a subscriber master record 700 which had a DID number 701 which had the same last four digits as the digits 'cdef' detected in step 3706, and also had the same PIN code 702 as that detected in step 3727. Control passes to step 3730 where the subscriber is prompted: "Please enter your new mode memory number." Control then passes to step 3731 where a determination is made as to whether the subscriber has entered a memory number '10 to '99', and if such a number has not been entered, then control remains at step 3731. If a mode memory number is entered, then control passes to step 3732 where a 'mode memory inquiry' message is constructed using the DID number 701 and the and the memory number as detected in step 3731, and the message is sent via data network interface 3507 to the CPF 100. Control then passes to step 3733 where a determination is made as to whether a response has been received to the 'mode memory inquiry' message, and if such a response is not received control remains at step 3733. If the response message is received by network interface 3507, then control passes to step 3734 where a determination is made as to whether a valid mode memory 800 is included in the returned message. If a valid mode memory 800 is not included, then control passes to the connector labelled "SAF REORDER" shown at reference 3745. If at step 3734 a valid mode memory 800 is found, then control passes to step 3735, where a determination is made as to whether the mode memory which was received in the message requires an 'externally entered' memory. This is determined by inspecting the transfer number field of the mode memory 800. If it has an 'externally entered number' tag, then the mode memory does require an 'externally entered number'. If at step 3735 it is determined that an 'externally entered number' is not required, then control passes to a connector labelled "SAF UPDATE" at reference 3736, which as described earlier causes the mode memory requested to be invoked. If at step 3735 it is determined that an 'externally entered number' is required, then control passes to step 3737 where the subscriber master record 700 as acquired at step 3750 is retrieved from memory. Control then passes to step 3738 where the mode memory 800 that was acquired in step 3734 is copied field-by-field to the subscriber master record. Control then passes to step 3740, where the ANI number acquired in step 3703 is copied to the transfer number field 707 of the subscriber master record 700. Control then passes to step 3741 where an 'update master record' message is constructed using this master record, and the message is sent via data network interface 3507 to CPF 100. Control then passes to step 3742 where the subscriber is prompted: "Accepted, you have selected mode memory..xx", where the digits 'xx' are the digits entered by the subscriber at step 3731. Control then passes to step 3743 where the trunk 8 is placed on hook, and then control returns to the Subscriber Access Facility Main Task entry point as shown at reference 3748. If at step 3734, it is determined that the mode memory received in the message was not valid, then control passes to a connector labelled "SAF REORDER" at reference 3745, which causes control to pass to step 3746 where the call processor 3504 plays a 'reorder signal' to the subscriber, indicating that the call is being terminated. Control then passes to step 3747 where the trunk 8 is placed on hook. Control then returns to the Subscriber Access Facility Main Task entry point as shown at reference 3748.

A block diagram of the Communicator Access Facility (CAF) 125 is illustrated in FIG. 38. As shown, the CAF 125 contains a serial port interface module 3800 which connects to the serial datalink 10, and a data network interface 3801 which connects to the high speed data network 150. The functions of data network interface 3801 are well known in the art, and many products, such as the Model COM4i from Digiboard Corporation, exist commercially which can accomplish these functions. The operation of the CAF 125 is controlled by CPU module 3802, which consists of a microprocessor, a boot ROM, RAM, and disk. The boot ROM is used to initialize the CPU and load the control program into memory from disk. Operation of the CAF 125 is then controlled by the control program. The control program is described in the explanation which accompanies FIG. 39. Still referring to FIG. 38, the serial port 3800, data network interface 3801, and the CPU module 3802 are all shown connected to internal bus 3803.

A flowchart of the Communicator Access Facility Main Task is illustrated in FIG. 39. This program is loaded into memory and executed by CPU 3802 of the CAF 125. The Communicator Access Facility Main Task is entered at step 3900 and control passes to step 3901 where a determination is made as to whether a 'page message' is received via data network interface 3801, and if a 'page message' is received then control passes to step 3902 where the page message is sent via serial port 3800 and datalink 10 to packet radio transceiver 9. Control then passes to step 3903 where an 'acknowledge timer' is started, and the 'page message' just sent is saved and associated with this 'acknowledge timer'. Control then passes to step 3904. Control also passes to step 3904 if a 'page message' is not received as determined at step 3901. At step 3904, a determination is made as to whether a 'phone number' message has been received from a communicator 11 via the serial port interface 3800. If such a message has been received, then control passes to step 3905 where the subscriber's DID number 701 is retrieved form the message, and a 'request master record' message is created and sent to the CPF 100 via data network interface 3801. When the response is received from the CPF 100 via the data network interface 3801, the subscriber master record is retrieved from the response message. Control then passes to step 3906 where the transfer number field 707 of the subscriber master record 700 is updated per the phone number received from the communicator 11 at step 3904. Control then passes to step 3907 where an 'update master record' message is then constructed and sent back to the CPF 100 via data network interface 3801. Control then passes to step 3908, where an 'acknowledge message' is sent back to the communicator 11 via serial port 3800. Control then passes to step 3909. Control also passes to step 3909 if a 'phone number' message is not detected at step 3904. At step 3909 a determination is made as to whether a 'new mode memory' message is received from a communicator 11 via serial port 3800. If this message type has been received then control passes to step 3910 where a 'change to new mode memory' message is constructed and sent to CPF 100 via data network interface 3801, and then control passes to step 3908 where an 'acknowledge' message is sent back to the communicator 11 as described earlier. If at step 3909 a 'new mode memory request' message is not detected, then control passes to step 3911 where a determination is made as to whether a 'set dynamic mode assignment mode' message is received from a communicator 11 via serial port 3800. If this message type is received, then control passes to step 3912, where a 'set dynamic mode assignment flag' message is constructed and sent to CPF 100 via data network interface 3801. Control then passes to step 3908 where an 'acknowledge' message is sent back to the communicator 11 as described earlier. If at step 3911 a 'set dynamic mode assignment mode' message is not detected, then control passes to step 3913 where a determination is made as to whether a 'disable dynamic mode assignment mode' message is received from a communicator 11 via serial port 3800. If this message type is received, then control passes to step 3914, where a 'clear dynamic mode assignment flag' message is constructed and sent to CPF 100 via data network interface 3801. Control then passes to step 3908 where an 'acknowledge' message is sent back to the communicator 11 as described earlier. If at step 3913 a 'disable dynamic mode assignment mode' message is not detected, then control passes to step 3915 where a determination is made as to whether the 'acknowledge timer' has just expired. If the 'acknowledge timer' has just expired, as determined at step 3915, then control passes to step 3916 where the message which was associated with this 'acknowledge timer' is again sent to the communicator 11 via serial port interface 3800, and control then passes to step 3903. If at step 3915 it is determined that an 'acknowledge timer' has not just expired, then control passes to step 3917 where a determination is made as to whether an 'acknowledge' message is received from a communicator via serial port interface 3800, and if an 'acknowledge' message is not received, then control returns to step 3901. If an 'acknowledge' message is received, as determined at step 3917, then the 'acknowledge' timer which is associated with the last message sent to the communicator 11 identified in the 'acknowledge' message is cleared. Control then returns to step 3901.

A flowchart of the Communicator Main Task is illustrated in FIG. 40. This task is executed by microprocessor 200, and controls all operations of the communicator 11. The Communicator Main Task is entered at step 4000 and control passes to step 4001 where a determination is made as to whether a 'page' message is detected at the receive data input 245. If a 'page' message is detected, and the message contains the pager number which corresponds to this communicator, then control passes to step 4002, where a signal is sent via output port 250 causing the beeper 260 to generate an alerting sound. Control then passes to step 4003 where a display number is retrieved from the decoded 'page' message, and is output to the display 210. Control then passes to step 4004 where a brief 'acknowledge' message is sent via transmit data port 215 to packet data encoder 220 and then to rf transmitter 225. All RF transmissions from communicator 11 are sent as brief packet 'bursts', thus maintaining a longer life for battery 290. Control then returns to step 4001. If at step 4001 it is determined that a 'page' message is not received, then control passes to step 4005 where a determination is made as to whether a 'manual phone number flag' is set, and if the flag is not set control passes to step 4006 where a determination is made as to whether a data message is received from the ultrasonic transmitter 12 via the serial data input 280, and if such a message is received then control passes to step 4007. At step 4007 a determination is made as to whether a 'auto phone number flag' is set and if the flag is not set control passes to step 4009 where the 'auto phone number flag' is set. Control then passes to step 4010 where the phone number which was embedded in the message received at step 4006 is stored in the memory of microprocessor 200. Control then passes to step 4011 where a 'phone number' message is constructed using the phone number of step 4006, and the subscriber DID number 701, as prestored in the RAM of microprocessor 200. This message is then sent to transmit data port 215. Control then passes to step 4012 where an 'acknowledge timer' is started. Control then passes to step 4013 where a '20 second ultrasonic data timer' is started. Control then returns to step 4001. If at step 4007 it is determined that the 'auto phone number flag' is set, then control passes to step 4008, where a determination is made as to whether the phone number received embedded in the message at step 4006 is the same as the phone number currently in memory as stored at step 4010. If the phone numbers match then control passes to step 4013. If at step 4008 it is determined that the phone numbers are different, indicating that the subscriber has moved to a different room, then control passes to step 4009. If at step 4005 it is determined that the 'manual phone number flag' is set, or if at step 4006 it is determined that a data message is not being received via input port 280, then control passes to step 4014 where a determination is made as to whether the '20 second ultrasonic data timer' has just expired, and if it has, indicating that the subscriber is no longer near an ultrasonic transmitter 12, then control passes to step 4015. At step 4015 the 'auto phone number flag' is cleared and control passes to step 4016 where the a 'new mode memory request' message is constructed using the stored subscriber DID number 701, and the stored 'default mode memory'. Control then passes to step 4017 where an 'acknowledge timer' is started, and then control returns to step 4001. If at step 4014 it is determined that the '20 second ultrasonic data timer' has not just expired, then control passes to step 4018 where a determination is made as to whether the 'acknowledge timer' has just expired, and if it has just expired control passes to step 4019 where the message is re-sent via transmit data port 215. Then at step 4020, an 'acknowledge timer' is started, and then control returns to step 4001. If at step 4018 it is determined that the 'acknowledge timer' has not expired, then control passes to step 4021 where a determination is made as to whether an 'acknowledge' message with a pager number that corresponds to this communicator 11 is received via the receive data input 245, and if such a message is received then control passes to step 4022 where the 'acknowledge timer' is cleared. Control the returns to step 4001. If at step 4021 it is determined that an 'acknowledge' message is not received, then control passes to step 4023 where a determination is made as to whether the subscriber is entering data via the keypad 205, and if the subscriber is not entering data, then control returns to step 4001. If the subscriber is entering data via the keypad, as determined at step 4023, then control passes to step 4024 where a determination is made as to whether the subscriber wishes to program the Telephone Control System 1 to operate under control of a new mode memory. If this is the case, then control passes to step 4025 where a 'new mode memory request message' is constructed and sent to transmit data port 215, and then control passes to step 4033 where an 'acknowledge timer' is started, and then control returns to step 4001. If at step 4024 it is determined that the subscriber is not selecting a new mode memory, then control passes to step 4026 where a determination is made as to whether the subscriber is selecting the dynamic mode assignment feature, and if this is the case, then control passes to step 4027 where a 'set dynamic mode assignment mode' message is constructed and sent to transmit data port 215, and then control passes to step 4033. If at step 4026 it is determined that the subscriber is not selecting the dynamic mode assignment feature, then control passes to step 4028 where a determination is made as to whether the subscriber is disabling the dynamic mode assignment feature, and if this is the case, then control passes to step 4029 where a 'disable dynamic mode assignment mode' message is constructed and sent to transmit data port 215, and then control passes to step 4033. If at step 4028 it is determined that the subscriber is not disabling the dynamic mode assignment mode, then control passes to step 4030 where a determination is made as to whether the subscriber is entering a phone number, and if a phone number is being entered then control passes to step 4031 where the 'manual phone number flag' is set. Control then passes to step 4032 where a 'phone number' message is constructed using the phone number entered by the subscriber at step 4030, and this message is transmitted via transmit data port 215. Control then passes to step 4033. If at step 4030 it is determined that the subscriber is not entering a phone number, then control passes to step 4034 where a determination is made as to whether the subscriber is selecting the auto-phone number mode, and if this is the case, then control passes to step 4035 where the 'manual phone number flag' is cleared, and control then passes to step 4015. If at step 4034 it is determined that the subscriber is not selecting the auto-phone number mode, then control passes to step 4036, where a determination is made as to whether the subscriber is entering new program data such a the stored DID number, the stored pager number, and the stored default mode memory. If the subscriber is attempting to modify any of these parameters, then control passes to step 4037 where the new data is stored in the RAM of microprocessor 200. Otherwise, control returns to step 4001. As can be understood from the explanation above, one of the primary functions of the Communicator 11 and the Communicator Access Facility 125 is to take the phone number identifying the location of the subscriber, as transmitted by ultrasonic transmitter 12, and cause that number to be used by the Telephone Control System

1 as the forwarding number for the subscriber. It should also be understood that in a similar fashion the ultrasonic transmitter 12 may transmit a 'mode memory number' which, if sent to the Telephone Control System 1 via the Communicator 11 and the Communicator Access Facility 125, would allow that mode memory to be used by the Telephone Control System 1 to specify the call handling mode for the subscriber. In this case, the Communicator 11 receives a data message from an ultrasonic transmitter 12 and determines that the message contains a mode memory. The Communicator 11 then transmits a 'new mode memory request' message, which includes the subscriber's DID number 701, via its RF transmitter 225. This message is received by packet radio transceiver 9 and sent to the CAF 125 via data line 10. The CAF 125 then sends a "change to new mode memory" message to CPF 100 via data network interface 3801. CPF 100 then copies the mode memory referred to in the message to this subscriber's "subscriber master record" 700. To further illustrate this process, consider the example of a hospital operating room where an ultrasonic transmitter 12 is transmitting a message containing a mode memory number which corresponds to the "message center" call handling mode. If a doctor, carrying a Communicator 11 enters the operating room, then the Telephone Control System is automatically programmed to send his calls to the "message center."

A block diagram of the Pager Dialing Facility (PDF) 105 is illustrated in FIG. 41. Standard tip-ring line 5 is shown connected to call processor 4100, which contains a tip-ring interface, DTMF generators, call progress detectors. The functions of call processor 4100 are well known in the art, and many products, such as the Model D41B manufactured by Dialogic Corporation, exist commercially which can accomplish these functions. The PDF 105 also contains a CPU 4101 which contains a microprocessor, a boot ROM, a RAM, and a disk. The PDF 105 also contains a data network interface module 4103 which connects to the high speed data network 150. The functions of data network interface 4103 are well known in the art, and many products, such as the Model COM4i from Digiboard Corporation, exist commercially which can accomplish these functions. The call processor 4100, the CPU 4101, and the data network interface 4103 are all shown connected to an internal data bus 4102. The CPU 4101 initializes itself at power-up using the boot ROM and then loads a control program into memory which it then executes.

A flowchart of the Pager Dialing Facility Main Task is illustrated in FIG. 42. This program is loaded into memory and executed by CPU 4101 of the PDF 105. The Pager Dialing Facility Main Task is entered at step 4200 and control passes to step 4201 where a determination is made as to whether a 'page' message is received from CPF 100 via data network interface 4201, and if the message is not received then control remains at step 4201. If a 'page' message is received, then the 'pager number' and the 'display digits' are retrieved from the message, and control passes to step 4202 where an 'attempt count' is set to a value of 1. Control then passes to step 4203, where line 5 is taken off hook, and then control passes to step 4204 where call processor 4100 dials the 'pager number'. Control then passes to step 4205 where a determination is made as to whether the call has not been answered due to a time-out or a non-answer signal such as operator intercept, busy, or reorder. If such a signal or time-out condition is not detected then control passes to step 4206 where a determination is made as to whether the call has been answered by the paging terminal, and if the call has not been answered, control returns to step 4205. If at step 4206 it is determined that the call is answered, then control passes to step 4207 where a 1 second pause is initiated, and then control passes to step 4208 where the 'display digits' are dialed by call processor 4100. Control then passes to step 4209 where the pager termination digit '#' is dialed, and then control passes to step 4210 where the line 5 is placed on hook. Control then passes to step 4211 where a 2 second delay is initiated before returning control to step 4201. If at step 4205 it is determined that a time-out or non-answer signal is detected, then control passes to step 4212 where the 'attempt count' is incremented. Control then passes to step 4213 where the 'attempt count' is checked and if it is found to be not equal to ten then control passes to step 4214 where the line 5 is placed on hook and then after a 2 second pause a step 4215, control returns to step 4203 to make another attempt at dialing this number. If at step 4213 it is found that the 'attempt count' is now equal to 10, then this page is abandoned by returning control to step 4201.

A block diagram of the Client Services Facility (CSF) 130 is illustrated in FIG. 43. The Client Services Facility (CSF) 130 is used by the service bureau which provides the Telephone Control System service to it's subscribers. The CSF 130 allows a client services representative to gain access to the database contained in the CPF 100, and thus be able to review and modify the subscriber master records 700 and mode memories 800 of the subscribers. The CSF 130 contains a CPU 4300 which contains a microprocessor, a boot ROM, a RAM, and a disk. The CSF 130 also contains a data network interface module 4301 which connects to the high speed data network 150. The functions of data network interface 4301 are well known in the art, and many products, such as the Model COM4i from Digiboard Corporation, exist commercially which can accomplish these functions. Also shown is a display monitor 4302, and a keyboard 4303. The CPU 4300 initializes itself at power-up using the boot ROM and then loads a control program into memory which it then executes.

A flowchart of the Client Services Facility Main Program is illustrated in FIG. 44. This program is loaded into memory and executed by CPU 4300 of the CSF 130. The Client Services Facility Main Program is entered at step 4400 and control passes to step 4401 where a determination is made as to whether the client services representative has entered the DID number for a particular subscriber and has requested a subscriber master record 700, and if this is the case then control passes to step 4402 where a 'request master record' message is sent via data network interface 4301 to CPF 100. Control then passes to step 4403 where the subscriber master record 700 is removed from the response message from the CPF 100, and is displayed on monitor 4302. Then at step 4404, the client services representative is allowed to review and modify the contents of the subscriber master record 700 using monitor 4302 and keyboard 4303. Then at step 4405 a determination is made as to whether the client services representative is completed with this operation, and if not, then control returns to step 4404. When the operation is complete, then control passes to step 4406 where an 'update master record' message is constructed and sent to CPF 100 via data network interface 4301. Control then returns to step 4401. If at step 4401 it is determined that the client services representative is not requesting a subscriber master record 700, then control passes to step 4407 where a determination is made as to whether the client services representative has entered the DID number for a particular subscriber and has requested a subscriber mode memory 800, and if this is the case then control passes to step 4408 where a 'mode memory inquiry' message is sent via data network interface 4301 to CPF 100. Control then passes to step 4409 where the mode memory 800 is removed from the response message from the CPF 100, and is displayed on monitor 4302. Then at step 4410, the client services representative is allowed to review and modify the contents of the mode memory 800 using monitor 4302 and keyboard 4303. Then at step 4411 a determination is made as to whether the client services representative is completed with this operation, and if not, then control returns to step 4410. When the operation is complete, then control passes to step 4412 where an 'update mode memory' message is constructed and sent to CPF 100 via data network interface 4301. Control then returns to step 4401. If at step 4407 it is determined that the client services representative is not requesting a new mode memory, the control passes to step 4413 where a determination is made as to whether the client services representative has entered a DID number and wishes to activate a new subscriber for this number. If this is the case then control passes to step 4414 where a 'create a new subscriber message' is generated with this DID number and the message is sent to CPF 100 via data network interface 4301. Control then returns to step 4401. If at step 4413 it is determined that the client services representative does not wish to create a new subscriber, then control returns to step 4401.

While a preferred embodiment of the invention has been described in detail, it should be apparent that many modifications and variations thereto are possible, all of which fall within the true spirit and scope of the invention. For example, while the preferred embodiment of the control system provides voice synthesized type courtesy messages, any appropriate tones, beeps, etc. would serve as a courtesy message and such is the use of that term throughout the claims appended hereto. In addition, the term "line" as used herein and in the claims appended hereto includes both lines and trunks. In addition, whereas the preferred embodiment of the invention uses the term "line" to describe the interconnecting medium between the control system and the central exchange, it should be understood throughout the specification and claims that "line" refers to tip and ring pairs, trunks or any other form of connecting circuits.

Claims:

1. A control system adapted for connection to a telephone exchange for receiving and processing calls from a caller to a user, the control system comprising: input/output means adapted for connection to telephone exchange lines to input and output telephone calls; switching control means for controllably connecting a call on one line to a different line; memory means for controllably storing and recalling electronic signals; and electronic processing means for accessing said memory means, switching control means and input/output means to direct the flow of input and output calls, said electronic processing means including means for:

a) processing calls to the system directed to a specific user in a selected one of at least two distinct modes, one such mode being call forwarding in which case the processing means proceeds with items b) through d),

b) recalling from said memory means a forwarding number for said user,

c) implementing a call to said forwarding number,

d) switching the caller's call to said forwarding number,

e) identifying a call to said system from a specific user,

f) changing said specific user's memory means stored forwarding number responsive to a command from said specific user, and

g) changing a users call processing mode responsive a command from said user.

2. The control system of claim 1 wherein the electronic processing means further comprises means for recalling from said memory means a courtesy message indicating that a call transfer is being effectuated and transmitting said message to said caller.

3. The control system of claim 1 including a message center mode wherein the electronic processing means further comprises means for: storing as a user mode control signal a message center mode signal that represents the user not being available and, upon recalling said message center mode signal in processing an

incoming call, recalling from said memory means a courtesy message indicating that the user is not available and that the caller can leave a message, transmitting said courtesy message to said caller, recording a message sent by the caller and allowing the specific user to call the system and access such message.

4. The control system of claim 1 including a priority call screening mode wherein the electronic processing means further comprises means for: storing as a user mode control signal a priority screening mode signal that represents the user not being available except for a priority call and, upon recalling said priority screening mode signal in processing an incoming call, recalling from said memory means a message indicating that the user is not available unless the call is a priority call in which event the caller should transmit a specific command from his calling station, transmitting said message to said caller and, responsive to said caller transmitting said specific command, proceeding with steps b) - d) to forward said call.

5. The control system of claim 4 wherein the electronic processing means further comprises means for: responding to said caller not transmitting said specific command to process said caller's call in accordance with a predetermined default processing mode.

6. The control system of claim 4 wherein the electronic processing means further comprises means for: responding to said caller not transmitting said specific command by recalling from said memory means a courtesy message indicating that the caller can leave a message, transmitting said message to said caller, recording a message sent by the caller, and allowing the specific user to call the system and access such message.

7. The control system of claim 1 including a VIP screening mode wherein the electronic processing means further comprises means for: storing as a user mode control signal a VIP screening mode signal that represents the user not being available except for callers having a VIP code and, upon recalling said VIP screening mode signal in processing an incoming call, recalling from said memory means a message indicating that the user is not available unless the caller has said VIP code in which event the caller should transmit said VIP code from his calling station,

transmitting said message to said caller and, responsive to said caller transmitting said VIP code, proceeding with steps b) - d) to forward said call.

8. The control system of claim 7 wherein the electronic processing means further comprises means for: responding to said caller not entering said VIP code to process said caller's call in accordance with a predetermined default processing mode.

9. The control system of claim 7 wherein the electronic processing means further comprises means for: responding to said caller not entering said VIP code by recalling from said memory means a courtesy message indicating that the caller can leave a message, transmitting said message to said caller, recording a message sent by the caller, and allowing the specific user to call the system and access such message.

10. The control system of claim 1 including a branch routing mode wherein the electronic control means further comprises means for: responding to an incoming call to a specific user to recall from said memory means a courtesy message indicating to the caller that his call could be branched to any of a plurality of options responsive to the caller transmitting an appropriate command corresponding to the desired branch, transmitting said courtesy message to said caller and, responsive to the caller transmitting said command, proceeding with steps b) -d) to forward said call.

11. The control system of claim 1 including a voice-screening mode wherein the electronic control system comprises means for: storing as a user mode control signal a voice-screen mode signal and, upon recalling said voice-screening mode signal in processing an incoming call, recalling from said memory means a message requesting that the caller transmit specific identification information, transmitting said message to said caller, recording the caller's response to said message, implementing a call to the user's forwarding number, transmitting the caller's response, and responding to a command response from the user to appropriately dispose of the caller's call.

12. The control system of claim 11 wherein the electronic control system

further comprises means for: upon completing said call to said user's forwarding number, recalling a courtesy message from said memory means indicating that the user is being called by a caller, interleaving with said courtesy message the caller's recorded identification information, said courtesy message further including the user's option to transmit a first control command signal to connect the user to the caller or an alternate control command signal to recall a second courtesy message from said memory means and send such message to said caller, transmitting said courtesy message to said user, and responding to the user transmitting a control command to appropriately dispose of the caller's call.

13. The control system of claim 11 wherein the electronic processing means further comprises means for: responding to a second control command from said user to recall a terminating message from said memory means, send said terminating message to said caller and terminate the connection to said caller, responding to a third control signal from said user to recall a leave-a-message signal, transmitting said leave-a-message signal to said caller and storing the caller's message in said memory means, and responding to a fourth control signal from said user to recall a predetermined phone number from said memory means, call said number and connect said caller to said predetermined number.

14. The control system of claim 1 including a meet-me mode wherein the electronic processing means further comprises means for: storing as a user mode control signal a meet-me mode signal and, upon recalling said meet-me mode signal in processing an incoming call, recalling from said memory means a pager access control procedure, controlling a paging system to page the user with an indication that he has a call waiting, identifying a call to said system from said user responsive to said page and, in the event the caller has waited on the line, switching the caller to the user.

15. The control system of claim 14 wherein the electronic processing means further comprises means for: in the event the caller has not waited on the line, so notifying the user by recalling a predetermined courtesy message and transmitting said courtesy message to the user.

16. The control system of claim 14 wherein the electronic processing means further comprises means for: recalling from said memory means a courtesy signal notifying the caller that the system must page the user and informing the caller that by transmitting a command the caller would be allowed to record a message, transmitting said courtesy signal to the caller before initiating the page to the user and, responsive to receipt of said command from the caller, recording the caller's message.

17. The control system of claim 15 wherein the electronic processing means further comprises means for: recalling from said memory means a courtesy signal notifying the caller that the system must page the user and informing the caller that by entering a command the caller would be allowed to record a message, transmitting said courtesy message to the caller before initiating the page to the user and, responsive to receipt of said command from the caller, recording the caller's message.

18. The control system of claim 1 in combination with a paging system wherein each user carries a paging device adapted to receive radio frequency signals broadcast from a central paging system with a unique identification code assigned to each paging device, and wherein: each paging device includes: a radio frequency transmitter; and control means for controlling said radio frequency transmitter such that a data signal may be transmitted to the control system to revise said control system memory means.

19. The combination of claim 18 wherein said controller means comprises a keypad and associated electronics allowing the user to manually revise said control system memory means stored signals.

20. The combination of claim 18 in further combination with a locater system wherein said locater system comprises a plurality of predeterminedly located locater transmitters, with each locater transmitter transmitting to a proximate paging device a signal corresponding to the telephone number of a nearby phone, and wherein said control means controls said radio frequency transmitter to transmit a signal related to said telephone number signal to said control system to revise said control system

memory means, whereby the control system will automatically forward incoming calls to a telephone near said user.

21. The combination of claim 20 wherein: each locater transmitter transmits an ultrasonic signal; and wherein: each paging device includes an ultrasonic receiver for receiving said ultrasonic signal.

22. The combination of claim 21 wherein each locater transmitter transmits an ultrasonic signal of approximately 40 kiloHertz modulated by said locater signal at a rate of approximately 75 baud.

23. The control system of claim 18 wherein each paging device includes a keypad for entry of commands by the user.

24. The control system of any one of claims 1 through 23 wherein a command from said user is entered via the user's telephone keypad.

25. The control system of any one of claims 1 through 23 wherein a command from said user is entered via a user voice command.

26. The control system of claim 1 wherein the electronic processing means further comprises means for: identifying a call to said system from a specific user requesting the present mode by which the system processes incoming calls directed to said specific user, recalling from said memory means the current mode by which the system processes incoming calls directed to said specific user, recalling from said memory means a courtesy message which indicates said current mode and informs the user as to how to enter a mode command to change said mode, transmitting said courtesy message to said user, and responding to said user transmitting a specific mode command to appropriately change the mode by which the system processes subsequent incoming calls directed to said user.

27. The control system of claim 26 wherein said user mode commands are entered via the user's telephone keypad.

28. The control system of claim 26 wherein said user mode commands are entered via voice commands from said user.

29. The control system of claim 1 including a schedule mode wherein said electronic processing means further comprises means for: responding to said user

transmitting a sequence of schedule command modes and corresponding times for each such command mode to appropriately automatically change to the commanded modes at the commanded times.

30. The control system of claim 1 including a timer mode wherein said electronic processing means further comprises means for: responding to said user entering a timer command mode and a corresponding time interval command to appropriately change to the commanded mode for said commanded interval, thereafter reverting to a predetermined default mode.

31. The control system of claim 1 wherein said telephone exchange provides an automatic number identification of a caller (ANI) to said control system, and wherein the electronic processing means further comprises means for: identifying a call to said system by a user; and utilizing said ANI as said user's memory means stored forwarding number, whereby the user can update his forwarding number without manually entering the number of the telephone at his present location.

32. The control system of claim 31 wherein calls to said system are made by dialing a sequence which includes digits to identify one of a plurality of call processing modes and wherein the electronic processing means further comprises means for: identifying a call to said system by a specific user by matching the ANI with a memory means stored number for said user and processing said digits as a specific mode command control signal from said user.

33. The control system of claim 32 wherein said user's telephone station includes a telephone exchange provided speed calling service wherein multi-digit numbers can be recalled and dialed by the user entering a limited digit code, and wherein a sequence of said speed calling numbers are programmed as said specific mode command control signals, whereby a user can change the system mode control for processing his calls by entering the appropriate limited digit code.

34. The control system of claim 1 wherein said electronic processing means further includes means for: identifying a message center control signal from said caller and, regardless of the user's present call processing mode, permitting said caller to access the user's provided message center memory and store a message for

later access by the user.

35. The control system of claim 1 wherein said electronic processing means further includes means in implementing step c) for:upon implementing a call to said forwarding number, identifying if the user's number is busy or not answered and, in such event, disposing of said call in a predetermined manner.

36. The control system of claim 1 wherein said electronic processing means further includes means in implementing step c) for: upon implementing a call to said forwarding number, initiating a predetermined time-out interval, identifying the condition of the user's line not being busy or not ringing within said time-out interval to thereby implement step d), or identifying the condition of the user's phone ringing within said time-out interval to allow the user's phone to ring for a predetermined sequence and, if the user does not answer within said sequence, terminating the attempt to forward the call or identifying the condition of the user's phone being busy within said time-out interval to thereby terminate the attempt to forward the call.

37. The control system of claim 36 wherein the duration of said time-out interval is predeterminedly a function of whether the call from the caller is detected as local or long distance.

38. The control system of claim 36 wherein the electronic processing means further comprises means responsive to terminating the attempt to forward the call to recall from said memory means a courtesy message indicating that the caller can leave a message, transmitting said message to the caller, recording a message sent by the caller and allowing the user to call the system and access such message.

39. The control system of claim 1 wherein the electronic processing means further comprises means in implementing step for: identifying a command from said user requesting a desired call processing mode and responding to a predetermined help command from said user to recall from said memory means a courtesy message related to said specific call processing mode and transmitting said courtesy message to said user.

40. The control system of claim 1 wherein the electronic processing means further comprises means for: identifying a predetermined client service command

transmitted by said user and, responsive thereto, connecting said user to a client service operator.

41. The control system of claim 1 wherein the electronic processing means further comprises means following step e) for: identifying a command from said user indicating that the user desires to place a call to a predetermined number, calling said number, connecting said user to said called number while maintaining the user to system connection such that upon completion of said user's call the user is returned to the system to enter further commands.

42. The control system of claim 3 wherein the electronic processing means further comprises means for: upon said user calling the system and accessing a recorded message from a caller, identifying a command from said user indicating that the user desires to place a call to a predetermined number, calling said number, connecting the user to said called number while maintaining the user to system connection such that upon completion of said user's call the user is returned to his recorded messages in the system.

43. The control system of claim 1 wherein the electronic processing means step g) further comprises means for: implementing user call processing mode changes responsive to predetermined command sequences and further including user controls which, when transmitted, automatically implement said predetermined command sequences.

44. The control system of claim 1 wherein the electronic processing means further comprises means for: upon implementing call forwarding in accordance with steps b) through d), generating predetermined signals which to a caller indicate normal call processing operation but to a user indicate that the user can transmit his predetermined identification code to gain access to his system provided functions and, upon receipt of said predetermined identification code, terminating the call forwarding and transferring the user to the system's command mode.

45. The control system of claim 1 wherein the electronic processing means further comprises means for: upon implementing call forwarding in accordance with steps b) through d), recalling from said memory means a predetermined courtesy

message which directs the caller to ask for the called party by name upon completion of the call and transmitting said courtesy message to the caller.

46. The control system of claim 1 wherein the electronic processing means further comprises means for: upon implementing call forwarding in accordance with steps b) through d), recalling from said memory means a predetermined courtesy message which directs the caller to ask for the called party's extension upon completion of the call and transmitting said courtesy message to the caller.

47. The control system of either one of claims 45 or 46 wherein the particular courtesy message recalled from the memory means and transmitted to the caller is controllable by command from the user.

48. The control system of claim 1 wherein the electronic processing means further comprises means for: identifying a predetermined external message center command transmitted by said user and, responsive thereto, completing a call to a message center external of the control system and connecting the user to said external message center.

49. A control system for receiving and processing telephone calls from a caller to a user, the system comprising: detector means for detecting signals over a telephone line; signal generating means, responsive to control signals, for producing predetermined signals and transmitting said signals over a telephone line; memory means for controllably storing and recalling electronic signals; switching means for connecting a call on one line to another line; and central processing means for monitoring signals on a telephone line, accessing said memory means and controlling said signal generator means and switching means, said central processing means including means for:

a) identifying a call to said system directed to a user,

b) recalling from said memory means a user controlled call processing signal corresponding to a user selected one of a plurality of call processing modes and, responsive to said user controlled call processing signal dictating a voice-screen mode, proceeding by

c) recalling from said memory means a predetermined identification message requesting caller identification information,

d) controlling said signal generating means to send said identification message to said caller,

e) storing in said memory means the caller's response to said identification message,

f) recalling the user's number from said memory means and controlling said signal generating means to call said user,

g) recalling from said memory means the caller's response and sending said response to said user, and

h) responding to a control command from said user to dispose of the call from the caller in a selected one of a plurality of options.

50. The control system of claim 49 wherein the central processing means further includes means for:

h) i) responding to a first control command from said user to connect said caller to said user.

51. The control system of claim 50 wherein the central processing means further includes means for:

h) ii) responding to an alternate control command from said user to recall a predetermined message from said memory means and send said message to said caller.

52. The control system of claim 50 wherein the central processing means further includes means for:

h) ii) responding to a second control command from said user to recall a terminating message from said memory means, send said terminating message to said caller and terminate the connection to said caller.

53. The control system of claim 50 wherein the central processing means further includes means for:

h) ii) responding to a second control command from said user to recall a leave-a-message signal, sending said leave-a-message signal to said caller and storing the caller's message in said memory means.

54. The control system of claim 50 wherein the central processing means further includes means for:

h) ii) responding to a second control command from said user to recall a predetermined phone number from said memory means, call said number and connect said user to said predetermined number.

55. The control system of claim 49 wherein the central processing means further includes means for:

a) i) recognizing a predetermined signal transmitted by the caller and, responsive thereto, immediately proceeding with step f) to call said user and then connect said caller to said user.

56. A control system for receiving and processing telephone calls from a caller to a user, the system comprising: detector means for detecting signals over a telephone line; signal generating means, responsive to control signals, for producing predetermined signals and transmitting said signals over a telephone line; memory means for controllably storing and recalling electronic signals; switching means for connecting a call on one line to another line; and electronic processing means for monitoring signals on a telephone line as detected by said detection means, accessing said memory means and producing and applying predetermined control signals to said signal generating means and switching means, said central processing means including means for:

a) identifying a call to said system directed to a user,

b) recalling from said memory means a predetermined paging system control procedure for paging said user,

c) causing said signal generating means to control said paging system to page said user with an indication that the user has a call waiting,

d) identifying a call to said system from said user,

e) in the event said caller has waited on the line, connecting the caller to the user, and

f) in the event the caller has not waited on the line, so notifying the user by recalling a predetermined message from said memory means and causing said signal generating means to transmit said message to said user.

57. The control system of claim 56 wherein the electronic processing means further includes means for: recalling from said memory means a courtesy message notifying the caller that the system must page the user and allowing the user the option of recording a message in said memory means, transmitting said courtesy message to said caller and, responsive to said caller implementing the message option, recording said message in said memory means and allowing the user to access said message after responding to the page.

58. A paging system wherein each user carries a paging device adapted to receive radio frequency broadcast signals from a central paging system, with a unique identification code assigned to each paging device, characterized by:

a plurality of predeterminedly located ultrasonic transmitters, each transmitter transmitting and information signal; and

each paging device including:

an ultrasonic receiver for receiving the signal from a proximate ultrasonic transmitter, and

a radio frequency transmitter for transmitting to the central paging system a signal corresponding to the paging device received ultrasonic signal and the paging device assigned identification code,

whereby the paging system is provided with related information for each paging device.

59. The paging system of claim 58 wherein: said information signal is the phone number of a proximately located telephone, and said paging device radio frequency transmitter transmits said phone number related signal to said central paging system, whereby the central paging system is provided with the phone number of a phone proximate to said user.

60. The paging system of claim 59 in combination with a control system, wherein the control system comprises: means for detecting a call from a caller to said control system directed to a user, and processing means for accessing said central paging system and causing said central paging system to transmit a control signal to

the paging device of said user, which control signal causes said paging device to transmit back to said central paging system the proximate phone number signal, and implementing a call to said proximate phone number.

61. The system of claim 60 further comprising: means for connecting the caller to said proximate phone number.

62. The system of claim 60 wherein: each paging device includes user controllable means for transmitting telephone call processing control signals to said control system, and wherein each control system includes means responsive to said control signals to process incoming calls to said user in accordance therewith.

63. The system of claim 62 wherein one telephone call processing control signal causes said control system to forward to the user only selected calls.

64. The system of claim 59 wherein each locater transmitter transmits an ultrasonic signal of approximately 40 kiloHertz modulated by said locater signal at a rate of approximately 75 band.

65. A control system connected to the telephone exchange wherein each user of the system is assigned a unique telephone number and said control system routes a call to a user controlled number, the control system comprising:

input/output means adapted for connection to telephone exchange lines to input and output telephone calls;

switching control means for controllably connecting a call on one line to a different line;

memory means for controllably storing and recalling electronic signals; and

electronic processing means for accessing said memory means, switching control means and input/output means to direct the flow of input and output calls, said electronic processing means including means for responding to an incoming call to a specific user to recall from said memory means a courtesy message indicating to the caller that his call could be branched to any of a plurality of options responsive to the caller transmitting an appropriate command corresponding to the desired branch, transmitting said courtesy message to said caller and, responsive to the caller transmitting said implementing the commanded option.

