


PSTN
104
106
INTERNET
CALLED PARTY
COMPUTER
110
116
CALLED PARTY
TELEPHONE
112
CALLING
PARTY
102
124
IAM
SYSTEM
1.
2.
3.
4.
5.
6.
7.
8.
9.
10.
11.
12.
13.
14.
15.
16.
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19.

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

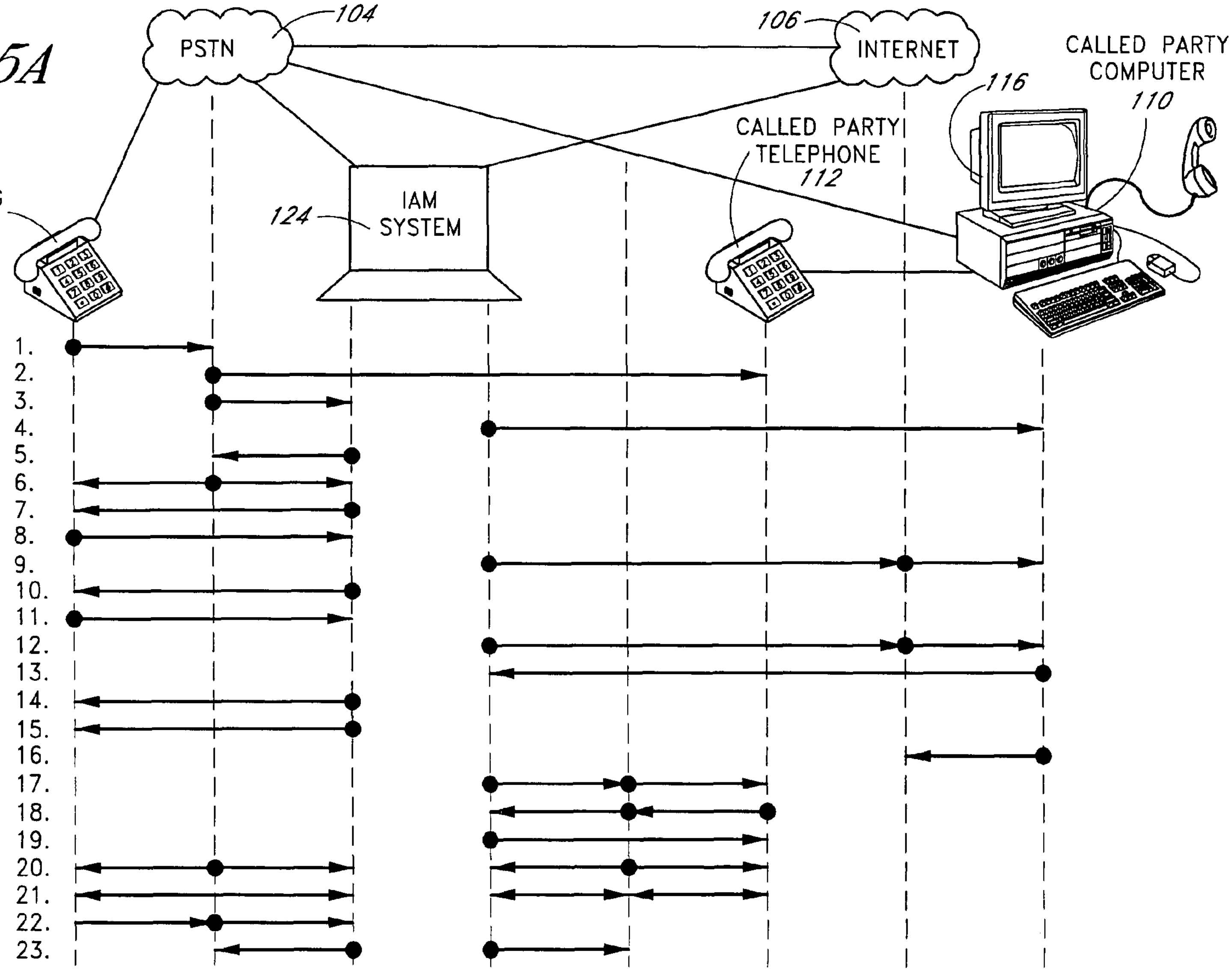
417. (Normal 2-way conversation between Calling Party telephone 102 and Called Party Computer110)

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

FIG. 5A


PSTN
104
106
INTERNET
116
CALLED PARTY
COMPUTER
110
CALLED PARTY
TELEPHONE
112
IAM
SYSTEM
124
CALLING
PARTY
102
1.
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20.
21.
22.
23.