66. A paging system wherein each user carries a paging device adapted to receive radio frequency signals broadcast from a central paging system with a unique identification code assigned to each paging device, and characterized in that:

each paging device includes:

a radio frequency transmitter, and controller means for controlling said radio frequency transmitter, said controller means operable in at least two distinct user selectable modes, the first such mode being responsive to receiving a signal broadcast by the central paging system and responding by transmitting predetermined data back to said central paging system.

67. The system of claim 66 wherein said controller means comprises a keypad and associated electronics allowing a user to manually enter data.

68. The system of claim 66 in combination with a locater system comprising a plurality of predeterminedly located locater transmitters with each locater transmitter transmitting a unique locater signal including locating information, and wherein said controller means includes means for receiving the signal from a proximate locater transmitter and controlling said radio frequency transmitter to re-transmit said locating information to the central paging system.

69. The system of claim 68 wherein each locater transmitter locater signal corresponds to the number of a nearby telephone and wherein said controller controls said radio frequency transmitter to transmit said telephone number related signal to said central paging system.

70. The system of claim 68 in combination with a telephone control system for predeterminedly controlling incoming calls to a user in a selected one of at least two distinct modes responsive to a mode control signal wherein a locater transmitter transmits a predetermined mode signal and wherein said paging device transmits a signal responsive to said mode control signal through said paging system to said telephone control system to control the mode thereof.

71. A control system for receiving and processing telephone calls from a caller to a user, the system comprising: detector means for detecting signals over a telephone line; signal generating means, responsive to control signals, for producing

predetermined signals and transmitting said signals over a telephone line; memory means for controllably storing and recalling electronic signals; switching means for connecting a call on one line to another line; and central processing means for monitoring signals on a telephone line, accessing said memory means and controlling said signal generator means and switching means, said central processing means including means for:

a) identifying a call to said system directed to a user,

b) recognizing a predetermined signal transmitted by the caller and, responsive thereto, immediately proceeding with step g) to call said user, otherwise proceeding to step c);

c) recalling from said memory means a user controlled call processing signal corresponding to a call processing mode and, responsive to said user controlled call processing signal dictating a voice-screen mode, proceeding by

d) recalling from said memory means a predetermined identification message requesting caller identification information,

e) controlling said signal generating means to send said identification message to said caller,

f) storing in said memory means the caller's response to said identification message,

g) recalling the user's number from said memory means and controlling said signal generating means to call said user,

h) recalling from said memory means the caller's response and sending said response to said user, and

i) responding to a control command from said user to dispose of the call from the caller in a selected one of a plurality of options.

72. The control system of claim 71 wherein the central processing means further includes means for:

i) i) responding to a first control command from said user to connect said caller to said user.

73. The control system of claim 72 wherein the central processing means

further includes means for:

i) ii) responding to an alternate control command from said user to recall a predetermined message from said memory means and send said message to said caller.

74. The control system of claim 73 wherein the central processing means further includes means for:

i) ii) a) responding to a second control command from said user to recall a terminating message from said memory means, send said terminating message to said caller and terminate the connection to said caller.

75. The control system of claim 74 wherein the central processing means further includes means for:

i) ii) b) responding to a third control command from said user to recall a leave-a-message signal, sending said leave-a-message signal to said caller and storing the caller's message in said memory means.

76. The control system of claim 75 wherein the central processing means further includes means for:

i) ii) c) responding to a fourth control command from said user to recall a predetermined phone number from said memory means, call said number and connect said user to said predetermined number.

77. The combination of claim 18 in further combination with a locater system wherein said locater system comprises a plurality of predeterminedly located locater transmitters, with each locater transmitter transmitting to a proximate paging device a signal corresponding to a call processing mode, and wherein said control means controls said radio frequency transmitter to transmit a signal related to said call processing mode to said control system to revise said control system memory means, whereby the control system will automatically change a user's call processing mode.

78. The combination of claim 77 wherein: each locater transmitter transmits an ultrasonic signal; and wherein: each locater transmitter transmits an ultrasonic receiver for receiving said ultrasonic signal.

79. The combination of claim 78 wherein each locater transmitter transmits

an ultrasonic signal of approximately 40 kilohertz modulated by said locater signal at a rate of approximately 75 baud.

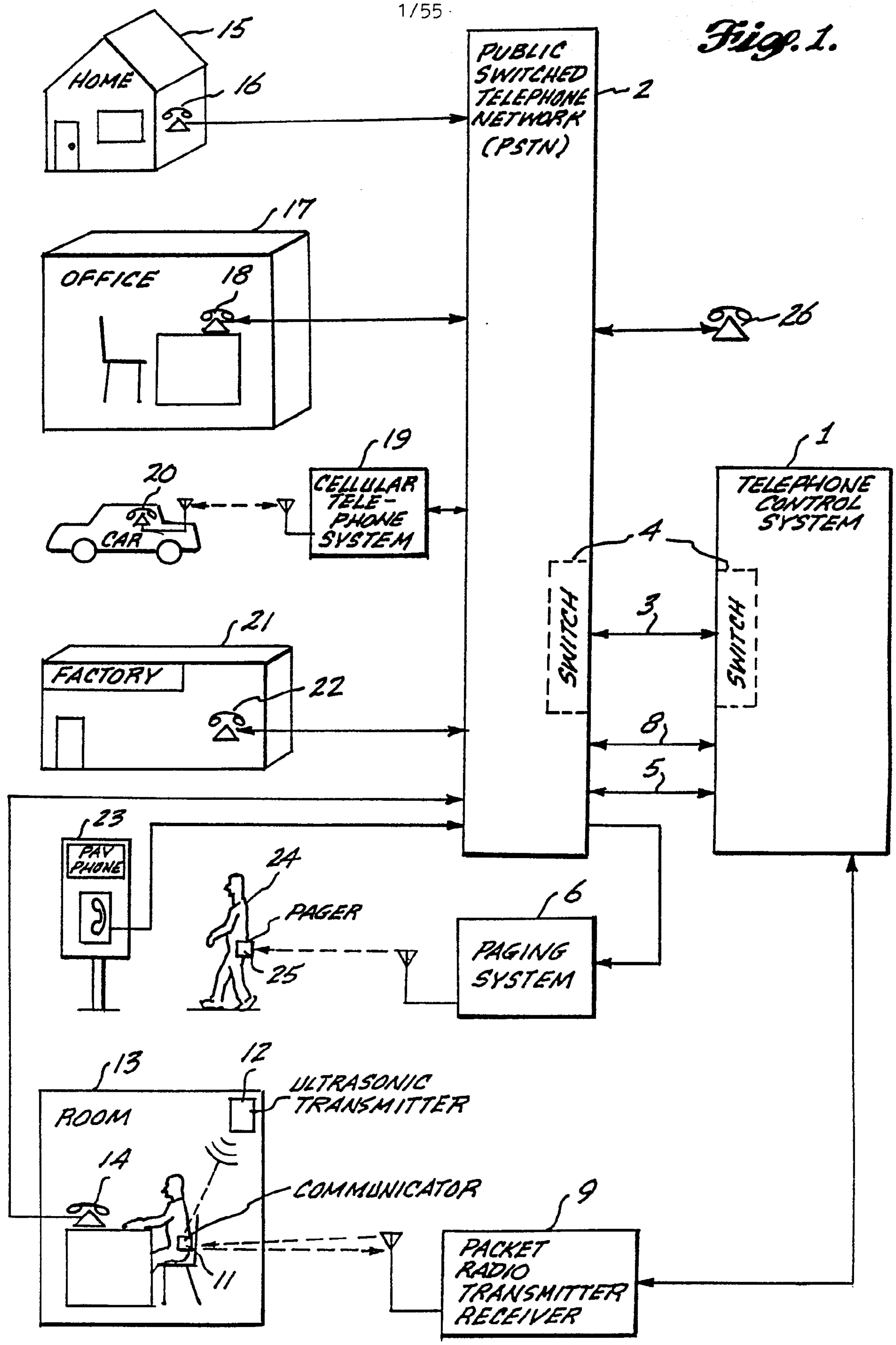
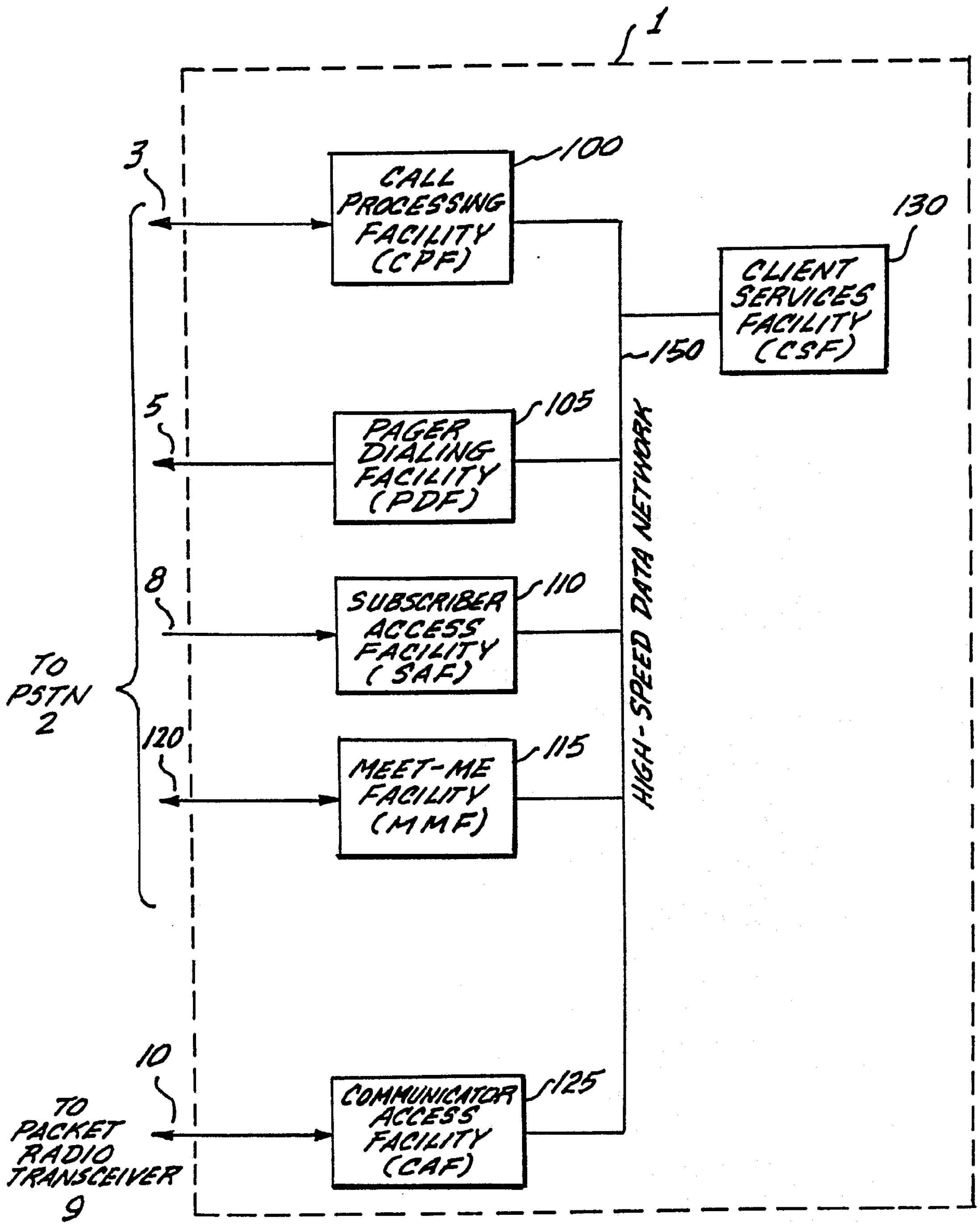


Fig. 1.
HOME
15
16
PUBLIC SWITCHED TELEPHONE NETWORK (PSTN)
2
17
OFFICE
18
26
19
20
CELLULAR TELE-PHONE SYSTEM
CAR
1
TELEPHONE CONTROL SYSTEM
4
SWITCH
3
SWITCH
21
FACTORY
22
8
5
23
PAY PHONE
24
PAGER
25
6
PAGING SYSTEM
13
12
ULTRASONIC TRANSMITTER
ROOM
14
COMMUNICATOR
9
11
PACKET RADIO TRANSMITTER RECEIVER

Fig. 2.

TELEPHONE CONTROL SYSTEM



SUBSTITUTE SHEET

Fig. 3.

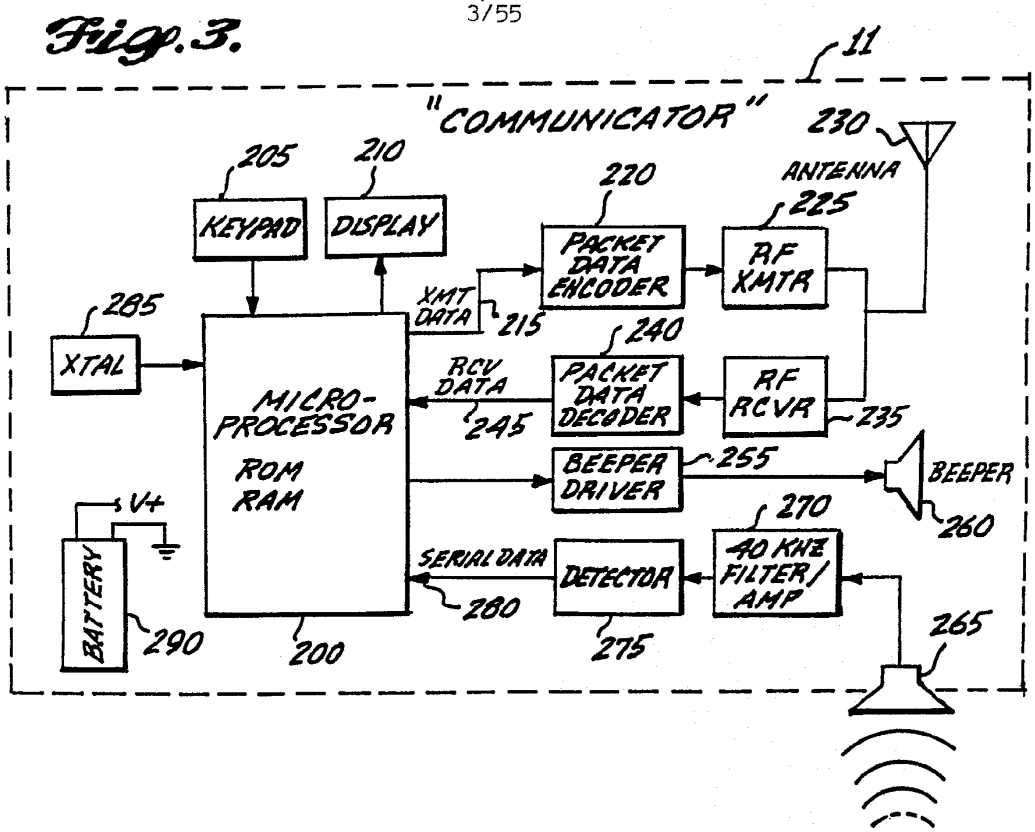
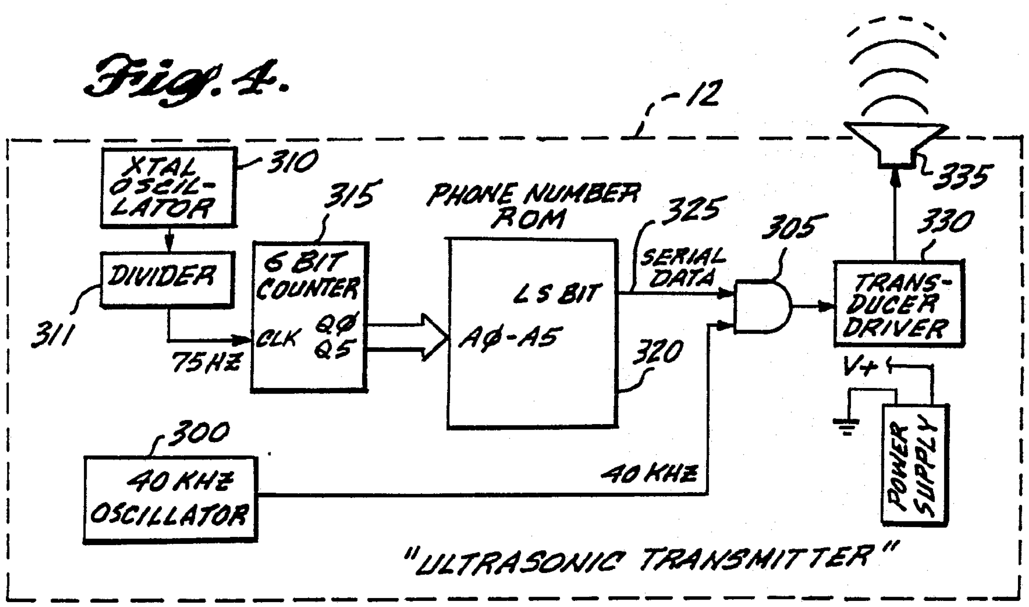

11
"COMMUNICATOR"
230
ANTENNA
205
KEYPAD
210
DISPLAY
220
PACKET DATA ENCODER
225
RF XMTR
XMT DATA
215
285
XTAL
MICRO-PROCESSOR
ROM
RAM
240
PACKET DATA DECODER
RCV DATA
245
RF RCVR
235
BEEPER DRIVER
255
BEEPER
260
V+
BATTERY
290
200
SERIAL DATA
280
DETECTOR
275
270
40 KHZ FILTER/ AMP
265

Fig. 4.


12
XTAL OSCIL-LATOR
310
315
6 BIT COUNTER
CLK
Q0
Q5
PHONE NUMBER ROM
325
LS BIT
A0-A5
SERIAL DATA
305
335
330
TRANS-DUCER DRIVER
DIVIDER
311
75HZ
320
V+
POWER SUPPLY
300
40 KHZ OSCILLATOR
40 KHZ
"ULTRASONIC TRANSMITTER"

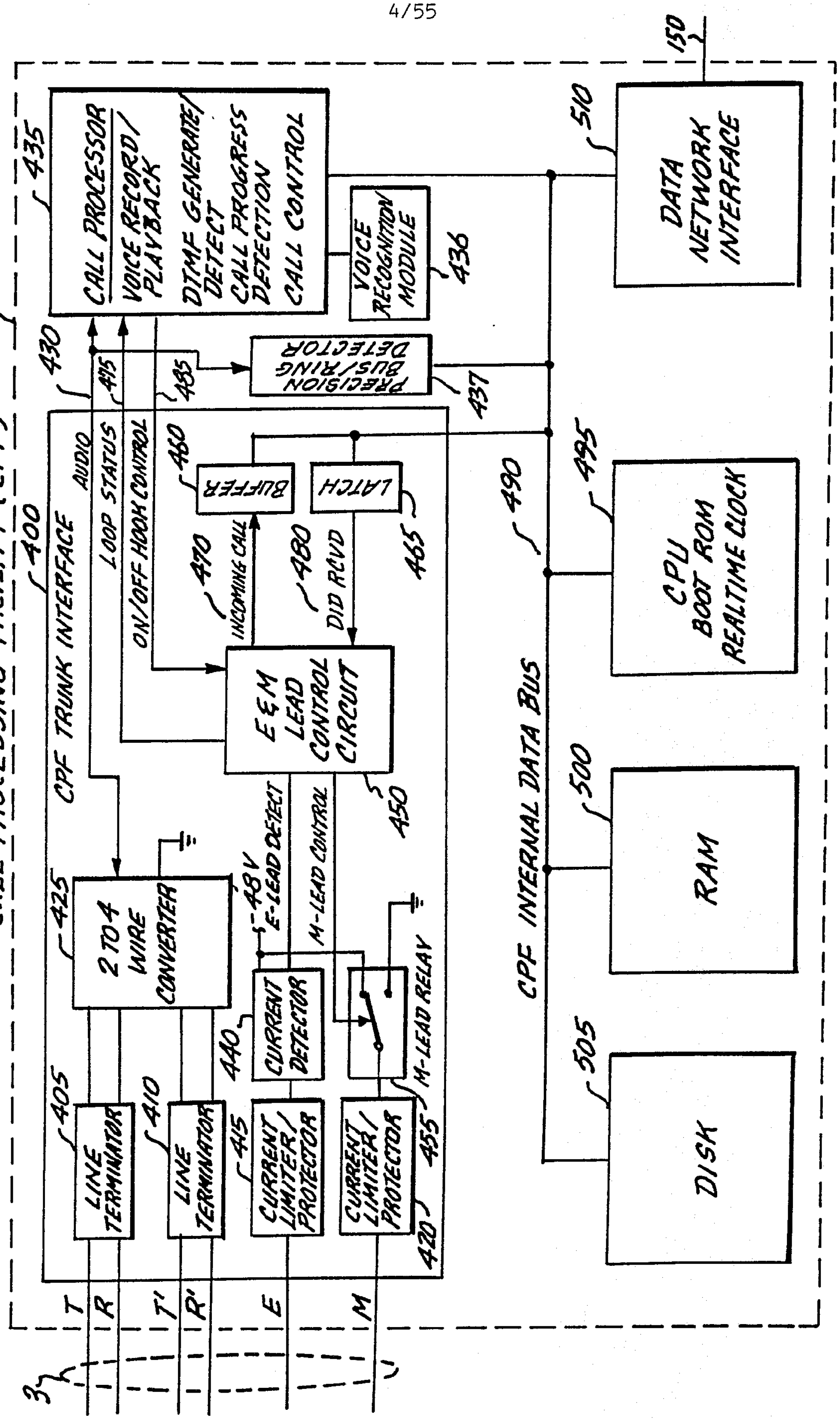

CALL PROCESSING FACILITY (CPF)
100
CALL PROCESSOR
VOICE RECORD/ PLAYBACK
DTMF GENERATE/ DETECT
CALL PROGRESS DETECTION
CALL CONTROL
435
VOICE RECOGNITION MODULE
436
PRECISION BUS/RING DETECTOR
437
DATA NETWORK INTERFACE
510
150
CPF TRUNK INTERFACE
400
AUDIO
LOOP STATUS
ON/OFF HOOK CONTROL
430
475
485
BUFFER
460
LATCH
465
INCOMING CALL
470
DID RCVD
480
E&M LEAD CONTROL CIRCUIT
450
CPF INTERNAL DATA BUS
490
CPU
BOOT ROM
REALTIME CLOCK
495
RAM
500
DISK
505
2 TO 4 WIRE CONVERTER
425
-48V
E-LEAD DETECT
M-LEAD CONTROL
CURRENT DETECTOR
440
M-LEAD RELAY
LINE TERMINATOR
405
LINE TERMINATOR
410
CURRENT LIMITER/ PROTECTOR
415
CURRENT LIMITER/ PROTECTOR
420
455
T
R
T'
R'
E
M
3

Fig. 5.

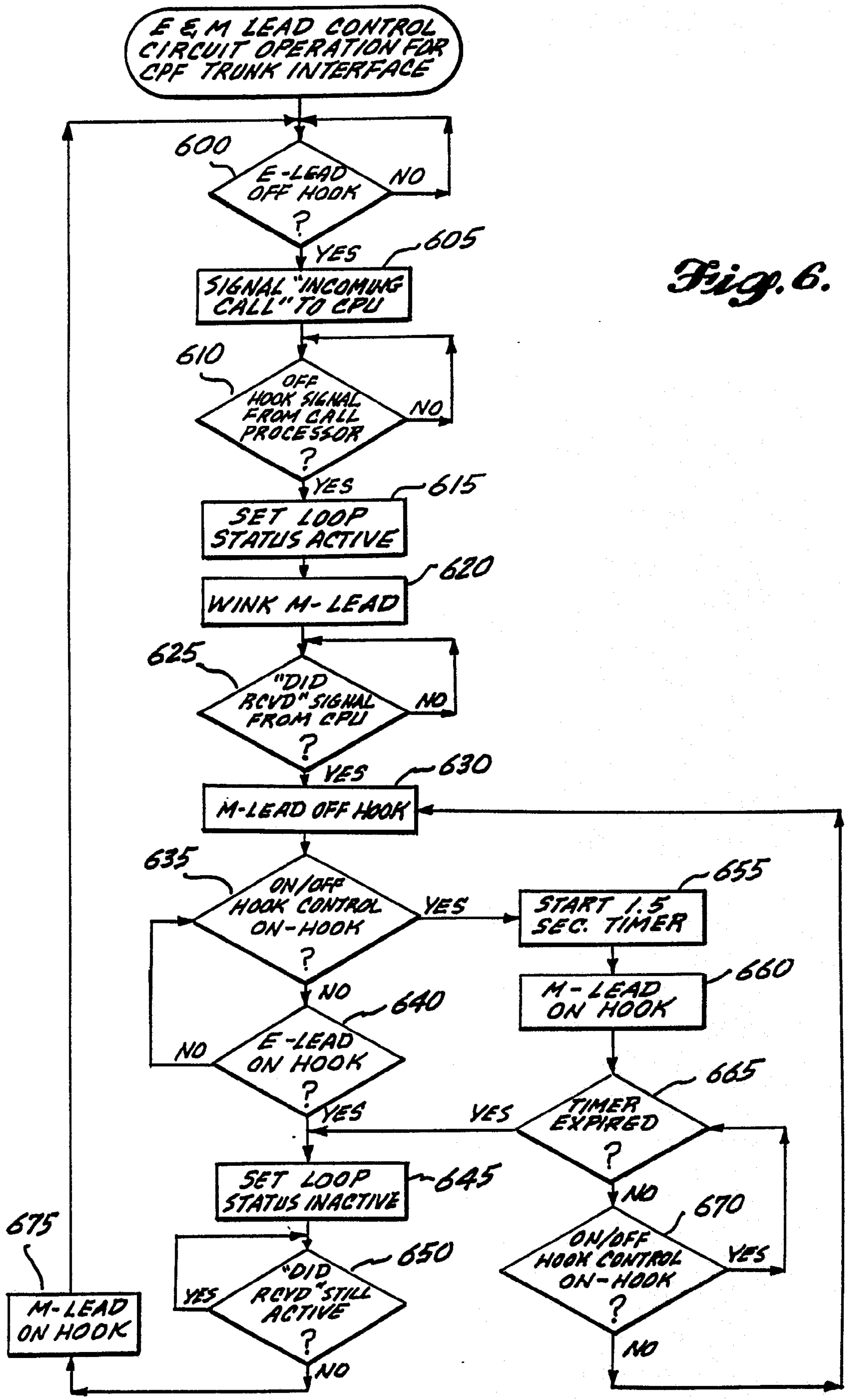

E & M LEAD CONTROL CIRCUIT OPERATION FOR CPF TRUNK INTERFACE
600
E-LEAD OFF HOOK ?
NO
YES
605
SIGNAL "INCOMING CALL" TO CPU
610
OFF HOOK SIGNAL FROM CALL PROCESSOR ?
NO
YES
615
SET LOOP STATUS ACTIVE
620
WINK M-LEAD
625
"DID RCVD" SIGNAL FROM CPU ?
NO
YES
630
M-LEAD OFF HOOK
635
ON/OFF HOOK CONTROL ON-HOOK ?
YES
655
START 1.5 SEC. TIMER
NO
640
E-LEAD ON HOOK ?
NO
660
M-LEAD ON HOOK
YES
665
TIMER EXPIRED ?
YES
645
SET LOOP STATUS INACTIVE
NO
670
ON/OFF HOOK CONTROL ON-HOOK ?
YES
675
M-LEAD ON HOOK
650
"DID RCVD" STILL ACTIVE ?
YES
NO
NO

Fig. 6.

Fig. 7.

SUBSCRIBER MASTER RECORD 700

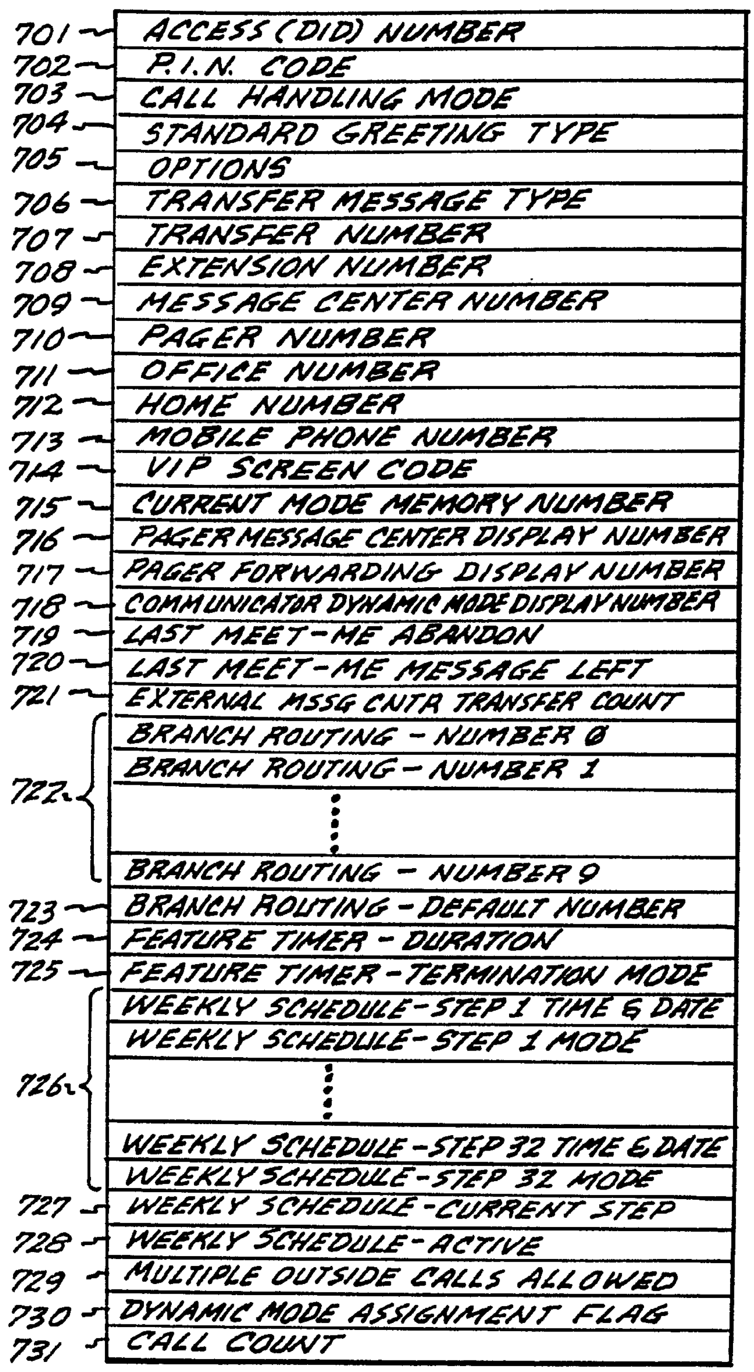
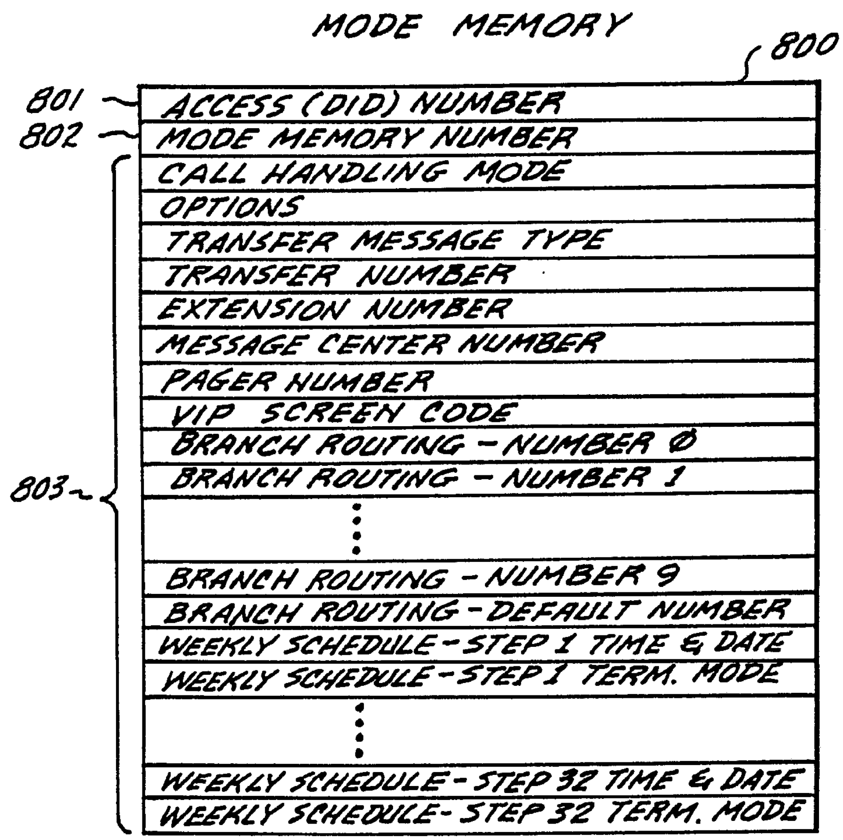


Fig. 8.


MODE MEMORY
800
801
ACCESS (DID) NUMBER
802
MODE MEMORY NUMBER
CALL HANDLING MODE
OPTIONS
TRANSFER MESSAGE TYPE
TRANSFER NUMBER
EXTENSION NUMBER
MESSAGE CENTER NUMBER
PAGER NUMBER
VIP SCREEN CODE
BRANCH ROUTING - NUMBER Ø
BRANCH ROUTING - NUMBER 1
803
BRANCH ROUTING - NUMBER 9
BRANCH ROUTING - DEFAULT NUMBER
WEEKLY SCHEDULE - STEP 1 TIME & DATE
WEEKLY SCHEDULE - STEP 1 TERM. MODE
WEEKLY SCHEDULE - STEP 32 TIME & DATE
WEEKLY SCHEDULE - STEP 32 TERM. MODE

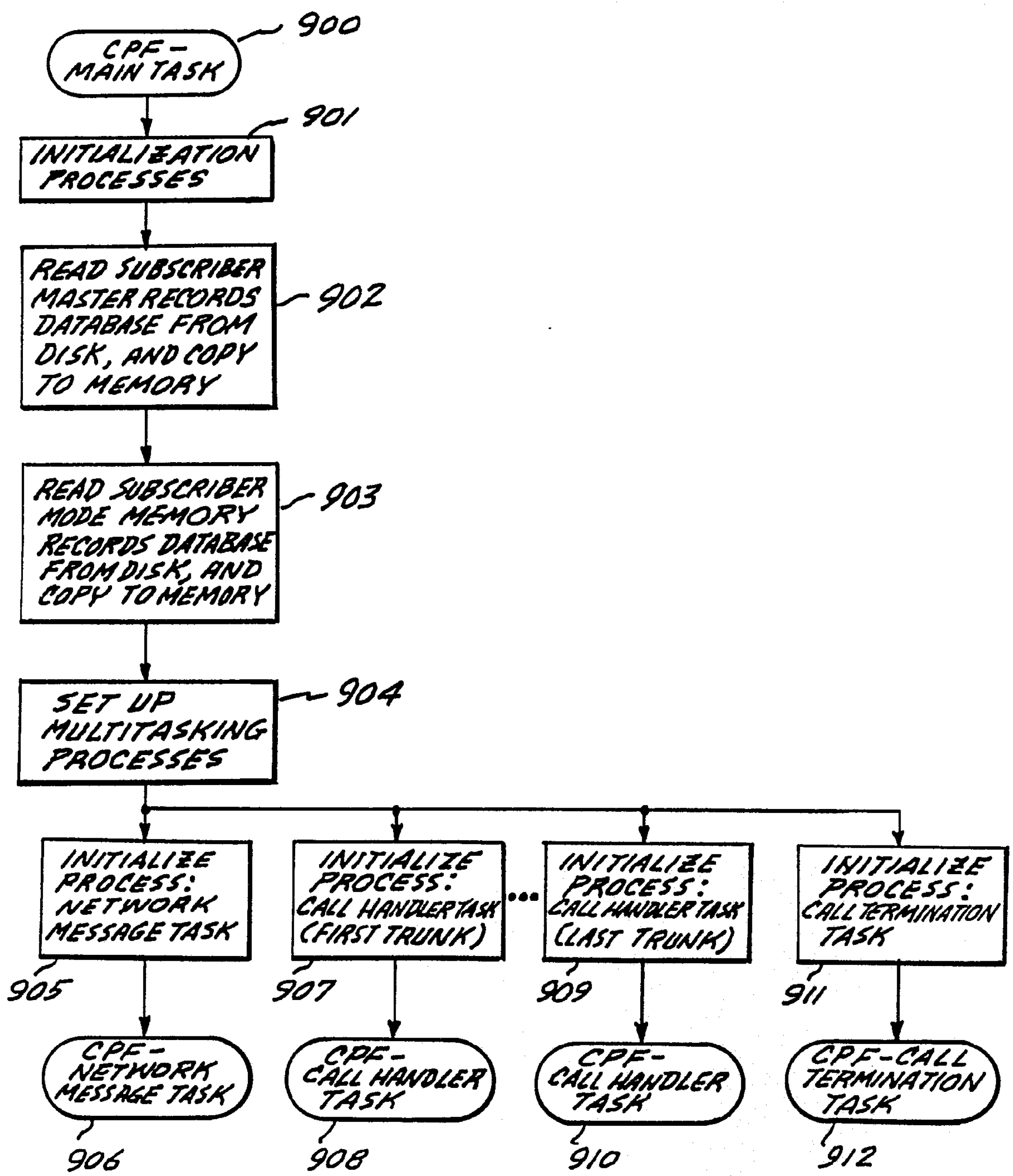

CPF - MAIN TASK
900
INITIALIZATION PROCESSES
901
READ SUBSCRIBER MASTER RECORDS DATABASE FROM DISK, AND COPY TO MEMORY
902
READ SUBSCRIBER MODE MEMORY RECORDS DATABASE FROM DISK, AND COPY TO MEMORY
903
SET UP MULTITASKING PROCESSES
904
INITIALIZE PROCESS: NETWORK MESSAGE TASK
905
INITIALIZE PROCESS: CALL HANDLER TASK (FIRST TRUNK)
907
INITIALIZE PROCESS: CALL HANDLER TASK (LAST TRUNK)
909
INITIALIZE PROCESS: CALL TERMINATION TASK
911
CPF - NETWORK MESSAGE TASK
906
CPF - CALL HANDLER TASK
908
CPF - CALL HANDLER TASK
910
CPF - CALL TERMINATION TASK
912

Fig. 9.