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

SYSTEMS AND METHODS FOR CALL SCREENING

PRIORITY CLAIM

This application is a continuation of U.S. patent application Ser. No. 10/439,601, filed May 16, 2003, now U.S. Pat. No. 7,103,167which claims the benefit under 35 U.S.C. 119 (e) of U.S. Provisional Application No. 60/382,257, filed May 20, 2002, the contents of which are incorporated herein in its entirety.

BACKGROUND OF THE INVENTION

1. Field of the Invention

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

2. 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 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.

Alternatively, many residential telephone customers equip their homes with a Telephone Answering Machine (IAM) 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 IAM 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 IAM provides a MWI notification, usually by illuminating a lamp on the IAM 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 IAM allows screening of unanswered calls but does not handle busy calls. 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 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 a 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 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 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.

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 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.

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 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.

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

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

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

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

FIGS. **5**A-**5**B 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.

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.

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 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 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.

FIG. **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**.

FIG. **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 Ser. No. 09/539,375, filed Mar. 31, 2000, now U.S. Pat. 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 FIG. **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**. 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**.

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 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.

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 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) 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 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). Advantageously, H.323 provides 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.

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 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, 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 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 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-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.

FIGS. **4A-4B** illustrate one example embodiment of the present invention, including an abstraction 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 FIGS. **1** and **2** are not shown in FIG. **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 regarding the call.

With reference to FIG. **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 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 FIG. **3** for example, and listening to the corresponding streaming message, the called party is presented with one or more of the following options (see FIG. **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 FIG. **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 FIG. **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**.

FIGS. **4**A-**4**B 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 FIGS. **4**A-**4**B, 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 FIGS. **4**A-**4**B, 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 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 FIG. **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 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, intrastate, 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 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 Intrastate.

When the called party's phone line **114** is answered a brief announcement is played to the called 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 processing in accordance with a corresponding specified 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 name)

Called Party Number

Subscriber presence (IP device)

Telephony presence (phone)

Treatments:

Take a voice message (using selective greeting(s))

Take the call on home PC

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.

FIGS. **5**A-**5**B 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 FIGS. **5**A-**5**B, 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 FIGS. **4**A-**4**B, 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 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 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 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 the call, the called party signals the IAM system **124** by, for example; activating the "TALK REMOTELY" button option illustrated in FIG. **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 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 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 in this manner.

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

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

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 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 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 FIG. **3**, to indicate a desire to remotely screen 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 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 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 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 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 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 **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 "BLOCK CALL" option illustrated in FIG. **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 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 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 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 **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.

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 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**.

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 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 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**.

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, 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 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 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 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;

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.

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 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 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.

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.

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:

receiving 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;

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

transmitting, while the first call is in progress, a call alert message, including at least a portion of the signaling information, to a terminal associated with the called party;

recording speech from the caller and transmitting the speech to the terminal in substantially real time, wherein the speech is intended to be played to the called party via the terminal;

receiving a connection instruction via the terminal, and causing the caller to be connected to at least one of a plurality of potential destinations associated with the called party;

placing an outcall from the call processing system to at least one destination, wherein the at least one destination is a phone;

bridging the first call and the outcall; and

transmitting status information related to the bridged first call and outcall to the terminal.

2. The method as defined in claim **1**, wherein the called party communicates with the caller using Voice over Internet Protocol (VoIP).

3. The method as defined in claim **1**, wherein the act of causing the caller to be connected to at least one of a plurality of potential destinations includes placing an outcall from the call processing system and bridging the first call and the outcall.

4. The method as defined in claim **1**, further comprising:

placing an outcall from the call processing system to at least one destination;

bridging the first call and the outcall; and

automatically interrupting the bridged first call and outcall after a first time period.

5. The method as defined in claim **1**, wherein the terminal is a computer.

6. A method of providing call screening, the method comprising:

receiving 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;

transmitting over the Internet to a networked computer associated with the called party in substantially real time at least a portion of the signaling information included in the first call for display to the called party;

placing a second call from the call processing system to a destination associated with the called party, the second call including signaling information having at least a selected portion of the phone number of the caller;

recording speech from the caller and transmitting the speech in substantially real time to the destination and/or over the Internet to the networked computer to be played by the networked computer to the called party; and

receiving an instruction from the called party via the networked computer or a destination phone associated with the called party, and at least partly in response to the received instruction, causing the first call to be connected to one of a plurality of potential terminal destinations;

wherein the act of causing the first call to be connected to one of a plurality of potential terminal destinations further comprises:

placing an outcall from the call processing system to at least one destination, wherein the at least one destination is a phone;

bridging the first call and the outcall; and

transmitting status information related to the bridged first call and outcall to the terminal.

7. The method as defined in claim **6**, wherein the called party communicates with the caller using Voice over Internet Protocol (VoIP).

8. The method as defined in claim **6**, wherein the act of causing the first call to be connected to one of a plurality of potential terminal destinations includes placing an outcall from the call processing system and bridging the first call and the outcall.

9. The method as defined in claim **6**, further comprising:

placing an outcall from the call processing system to at least one destination;

bridging the first call and the outcall; and

automatically interrupting the bridged first call and outcall after a first time period.

10. A method of processing calls, comprising: receiving over a network at a first 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;

placing a second call from the first system over the network to a first terminal associated with the called party, the second 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;

receiving an instruction from the called party to forward the caller to a second terminal; and

at least partly in response to the instruction, placing a third call from the first system over the switched network to the second terminal.

11. The method as defined in claim **10**, further comprising causing the first call and the third call to be bridged.

12. The method as defined in claim **10**, wherein the called party communicates with the caller using Voice over Internet Protocol (VoIP).

13. The method as defined in claim **10**, wherein the second terminal is a cellular phone.

14. The method as defined in claim **10**, further comprising transmitting status information regarding first call and outcall to the first terminal.

15. The method as defined in claim **10**, wherein the first terminal is a computer.

16. A method of processing calls, comprising:

receiving over a network at a call processing system a first call from a caller intended for a called party;

placing a second call from the call processing system over the network to a terminal associated with the called party, the second 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;

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.

17. The method as defined in claim **10**, further comprising causing the first call and the third call to be bridged.

18. The method as defined in claim **10**, wherein the called party communicates with the caller using Voice over Internet Protocol (VoIP).

19. The method as defined in claim **10**, wherein the second terminal is a cellular phone.

20. The method as defined in claim **10**, further comprising transmitting status information regarding first call to the first terminal.

21. The method as defined in claim **10**, wherein the first terminal is a computer.

22. A method of processing calls via a wireless phone, comprising:

receiving over a network at a call processing system a first call from a caller intended for a called party;

answering the first call from the caller at the call processing system;

playing a greeting message to the caller via the call processing system;

receiving a voice message from the caller at the call processing system;

placing an outcall from the call processing system to a wireless phone associated with the called party,

wherein the outcall is performed at least in part over a wireless communications network;

streaming the voice message to the wireless phone via the outcall in substantially real time;

the call processing system automatically muting a return talk path to the caller to prevent sound traveling from the called party wireless phone to the caller to thereby allow the called party to perform call screening via the wireless phone while the caller is unaware that their call is being screened;

receiving a call acceptance instruction; and

enabling the caller to hear the called party speak at least partly in response to receiving the call acceptance instruction, wherein the called party is speaking into the wireless phone.

23. The method as defined in claim **22**, further comprising enabling the caller to converse with the called party in duplex mode at least partly in response to receiving the call acceptance instruction.

24. The method as defined in claim **22**, wherein the call acceptance instruction is from the wireless phone.

25. The method as defined in claim **22**, wherein the call acceptance instruction is from a personal computer associated with the called party.

26. The method as defined in claim **22**, further comprising enabling the called party to provide a call transfer instruction via the wireless phone while performing call screening.

27. The method as defined in claim **22**, further comprising enabling the called party to instruct the call processing system to play a call termination voice message to the caller and to terminate the first call.

28. The method as defined in claim **22**, further comprising bridging the outcall and the caller call.

29. The method as defined in claim **22**, wherein the wireless phone is a cellular phone.

30. The method as defined in claim **22**, wherein the call processing system is serviced by a different local exchange then the caller communication device and the wireless phone of the called party.