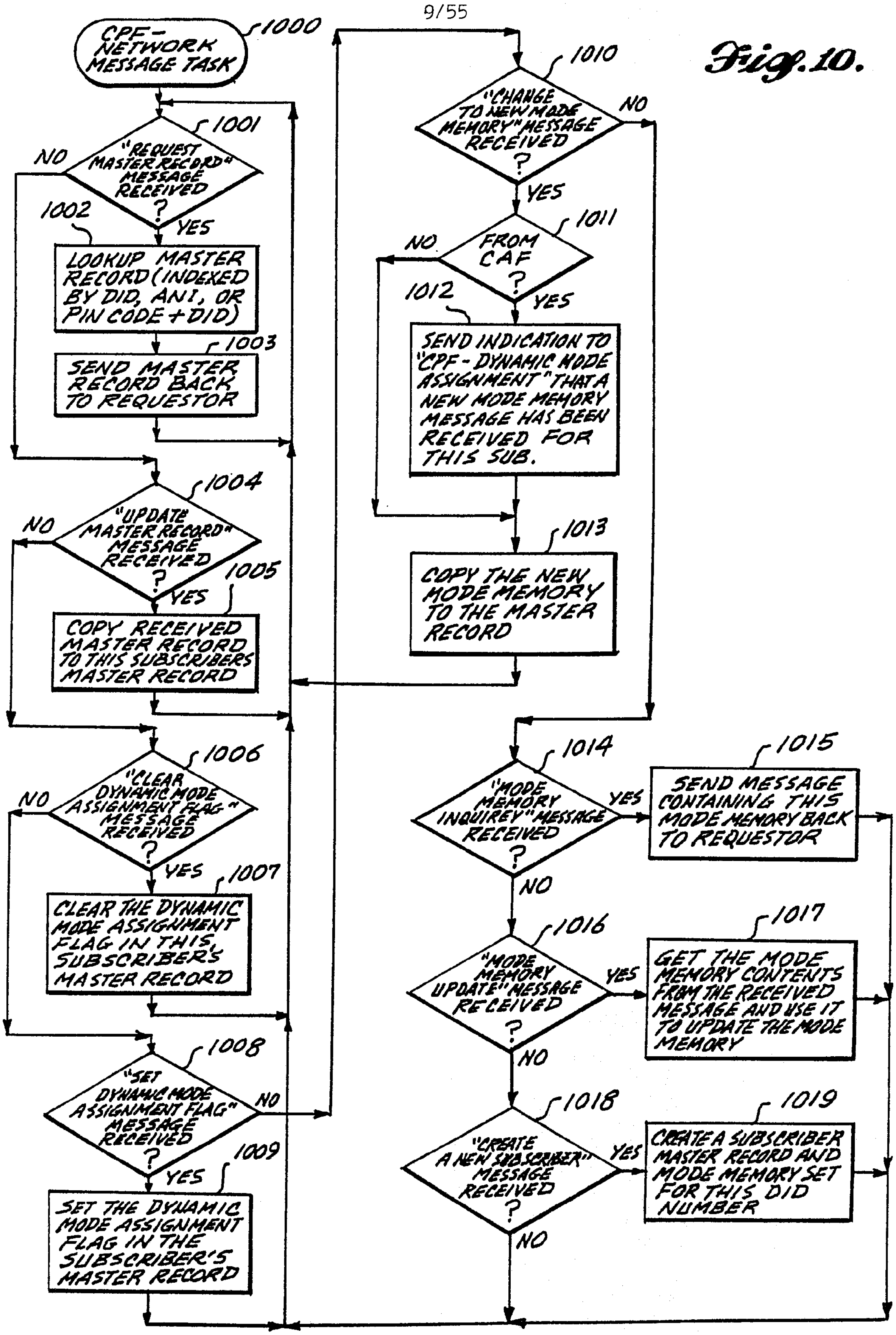


Fig. 10.
CPF - NETWORK MESSAGE TASK
1000
"REQUEST MASTER RECORD" MESSAGE RECEIVED ?
1001
NO
YES
1002
LOOKUP MASTER RECORD (INDEXED BY DID, ANI, OR PIN CODE + DID)
1003
SEND MASTER RECORD BACK TO REQUESTOR
"UPDATE MASTER RECORD" MESSAGE RECEIVED ?
1004
1005
COPY RECEIVED MASTER RECORD TO THIS SUBSCRIBERS MASTER RECORD
"CLEAR DYNAMIC MODE ASSIGNMENT FLAG" MESSAGE RECEIVED ?
1006
1007
CLEAR THE DYNAMIC MODE ASSIGNMENT FLAG IN THIS SUBSCRIBER'S MASTER RECORD
"SET DYNAMIC MODE ASSIGNMENT FLAG" MESSAGE RECEIVED ?
1008
1009
SET THE DYNAMIC MODE ASSIGNMENT FLAG IN THE SUBSCRIBER'S MASTER RECORD
"CHANGE TO NEW MODE MEMORY" MESSAGE RECEIVED ?
1010
FROM CAF ?
1011
1012
SEND INDICATION TO "CPF - DYNAMIC MODE ASSIGNMENT" THAT A NEW MODE MEMORY MESSAGE HAS BEEN RECEIVED FOR THIS SUB.
1013
COPY THE NEW MODE MEMORY TO THE MASTER RECORD
"MODE MEMORY INQUIREY" MESSAGE RECEIVED ?
1014
1015
SEND MESSAGE CONTAINING THIS MODE MEMORY BACK TO REQUESTOR
"MODE MEMORY UPDATE" MESSAGE RECEIVED ?
1016
1017
GET THE MODE MEMORY CONTENTS FROM THE RECEIVED MESSAGE AND USE IT TO UPDATE THE MODE MEMORY
"CREATE A NEW SUBSCRIBER" MESSAGE RECEIVED ?
1018
1019
CREATE A SUBSCRIBER MASTER RECORD AND MODE MEMORY SET FOR THIS DID NUMBER

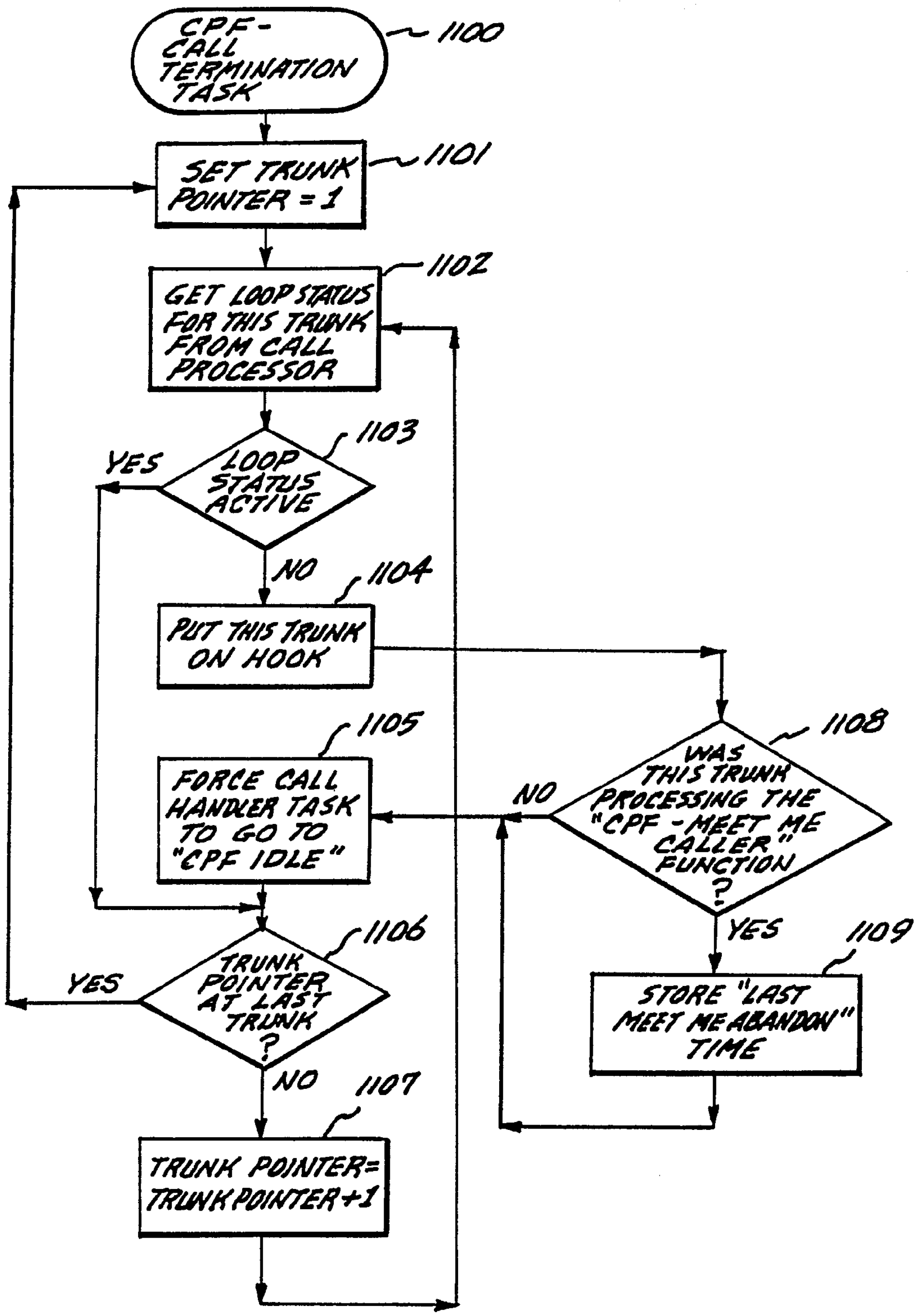

CPF-CALL TERMINATION TASK
1100
SET TRUNK POINTER = 1
1101
GET LOOP STATUS FOR THIS TRUNK FROM CALL PROCESSOR
1102
LOOP STATUS ACTIVE
1103
YES
NO
PUT THIS TRUNK ON HOOK
1104
FORCE CALL HANDLER TASK TO GO TO "CPF IDLE"
1105
TRUNK POINTER AT LAST TRUNK ?
1106
YES
NO
TRUNK POINTER= TRUNK POINTER+1
1107
WAS THIS TRUNK PROCESSING THE "CPF - MEET ME CALLER" FUNCTION ?
1108
NO
YES
STORE "LAST MEET ME ABANDON" TIME
1109

Fig. 11.

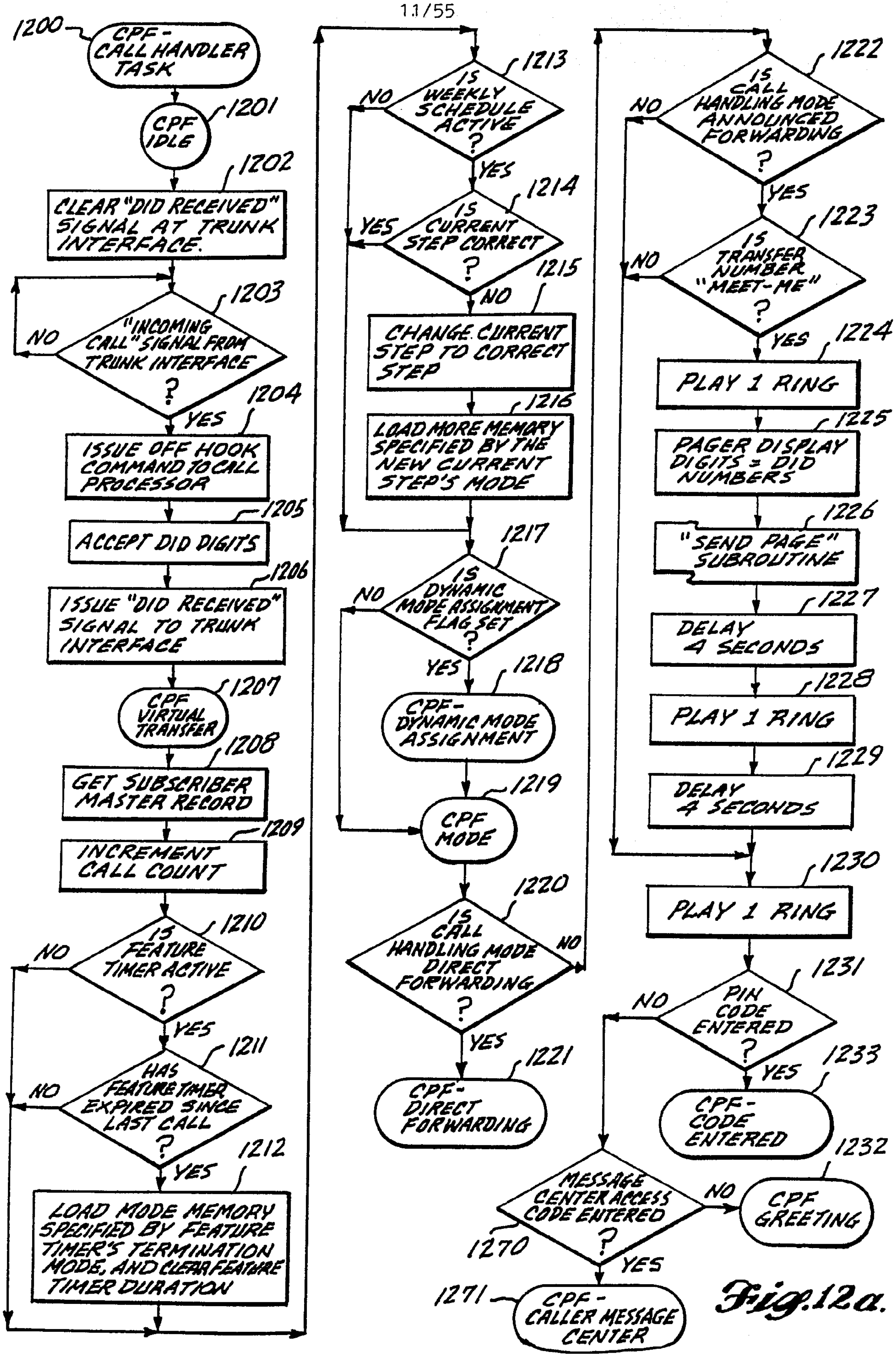

1200
CPF - CALL HANDLER TASK
1201
CPF IDLE
1202
CLEAR "DID RECEIVED" SIGNAL AT TRUNK INTERFACE.
1203
"INCOMING CALL" SIGNAL FROM TRUNK INTERFACE ?
NO
YES
1204
ISSUE OFF HOOK COMMAND TO CALL PROCESSOR
1205
ACCEPT DID DIGITS
1206
ISSUE "DID RECEIVED" SIGNAL TO TRUNK INTERFACE
1207
CPF VIRTUAL TRANSFER
1208
GET SUBSCRIBER MASTER RECORD
1209
INCREMENT CALL COUNT
1210
IS FEATURE TIMER ACTIVE ?
NO
YES
1211
HAS FEATURE TIMER EXPIRED SINCE LAST CALL ?
NO
YES
1212
LOAD MODE MEMORY SPECIFIED BY FEATURE TIMER'S TERMINATION MODE, AND CLEAR FEATURE TIMER DURATION
1213
IS WEEKLY SCHEDULE ACTIVE ?
NO
YES
1214
IS CURRENT STEP CORRECT ?
YES
NO
1215
CHANGE CURRENT STEP TO CORRECT STEP
1216
LOAD MORE MEMORY SPECIFIED BY THE NEW CURRENT STEP'S MODE
1217
IS DYNAMIC MODE ASSIGNMENT FLAG SET ?
NO
YES
1218
CPF-DYNAMIC MODE ASSIGNMENT
1219
CPF MODE
1220
IS CALL HANDLING MODE DIRECT FORWARDING ?
NO
YES
1221
CPF-DIRECT FORWARDING
1222
IS CALL HANDLING MODE ANNOUNCED FORWARDING ?
NO
YES
1223
IS TRANSFER NUMBER "MEET-ME" ?
NO
YES
1224
PLAY 1 RING
1225
PAGER DISPLAY DIGITS = DID NUMBERS
1226
"SEND PAGE" SUBROUTINE
1227
DELAY 4 SECONDS
1228
PLAY 1 RING
1229
DELAY 4 SECONDS
1230
PLAY 1 RING
1231
PIN CODE ENTERED ?
NO
YES
1233
CPF-CODE ENTERED
1232
CPF GREETING
1270
MESSAGE CENTER ACCESS CODE ENTERED ?
NO
YES
1271
CPF-CALLER MESSAGE CENTER

Fig.12a.

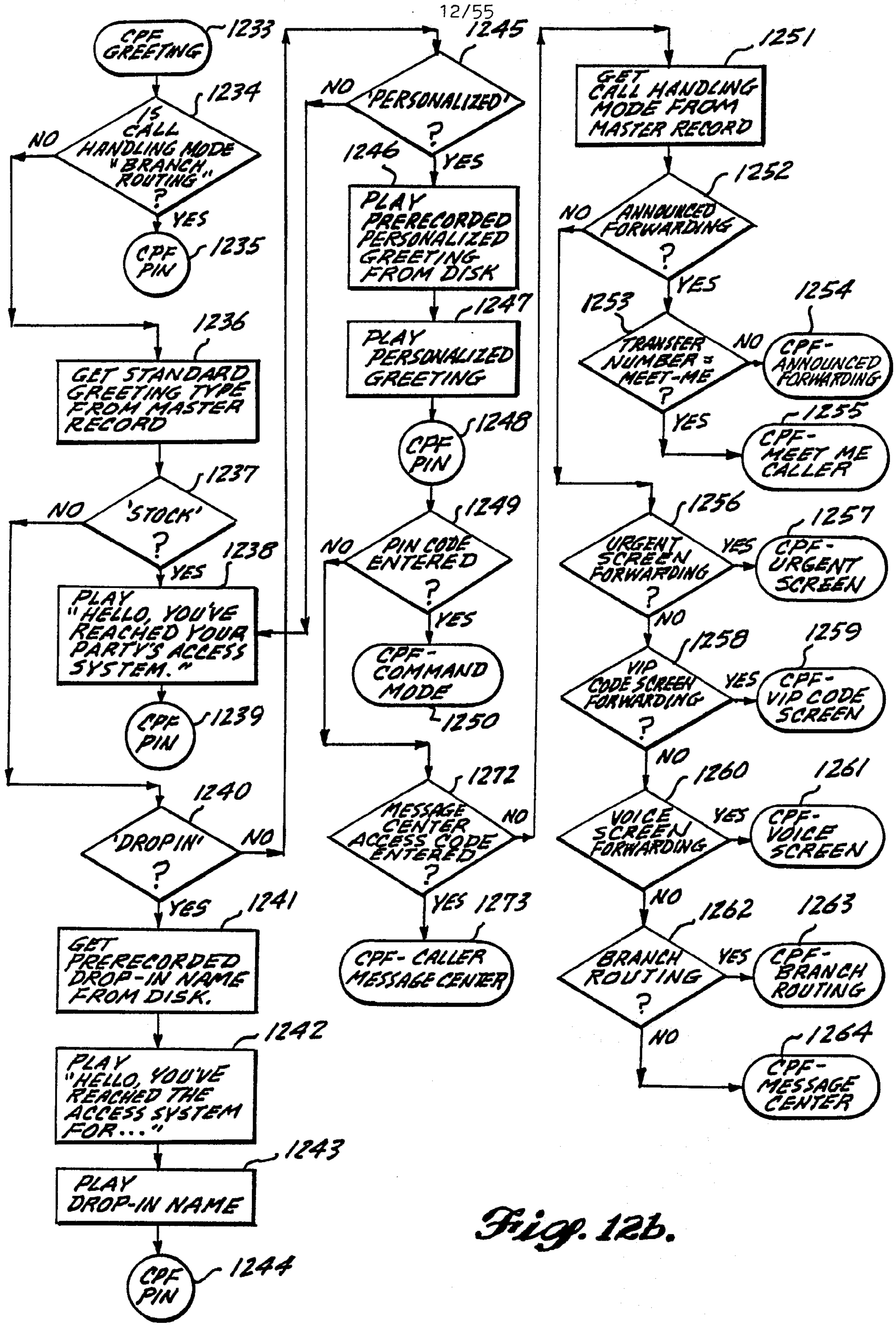

CPF GREETING
1233
IS CALL HANDLING MODE "BRANCH ROUTING" ?
1234
NO
YES
CPF PIN
1235
1236
GET STANDARD GREETING TYPE FROM MASTER RECORD
'STOCK' ?
1237
NO
YES
1238
PLAY "HELLO, YOU'VE REACHED YOUR PARTY'S ACCESS SYSTEM."
CPF PIN
1239
'DROPIN' ?
1240
NO
YES
1241
GET PRERECORDED DROP-IN NAME FROM DISK.
1242
PLAY "HELLO, YOU'VE REACHED THE ACCESS SYSTEM FOR..."
1243
PLAY DROP-IN NAME
CPF PIN
1244
1245
'PERSONALIZED' ?
NO
1246
YES
PLAY PRERECORDED PERSONALIZED GREETING FROM DISK
1247
PLAY PERSONALIZED GREETING
CPF PIN
1248
1249
PIN CODE ENTERED ?
NO
YES
CPF-COMMAND MODE
1250
1272
MESSAGE CENTER ACCESS CODE ENTERED ?
NO
YES
1273
CPF-CALLER MESSAGE CENTER
1251
GET CALL HANDLING MODE FROM MASTER RECORD
1252
ANNOUNCED FORWARDING ?
NO
YES
1253
TRANSFER NUMBER = MEET-ME ?
NO
1254
CPF-ANNOUNCED FORWARDING
YES
1255
CPF-MEET ME CALLER
1256
URGENT SCREEN FORWARDING ?
YES
1257
CPF-URGENT SCREEN
NO
1258
VIP CODE SCREEN FORWARDING ?
YES
1259
CPF-VIP CODE SCREEN
NO
1260
VOICE SCREEN FORWARDING ?
YES
1261
CPF-VOICE SCREEN
NO
1262
BRANCH ROUTING ?
YES
1263
CPF-BRANCH ROUTING
NO
1264
CPF-MESSAGE CENTER

Fig. 12b.

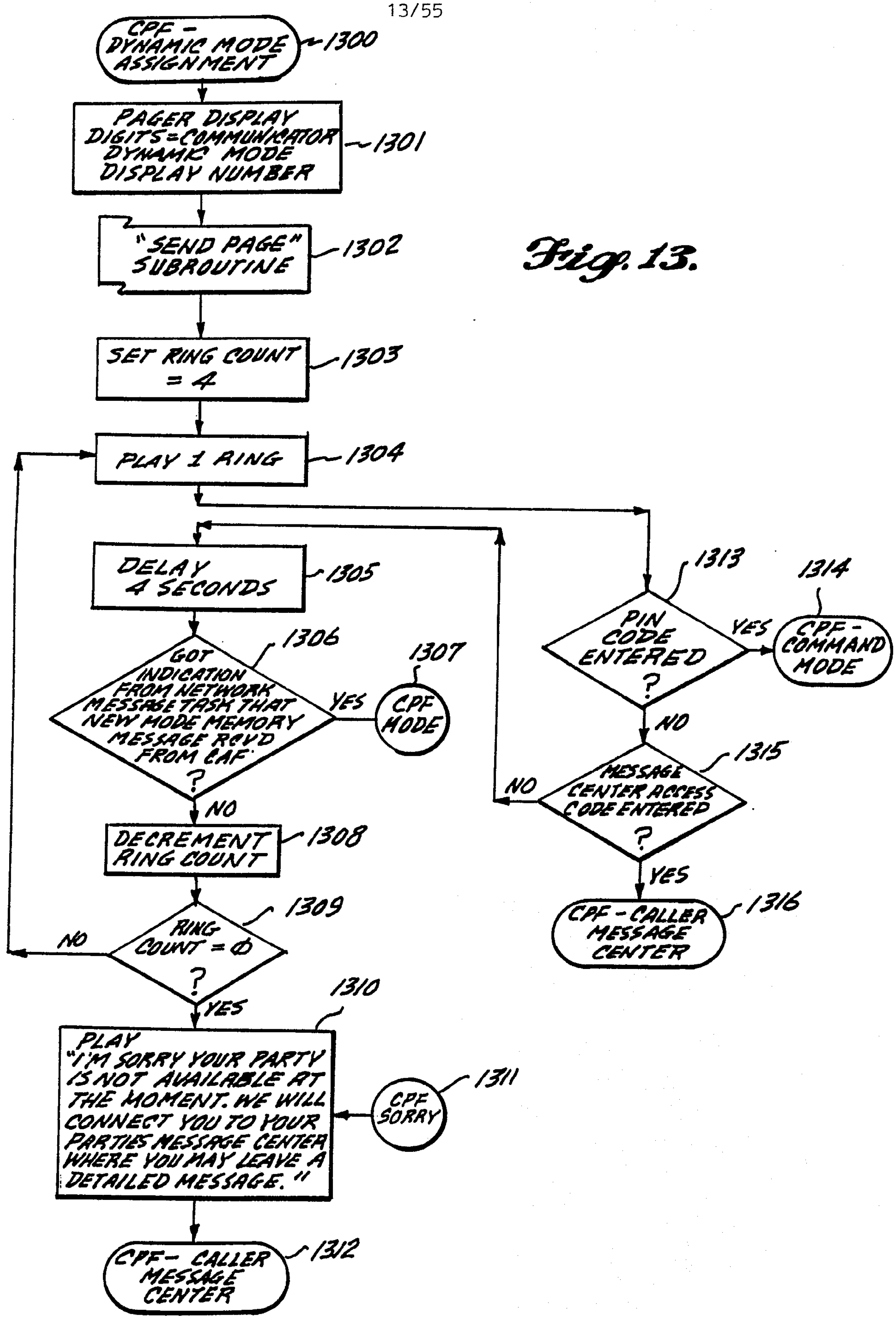

CPF - DYNAMIC MODE ASSIGNMENT
1300
PAGER DISPLAY DIGITS=COMMUNICATOR DYNAMIC MODE DISPLAY NUMBER
1301
"SEND PAGE" SUBROUTINE
1302
SET RING COUNT = 4
1303
PLAY 1 RING
1304
DELAY 4 SECONDS
1305
GOT INDICATION FROM NETWORK MESSAGE TASK THAT NEW MODE MEMORY MESSAGE RCVD FROM CAF ?
1306
YES
CPF MODE
1307
NO
DECREMENT RING COUNT
1308
RING COUNT = Φ ?
1309
NO
YES
PLAY "I'M SORRY YOUR PARTY IS NOT AVAILABLE AT THE MOMENT. WE WILL CONNECT YOU TO YOUR PARTIES MESSAGE CENTER WHERE YOU MAY LEAVE A DETAILED MESSAGE."
1310
CPF SORRY
1311
CPF- CALLER MESSAGE CENTER
1312
PIN CODE ENTERED ?
1313
YES
CPF- COMMAND MODE
1314
NO
MESSAGE CENTER ACCESS CODE ENTERED ?
1315
NO
YES
CPF - CALLER MESSAGE CENTER
1316

Fig. 13.

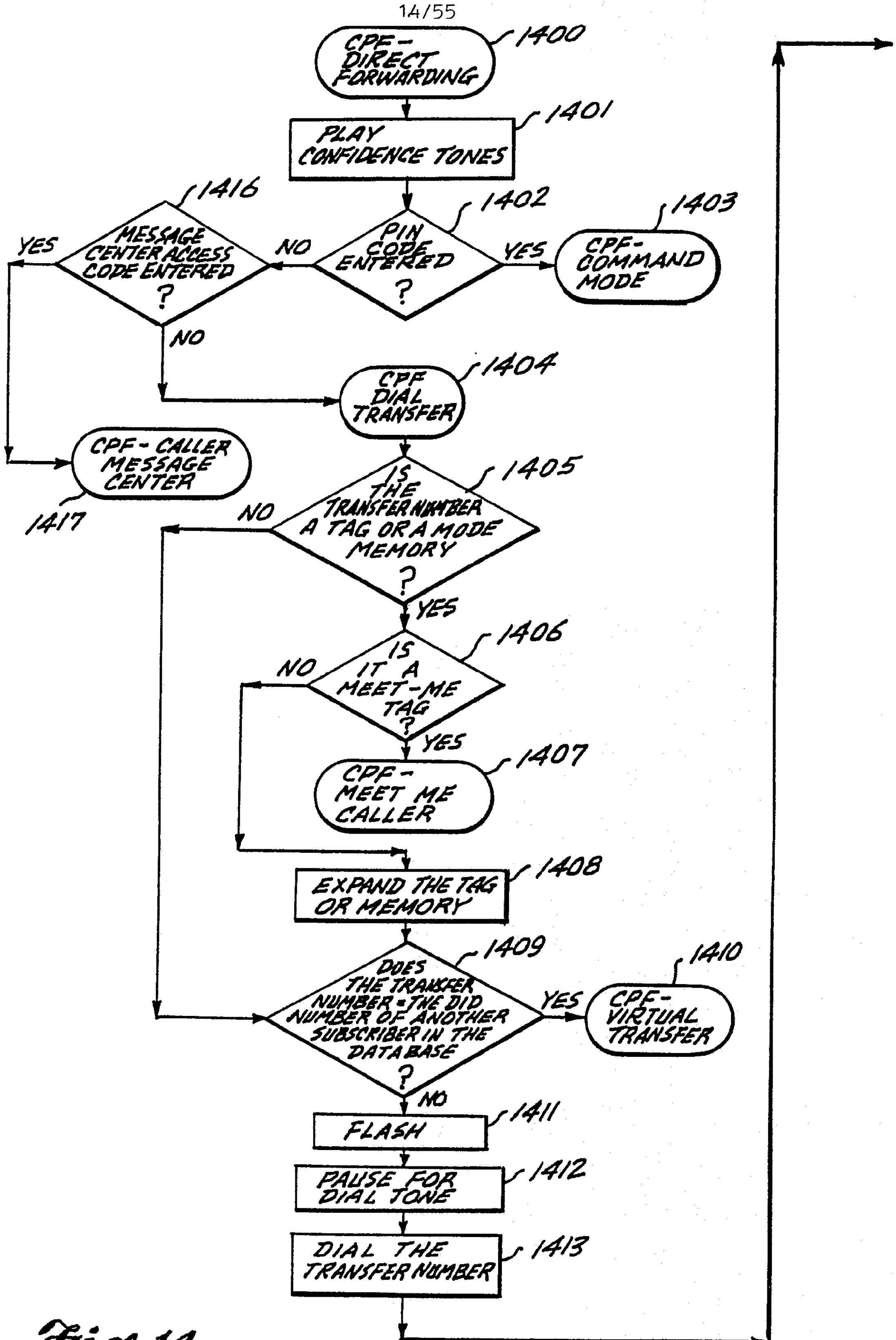
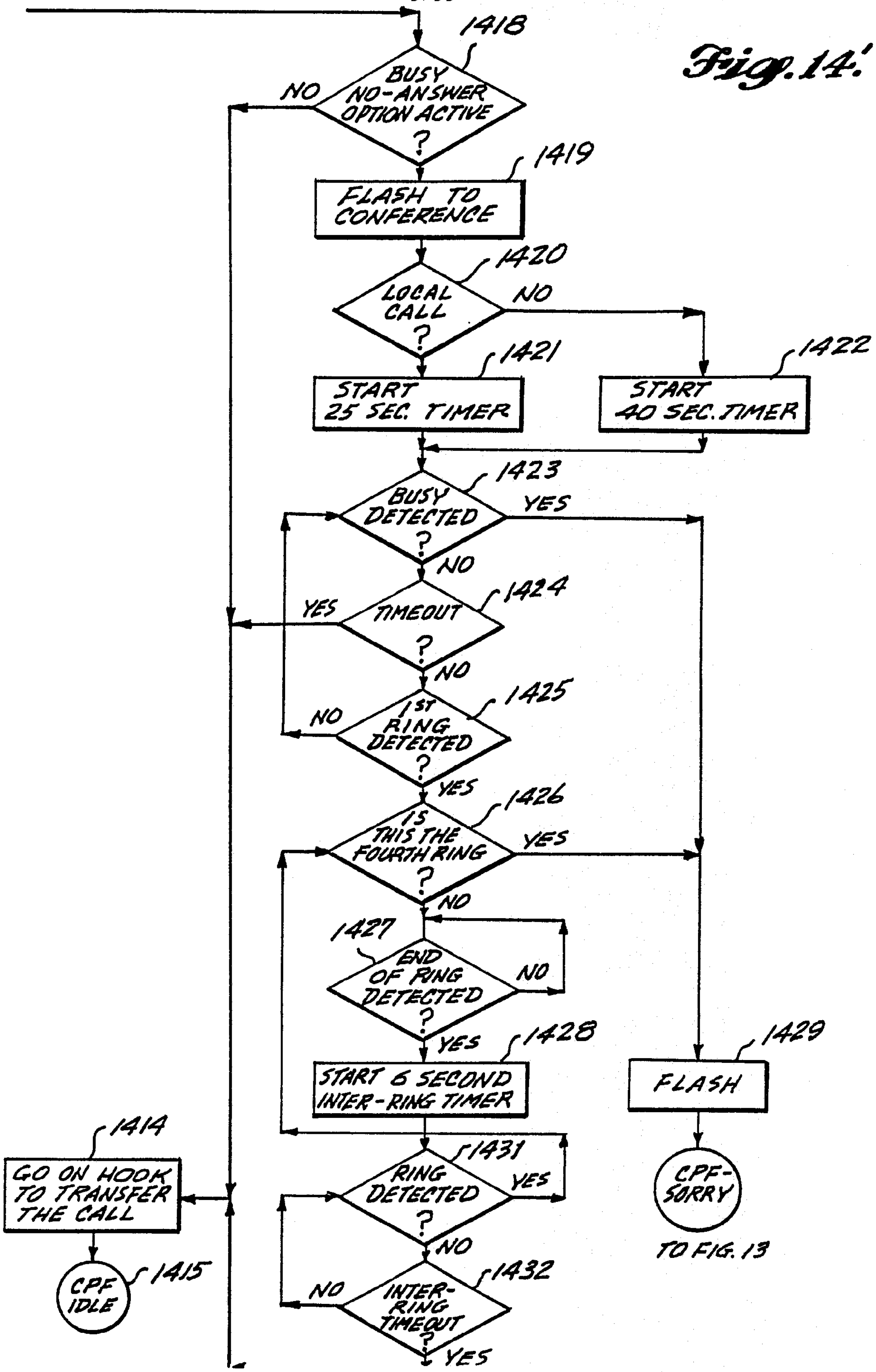

CPE-DIRECT FORWARDING
1400
PLAY CONFIDENCE TONES
1401
PIN CODE ENTERED ?
1402
YES
CPF-COMMAND MODE
1403
NO
MESSAGE CENTER ACCESS CODE ENTERED ?
1416
YES
NO
CPF DIAL TRANSFER
1404
CPF-CALLER MESSAGE CENTER
1417
IS THE TRANSFER NUMBER A TAG OR A MODE MEMORY ?
1405
NO
YES
IS IT A MEET-ME TAG ?
1406
NO
YES
CPF-MEET ME CALLER
1407
EXPAND THE TAG OR MEMORY
1408
DOES THE TRANSFER NUMBER = THE DID NUMBER OF ANOTHER SUBSCRIBER IN THE DATABASE ?
1409
YES
CPF-VIRTUAL TRANSFER
1410
NO
FLASH
1411
PAUSE FOR DIAL TONE
1412
DIAL THE TRANSFER NUMBER
1413

Fig.14.

Fig. 14.


1418
BUSY NO-ANSWER OPTION ACTIVE ?
NO
1419
FLASH TO CONFERENCE
1420
LOCAL CALL ?
NO
1421
START 25 SEC. TIMER
1422
START 40 SEC. TIMER
1423
BUSY DETECTED ?
YES
NO
1424
TIMEOUT ?
YES
NO
1425
1ST RING DETECTED ?
NO
YES
1426
IS THIS THE FOURTH RING ?
YES
NO
1427
END OF RING DETECTED ?
NO
YES
1428
START 6 SECOND INTER-RING TIMER
1429
FLASH
CPF-SORRY
TO FIG. 13
1414
GO ON HOOK TO TRANSFER THE CALL
1415
CPF IDLE
1431
RING DETECTED ?
YES
NO
1432
INTER-RING TIMEOUT ?
NO
YES

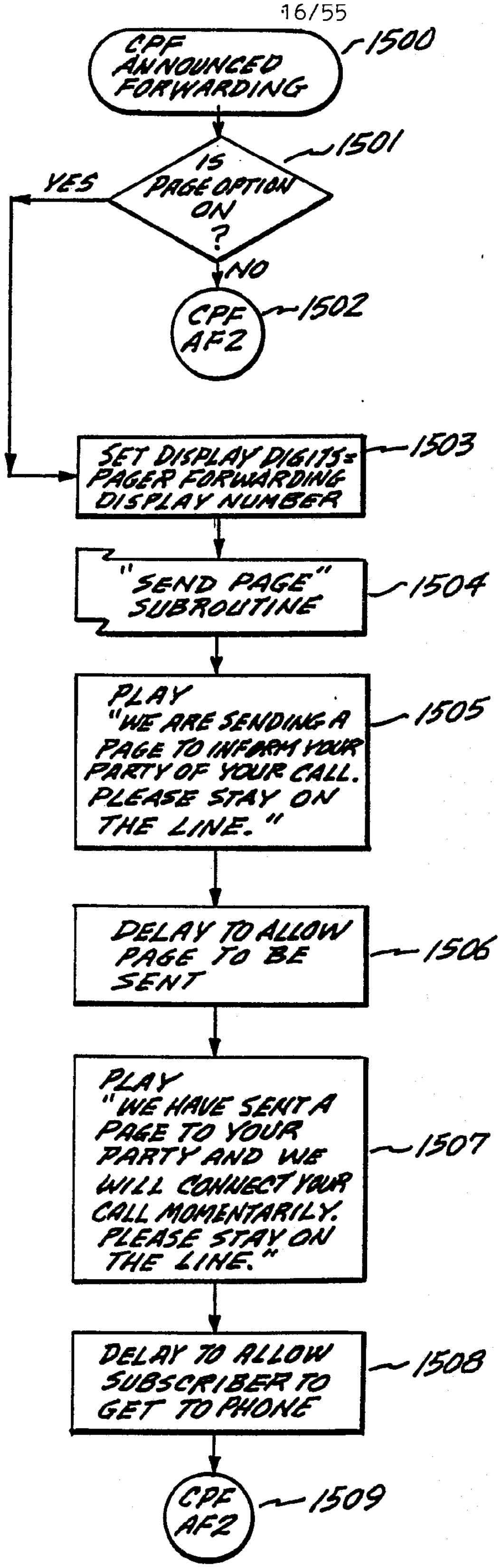

CPF ANNOUNCED FORWARDING
1500
IS PAGE OPTION ON ?
1501
YES
NO
CPF AF2
1502
SET DISPLAY DIGITS= PAGER FORWARDING DISPLAY NUMBER
1503
"SEND PAGE" SUBROUTINE
1504
PLAY "WE ARE SENDING A PAGE TO INFORM YOUR PARTY OF YOUR CALL. PLEASE STAY ON THE LINE."
1505
DELAY TO ALLOW PAGE TO BE SENT
1506
PLAY "WE HAVE SENT A PAGE TO YOUR PARTY AND WE WILL CONNECT YOUR CALL MOMENTARILY. PLEASE STAY ON THE LINE."
1507
DELAY TO ALLOW SUBSCRIBER TO GET TO PHONE
1508
CPF AF2
1509

Fig. 15a.

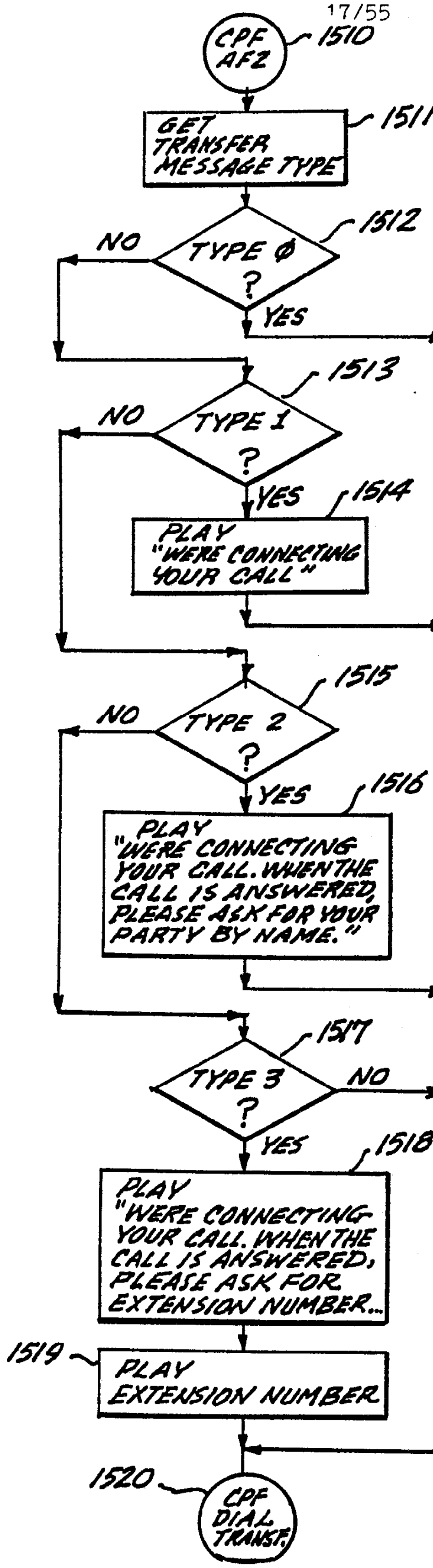

CPF AFZ
1510
GET TRANSFER MESSAGE TYPE
1511
TYPE Ø ?
1512
NO
YES
TYPE 1 ?
1513
NO
YES
PLAY "WERE CONNECTING YOUR CALL"
1514
TYPE 2 ?
1515
NO
YES
"PLAY "WERE CONNECTING YOUR CALL. WHEN THE CALL IS ANSWERED, PLEASE ASK FOR YOUR PARTY BY NAME."
1516
TYPE 3 ?
1517
NO
YES
PLAY "WERE CONNECTING YOUR CALL. WHEN THE CALL IS ANSWERED, PLEASE ASK FOR EXTENSION NUMBER...
1518
1519
PLAY EXTENSION NUMBER
1520
CPF DIAL TRANSF.

Fig. 15b.

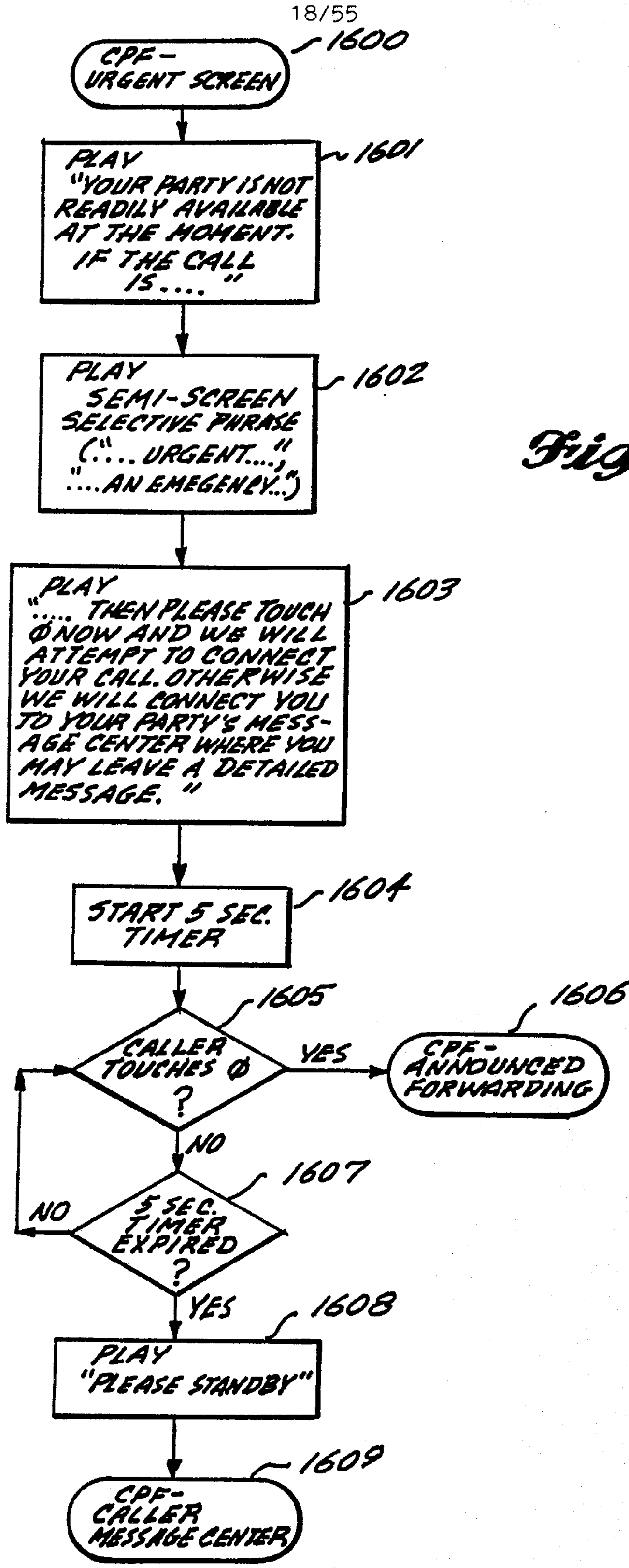

CPF-
URGENT SCREEN
1600
PLAY
"YOUR PARTY IS NOT READILY AVAILABLE AT THE MOMENT. IF THE CALL IS...."
1601
PLAY
SEMI-SCREEN SELECTIVE PHRASE
("...URGENT...", "...AN EMEGENCY...")
1602
PLAY
"..... THEN PLEASE TOUCH Ø NOW AND WE WILL ATTEMPT TO CONNECT YOUR CALL. OTHERWISE WE WILL CONNECT YOU TO YOUR PARTY'S MESSAGE CENTER WHERE YOU MAY LEAVE A DETAILED MESSAGE."
1603
START 5 SEC. TIMER
1604
CALLER TOUCHES Ø ?
1605
YES
CPF-
ANNOUNCED FORWARDING
1606
NO
5 SEC. TIMER EXPIRED ?
1607
NO
YES
PLAY
"PLEASE STANDBY"
1608
CPF-
CALLER MESSAGE CENTER
1609

Fig. 16.

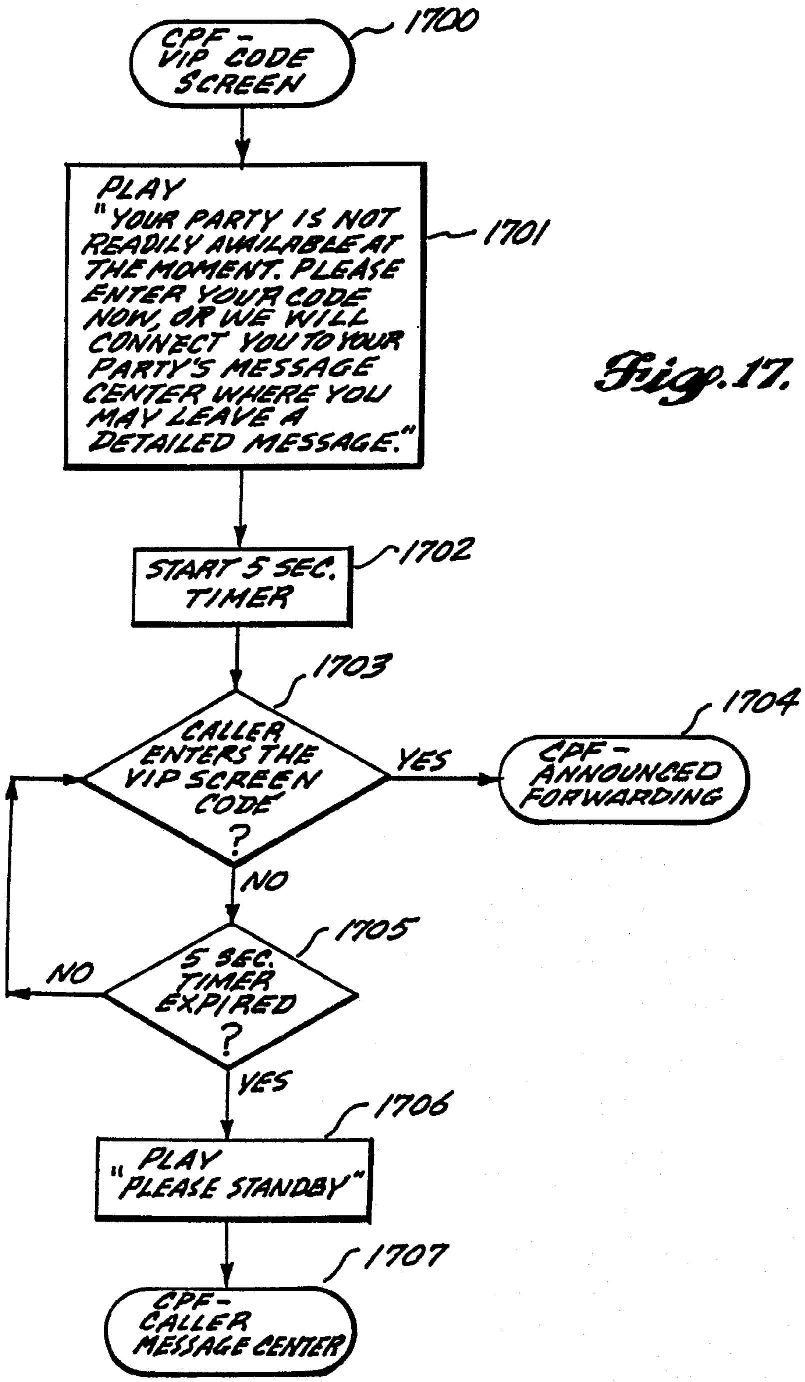

CPF - VIP CODE SCREEN
1700
PLAY "YOUR PARTY IS NOT READILY AVAILABLE AT THE MOMENT. PLEASE ENTER YOUR CODE NOW, OR WE WILL CONNECT YOU TO YOUR PARTY'S MESSAGE CENTER WHERE YOU MAY LEAVE A DETAILED MESSAGE."
1701
START 5 SEC. TIMER
1702
CALLER ENTERS THE VIP SCREEN CODE ?
1703
YES
CPF - ANNOUNCED FORWARDING
1704
NO
5 SEC. TIMER EXPIRED ?
1705
NO
YES
PLAY "PLEASE STANDBY"
1706
CPF - CALLER MESSAGE CENTER
1707

Fig. 17.

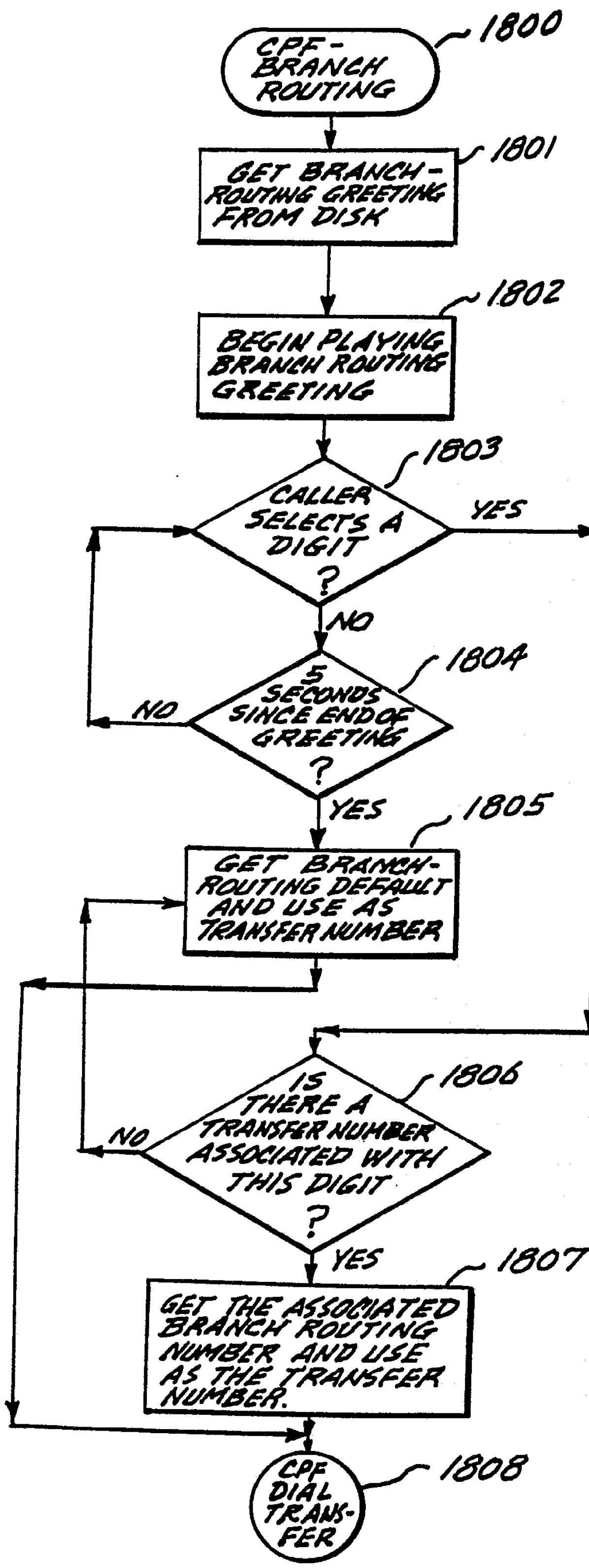


Fig. 18.

SUBSTITUTE SHEET

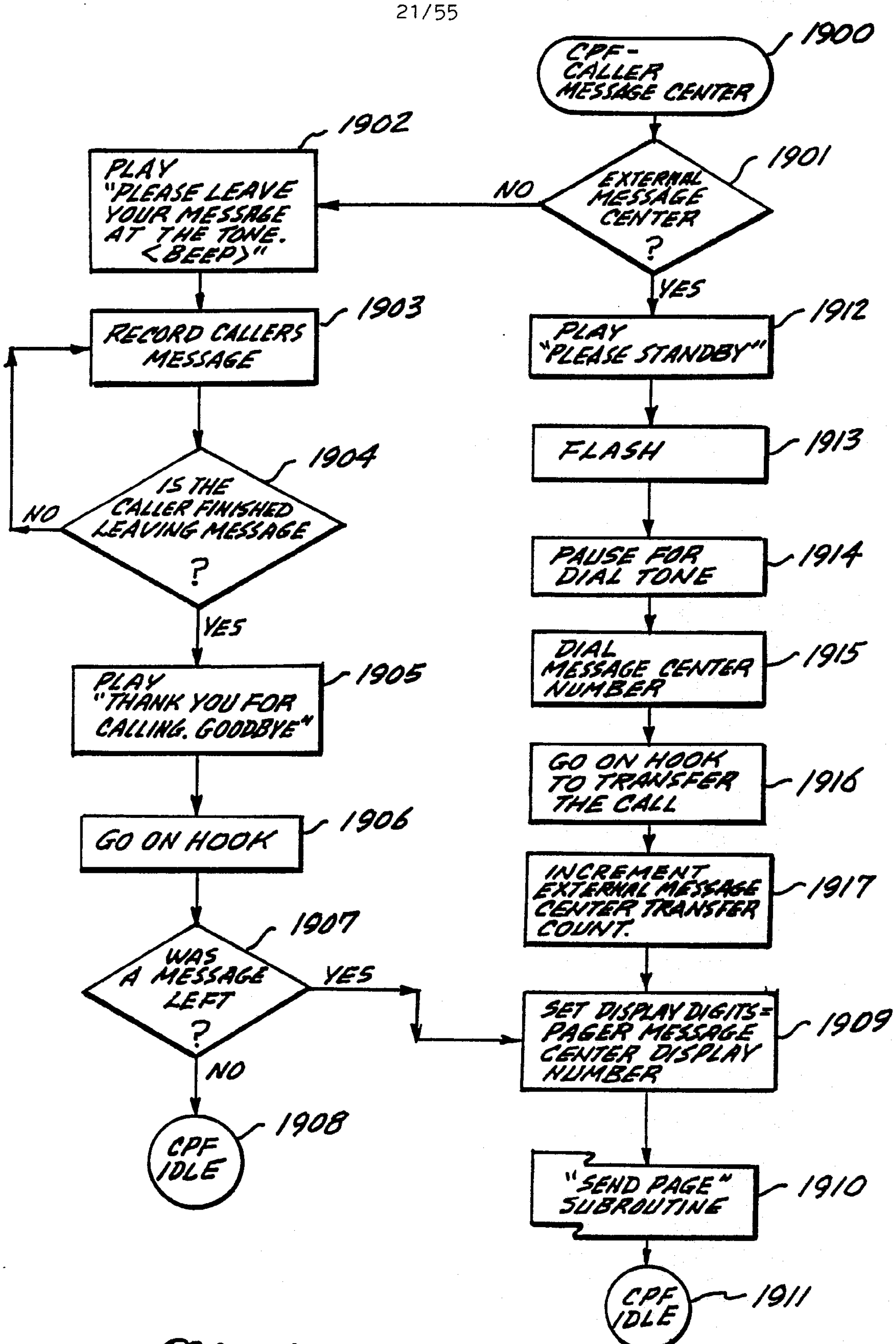

CPF - CALLER MESSAGE CENTER
1900
EXTERNAL MESSAGE CENTER ?
1901
NO
YES
PLAY "PLEASE LEAVE YOUR MESSAGE AT THE TONE. <BEEP>"
1902
RECORD CALLERS MESSAGE
1903
IS THE CALLER FINISHED LEAVING MESSAGE ?
1904
NO
YES
PLAY "THANK YOU FOR CALLING. GOODBYE"
1905
GO ON HOOK
1906
WAS A MESSAGE LEFT ?
1907
YES
NO
CPF IDLE
1908
PLAY "PLEASE STANDBY"
1912
FLASH
1913
PAUSE FOR DIAL TONE
1914
DIAL MESSAGE CENTER NUMBER
1915
GO ON HOOK TO TRANSFER THE CALL
1916
INCREMENT EXTERNAL MESSAGE CENTER TRANSFER COUNT.
1917
SET DISPLAY DIGITS = PAGER MESSAGE CENTER DISPLAY NUMBER
1909
"SEND PAGE" SUBROUTINE
1910
CPF IDLE
1911

Fig. 19.

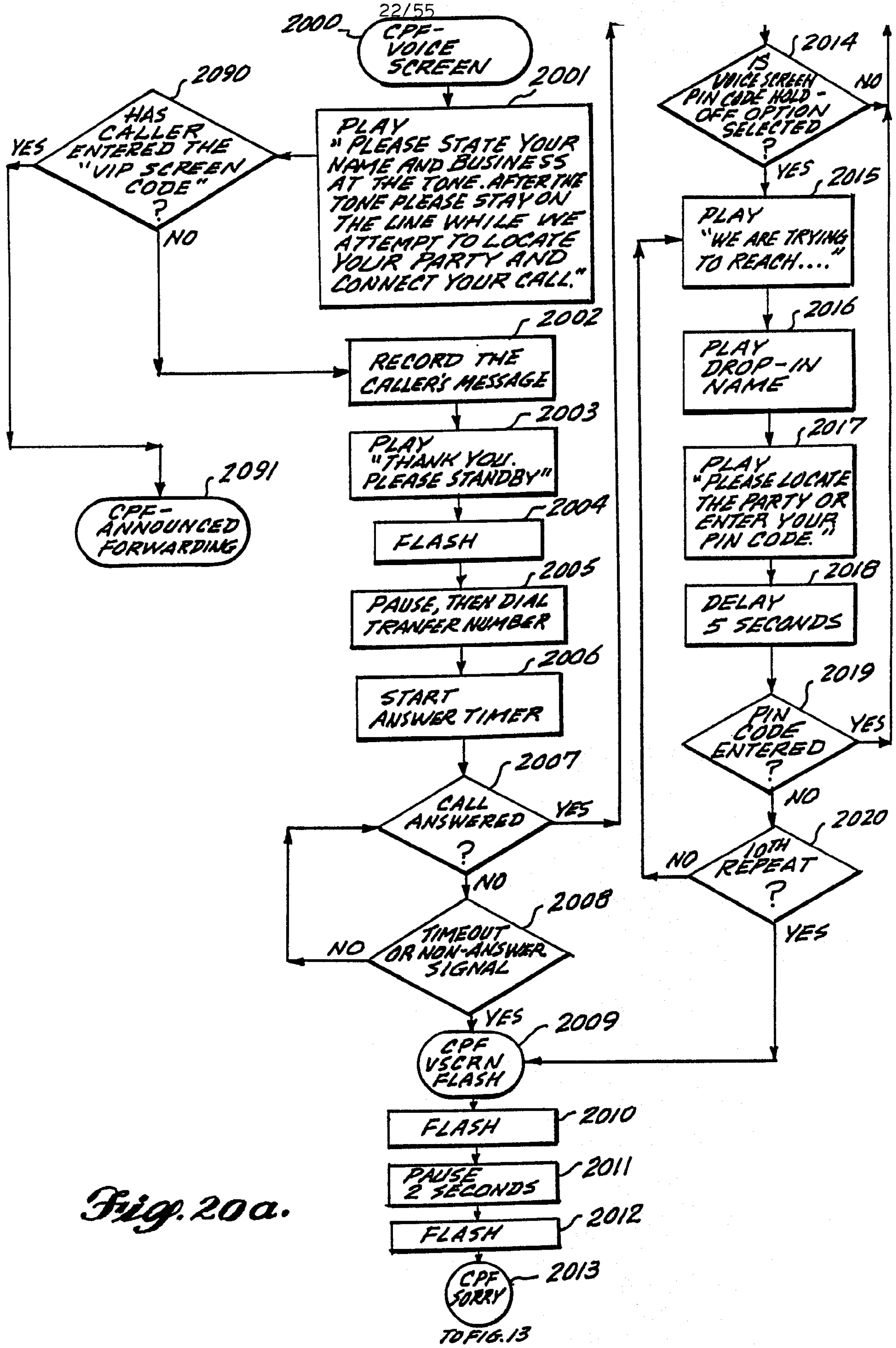


Fig. 20a.

SUBSTITUTE SHEET

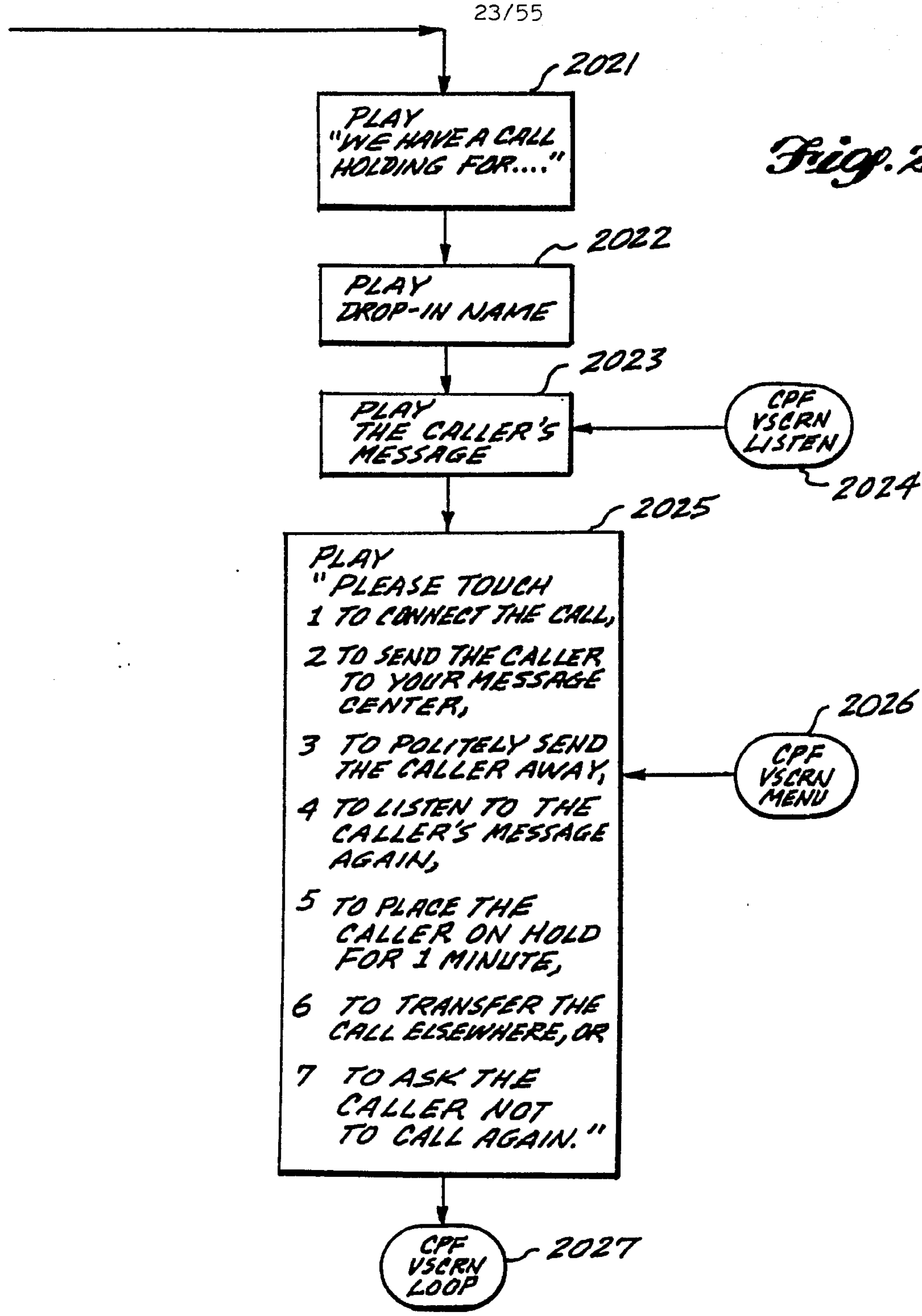

2021
PLAY "WE HAVE A CALL HOLDING FOR...."
2022
PLAY DROP-IN NAME
2023
PLAY THE CALLER'S MESSAGE
CPF VSCRN LISTEN
2024
2025
PLAY "PLEASE TOUCH
1 TO CONNECT THE CALL,
2 TO SEND THE CALLER TO YOUR MESSAGE CENTER,
3 TO POLITELY SEND THE CALLER AWAY,
4 TO LISTEN TO THE CALLER'S MESSAGE AGAIN,
5 TO PLACE THE CALLER ON HOLD FOR 1 MINUTE,
6 TO TRANSFER THE CALL ELSEWHERE, OR
7 TO ASK THE CALLER NOT TO CALL AGAIN."
2026
CPF VSCRN MENU
CPF VSCRN LOOP
2027

Fig. 20a.

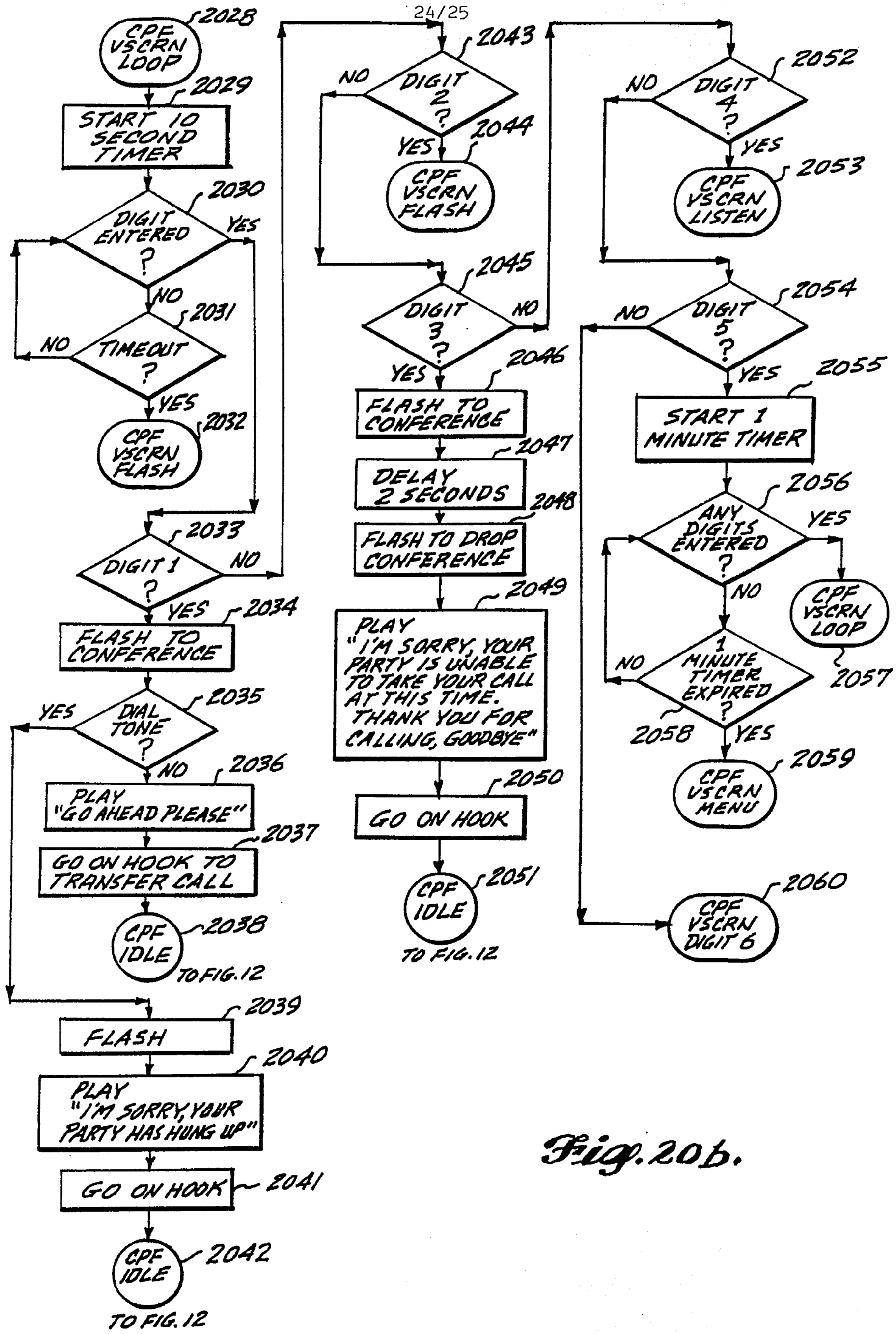


Fig. 20b.

SUBSTITUTE SHEET

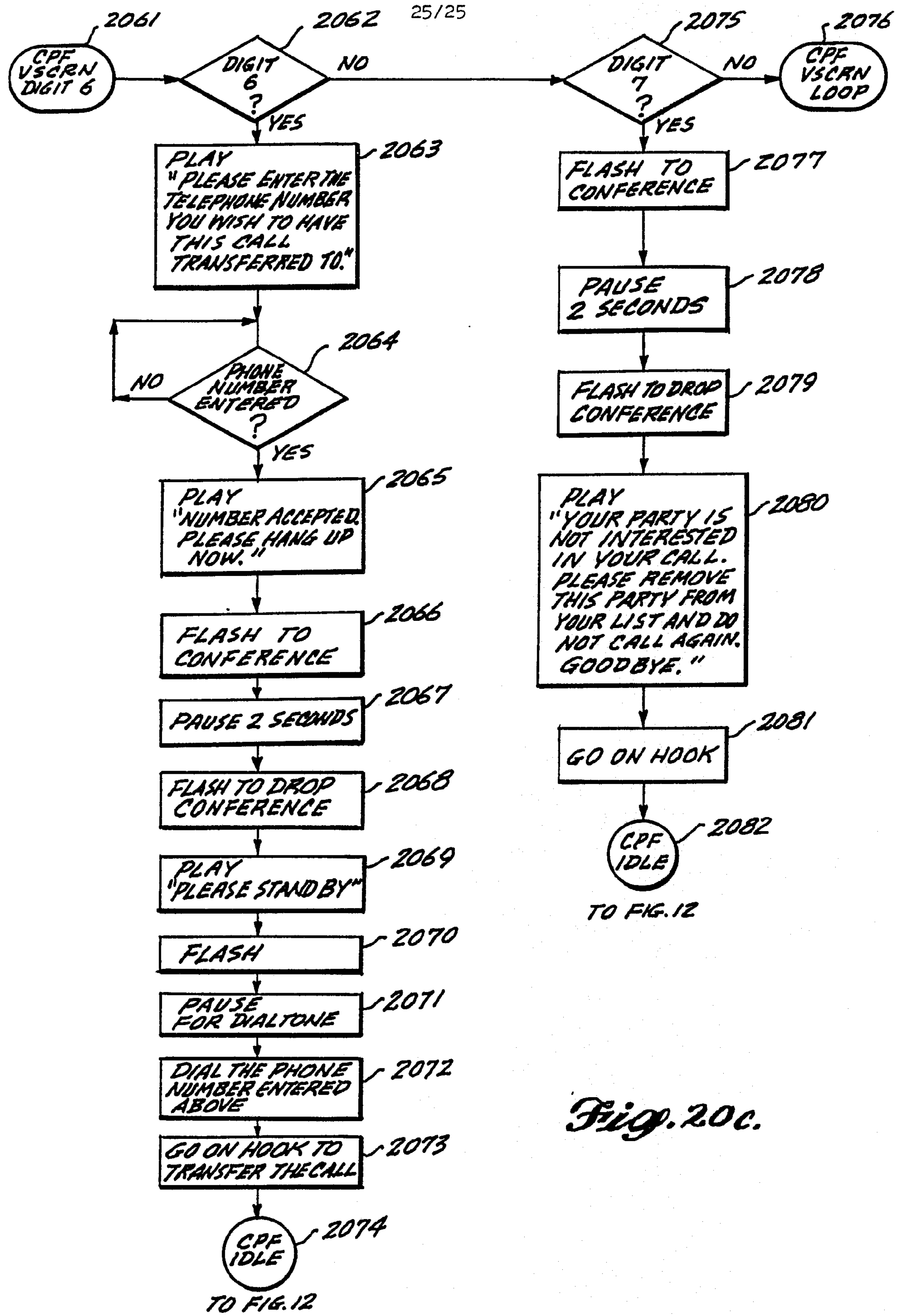
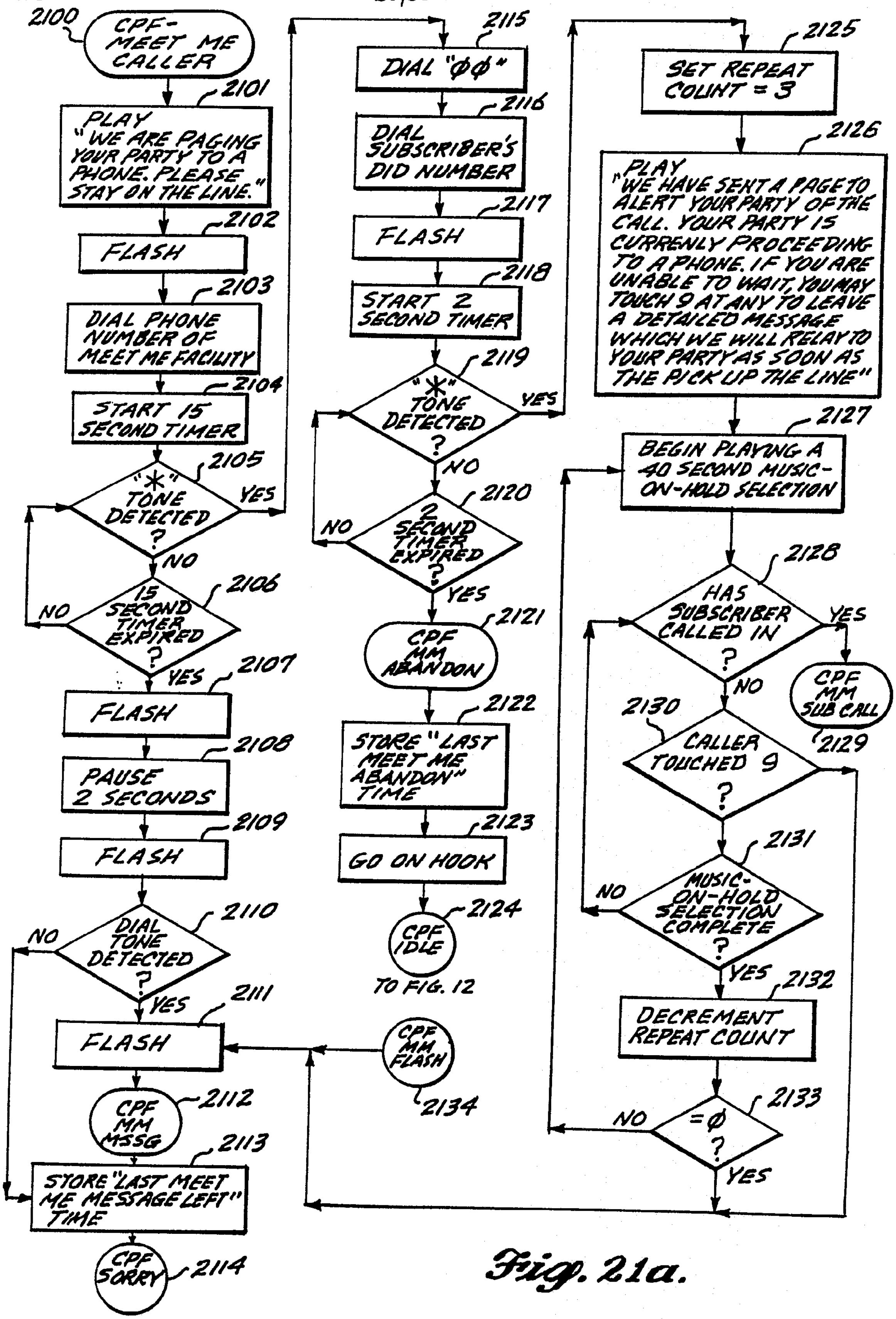

2061
CPF VSCRN DIGIT 6
2062
DIGIT 6 ?
NO
YES
2063
PLAY "PLEASE ENTER THE TELEPHONE NUMBER YOU WISH TO HAVE THIS CALL TRANSFERRED TO."
2064
PHONE NUMBER ENTERED ?
NO
YES
2065
PLAY "NUMBER ACCEPTED. PLEASE HANG UP NOW."
2066
FLASH TO CONFERENCE
2067
PAUSE 2 SECONDS
2068
FLASH TO DROP CONFERENCE
2069
PLAY "PLEASE STAND BY"
2070
FLASH
2071
PAUSE FOR DIALTONE
2072
DIAL THE PHONE NUMBER ENTERED ABOVE
2073
GO ON HOOK TO TRANSFER THE CALL
2074
CPF IDLE
TO FIG. 12
2075
DIGIT 7 ?
NO
YES
2076
CPF VSCRN LOOP
2077
FLASH TO CONFERENCE
2078
PAUSE 2 SECONDS
2079
FLASH TO DROP CONFERENCE
2080
PLAY "YOUR PARTY IS NOT INTERESTED IN YOUR CALL. PLEASE REMOVE THIS PARTY FROM YOUR LIST AND DO NOT CALL AGAIN. GOODBYE."
2081
GO ON HOOK
2082
CPF IDLE
TO FIG. 12

Fig. 20c.