31. A method of processing calls via a wireless phone, comprising:

receiving over a network at a processing system a first call from a calling party intended for a called party;

providing a greeting to the calling party via the processing system;

receiving a voice communication from the calling party at the processing system;

streaming at least a first portion of the voice communication via the processing system to a wireless phone associated with the called party in substantially real time while automatically ensuring that sound does not travel back to the calling party from the wireless phone to thereby allow the called party to screen the call via the wireless phone while the calling party is unaware that their call is being screened;

providing a first control via the wireless phone that, when selected by the called party, causes duplex communication to be provided so that the called party can converse with the calling party; and

providing a second control via the wireless phone that, when selected by the called party, causes at least a second portion of the voice communication to be streamed to another communication device in substantially real time.

32. The method as defined in claim **31**, further comprising enabling the calling party to converse with the called party in duplex mode.

33. The method as defined in claim **31**, wherein the communication device is another wireless phone.

34. The method as defined in claim **31**, wherein the communication device is a home phone or a work phone.

35. The method as defined in claim **31**, wherein the communication device is a computer.

36. The method as defined in claim **31**, further comprising placing an outcall from the processing system to the wireless phone and bridging the first call with the outcall.

37. The method as defined in claim **31**, further comprising:

placing an outcall from the processing system to the wireless phone and bridging the first call with the outcall; and

based on a first call time limit, providing a voice notification to the called party that a call termination will be performed.

38. The method as defined in claim **31**, wherein the called party selects the first control by selecting a wireless phone key.

39. The method as defined in claim **31**, wherein the called party selects the first control by a voice command.

40. The method as defined in claim **31**, further comprising providing the called party with a third control, which when selected by the called party,

causes a call termination voice message to be played to the calling party, and

causes the first call to be terminated.

41. The method as defined in claim **31**, wherein the wireless phone is a cellular phone.

42. A method of processing calls, comprising:

receiving over a communications network at a processing system a first call from a calling party intended for a called party;

providing a greeting to the calling party;

receiving a voice communication from the calling party;

transmitting in substantially real time from the processing system at least a first portion of the voice communication to a telephone associated with the called party to allow the called party to screen the call via the telephone while preventing the calling party from hearing sounds received via the called party telephone;

receiving via the telephone a first command to continue transmitting the voice communication to a second communication device; and

at least partly in response to the first command, transmitting in substantially real time at least a second portion of the voice communication from the calling party to the second communication device.

43. The method as defined in claim **42**, further muting a communication path between the telephone and a calling party communication device.

44. The method as defined in claim **42**, further comprising enabling the calling party to converse with the called party in duplex mode.

45. The method as defined in claim **42**, where the telephone is a cellular telephone.

46. The method as defined in claim **42**, where the second communication device is a home telephone or a work telephone.

47. The method as defined in claim **42**, where the second communication device is a computer.

48. The method as defined in claim **42**, where the second communication device is a voice over Internet protocol terminal.

49. The method as defined in claim **42**, where the second communication device hosts a communication application client.

50. The method as defined in claim **42**, further comprising, based at least in part on a first time limit, providing a voice notification to the called party that a call termination will be performed.

51. The method as defined in claim **42**, wherein the called party provides the first command by selecting a telephone key or by providing a voice command.

52. The method as defined in claim **42**, further comprising providing the called party with a call termination user interface, which when activated by the called party,

causes a call termination voice message to be played to the calling party, and

causes the first call to be terminated.

53. The method as defined in claim **42**, wherein the communications network includes a switched circuit network.

54. The method as defined in claim **42**, wherein the communications network includes a packet switched network.

55. The method as defined in claim **42**, wherein the first call is forwarded to the processing system.

56. A method of processing calls via a wireless phone, comprising:

receiving over a network at a call processing system a first call from a caller intended for a called party;

answering the first call from the caller at the call processing system;

playing a greeting message to the caller via the call processing system;

receiving a voice message from the caller at the call processing system;

streaming the voice message over the Internet to a terminal associated with the called party;

the call processing system automatically muting a return talk path to the caller to prevent sound traveling from the called party's terminal to the caller to thereby allow the called party to perform call screening via the terminal while the caller is unaware that their call is being screened;

receiving a call acceptance instruction; and

enabling the caller to hear the called party speak at least partly in response to receiving the call acceptance instruction, wherein the called party is speaking into a microphone associated with the terminal.