2100 CPF-MEET ME CALLER
2101 PLAY "WE ARE PAGING YOUR PARTY TO A PHONE. PLEASE STAY ON THE LINE."
2102 FLASH
2103 DIAL PHONE NUMBER OF MEET ME FACILITY
2104 START 15 SECOND TIMER
2105 "*" TONE DETECTED ?
YES
NO
2106 15 SECOND TIMER EXPIRED ?
NO
YES
2107 FLASH
2108 PAUSE 2 SECONDS
2109 FLASH
2110 DIAL TONE DETECTED ?
NO
YES
2111 FLASH
2112 CPF MM MSSG
2113 STORE "LAST MEET ME MESSAGE LEFT" TIME
2114 CPF SORRY
2115 DIAL "ØØ"
2116 DIAL SUBSCRIBER'S DID NUMBER
2117 FLASH
2118 START 2 SECOND TIMER
2119 "*" TONE DETECTED ?
YES
NO
2120 2 SECOND TIMER EXPIRED ?
NO
YES
2121 CPF MM ABANDON
2122 STORE "LAST MEET ME ABANDON" TIME
2123 GO ON HOOK
2124 CPF IDLE
TO FIG. 12
2134 CPF MM FLASH
2125 SET REPEAT COUNT = 3
2126 PLAY "WE HAVE SENT A PAGE TO ALERT YOUR PARTY OF THE CALL. YOUR PARTY IS CURRENLY PROCEEDING TO A PHONE. IF YOU ARE UNABLE TO WAIT, YOU MAY TOUCH 9 AT ANY TO LEAVE A DETAILED MESSAGE WHICH WE WILL RELAY TO YOUR PARTY AS SOON AS THE PICK UP THE LINE"
2127 BEGIN PLAYING A 40 SECOND MUSIC-ON-HOLD SELECTION
2128 HAS SUBSCRIBER CALLED IN ?
YES
NO
2129 CPF MM SUB CALL
2130 CALLER TOUCHED 9 ?
2131 MUSIC-ON-HOLD SELECTION COMPLETE ?
NO
YES
2132 DECREMENT REPEAT COUNT
2133 =Ø ?
NO
YES

Fig. 21a.

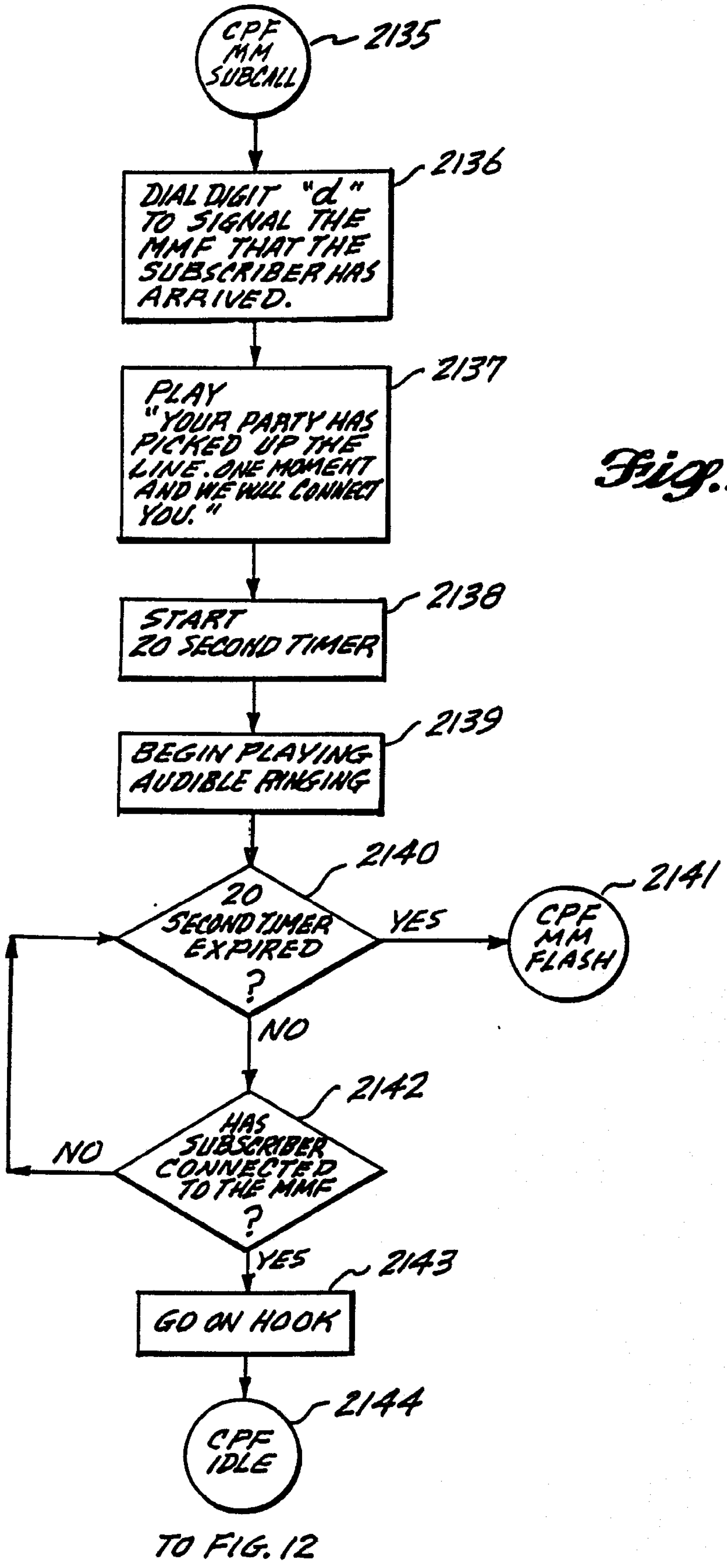

CPF MM SUBCALL
2135
DIAL DIGIT "d" TO SIGNAL THE MMF THAT THE SUBSCRIBER HAS ARRIVED.
2136
PLAY "YOUR PARTY HAS PICKED UP THE LINE. ONE MOMENT AND WE WILL CONNECT YOU."
2137
START 20 SECOND TIMER
2138
BEGIN PLAYING AUDIBLE RINGING
2139
20 SECOND TIMER EXPIRED ?
2140
YES
CPF MM FLASH
2141
NO
HAS SUBSCRIBER CONNECTED TO THE MMF ?
2142
NO
YES
GO ON HOOK
2143
CPF IDLE
2144
TO FIG. 12

Fig. 21b.

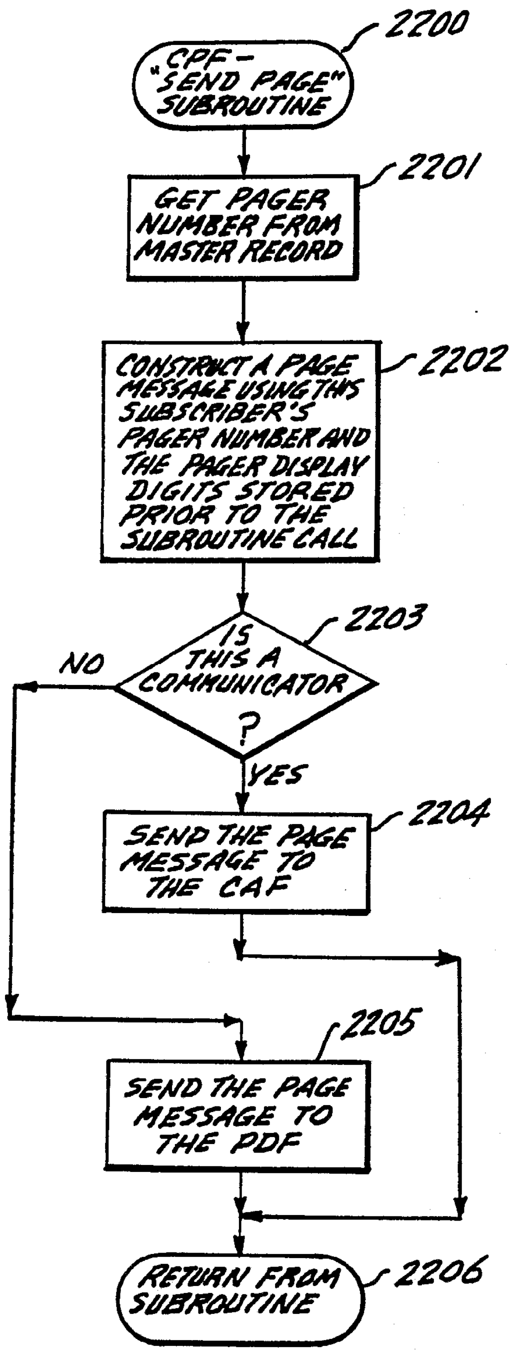

2200
CPF - "SEND PAGE" SUBROUTINE
2201
GET PAGER NUMBER FROM MASTER RECORD
2202
CONSTRUCT A PAGE MESSAGE USING THIS SUBSCRIBER'S PAGER NUMBER AND THE PAGER DISPLAY DIGITS STORED PRIOR TO THE SUBROUTINE CALL
2203
IS THIS A COMMUNICATOR ?
NO
YES
2204
SEND THE PAGE MESSAGE TO THE CAF
2205
SEND THE PAGE MESSAGE TO THE PDF
2206
RETURN FROM SUBROUTINE

Fig. 22.

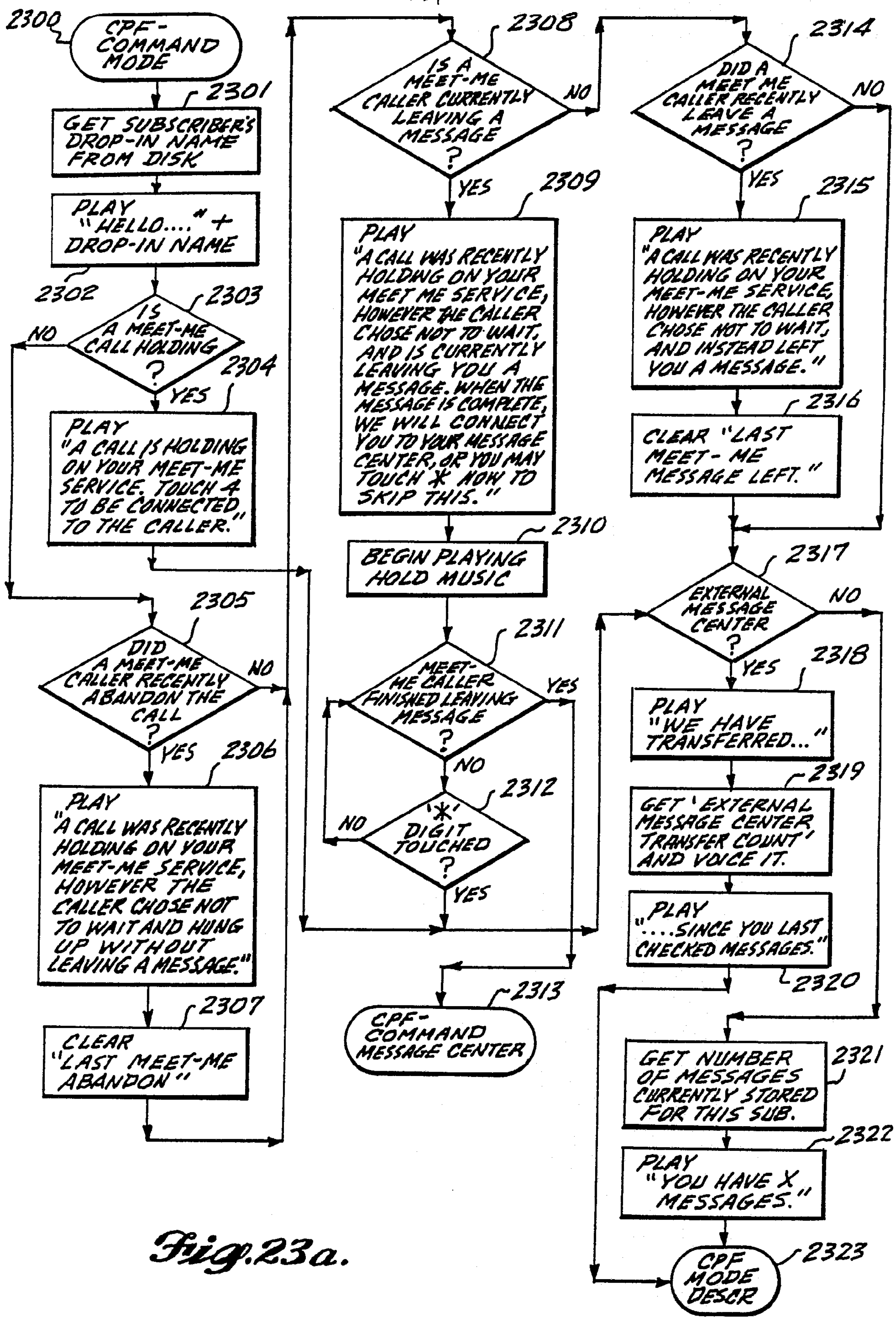

2300
CPF-COMMAND MODE
2301
GET SUBSCRIBER'S DROP-IN NAME FROM DISK
PLAY "HELLO...." + DROP-IN NAME
2302
2303
IS A MEET-ME CALL HOLDING ?
NO
YES
2304
PLAY "A CALL IS HOLDING ON YOUR MEET-ME SERVICE. TOUCH 4 TO BE CONNECTED TO THE CALLER."
2305
DID A MEET-ME CALLER RECENTLY ABANDON THE CALL ?
NO
YES
2306
PLAY "A CALL WAS RECENTLY HOLDING ON YOUR MEET-ME SERVICE, HOWEVER THE CALLER CHOSE NOT TO WAIT AND HUNG UP WITHOUT LEAVING A MESSAGE."
2307
CLEAR "LAST MEET-ME ABANDON"
2308
IS A MEET-ME CALLER CURRENTLY LEAVING A MESSAGE ?
NO
YES
2309
PLAY "A CALL WAS RECENTLY HOLDING ON YOUR MEET ME SERVICE, HOWEVER THE CALLER CHOSE NOT TO WAIT, AND IS CURRENTLY LEAVING YOU A MESSAGE. WHEN THE MESSAGE IS COMPLETE, WE WILL CONNECT YOU TO YOUR MESSAGE CENTER, OR YOU MAY TOUCH * NOW TO SKIP THIS."
2310
BEGIN PLAYING HOLD MUSIC
2311
MEET-ME CALLER FINISHED LEAVING MESSAGE ?
YES
NO
2312
'*' DIGIT TOUCHED ?
NO
YES
2313
CPF-COMMAND MESSAGE CENTER
2314
DID A MEET ME CALLER RECENTLY LEAVE A MESSAGE ?
NO
YES
2315
PLAY "A CALL WAS RECENTLY HOLDING ON YOUR MEET-ME SERVICE, HOWEVER THE CALLER CHOSE NOT TO WAIT, AND INSTEAD LEFT YOU A MESSAGE."
2316
CLEAR "LAST MEET-ME MESSAGE LEFT."
2317
EXTERNAL MESSAGE CENTER ?
NO
YES
2318
PLAY "WE HAVE TRANSFERRED..."
2319
GET 'EXTERNAL MESSAGE CENTER TRANSFER COUNT' AND VOICE IT.
PLAY "....SINCE YOU LAST CHECKED MESSAGES."
2320
GET NUMBER OF MESSAGES CURRENTLY STORED FOR THIS SUB.
2321
PLAY "YOU HAVE X MESSAGES."
2322
CPF MODE DESCR
2323

Fig.23a.

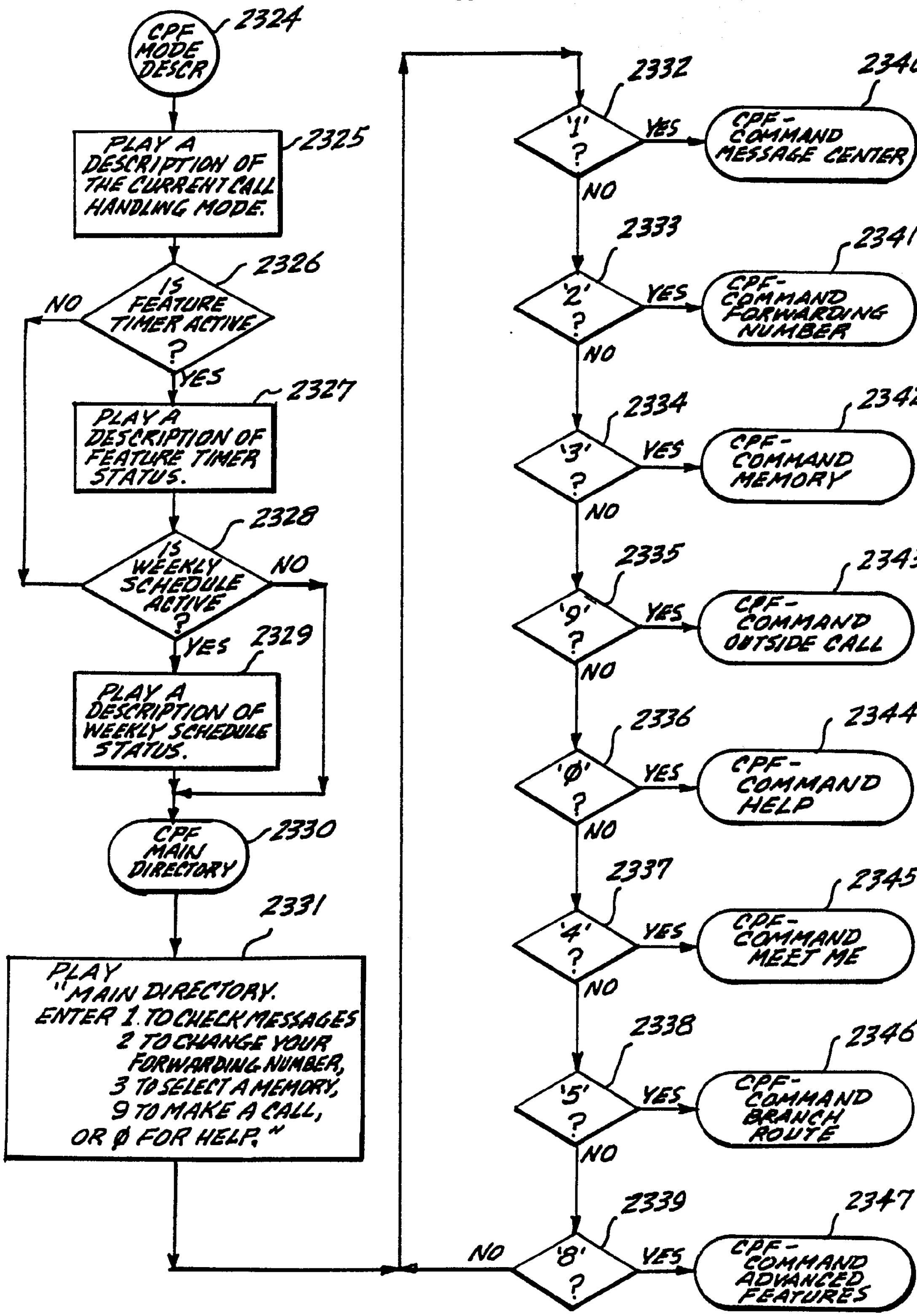

CPF MODE DESCR
2324
PLAY A DESCRIPTION OF THE CURRENT CALL HANDLING MODE.
2325
IS FEATURE TIMER ACTIVE ?
2326
NO
YES
PLAY A DESCRIPTION OF FEATURE TIMER STATUS.
2327
IS WEEKLY SCHEDULE ACTIVE ?
2328
NO
YES
2329
PLAY A DESCRIPTION OF WEEKLY SCHEDULE STATUS.
CPF MAIN DIRECTORY
2330
2331
PLAY "MAIN DIRECTORY. ENTER 1 TO CHECK MESSAGES 2 TO CHANGE YOUR FORWARDING NUMBER, 3 TO SELECT A MEMORY, 9 TO MAKE A CALL, OR Ø FOR HELP."
2332
'1' ?
YES
NO
2340
CPF-COMMAND MESSAGE CENTER
2333
'2' ?
2341
CPF-COMMAND FORWARDING NUMBER
2334
'3' ?
2342
CPF-COMMAND MEMORY
2335
'9' ?
2343
CPF-COMMAND OUTSIDE CALL
2336
'Ø' ?
2344
CPF-COMMAND HELP
2337
'4' ?
2345
CPF-COMMAND MEET ME
2338
'5' ?
2346
CPF-COMMAND BRANCH ROUTE
2339
'8' ?
2347
CPF-COMMAND ADVANCED FEATURES

Fig. 23b.

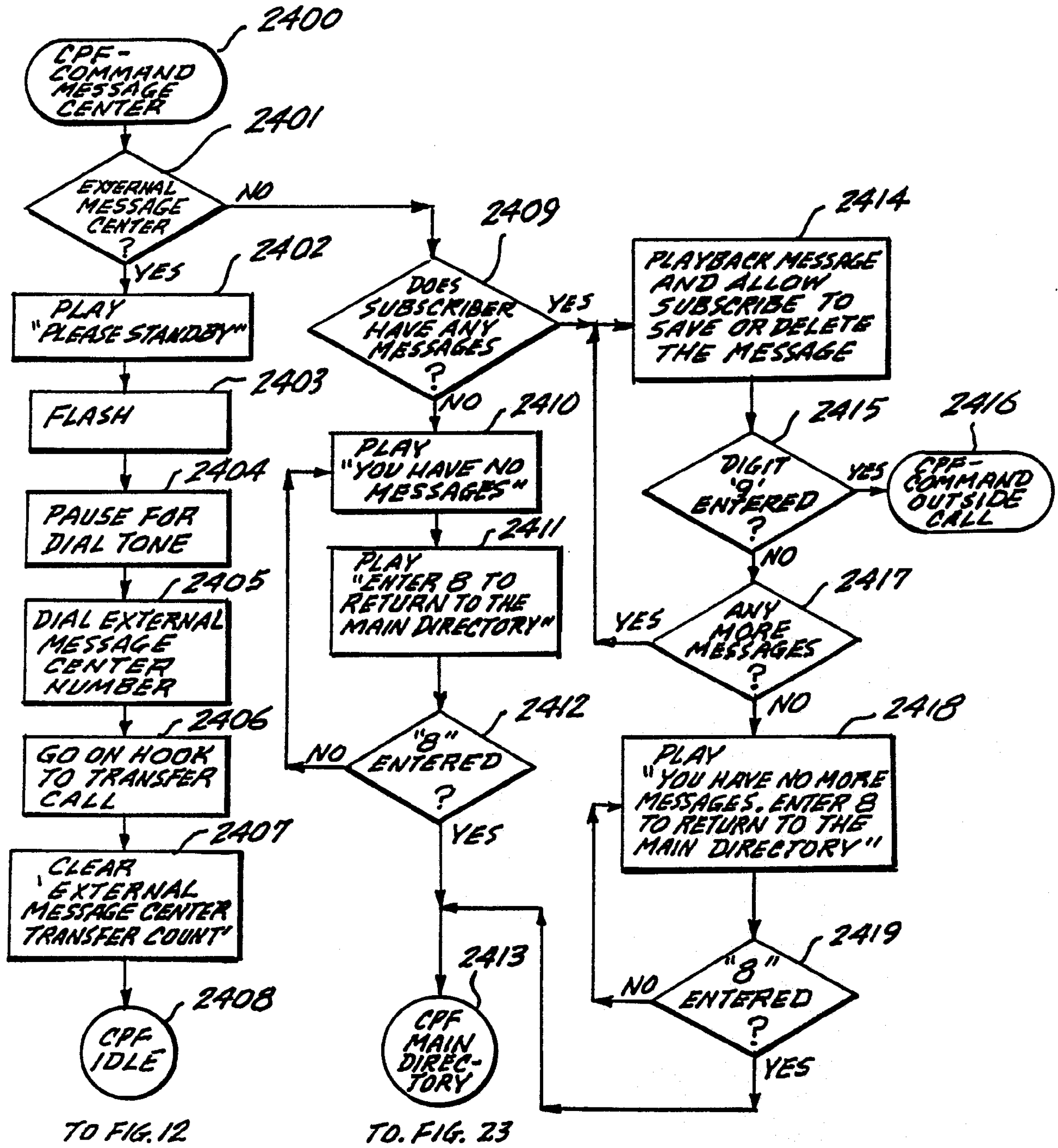

2400
CPF-COMMAND MESSAGE CENTER
2401
EXTERNAL MESSAGE CENTER ?
NO
YES
2402
PLAY "PLEASE STANDBY"
2403
FLASH
2404
PAUSE FOR DIAL TONE
2405
DIAL EXTERNAL MESSAGE CENTER NUMBER
2406
GO ON HOOK TO TRANSFER CALL
2407
CLEAR 'EXTERNAL MESSAGE CENTER TRANSFER COUNT'
2408
CPF IDLE
TO FIG. 12
2409
DOES SUBSCRIBER HAVE ANY MESSAGES ?
YES
NO
2410
PLAY "YOU HAVE NO MESSAGES"
2411
PLAY "ENTER 8 TO RETURN TO THE MAIN DIRECTORY"
2412
"8" ENTERED ?
NO
YES
2413
CPF MAIN DIREC-TORY
TO. FIG. 23
2414
PLAYBACK MESSAGE AND ALLOW SUBSCRIBE TO SAVE OR DELETE THE MESSAGE
2415
DIGIT '9' ENTERED ?
YES
NO
2416
CPF-COMMAND OUTSIDE CALL
2417
ANY MORE MESSAGES ?
YES
NO
2418
PLAY "YOU HAVE NO MORE MESSAGES. ENTER 8 TO RETURN TO THE MAIN DIRECTORY"
2419
"8" ENTERED ?
NO
YES

Fig. 24.

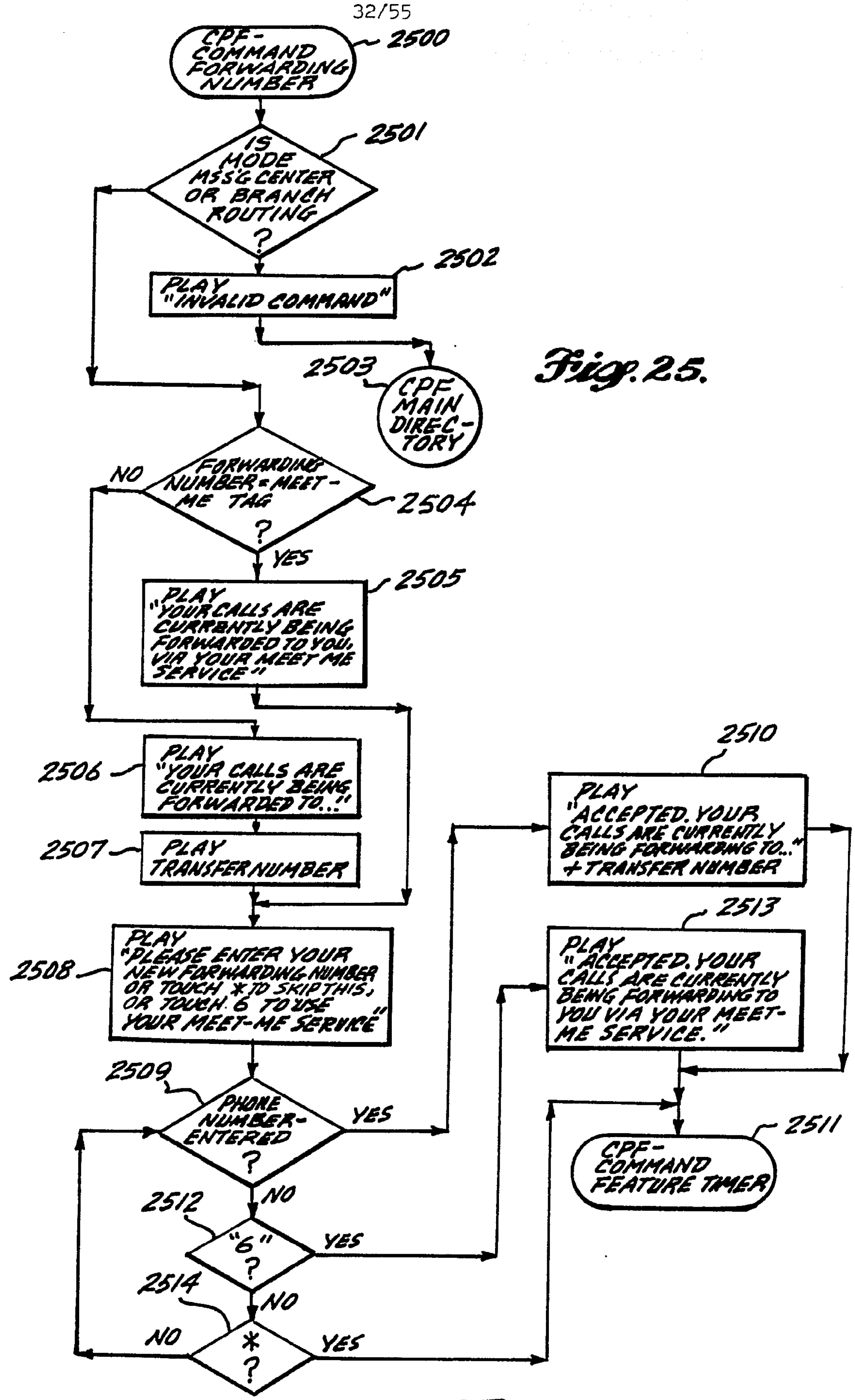


Fig. 25.

SUBSTITUTE SHEET

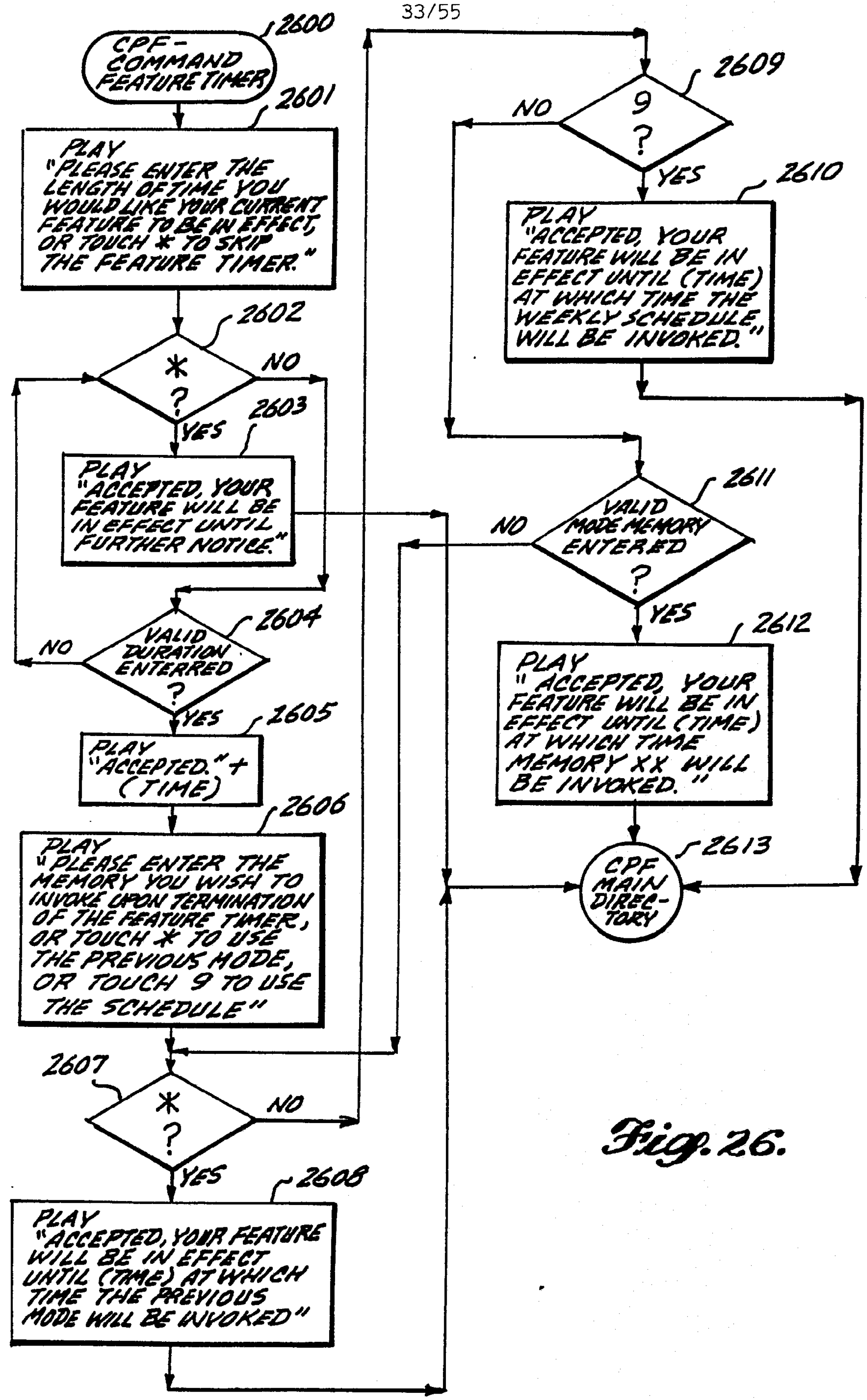

2600
CPF-COMMAND FEATURE TIMER
2601
PLAY "PLEASE ENTER THE LENGTH OF TIME YOU WOULD LIKE YOUR CURRENT FEATURE TO BE IN EFFECT, OR TOUCH * TO SKIP THE FEATURE TIMER."
2602
* ?
NO
YES
2603
PLAY "ACCEPTED, YOUR FEATURE WILL BE IN EFFECT UNTIL FURTHER NOTICE."
2604
VALID DURATION ENTERRED ?
NO
YES
2605
PLAY "ACCEPTED." + (TIME)
2606
PLAY "PLEASE ENTER THE MEMORY YOU WISH TO INVOKE UPON TERMINATION OF THE FEATURE TIMER, OR TOUCH * TO USE THE PREVIOUS MODE, OR TOUCH 9 TO USE THE SCHEDULE"
2607
* ?
NO
YES
2608
PLAY "ACCEPTED, YOUR FEATURE WILL BE IN EFFECT UNTIL (TIME) AT WHICH TIME THE PREVIOUS MODE WILL BE INVOKED"
2609
9 ?
NO
YES
2610
PLAY "ACCEPTED, YOUR FEATURE WILL BE IN EFFECT UNTIL (TIME) AT WHICH TIME THE WEEKLY SCHEDULE WILL BE INVOKED."
2611
VALID MODE MEMORY ENTERED ?
NO
YES
2612
PLAY "ACCEPTED, YOUR FEATURE WILL BE IN EFFECT UNTIL (TIME) AT WHICH TIME MEMORY XX WILL BE INVOKED."
2613
CPF MAIN DIREC-TORY

Fig. 26.

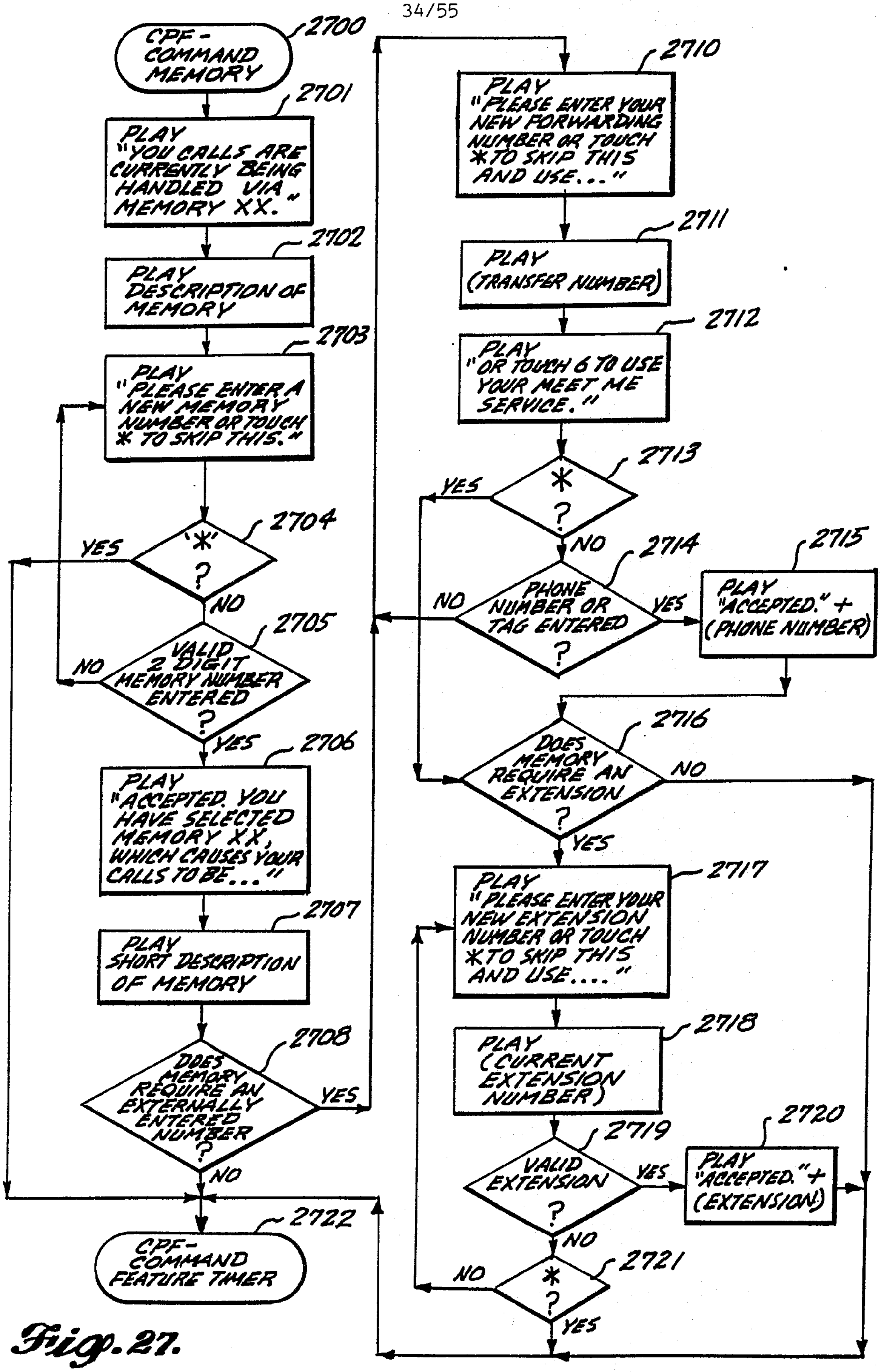

CPF - COMMAND MEMORY
2700
PLAY "YOU CALLS ARE CURRENTLY BEING HANDLED VIA MEMORY XX."
2701
PLAY DESCRIPTION OF MEMORY
2702
PLAY "PLEASE ENTER A NEW MEMORY NUMBER OR TOUCH * TO SKIP THIS."
2703
'*' ?
2704
YES
NO
VALID 2 DIGIT MEMORY NUMBER ENTERED ?
2705
NO
YES
PLAY "ACCEPTED. YOU HAVE SELECTED MEMORY XX, WHICH CAUSES YOUR CALLS TO BE..."
2706
PLAY SHORT DESCRIPTION OF MEMORY
2707
DOES MEMORY REQUIRE AN EXTERNALLY ENTERED NUMBER ?
2708
YES
NO
CPF- COMMAND FEATURE TIMER
2722
PLAY "PLEASE ENTER YOUR NEW FORWARDING NUMBER OR TOUCH *TO SKIP THIS AND USE..."
2710
PLAY (TRANSFER NUMBER)
2711
PLAY "OR TOUCH 6 TO USE YOUR MEET ME SERVICE."
2712
* ?
2713
YES
NO
PHONE NUMBER OR TAG ENTERED ?
2714
NO
YES
PLAY "ACCEPTED." + (PHONE NUMBER)
2715
DOES MEMORY REQUIRE AN EXTENSION ?
2716
NO
YES
PLAY "PLEASE ENTER YOUR NEW EXTENSION NUMBER OR TOUCH *TO SKIP THIS AND USE...."
2717
PLAY (CURRENT EXTENSION NUMBER)
2718
VALID EXTENSION ?
2719
YES
NO
PLAY "ACCEPTED." + (EXTENSION)
2720
* ?
2721
NO
YES

Fig.27.

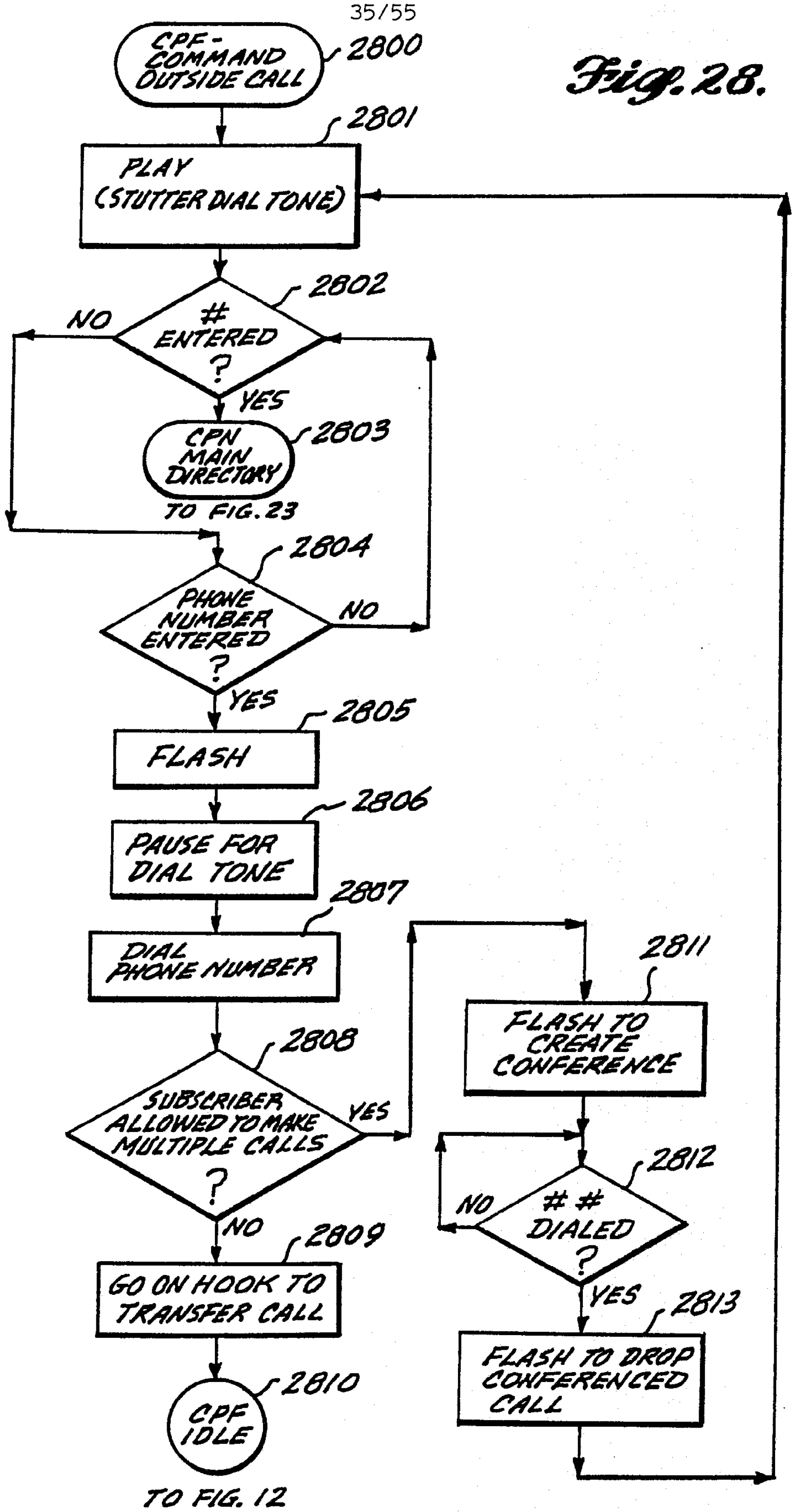

CPF-COMMAND OUTSIDE CALL
2800
2801
PLAY (STUTTER DIAL TONE)
2802
ENTERED ?
NO
YES
CPN MAIN DIRECTORY
2803
TO FIG. 23
2804
PHONE NUMBER ENTERED ?
NO
YES
2805
FLASH
2806
PAUSE FOR DIAL TONE
2807
DIAL PHONE NUMBER
2808
SUBSCRIBER ALLOWED TO MAKE MULTIPLE CALLS ?
YES
NO
2809
GO ON HOOK TO TRANSFER CALL
2810
CPF IDLE
TO FIG. 12
2811
FLASH TO CREATE CONFERENCE
2812
DIALED ?
NO
YES
2813
FLASH TO DROP CONFERENCED CALL

Fig. 28.

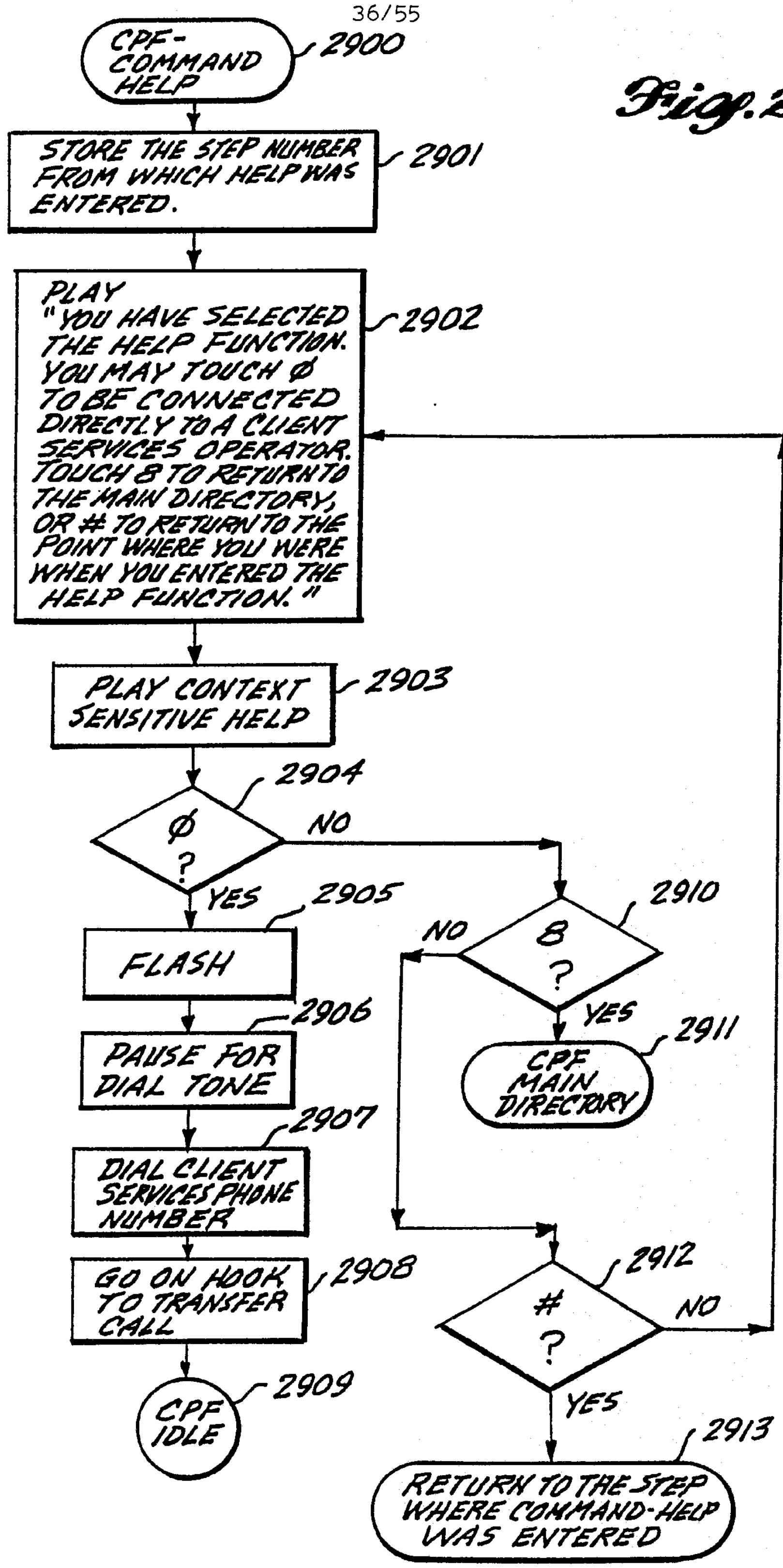

CPF-COMMAND HELP
2900
STORE THE STEP NUMBER FROM WHICH HELP WAS ENTERED.
2901
PLAY "YOU HAVE SELECTED THE HELP FUNCTION. YOU MAY TOUCH Ø TO BE CONNECTED DIRECTLY TO A CLIENT SERVICES OPERATOR. TOUCH 8 TO RETURN TO THE MAIN DIRECTORY, OR # TO RETURN TO THE POINT WHERE YOU WERE WHEN YOU ENTERED THE HELP FUNCTION."
2902
PLAY CONTEXT SENSITIVE HELP
2903
2904
Ø ?
NO
YES
2905
FLASH
2906
PAUSE FOR DIAL TONE
2907
DIAL CLIENT SERVICES PHONE NUMBER
2908
GO ON HOOK TO TRANSFER CALL
2909
CPF IDLE
2910
8 ?
NO
YES
2911
CPF MAIN DIRECTORY
2912
?
NO
YES
2913
RETURN TO THE STEP WHERE COMMAND-HELP WAS ENTERED

Fig. 29.

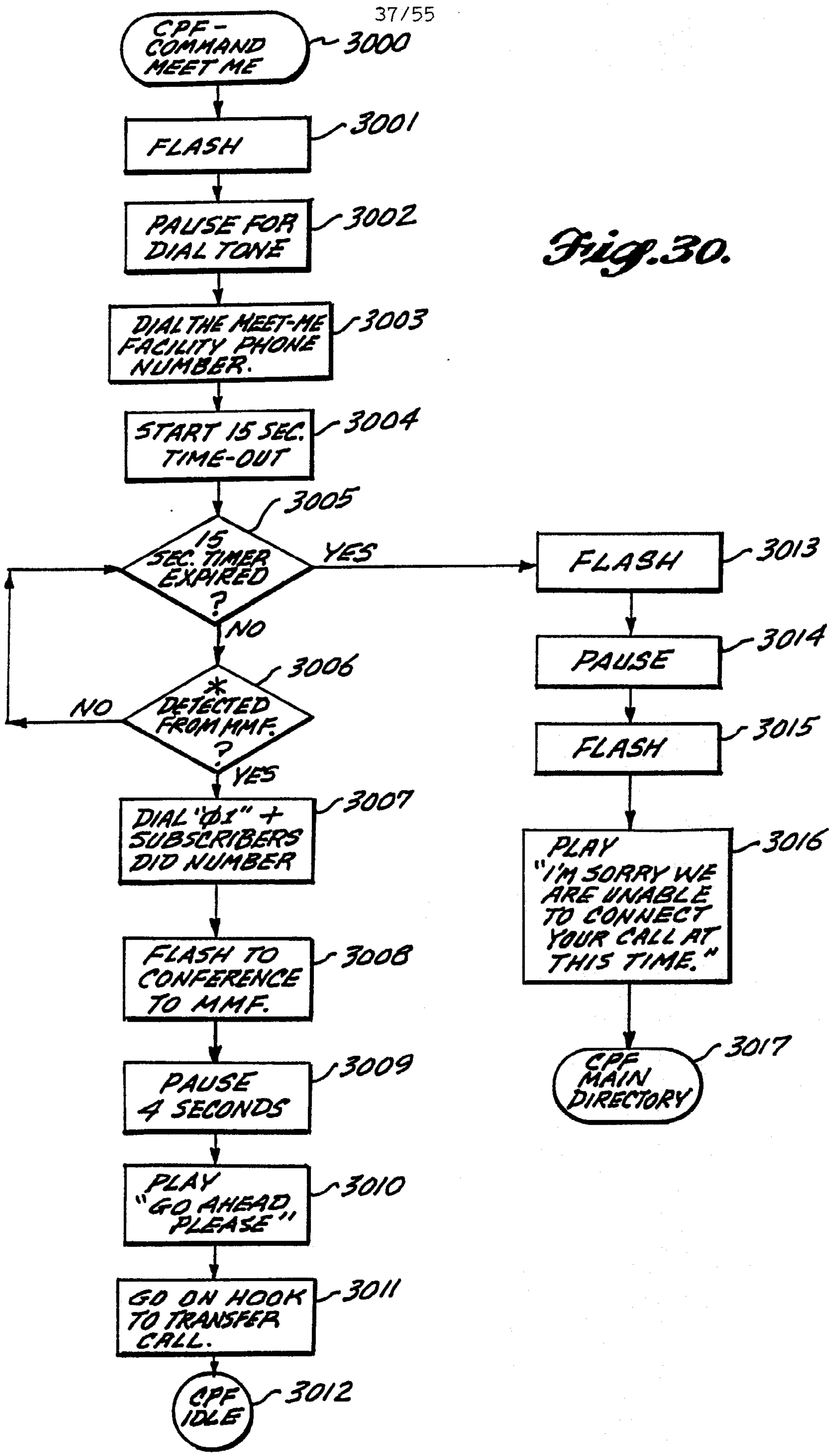

CPF - COMMAND MEET ME
3000
FLASH
3001
PAUSE FOR DIAL TONE
3002
DIAL THE MEET-ME FACILITY PHONE NUMBER.
3003
START 15 SEC. TIME-OUT
3004
15 SEC. TIMER EXPIRED ?
3005
YES
NO
* DETECTED FROM MMF. ?
3006
NO
YES
DIAL "Ø1" + SUBSCRIBERS DID NUMBER
3007
FLASH TO CONFERENCE TO MMF.
3008
PAUSE 4 SECONDS
3009
PLAY "GO AHEAD, PLEASE"
3010
GO ON HOOK TO TRANSFER CALL.
3011
CPF IDLE
3012
FLASH
3013
PAUSE
3014
FLASH
3015
PLAY "I'M SORRY WE ARE UNABLE TO CONNECT YOUR CALL AT THIS TIME."
3016
CPF MAIN DIRECTORY
3017

Fig.30.

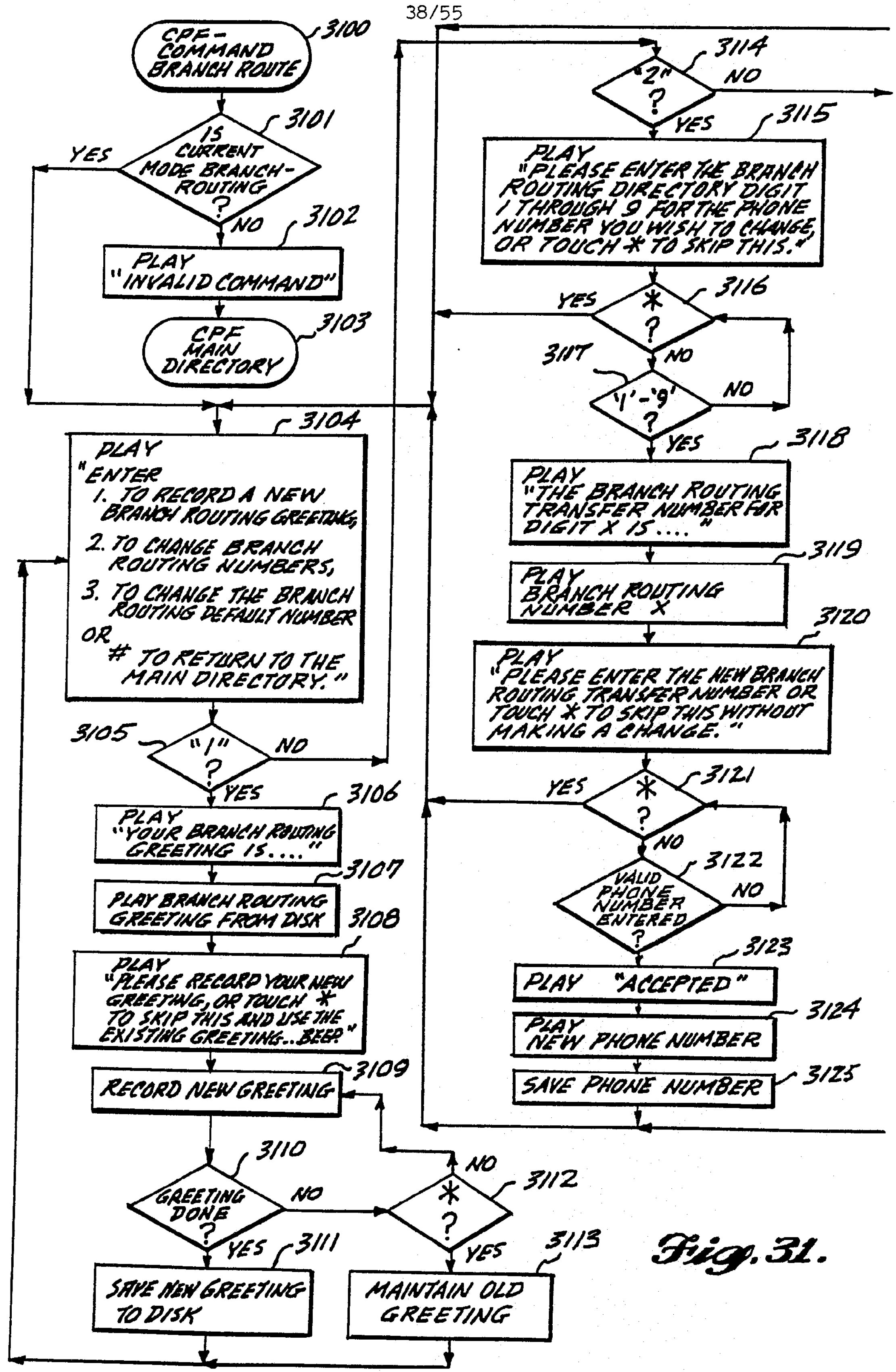

CPF-COMMAND BRANCH ROUTE
3100
IS CURRENT MODE BRANCH-ROUTING ?
3101
YES
NO
PLAY "INVALID COMMAND"
3102
CPF MAIN DIRECTORY
3103
PLAY "ENTER 1. TO RECORD A NEW BRANCH ROUTING GREETING, 2. TO CHANGE BRANCH ROUTING NUMBERS, 3. TO CHANGE THE BRANCH ROUTING DEFAULT NUMBER OR # TO RETURN TO THE MAIN DIRECTORY."
3104
"1" ?
3105
PLAY "YOUR BRANCH ROUTING GREETING IS...."
3106
PLAY BRANCH ROUTING GREETING FROM DISK
3107
PLAY "PLEASE RECORD YOUR NEW GREETING, OR TOUCH * TO SKIP THIS AND USE THE EXISTING GREETING...BEEP"
3108
RECORD NEW GREETING
3109
GREETING DONE ?
3110
SAVE NEW GREETING TO DISK
3111
* ?
3112
MAINTAIN OLD GREETING
3113
"2" ?
3114
PLAY "PLEASE ENTER THE BRANCH ROUTING DIRECTORY DIGIT 1 THROUGH 9 FOR THE PHONE NUMBER YOU WISH TO CHANGE, OR TOUCH * TO SKIP THIS."
3115
* ?
3116
'1'-'9' ?
3117
PLAY "THE BRANCH ROUTING TRANSFER NUMBER FOR DIGIT X IS...."
3118
PLAY BRANCH ROUTING NUMBER X
3119
PLAY "PLEASE ENTER THE NEW BRANCH ROUTING TRANSFER NUMBER OR TOUCH * TO SKIP THIS WITHOUT MAKING A CHANGE."
3120
* ?
3121
VALID PHONE NUMBER ENTERED ?
3122
PLAY "ACCEPTED"
3123
PLAY NEW PHONE NUMBER
3124
SAVE PHONE NUMBER
3125

Fig. 31.

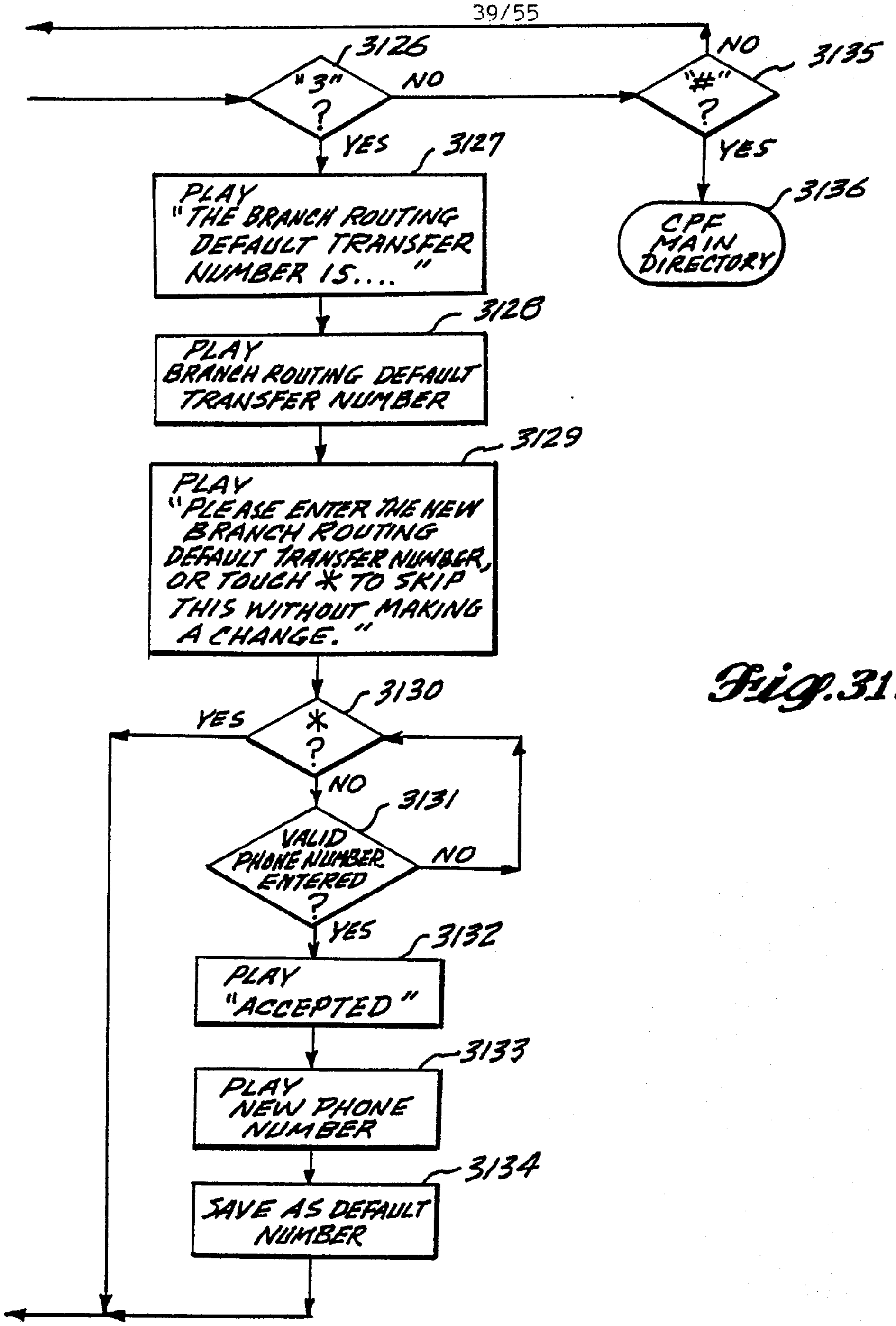
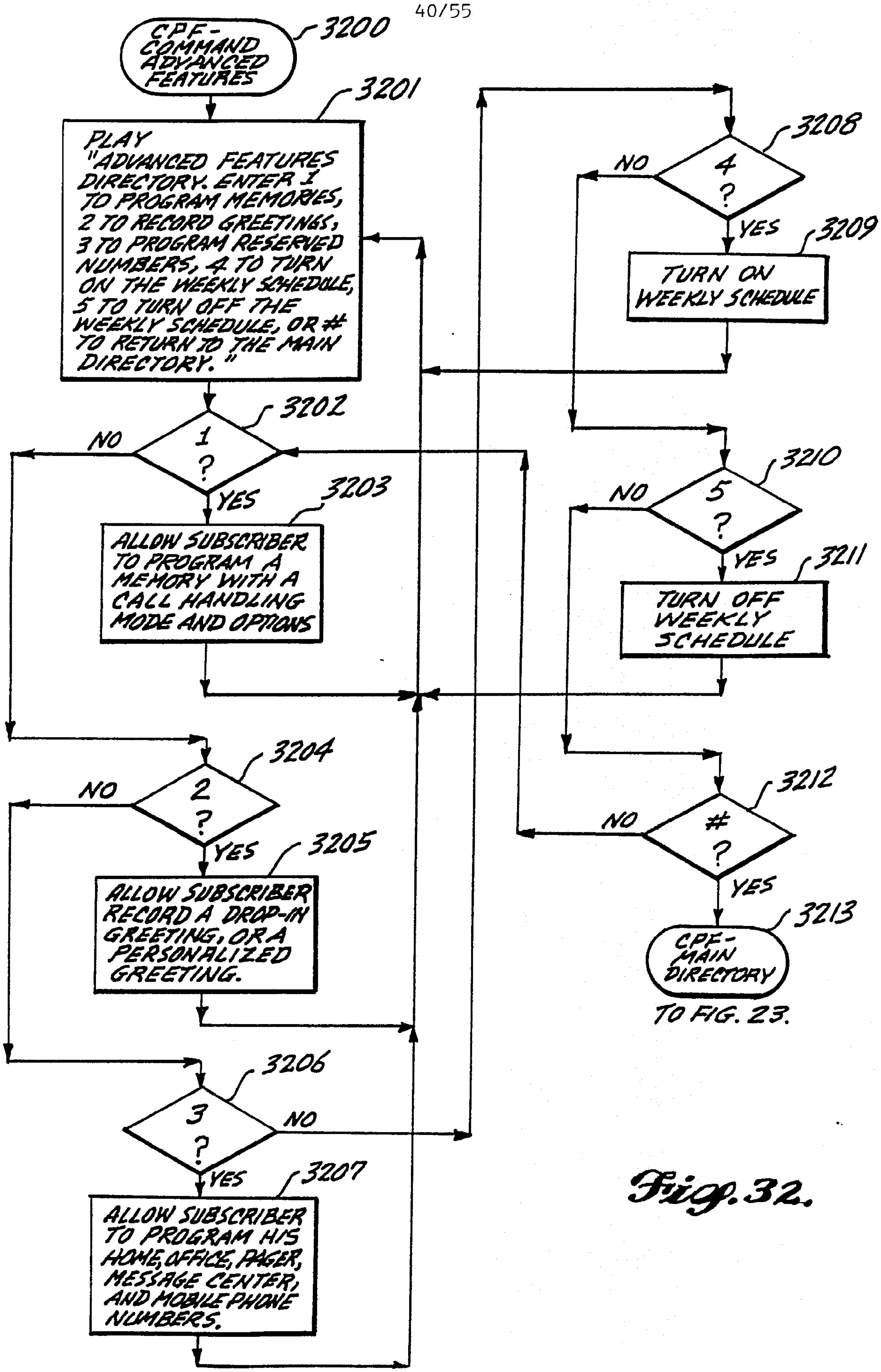

3126
"3" ?
NO
YES
3135
"#" ?
NO
YES
3136
CPF MAIN DIRECTORY
3127
PLAY "THE BRANCH ROUTING DEFAULT TRANSFER NUMBER IS.... "
3128
PLAY BRANCH ROUTING DEFAULT TRANSFER NUMBER
3129
PLAY "PLEASE ENTER THE NEW BRANCH ROUTING DEFAULT TRANSFER NUMBER, OR TOUCH * TO SKIP THIS WITHOUT MAKING A CHANGE."
3130
* ?
YES
NO
3131
VALID PHONE NUMBER ENTERED ?
NO
YES
3132
PLAY "ACCEPTED"
3133
PLAY NEW PHONE NUMBER
3134
SAVE AS DEFAULT NUMBER

Fig.31.



CPF-COMMAND ADVANCED FEATURES
3200
PLAY "ADVANCED FEATURES DIRECTORY. ENTER 1 TO PROGRAM MEMORIES, 2 TO RECORD GREETINGS, 3 TO PROGRAM RESERVED NUMBERS, 4 TO TURN ON THE WEEKLY SCHEDULE, 5 TO TURN OFF THE WEEKLY SCHEDULE, OR # TO RETURN TO THE MAIN DIRECTORY."
3201
1 ?
3202
NO
YES
ALLOW SUBSCRIBER TO PROGRAM A MEMORY WITH A CALL HANDLING MODE AND OPTIONS
3203
2 ?
3204
NO
YES
ALLOW SUBSCRIBER RECORD A DROP-IN GREETING, OR A PERSONALIZED GREETING.
3205
3 ?
3206
NO
YES
ALLOW SUBSCRIBER TO PROGRAM HIS HOME, OFFICE, PAGER, MESSAGE CENTER, AND MOBILE PHONE NUMBERS.
3207
4 ?
3208
NO
YES
TURN ON WEEKLY SCHEDULE
3209
5 ?
3210
NO
YES
TURN OFF WEEKLY SCHEDULE
3211
?
3212
NO
YES
CPF-MAIN DIRECTORY
3213
TO FIG. 23.

Fig. 32.

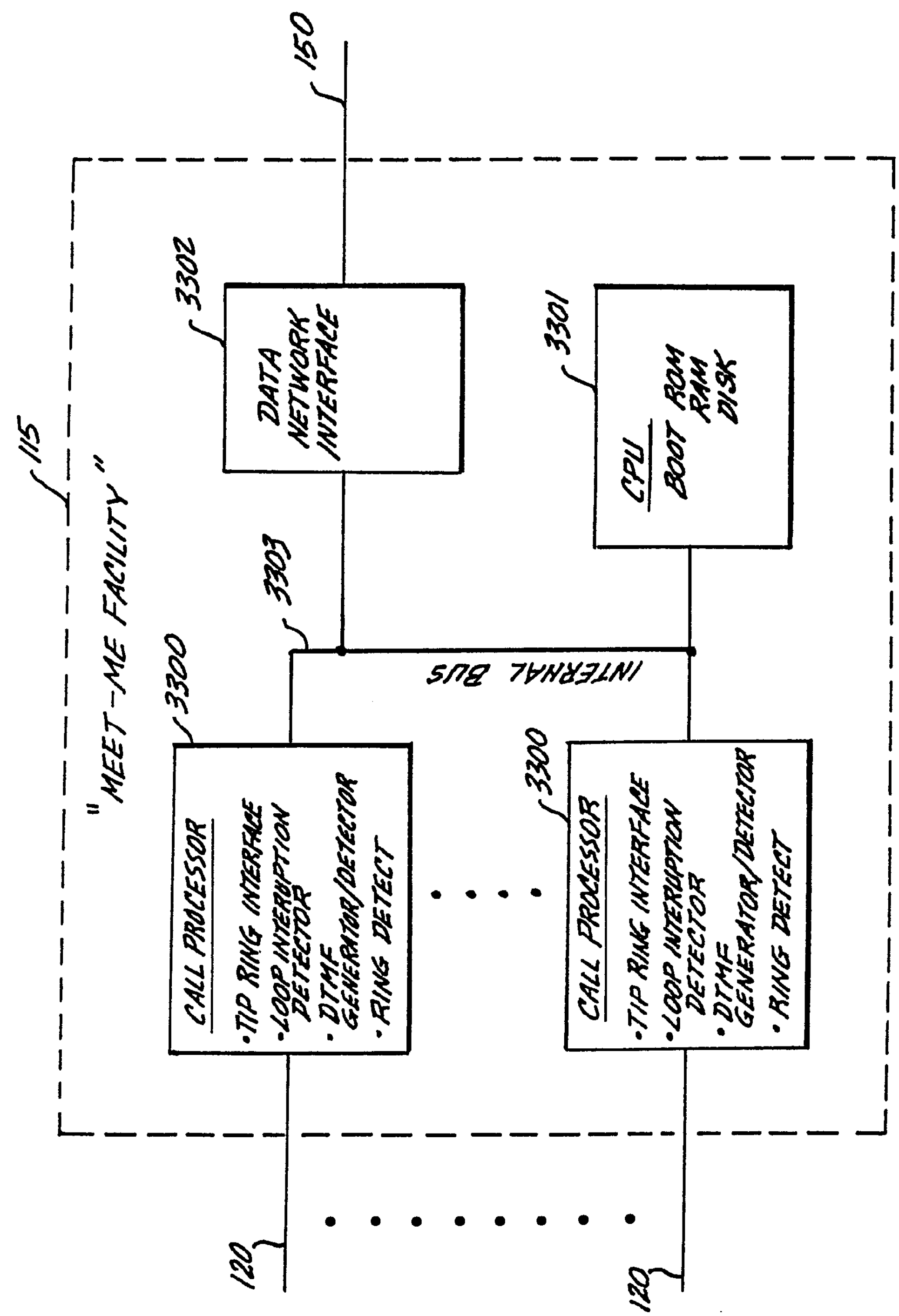

"MEET-ME FACILITY"
115
150
3302
DATA NETWORK INTERFACE
3301
CPU
BOOT ROM RAM DISK
3303
INTERNAL BUS
3300
CALL PROCESSOR
•TIP RING INTERFACE
•LOOP INTERRUPTION DETECTOR
•DTMF GENERATOR/DETECTOR
•RING DETECT
3300
CALL PROCESSOR
•TIP RING INTERFACE
•LOOP INTERRUPTION DETECTOR
•DTMF GENERATOR/DETECTOR
•RING DETECT
120
120

Fig. 33.

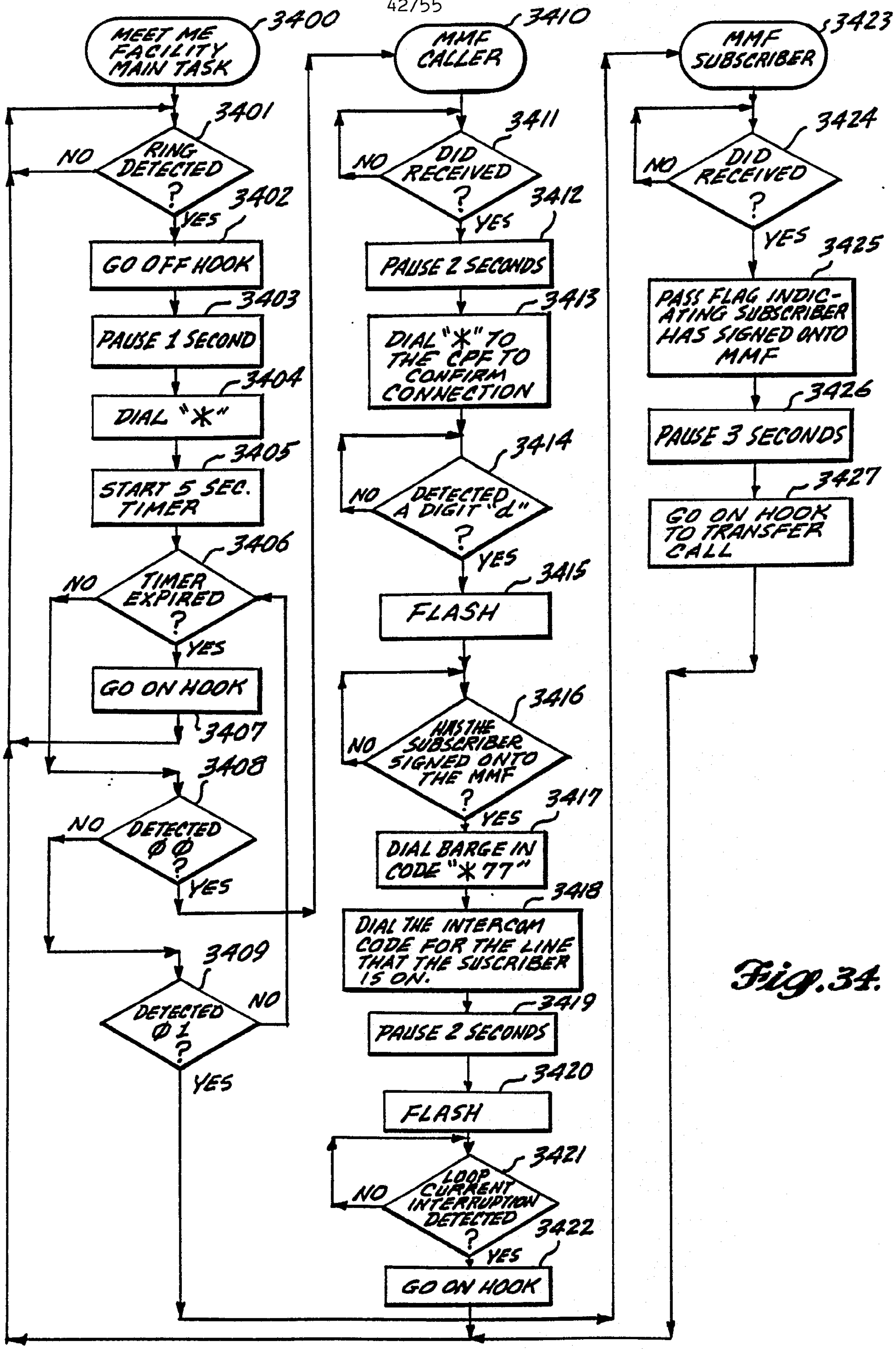

MEET ME FACILITY MAIN TASK
3400
RING DETECTED ?
3401
NO
YES
GO OFF HOOK
3402
PAUSE 1 SECOND
3403
DIAL "*"
3404
START 5 SEC. TIMER
3405
TIMER EXPIRED ?
3406
GO ON HOOK
3407
DETECTED ∅∅ ?
3408
DETECTED ∅1 ?
3409
MMF CALLER
3410
DID RECEIVED ?
3411
PAUSE 2 SECONDS
3412
DIAL "*" TO THE CPF TO CONFIRM CONNECTION
3413
DETECTED A DIGIT "d" ?
3414
FLASH
3415
HAS THE SUBSCRIBER SIGNED ONTO THE MMF ?
3416
DIAL BARGE IN CODE "*77"
3417
DIAL THE INTERCOM CODE FOR THE LINE THAT THE SUSCRIBER IS ON.
3418
PAUSE 2 SECONDS
3419
FLASH
3420
LOOP CURRENT INTERRUPTION DETECTED ?
3421
GO ON HOOK
3422
MMF SUBSCRIBER
3423
DID RECEIVED ?
3424
PASS FLAG INDICATING SUBSCRIBER HAS SIGNED ONTO MMF
3425
PAUSE 3 SECONDS
3426
GO ON HOOK TO TRANSFER CALL
3427

Fig.34.