57. The method as defined in claim **56**, further comprising enabling the caller to converse with the called party in duplex mode at least partly in response to receiving the call acceptance instruction.

58. The method as defined in claim **56**, wherein the call acceptance instruction is from the terminal.

59. The method as defined in claim **56**, wherein the terminal is a personal computer associated with the called party.

60. The method as defined in claim **56**, further comprising enabling the called party to provide a call transfer instruction while performing call screening.

61. The method as defined in claim **56**, further comprising enabling the called party to instruct the call processing system to play a call termination voice message to the caller and to terminate the first call.

62. The method as defined in claim **56**, further comprising bridging the outcall and the caller call.

63. A method of processing calls via a wireless phone, comprising:

receiving over a network at a call processing system a first call from a caller intended for a called party;

answering the first call from the caller at the call processing system;

playing a greeting message to the caller via the call processing system;

receiving a voice message from the caller at the call processing system;

placing an outcall from the call processing system to a wireless phone or networked computer associated with the called party,

wherein the outcall is performed at least in part over a wireless communications network or a data network;

streaming the voice message to the wireless phone or the networked computer via the outcall in substantially real time;

the call processing system automatically muting a return talk path to the caller to prevent sound traveling from the called party wireless phone or networked computer to the caller to thereby allow the called party to perform call screening while the caller is unaware that their call is being screened;

receiving a call acceptance instruction; and

enabling the caller to hear the called party speak at least partly in response to receiving the call acceptance instruction.

64. A method of processing calls, comprising:

receiving over a network at a first 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;

placing a second call from the first system over the network to a first terminal associated with the called party, the second 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;

placing a third call from the first system over the network after the second call to a third terminal associated with the called party;

playing a greeting to the calling party; and

streaming at least a first portion of the voice communication via the processing system to a networked computer associated with the called party, the terminal associated with the second call, and/or the terminal associated with the third call to thereby allow the called party to screen the call while automatically ensuring that sound does not travel back to the calling party from the networked computer or the terminal at least until a first action by the called party.

65. The method as defined in claim **64**, wherein the first action is the activation of a control on networked computer.

66. The method as defined in claim **64**, wherein terminal associated with the second call is a wireless phone, and the first action is a key press of a wireless phone key.

67. The method as defined in claim **64**, wherein terminal associated with the second call is a wireless phone, and the first action is a call acceptance instruction.

68. The method as defined in claim **64**, further comprising enabling the called party to provide a call transfer instruction while performing call screening.

69. The method as defined in claim **64**, further comprising bridging the first call and the second call.

70. The method as defined in claim **64**, further comprising bridging the first call and the third call.

71. A method of processing calls, comprising:

receiving over a network at a first 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;

placing a second call from the first system over the network to a first terminal associated with the called party, the second 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;

placing a third call from the first system over the network after the second call to a second terminal associated with the called party, the second call;

playing a greeting to the calling party;

streaming at least a first portion of the voice communication via the processing system to a networked computer associated with the called party, the terminal associated with the second call, and/or the terminal associated with the third call to thereby allow the called party to screen the call via the terminal or network computer while automatically ensuring that sound does not travel back to the calling party from the networked computer or the terminal; and

causing duplex communication to be provided so that the called party can converse with the calling party if the called party provides an instruction indicating that the called party wants to speak to the caller.

72. The method as defined in claim **71**, wherein the instruction is provided via the networked computer.

73. The method as defined in claim **71**, wherein the instruction is provided via a wireless phone.

74. The method as defined in claim **71**, wherein the instruction is provide via the terminal associated with the third call.

75. The method as defined in claim **71**, further comprising enabling the called party to provide a call transfer instruction while performing call screening.

76. The method as defined in claim **71**, further comprising bridging the first call and the second call.

77. The method as defined in claim **71**, further comprising bridging the first call and the third call.

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UNITED STATES PATENT AND TRADEMARK OFFICE

CERTIFICATE OF CORRECTION

PATENT NO. : 7,636,428 B2
APPLICATION NO. : 11/374390
DATED : December 22, 2009
INVENTOR(S) : Brahm et al.

It is certified that error appears in the above-identified patent and that said Letters Patent is hereby corrected as shown below:

On the Title Page:

The first or sole Notice should read --

Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 824 days.

Signed and Sealed this

Twenty-first Day of December, 2010

David J. Kappos

David J. Kappos
Director of the United States Patent and Trademark Office