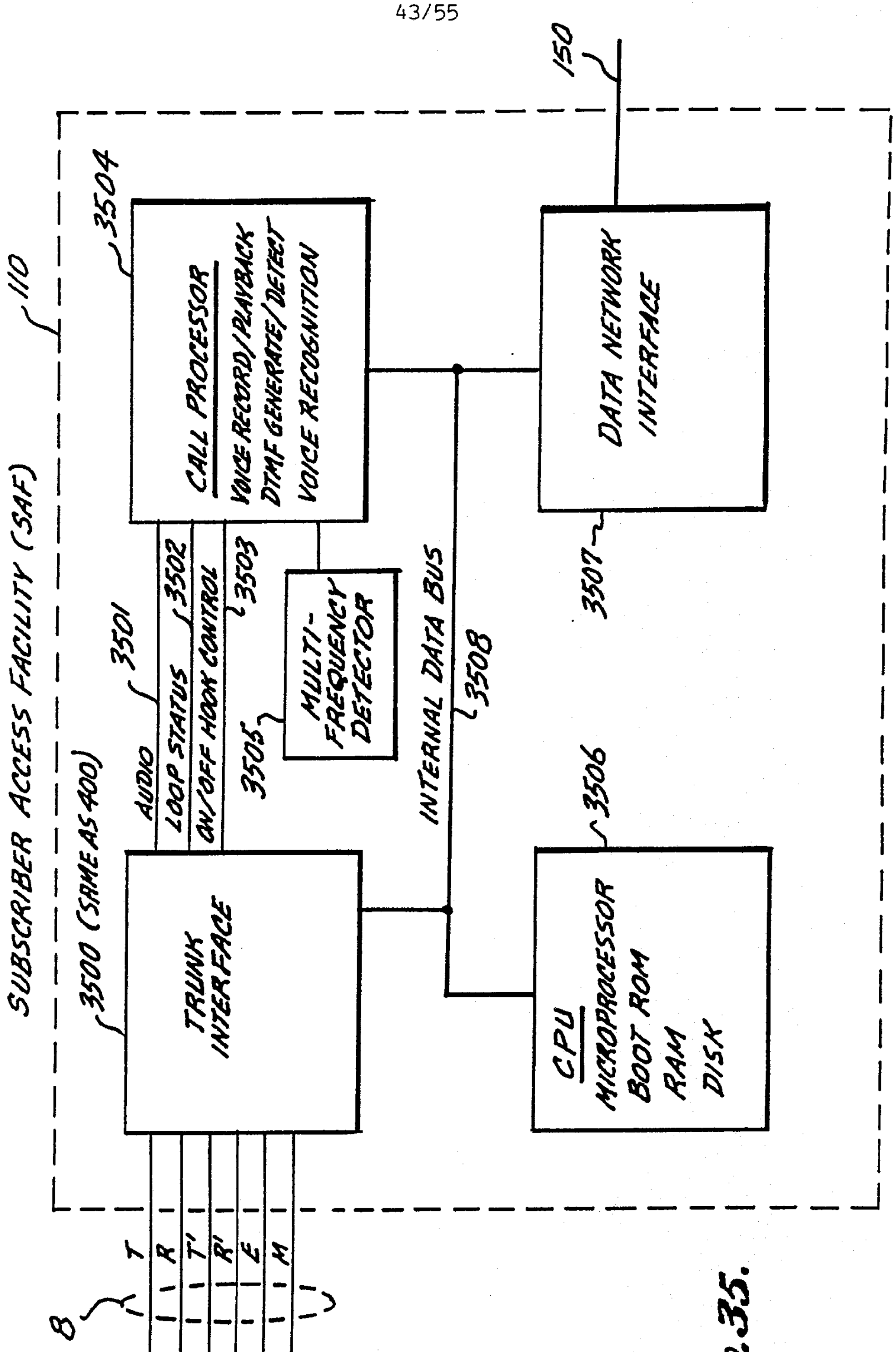


Fig. 35.

SUBSTITUTE SHEET

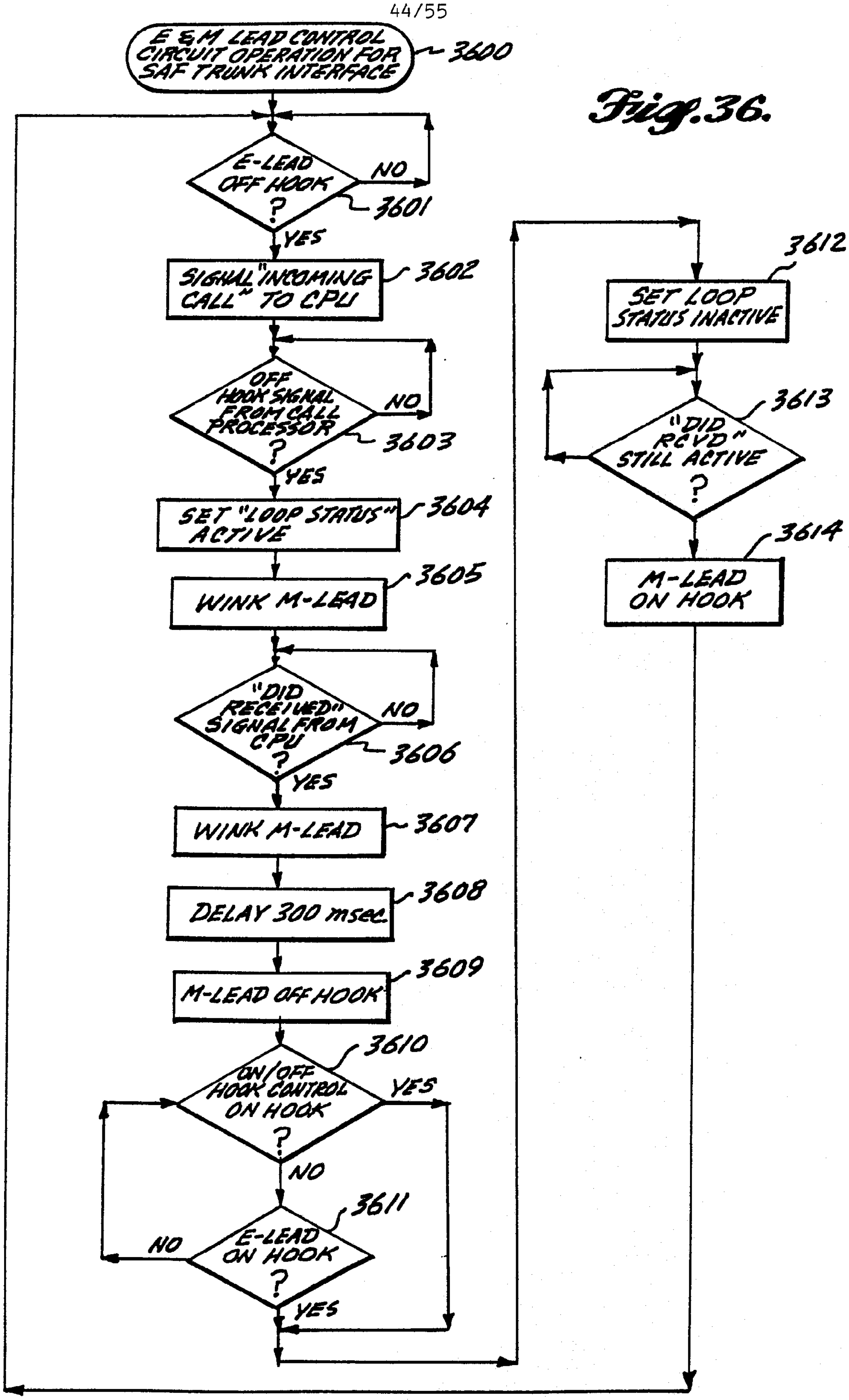

E & M LEAD CONTROL CIRCUIT OPERATION FOR SAF TRUNK INTERFACE
3600
E-LEAD OFF HOOK ?
NO
YES
3601
SIGNAL "INCOMING CALL" TO CPU
3602
OFF HOOK SIGNAL FROM CALL PROCESSOR ?
NO
YES
3603
SET "LOOP STATUS" ACTIVE
3604
WINK M-LEAD
3605
"DID RECEIVED" SIGNAL FROM CPU ?
NO
YES
3606
WINK M-LEAD
3607
DELAY 300 msec.
3608
M-LEAD OFF HOOK
3609
ON/OFF HOOK CONTROL ON HOOK ?
YES
NO
3610
E-LEAD ON HOOK ?
NO
YES
3611
SET LOOP STATUS INACTIVE
3612
"DID RCVD" STILL ACTIVE ?
3613
M-LEAD ON HOOK
3614

Fig. 36.

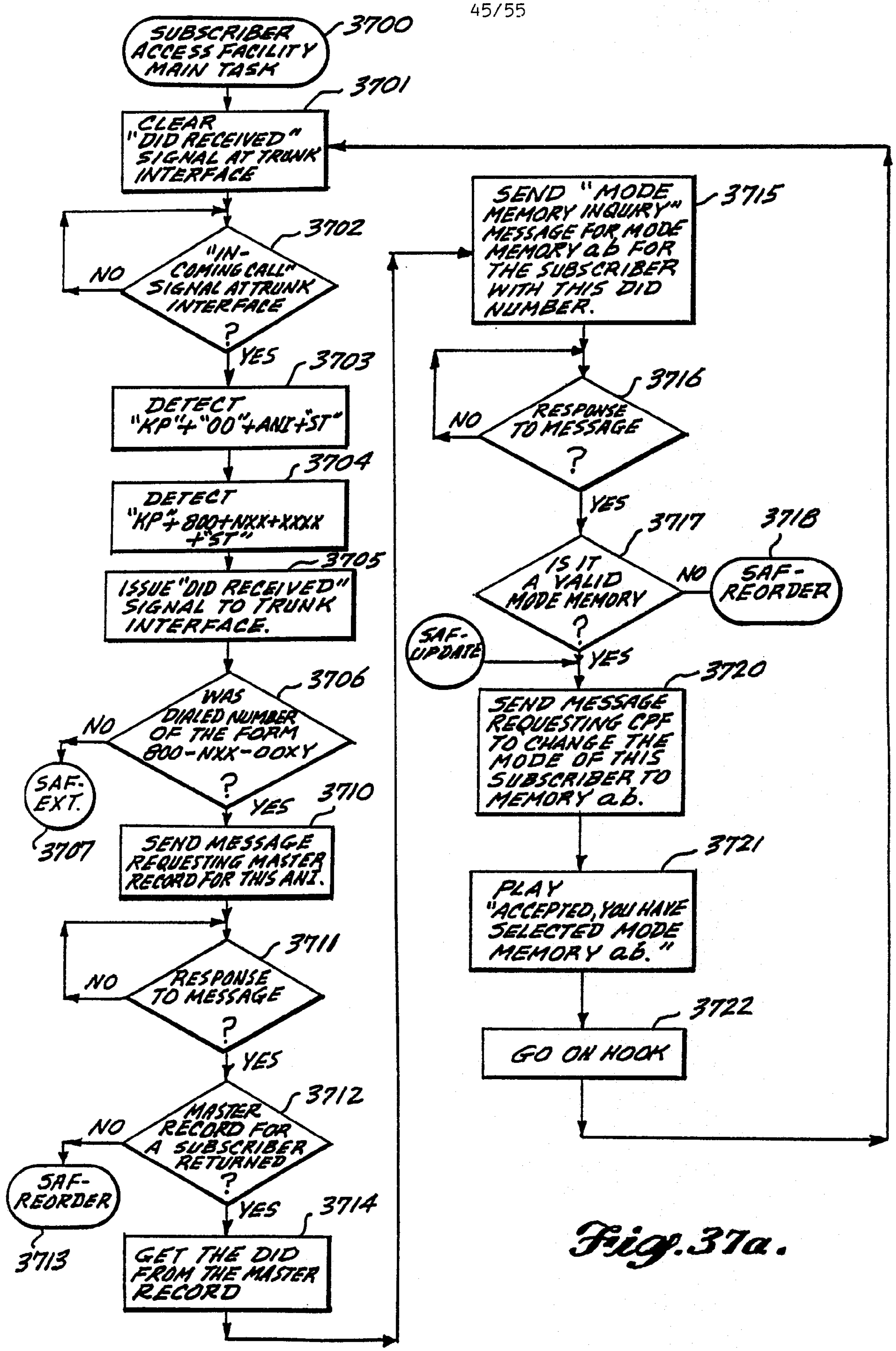

SUBSCRIBER ACCESS FACILITY MAIN TASK
3700
3701
CLEAR "DID RECEIVED" SIGNAL AT TRUNK INTERFACE
3702
"IN-COMING CALL" SIGNAL AT TRUNK INTERFACE ?
NO
YES
3703
DETECT "KP"+"00"+ANI+"ST"
3704
DETECT "KP"+800+NXX+XXXX +"ST"
3705
ISSUE "DID RECEIVED" SIGNAL TO TRUNK INTERFACE.
3706
WAS DIALED NUMBER OF THE FORM 800-NXX-00XY ?
NO
SAF-EXT.
3707
YES
3710
SEND MESSAGE REQUESTING MASTER RECORD FOR THIS ANI.
3711
RESPONSE TO MESSAGE ?
NO
YES
3712
MASTER RECORD FOR A SUBSCRIBER RETURNED ?
NO
SAF-REORDER
3713
YES
3714
GET THE DID FROM THE MASTER RECORD
3715
SEND "MODE MEMORY INQUIRY" MESSAGE FOR MODE MEMORY ab FOR THE SUBSCRIBER WITH THIS DID NUMBER.
3716
RESPONSE TO MESSAGE ?
NO
YES
3717
IS IT A VALID MODE MEMORY ?
NO
3718
SAF-REORDER
SAF-UPDATE
YES
3720
SEND MESSAGE REQUESTING CPF TO CHANGE THE MODE OF THIS SUBSCRIBER TO MEMORY ab.
3721
PLAY "ACCEPTED, YOU HAVE SELECTED MODE MEMORY ab."
3722
GO ON HOOK

Fig. 37a.

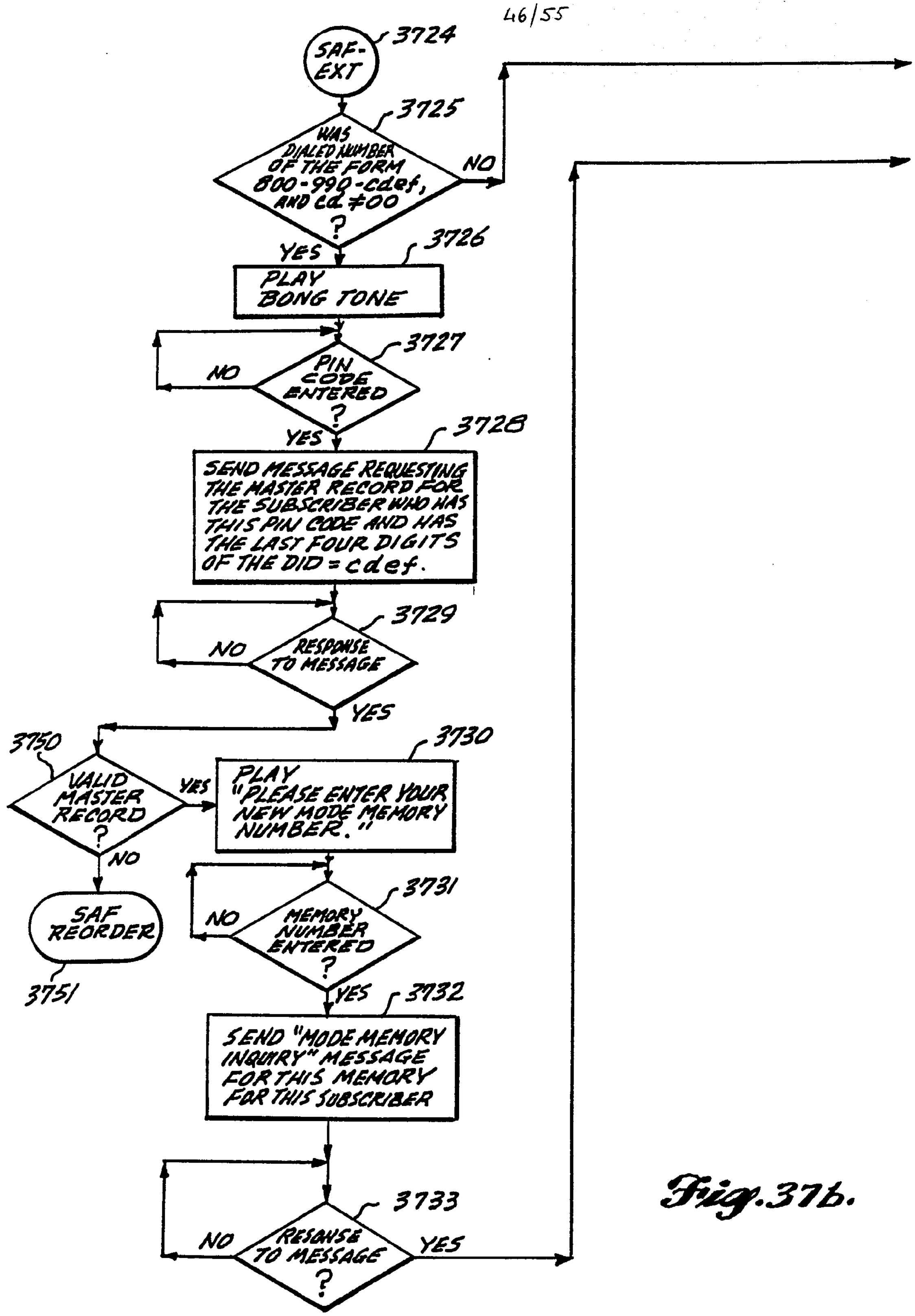
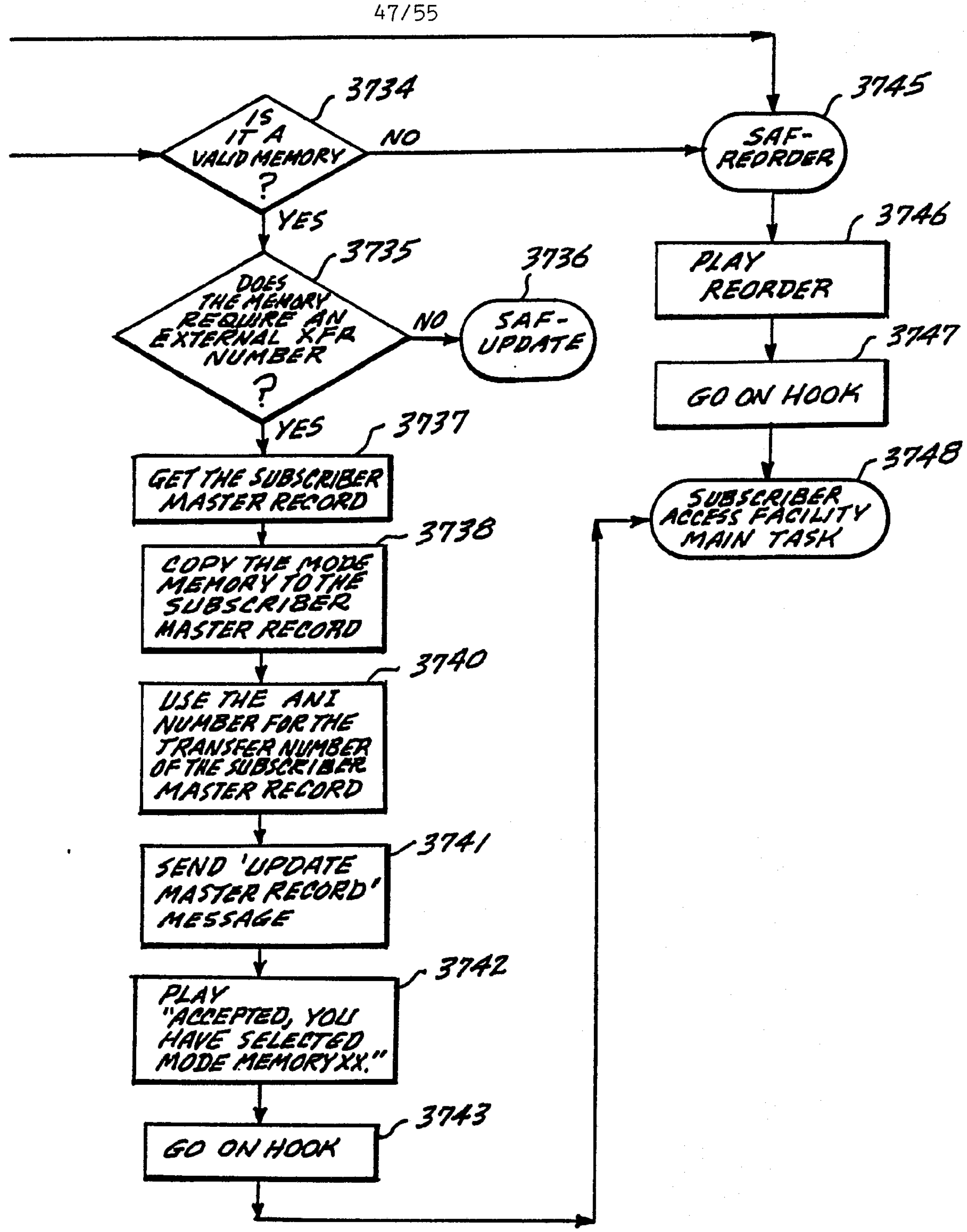

SAF-EXT
3724
WAS DIALED NUMBER OF THE FORM 800-990-cdef, AND cd ≠00 ?
3725
NO
YES
PLAY BONG TONE
3726
PIN CODE ENTERED ?
3727
NO
YES
SEND MESSAGE REQUESTING THE MASTER RECORD FOR THE SUBSCRIBER WHO HAS THIS PIN CODE AND HAS THE LAST FOUR DIGITS OF THE DID = cdef.
3728
RESPONSE TO MESSAGE
3729
NO
YES
3750
VALID MASTER RECORD ?
YES
NO
SAF REORDER
3751
PLAY "PLEASE ENTER YOUR NEW MODE MEMORY NUMBER."
3730
MEMORY NUMBER ENTERED ?
3731
NO
YES
SEND "MODE MEMORY INQUIRY" MESSAGE FOR THIS MEMORY FOR THIS SUBSCRIBER
3732
RESPONSE TO MESSAGE ?
3733
NO
YES

Fig.37b.



3734
IS IT A VALID MEMORY ?
NO
YES
3745
SAF-REORDER
3746
PLAY REORDER
3747
GO ON HOOK
3748
SUBSCRIBER ACCESS FACILITY MAIN TASK
3735
DOES THE MEMORY REQUIRE AN EXTERNAL XFR NUMBER ?
NO
3736
SAF-UPDATE
YES
3737
GET THE SUBSCRIBER MASTER RECORD
3738
COPY THE MODE MEMORY TO THE SUBSCRIBER MASTER RECORD
3740
USE THE ANI NUMBER FOR THE TRANSFER NUMBER OF THE SUBSCRIBER MASTER RECORD
3741
SEND 'UPDATE MASTER RECORD' MESSAGE
3742
PLAY "ACCEPTED, YOU HAVE SELECTED MODE MEMORY XX."
3743
GO ON HOOK

Fig.37b.

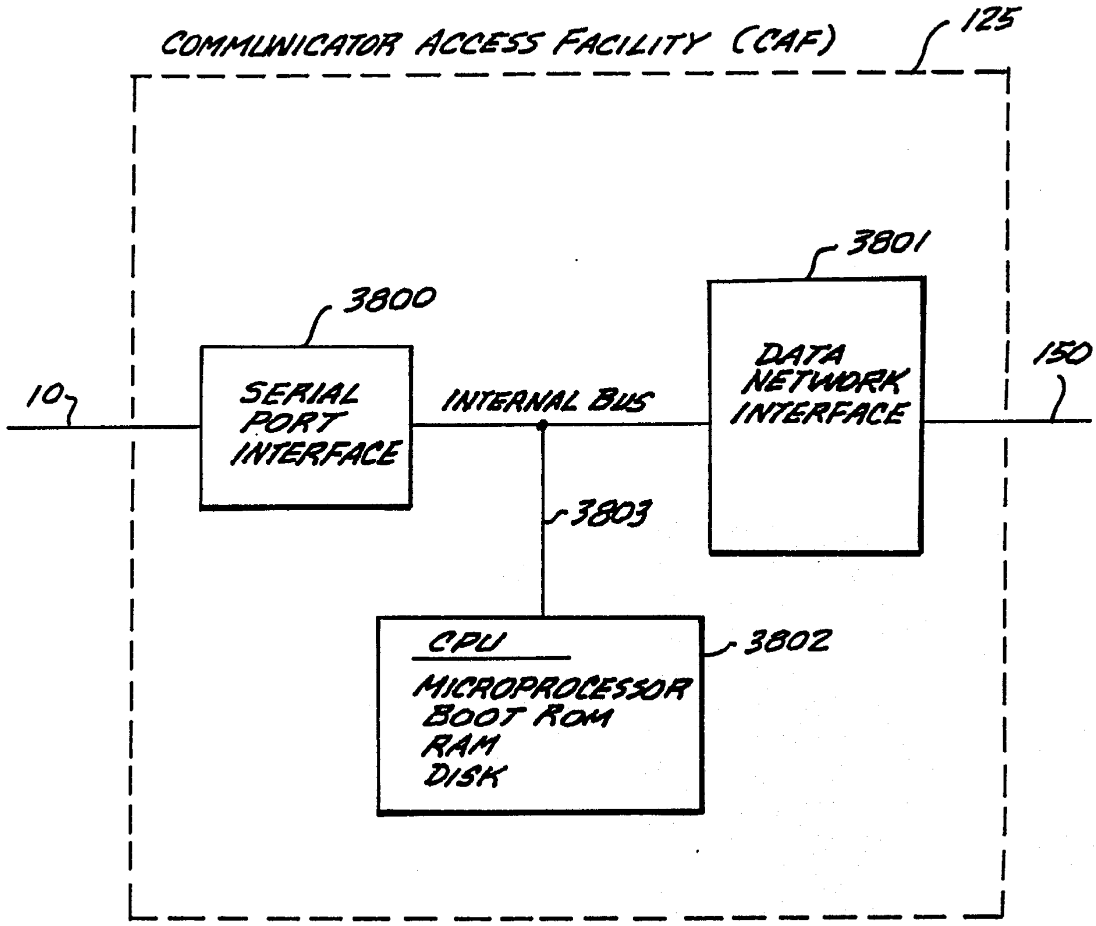

COMMUNICATOR ACCESS FACILITY (CAF)
125
3801
3800
DATA NETWORK INTERFACE
150
10
SERIAL PORT INTERFACE
INTERNAL BUS
3803
CPU
MICROPROCESSOR
BOOT ROM
RAM
DISK
3802

Fig. 38.

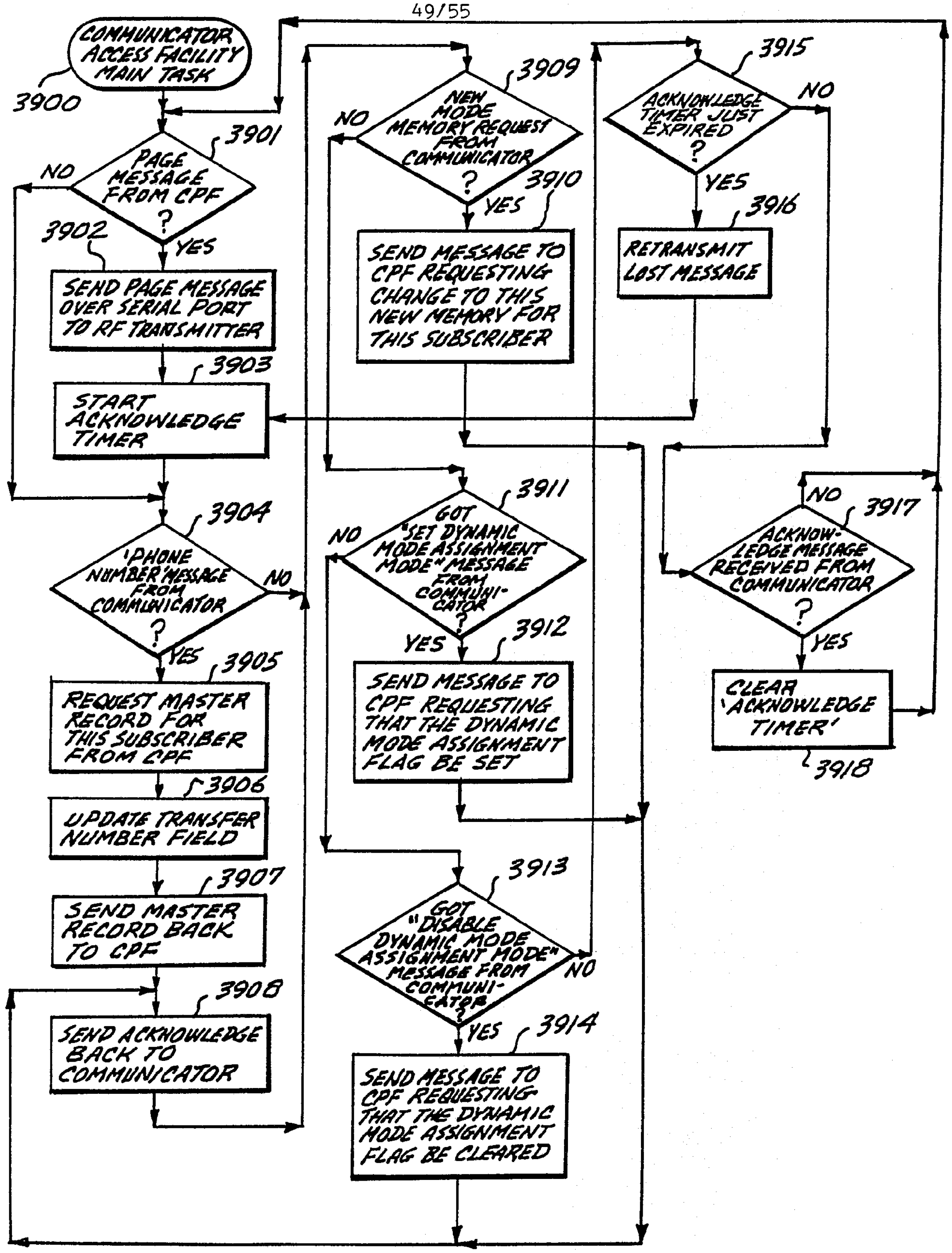


Fig. 39.

SUBSTITUTE SHEET

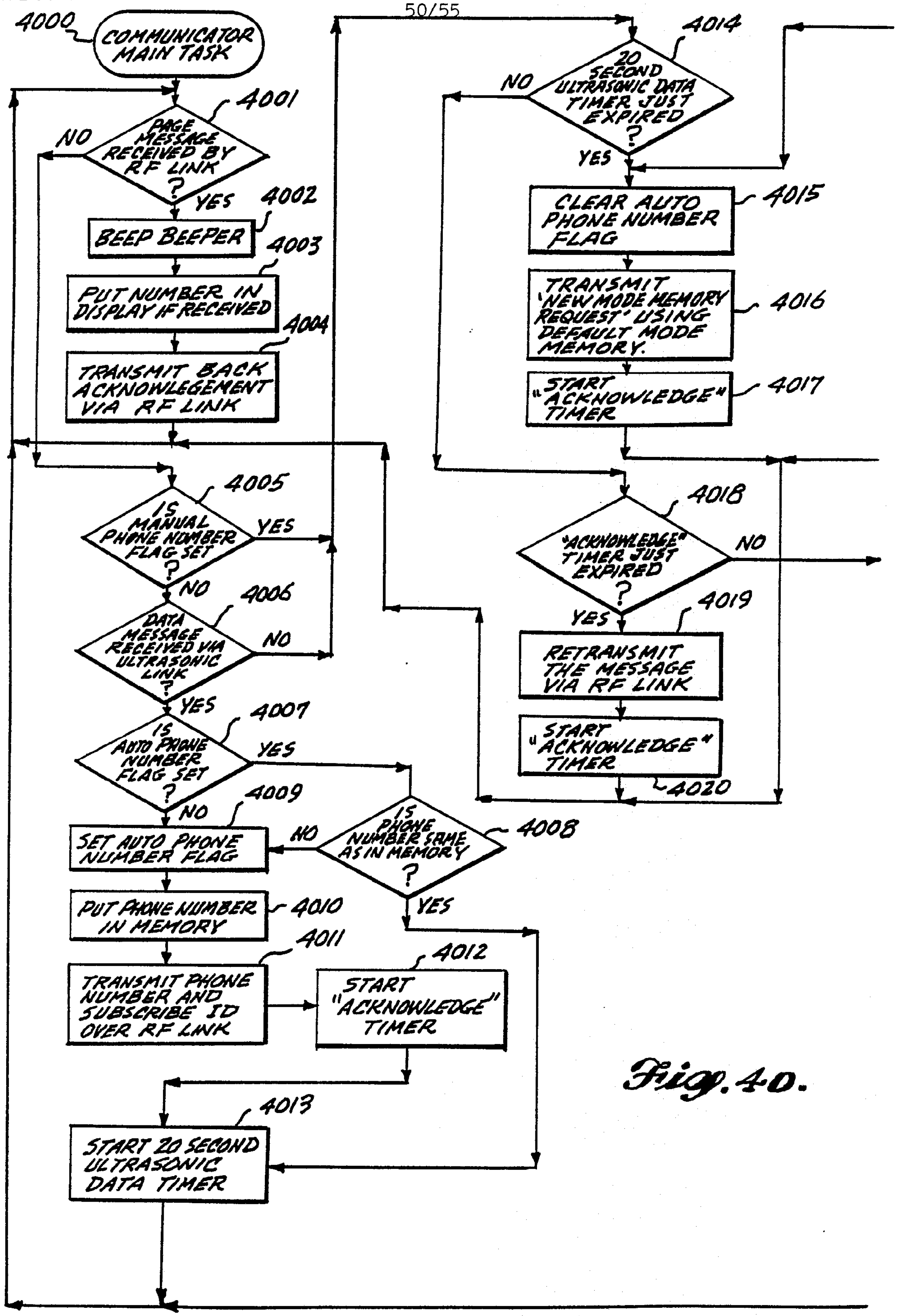


Fig. 40.

SUBSTITUTE SHEET

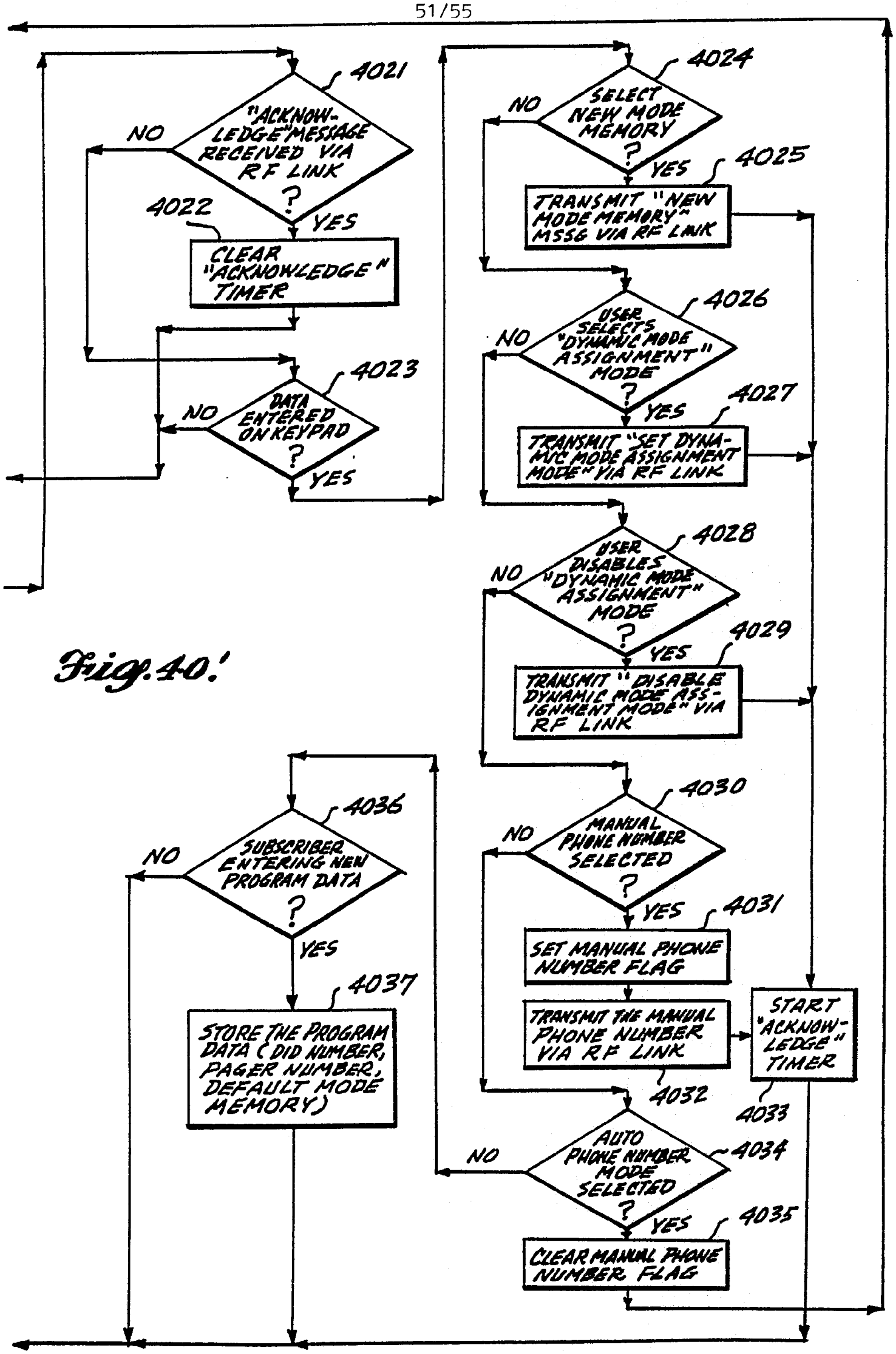

4021
"ACKNOWLEDGE" MESSAGE RECEIVED VIA RF LINK ?
NO
YES
4022
CLEAR "ACKNOWLEDGE" TIMER
4023
DATA ENTERED ON KEYPAD ?
NO
YES
4024
SELECT NEW MODE MEMORY ?
NO
YES
4025
TRANSMIT "NEW MODE MEMORY" MSSG VIA RF LINK
4026
USER SELECTS "DYNAMIC MODE ASSIGNMENT" MODE ?
NO
YES
4027
TRANSMIT "SET DYNAMIC MODE ASSIGNMENT MODE" VIA RF LINK
4028
USER DISABLES "DYNAMIC MODE ASSIGNMENT" MODE ?
NO
YES
4029
TRANSMIT "DISABLE DYNAMIC MODE ASSIGNMENT MODE" VIA RF LINK
4030
MANUAL PHONE NUMBER SELECTED ?
NO
YES
4031
SET MANUAL PHONE NUMBER FLAG
TRANSMIT THE MANUAL PHONE NUMBER VIA RF LINK
4032
START "ACKNOWLEDGE" TIMER
4033
4034
AUTO PHONE NUMBER MODE SELECTED ?
NO
YES
4035
CLEAR MANUAL PHONE NUMBER FLAG
4036
SUBSCRIBER ENTERING NEW PROGRAM DATA ?
NO
YES
4037
STORE THE PROGRAM DATA (DID NUMBER, PAGER NUMBER, DEFAULT MODE MEMORY)

Fig. 40.'

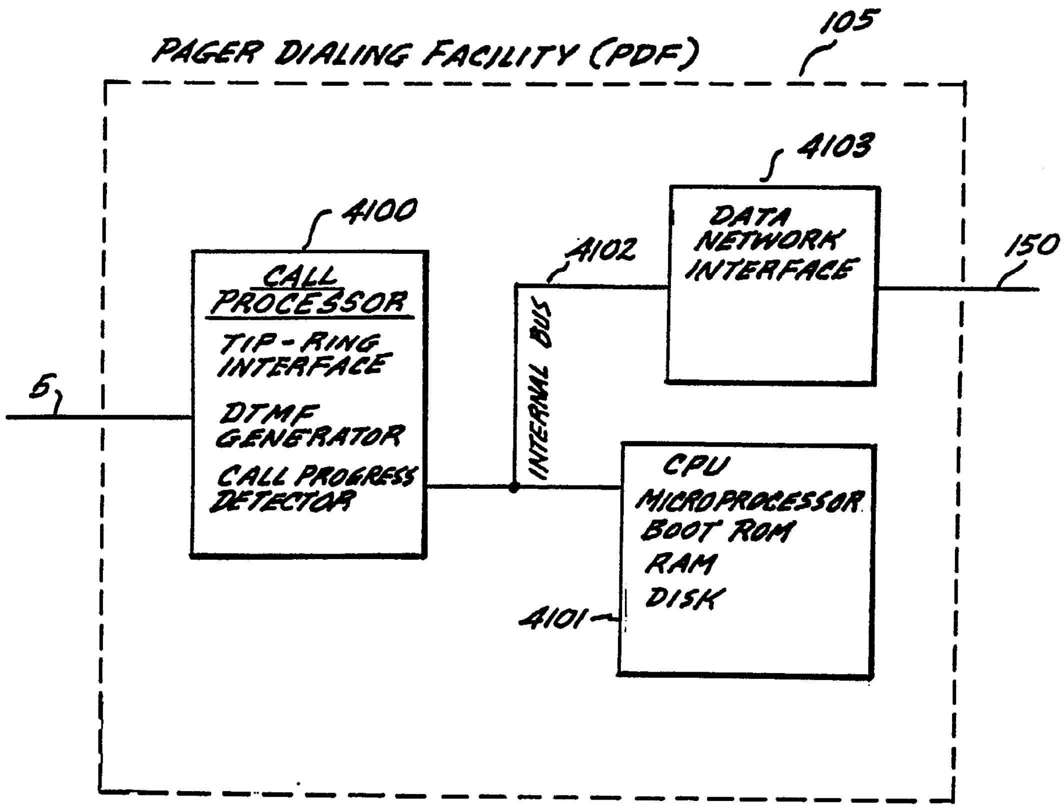
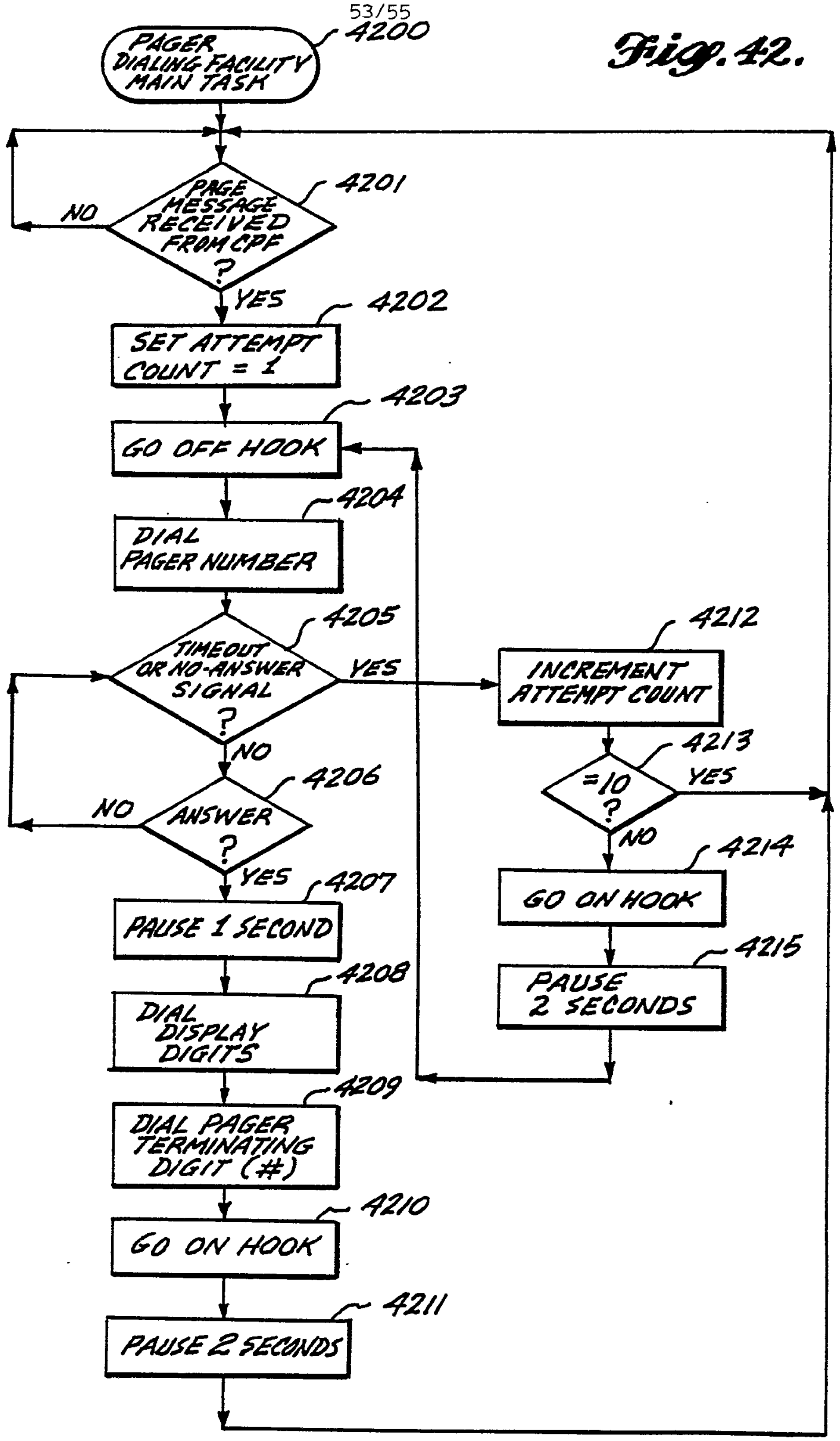

PAGER DIALING FACILITY (PDF)
105
4103
DATA NETWORK INTERFACE
4100
CALL PROCESSOR
TIP-RING INTERFACE
DTMF GENERATOR
CALL PROGRESS DETECTOR
4102
INTERNAL BUS
150
5
CPU
MICROPROCESSOR
BOOT ROM
RAM
DISK
4101

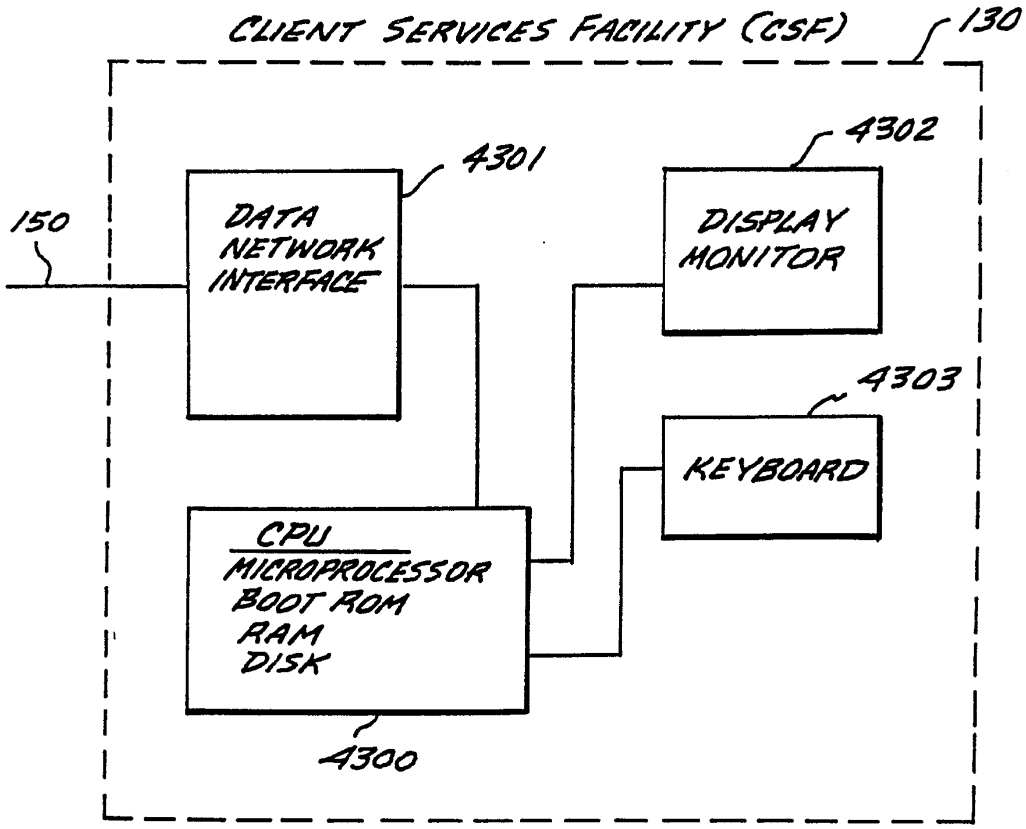
Fig. 41.

Fig. 42.



SUBSTITUTE SHEET

Fig. 43.


CLIENT SERVICES FACILITY (CSF)
130
150
DATA NETWORK INTERFACE
4301
DISPLAY MONITOR
4302
4303
KEYBOARD
CPU
MICROPROCESSOR
BOOT ROM
RAM
DISK
4300

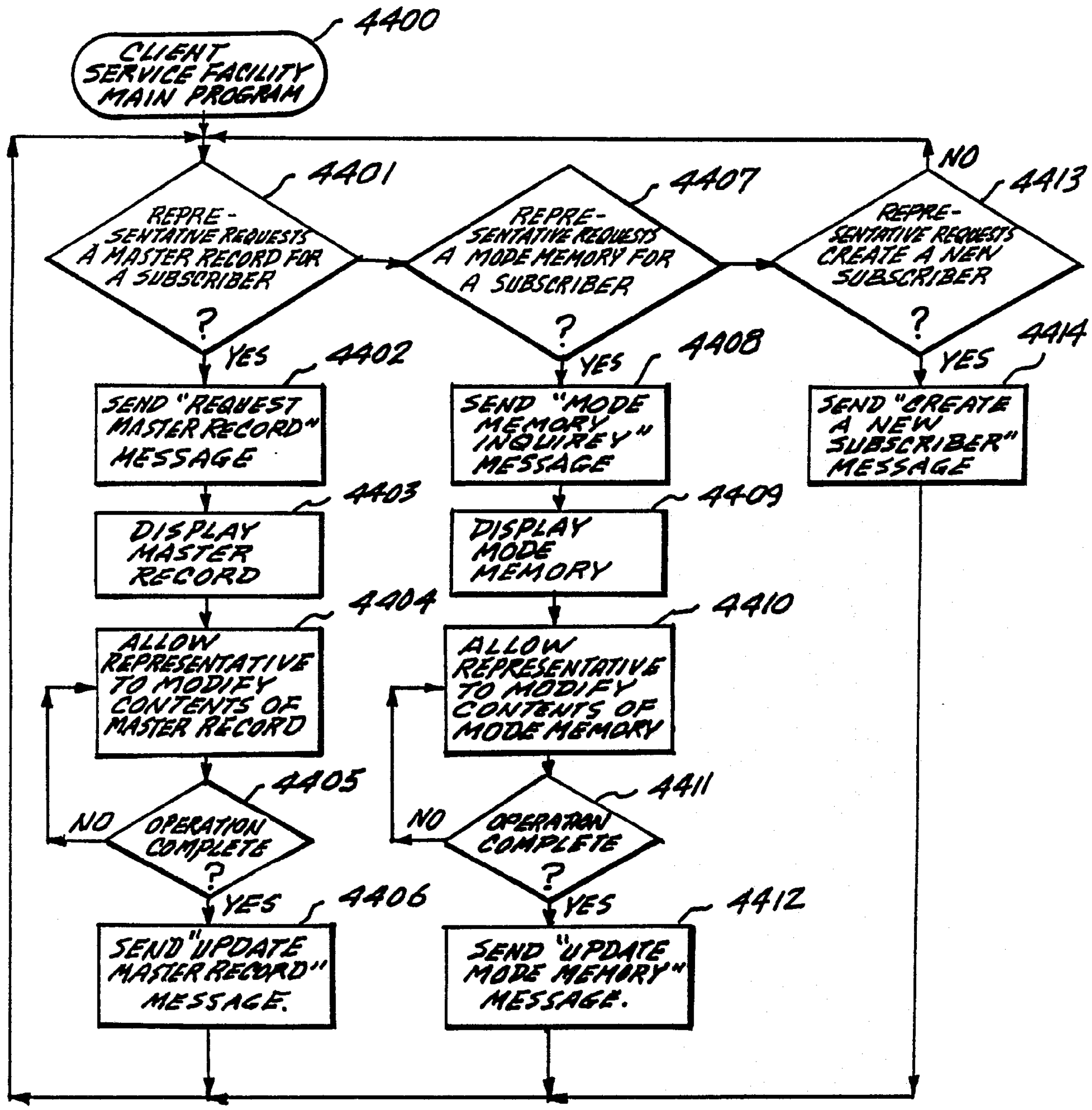

4400
CLIENT SERVICE FACILITY MAIN PROGRAM
4401
REPRE-SENTATIVE REQUESTS A MASTER RECORD FOR A SUBSCRIBER ?
YES
4402
SEND "REQUEST MASTER RECORD" MESSAGE
4403
DISPLAY MASTER RECORD
4404
ALLOW REPRESENTATIVE TO MODIFY CONTENTS OF MASTER RECORD
4405
OPERATION COMPLETE ?
NO
YES
4406
SEND "UPDATE MASTER RECORD" MESSAGE.
4407
REPRE-SENTATIVE REQUESTS A MODE MEMORY FOR A SUBSCRIBER ?
YES
4408
SEND "MODE MEMORY INQUIREY" MESSAGE
4409
DISPLAY MODE MEMORY
4410
ALLOW REPRESENTATIVE TO MODIFY CONTENTS OF MODE MEMORY
4411
OPERATION COMPLETE ?
NO
YES
4412
SEND "UPDATE MODE MEMORY" MESSAGE.
4413
REPRE-SENTATIVE REQUESTS CREATE A NEW SUBSCRIBER ?
NO
YES
4414
SEND "CREATE A NEW SUBSCRIBER" MESSAGE

Fig.44.

INTERNATIONAL SEARCH REPORT

International Application No PCT/US90/06729

I. CLASSIFICATION OF SUBJECT MATTER (if several classification symbols apply, indicate all) [3]

According to International Patent Classification (IPC) or to both National Classification and IPC

IPC(5): H04M 11/00, 1/64, 1/56, 1/66, 3/42

U.S. Cl. 379/57, 88, 142, 199, 211

II. FIELDS SEARCHED

Minimum Documentation Searched [4]

Classification System	Classification Symbols
U.S.	379/ 56, 57, 67, 88, 89, 142, 199, 200, 201, 210-212

Documentation Searched other than Minimum Documentation to the Extent that such Documents are Included in the Fields Searched [5]

III. DOCUMENTS CONSIDERED TO BE RELEVANT [14]

Category *	Citation of Document, [16] with indication, where appropriate, of the relevant passages [17]	Relevant to Claim No. [18]
Y	US, A, 4,028,498 (Mehaffey et al.) 07 June 1977, See abstract; Fig. 1; col.3, lines 13-18, 39-55; col. 11, lines 29-63; col. 13, lines 32-34, 48-51, 58, 59; col. 17, line 44-col. 118, line 12, 23-35.	1-28,31-36, 38-57,65, 71-79
Y	US, A, 4,313,035 (Jordan et al.) 26 January 1982, See col. 2, lines 29-39; col. 3, liens 16-31, 42-44; col.6, lines 25-45, 60-63; col. 7, lines 15-22; col.10, lines 1-10; col. 11, lines 43-46, line 62 through col. 12, line 25, 46-57; col. 13, lines 4-21.	1-28,31-36, 38-48,56,57, 77-79
Y	US, A, 4,625,081 (Lotito et al.) 25 November 1986, See abstract, lines 12-18; col. 3, lines 10-30, line 59 through col. 4, line 24; col. 7, lines 8-24.	2,3,10,24-28, 31036,38-48
Y	US, A, 4,674,115 (Kaleita et al.) 16 June 1987, See abstract; col. 1, lines 1-3, 27-32, 57-59; col. 2, lines 38-40; col. 5, lines 13-19; col 6, lines 43-46; col. 10, lines 46-52.	4-9,11-13, 49-55,71-76

* Special categories of cited documents: [15]

"A" document defining the general state of the art which is not considered to be of particular relevance

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"P" document published prior to the international filing date but later than the priority date claimed

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"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.

"&" document member of the same patent family

IV. CERTIFICATION

Date of the Actual Completion of the International Search [2]	Date of Mailing of this International Search Report [2]
28 FEBRUARY 1991	02 MAY 1991
International Searching Authority [1]	**Signature of Authorized Officer [20]**
ISA/US	Andre Robinson DWAYNE BOST

Form PCT/ISA/210 (second sheet) (May 1986)

International Application No. PCT/US90/06729

FURTHER INFORMATION CONTINUED FROM THE SECOND SHEET

Y,P	US, A, 4,893,329 (O'Brien) 09 January 1990, See abstract.	4-9,11-13
Y,P	US, A, 4,922,490 (Blakely) 01 May 1990, See Fig. 1; col. 5, lines 39-58; col. 6, lines 22-27.	4-9,11-13, 49-55,71-76
Y,P	US, A, 4,941,203 (Patsiokas et al.) 10 June 1990 See abstract.	4-9,11-13, 49-55,71-76
Y	US, A, 4,642,425 (Guinn, Jr. et al.) 10 February 1987, See abstract, col. 1, 2 and col. 3, lines 1-20	14-23,56,57, 77-79

V. ☐ OBSERVATIONS WHERE CERTAIN CLAIMS WERE FOUND UNSEARCHABLE [1]

This international search report has not been established in respect of certain claims under Article 17(2) (a) for the following reasons:

1. ☐ Claim numbers . . . , because they relate to subject matter [1] not required to be searched by this Authority, namely:

2. ☐ Claim numbers ... . .. , because they relate to parts of the international application that do not comply with the prescribed requirements to such an extent that no meaningful international search can be carried out [1], specifically:

3. ☐ Claim numbers__________, because they are dependent claims not drafted in accordance with the second and third sentences of PCT Rule 6.4(a).

VI. ☒ OBSERVATIONS WHERE UNITY OF INVENTION IS LACKING [2]

This International Searching Authority found multiple inventions in this international application as follows:

I. Claims 1-57, 65, 71-79 drawn to a telecommunication control system; class 379 subclass 201.

II. Claims 58-64, 66-70 drawn to a telephone paging system; class 379 subclass 57. SEE ATTACHMENT.

1. ☐ As all required additional search fees were timely paid by the applicant, this international search report covers ... searchable claims of the international application.

2. ☐ As only some of the required additional search fees were timely paid by the applicant, this international ... only those claims of the international application for which fees were paid, specifically claims:

3. ☒ No required additional search fees were timely paid by the applicant. Consequently, this international search ... the invention first mentioned in the claims; it is covered by claim numbers:

1-57, 65, & 71-79

4. ☐ As all searchable claims could be searched without effort justifying an additional fee, the International Searching Auth... invite payment of any additional fee.

Remark on Protest

☐ The additional search fees were accompanied by applicant's protest.

☐ No protest accompanied the payment of additional search fees.

Form PCT/ISA/210 (supplemental sheet (2) (Rev. 4-90)

International Application No. PCT/US90/06729

III. DOCUMENTS CONSIDERED TO BE RELEVANT (CONTINUED FROM THE SECOND SHEET)		
Category *	Citation of Document, [16] with indication, where appropriate, of the relevant passages [17]	Relevant to Claim No [18]
Y	US, A, 4,748,655 (Thrower et al.) 31 May 1988 See abstract; col. 5, lines 23-35.	14-23,77-79
&,P	US, A, 4,893,335 (Fuller et al.) 09 January 1990 See entrire document.	

Form PCT/ISA/210 (extra sheet) (May 1986)

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INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

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524,633 16 May 1990 (16.05.90) US

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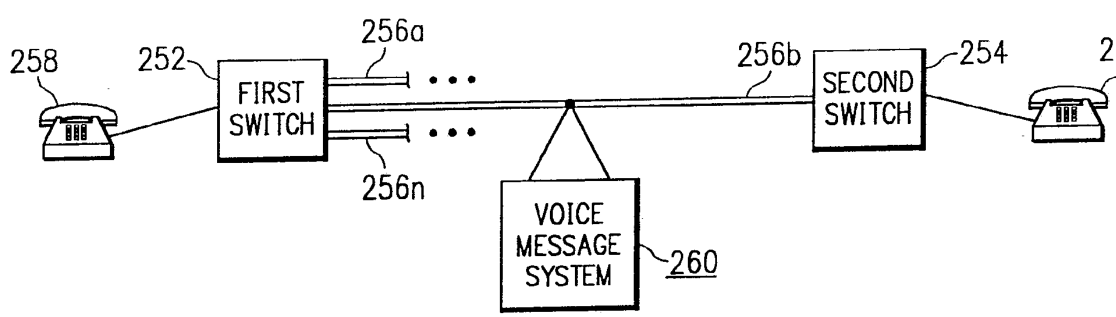
(74) Agent: JUDSON, David, H.; Hughes & Luce, 1717 Main Street, Suite 2800, Dallas, TX 75201 (US).

(81) Designated States: AT (European patent), AU, BE (European patent), CA, CH (European patent), DE (European patent), DK (European patent), ES (European patent), FR (European patent), GB (European patent), GR (European patent), IT (European patent), JP, LU (European patent), NL (European patent), SE (European patent).

Published
With international search report.

(54) Title: METHOD AND APPARATUS FOR PROVIDING PROACTIVE CALL SERVICES FOLLOWING CALL COMPLETION



(57) Abstract

An apparatus for use in a telephone network (250) having a calling station connectable to a first switch (252) means and a called station connectable to a second switch (254) means, with the first and second switch means being connectable by a link (257). The apparatus comprises an interface having a passive in-line monitor connected in the link for detecting entry of a predetermined service access code by a user of the calling station (258) or a user of the called station (259) after call completion between the calling station and the called station and before either of said users goes on-hook. Upon entry of predetermined service access code, a speech circuit issue a predetermined prompt to the user of the calling station and/or the user of the called station. Processor control (266) circuitry is then responsive to entry of predetermined signaling by one of the users following the issuance of the prompt for providing a predetermined service controlled and paid for by the user requesting the predetermined service.

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METHOD AND APPARATUS FOR PROVIDING PROACTIVE CALL SERVICES FOLLOWING CALL COMPLETION

The present invention relates to telephone systems and more particularly to a method and apparatus for providing a variety of caller-controlled proactive services after a caller has been connected to a called station.

It is known in the prior art to carry telephone calls between local telephone operating companies through the AT&T network or through one or more independent inter-exchange carriers such as MCI or Sprint. The local telephone operating companies operate within a so-called local access and transport area (LATA). When a long distance call is dialed, the call is usually transmitted through an operating company central office to a point of termination in the originating LATA at which it is picked up by the inter-exchange carrier and passed by that carrier on to a termination point in a distant LATA. Upon reaching the destination LATA, the call is then transferred by the inter-exchange carrier to the local operating company central office within that LATA for ultimate connection to the original called station therein. Typically, the termination points of each LATA include suitable switching circuits, e.g., an access tandem, that are interconnected by a digital serial link. Such digital links are also presently used to interconnect virtually all central offices as well as to interconnect operating company switching networks to one or more cell site control switches of a mobile telephone network.

It is also known in the prior art to provide "automatic voice messaging" where, upon the occurrence of a busy/ring-no- answer condition at a called station, the user of the calling station can be connected to a voice message facility for recording a voice message for subsequent delivery to the called station. The decision to accept or reject the automatic voice messaging service is determined by the caller. Automatic voice messaging operates essentially after a call has been initiated but before the call can be completed to the called station.

It would be desirable to extend the advantages of caller-controlled automatic voice messaging to facilitate the providing of enhanced proactive services after call completion.

It is an object of the invention to provide caller-controlled proactive telephone services to a caller after call completion.

It is another object to provide a unique system architecture that facilitates the offer/acceptance of various ancillary user services to the original caller at the calling station and/or the original called party at the called station after call completion between the calling station and called station.

It is a further object of the invention to describe an apparatus that passively monitors a line between calling and called stations following call completion, detects a request for an ancillary user service, and then controls the providing of such service at the request of either the calling party or the called party.

It is yet another object of the present invention to provide an apparatus having on-line monitoring capabilities for the selective offering and providing of various ancillary services under the control of, and at the cost to, one of the parties to the completed call.

These and other objects of the invention are achieved in a preferred embodiment of the invention describing an apparatus for use in a telephone network having a calling station connectable to a first switch means and a called station connectable to a second switch means, with the first and second switch means being connectable by a link. Generally, the apparatus comprises an interface having a passive in-line monitor connected in the link for detecting entry of a predetermined service access code by a user of the calling station or a user of the called station after call completion between the calling station and the called station and before either of said users goes on-hook. Upon entry of the predetermined service access code, a speech circuit of the apparatus issues a predetermined prompt to the user of the calling station and/or the user of the called station. Processor control circuitry of the apparatus is then responsive to entry of predetermined signaling by one of the users following the issuance of the prompt for providing a predetermined service controlled and paid for by the user requesting the predetermined service. Alternatively, if the user(s) are aware of the service offering, the speech circuit is not required in order to offer the service. In such alternative embodiments, the speech circuit can be used for a confirmation prompt.

For a more completed understanding of the present invention and the advantages thereof, reference is now made to the following Description taken in conjunction with the accompanying Drawings in which:

FIGURES 1A, 1B and 1C are block diagrams of a digital telephone network in which a proactive call services system is preferably incorporated; and

FIGURE 2 is a detailed block diagram of the preferred embodiment of the proactive call services system of FIGURE 1.

Referring now to FIGURE 1A, a block diagram is shown of a generic digital telephone network 250 in which an automatic voice message system is advantageously provided according to the teachings of copending application Serial No. 07/478,674. The telephone network generally includes a first switch 252, a second switch 254, and a plurality of digital communications links interconnecting the first and second switches 252 and 254 and designated by the reference numerals 256a-n. At least one digital link 256 is preferably a high speed (1.544 MHz) T-1 span over which conventional in-band signaling is provided in a serial fashion; of course, other higher speed links as DS/3 can be used. Link 256, alternatively, is a high speed digital serial link over which digital signals are provided using out-of-band signaling with other communications protocols, such as X.25 or common channel signaling (SS7).

For purposes of generalization, FIGURE 1A shows a calling station 258 connected (or connectable to via a central office or the like) to the first switch 252 and a called station 259 connected (or connectable) to the second switch 254. For the remainder of the discussion, it is assumed that a call to the called station 259 is initiated by a caller at the calling station 258. A proactive call services system 260 is placed across or in a digital link 256 for enabling the offering and acceptance of one or more predetermined call services under the control and at the expense of the caller at the calling station 258 or the caller at the called station.

Without limiting the foregoing, the first and second switches 252 and 254 are access tandems located at termination points between two LATAs.

Alternatively, the first switch 252 is an access tandem and the second switch 254 is a cellular tandem or cell site controller for a mobile telephone system. In this configuration, the network 250 is a cellular telephone network. The first switch 252 may be an operating company central office while the second switch 254 is an access tandem. The first and second switches can interconnect two central offices. The system 260 is bridged across the link.

Referring briefly to FIGURE 1B, in an alternate embodiment the system 260 is placed in a self-contained loop 256C from the first switch 252 (or the second switch 254 (not shown)). A "self-contained" loop means that the system 260 is located internally to the switch or as an adjunct thereto.

Referring now to FIGURE 2, a detailed block diagram is shown of the preferred embodiment of a proactive call services 260 for use in a digital network environment. System 260 preferably includes an interface means including a plurality of interface circuits 262a-n each connected to a multiplexer 264. The multiplexer includes a control bus connected to a control means comprising a processor 266, storage interface 268, storage device 270 and input/output device 272. The processor is controlled in a conventional manner by suitable application programs stored in the storage device 270. Input/output device is used to modify the system operation by entering suitable program commands to the control means.

The system 260 further includes a number of circuits for facilitating various monitor intercept, prompting, conferencing and redirect functions as will be described in more detail below. A scanner circuit 274 is provided to identify Feature Group D supervision or other similar information depending on the type of signaling used. A service circuit 276 includes a passive line monitor circuit as well as all necessary call progress (e.g., busy/ring-no-answer) detection circuits, speech generation circuits, and service acceptance (e.g., DTMF) detection circuits. The service circuit 276 also preferably includes appropriate circuitry for capturing ANI, DNI and billing information. A first communication interface 278 is provided to receive, interpret, format and transmit SS7 messages as will be described in more detail below. The system preferably also includes a second communication interface 280 connected to a network applications platform 282. The platform 282 includes a billing computer and other appropriate devices such as a database for transaction processing and accounting purposes. The system 260 advantageously includes its own voice/data storage unit 284 for storing voice messages, data or other call conversations as will be described. A storage 286, preferably a disk storage, is connected to the voice/ data storage unit 284. The voice message or data storage may take place either in the voice/data storage unit or in the network applications platform, as is appropriate, to facilitate subsequent delivery.

Each of the circuits 274, 276, 278, 280 and 284 are connected to the processor 266 via the control bus 290. Input/output channel buses 292 and 294 also interconnect these circuits to the multiplexer of the interface means. Preferably, the interface means includes twenty-eight (28) T-1 interface circuits, each of which is connected to two digital links. Each T-1 interface circuit includes first and second T-1 interface circuits 295 and 297, with the first interface circuit 295 connected to one of the digital links and the second interface circuit 297 connected to the other digital link. The first and second digital links are thus connectable to bypass their respective interface circuit if the first and second T-1 interfaces are interconnected by means of the switch 298. Given this architecture, i.e., with 28 T-1 spans each carrying twenty-four (24) channels, the buses 292 and 294 service 1344 channels.

To provide automatic voice messaging, the scanner 274, service circuit 276 and/or communications interface 278 monitor the received signaling to determine the state of the call progress. Of course, the actual circuit used depends on the type of signaling. If an SS7 protocol is used, communications interface 278 monitors the line. If Feature Group D signaling is used, the Feature Group D information is collected in and processed by service circuit 276; other types of signaling are collected and processed by scanner 274. When call processing is required due to a busy or ring/no answer condition, the processor 266 activates the service circuits 276 to thereby issue a prompt offering (e.g., "Your party is not available; if you would like to leave a message, please press the # key") and monitor the line for acceptance of the service. Processor 266 also controls the circuit 276 to capture ANI, DNI and billing information. If the service is accepted, the service circuit 276 notifies the processor 266, which then controls the service circuit to issue appropriate prompts to the caller to instruct the caller to begin recording the message. The message is then recorded by the voice storage unit 284, and the processor controls the service circuit 276 to transfer the ANI, DNI, and billing information to the network applications platform 282. Although not meant to be limiting, preferably voice messages are stored in the voice/data storage unit 284 or its associated disk storage, while ANI, billing and other management information resides in the network applications platform.

For message delivery, the network applications platform ships the message routing information (i.e., the ANI, etc.) back to the voice/data storage unit 284, and the processor 266 then locates an open channel on a link for outdialing to the original called station. The service circuit then dials the call. When the call is placed, the scanner 274 watches the call states for answer supervision or on-hook/off-hook detection. If off-hook is detected, the service circuit 276 issues a prompt announcing the message which is then delivered by the voice storage unit 284. When the message is delivered, the processor 266 notifies the network applications platform and the packet is deleted.

If desired, the system 260 is connectable to a remote host computer via a dedicated communications interface which in turn is connected to the remote host via an RS-232 link or the like. This enables messages to be transferred to another location for the subsequent outdial attempts. As an alternate embodiment, the network applications platform 282 is set up to control billing and delivery attempts while the remote host issues the prompt announcing the message and other voice functions. The remote host can alternatively retain all billing information with the voice messager or other call information for a short time; and then passes off all such information to the platform 282 for further processing. It is also possible to have the voice/data storage unit 284 pass a "packet" (comprising the billing information) downstream to a platform 282, e.g., in a destination delivery area (a destination LATA).

The architecture of FIGURE 2 is quite useful in providing various ancillary "instant" services to the calling station user and/or the called station user after the call between these parties has been completed. As used herein, "call completion" means the calling station has been successfully connected to the called station, i.e., a busy/ring-no-answer has not been encountered by the calling station user upon call initiation to the called station. According to the present invention, one or more such ancillary services are provided after call completion but before either party goes back to an on-hook condition.

One such ancillary service is conference calling. During the call, if either party determines that a conference is needed, that party can enter a predetermined access code (e.g., "2,2") which is detected by the passive on-line monitor circuit in the service circuit 276 of the system. Detection of the predetermined access code causes the processor 266 to control the service circuit to issue a prompt, e.g., "if you would like to add another party to this call, please dial that party's number now" or "if you would like to add a party to this call, please press # and follow the instructions" or the like. Depending on the prompt offering, one of the parties then enters the necessary signaling (dialed digits or the # sign, etc.) to accept the service offering. This signaling is detected by the service circuit 276 at which point the processor seizes an available line and places the call to the party sought to be conferenced. Such call initiation continues while the originally completed call remains in progress between the calling station and the called station. The system can either mute the ringing signals to the third party or allow such signals to be heard by the parties. Upon call completion to the third party, the processor 266 bridges the calls together to provide the conference. The database in the network applications platform 282 could then provide billing validity information as well as storing the billing information. Generally, the party who accepts the service is billed.

The system of FIGURE 2 thus facilitates call conferencing in a proactive or caller-controlled manner which has heretofore never been available. From the caller's perspective, such ancillary services are provided to the call-in-process unobtrusively. If the parties are familiar with the service offering, a prompt offering may not be necessary following entry of the predetermined access code. In this embodiment, detection of the predetermined access code by the monitor circuit will result in the generation by the service circuit of a "beep" tone or superimposed dialtone (over the existing talkpath) instead of the prompt offering. One of the parties then dials the third party's telephone number or enters the appropriate code for acceptance of the service as the case may be. Confirmation of follow-up prompts may then be provided if necessary.

Yet another ancillary service available through the system of FIGURE 2 is call recording. During the call, one or both of the parties may decide to record the call. Upon detection of a predetermined access code (e.g., a 2,7) by the on-line monitor of the service circuit 276, the circuit signals the processor 266. Processor 266 in turn connects the voice storage unit on line to immediately begin recording the call. Alternatively, the parties are provided the prompt offering to determine whether the continued conversations are to be recorded. The recording of the call, and the associated information packet containing the requesting party and billing information, is then transferred to the network applications platform 282 after the pathway to the platform is established as previously described. The party requesting the recording can then recall the recorded conversation from the platform 282 using dialup parameters such an entry and verification of security codes. Other collateral services, such as transcription of the recorded call, can thus be arranged and billed to the requesting party.

Another ancillary service is silent call recording. In this embodiment, a predetermined service access code is detected by the passive on-line monitor but not sent down the line to the party at the called station. The recording of the call is then carried out privately without the other party's knowledge. With this service, there is no need to provide a prompt offering, however, a one-way confirmation (to either the calling party or the called party) superimposed over a muted line (to the other party) is provided if desired. Although not meant to be limiting, the silent recording feature alternatively can be invoked by using a second band of an ISDN telephone, by transmitting out-of-band information, or by transmitting a combination of in-band and out-of-band signaling to a special "notch" filter.

According to the present invention, either the calling party and/or the called party can invoke one or more of the above-identified services by entry of the appropriate code. Service may be offered at the destination either by subscription or as a basic service by the destination service provider. Generalizing, and with reference to FIGURE 1C, conceptually the service offerings can be provided in numerous locations in and around the network. In particular, FIGURE 1C shows a public telephone network with alternative pickup points, alternative central office switching points and alternate delivery processors. The system of FIGURE 2 can be implemented at the various positions indicated in FIGURE 1C.

Although not shown in detail, it should be appreciated that the architecture of FIGURE 2 is quite powerful and enables the service provider to provide numerous ancillary call services that have heretofore been unavailable to users except in only limited ways behind a private branch exchange or the like and without the capability of providing such services in a proactive, caller-controlled and billed manner as described herein.

It should be appreciated by those skilled in the art that the specific embodiments disclosed above may be readily utilized as a basis for modifying or designed other structures for carrying out the same purposes of the present invention. It should also be realized by those skilled in the art that such equivalent constructions do not depart from the spirit and scope of the invention as set forth in the appended claims.

CLAIMS

What is claimed is:

1. In a telephone network having a calling station connectable to a first switch means and a called station connectable to a second switch means, with the first and second switch means being connectable by a link, the improvement comprising:

an interface connected in the link to monitor for entry of a predetermined service access code by a user of the calling station or a user of the called station after call completion between the calling station and the called station and before either of said users goes on-hook; and

control means connected to the interface and responsive to entry of the predetermined access code for controlling offer and acceptance of one or more predetermined services controlled and paid for by the user requesting the service.

2. In the telephone network as described in Claim 1 wherein the service is call conferencing.

3. In the telephone network as described in Claim 1 wherein the service is call recording.

4. In the telephone network as described in Claim 1 wherein the service is silent call recording.

5. Apparatus for use in a telephone network having a calling station connectable to a first switch means and a called station connectable to a second switch means, with the first and second switch means being connectable by a link, comprising:

an interface connected in the link to monitor for entry of a predetermined service access code by a user of the calling station or a user of the called station after call completion between the calling station and the called station and before either of said users goes on-hook; and

speech means responsive to entry of the predetermined service access code for issuing a predetermined prompt to the user of the calling station and/or the user of the called station; and

control means connected to the interface and responsive to entry of predetermined signaling by one of the users following the issuance of the prompt for providing a predetermined service controlled and paid for by the user requesting the predetermined service.

6. Apparatus for use in a telephone network having a calling station connectable to a first switch means and a called station connectable to a second switch means, with the first and second switch means being connectable by a link, comprising:

an interface connected in the link to monitor for entry of a predetermined service access code by a user of the calling station or a user of the called station after call completion between the calling station and the called station and before either of said users goes on-hook; and

control means connected to the interface and responsive to entry of predetermined signaling by one of the users for providing a predetermined service controlled and paid for by the user requesting the predetermined service.

7. Apparatus for use in a telephone network having a calling station connectable to a first switch means and a called station connectable to a second switch means, with the first and second switch means being connectable by a link, comprising:

an interface in the first switch means to monitor for entry of a predetermined service access code by a user of the calling station or a user of the called station after call completion between the calling station and the called station and before either of said users goes on-hook; and

control means connected to the interface and responsive to entry of predetermined signaling by one of the users for providing a predetermined service controlled and paid for by the user requesting the predetermined service.

1/2

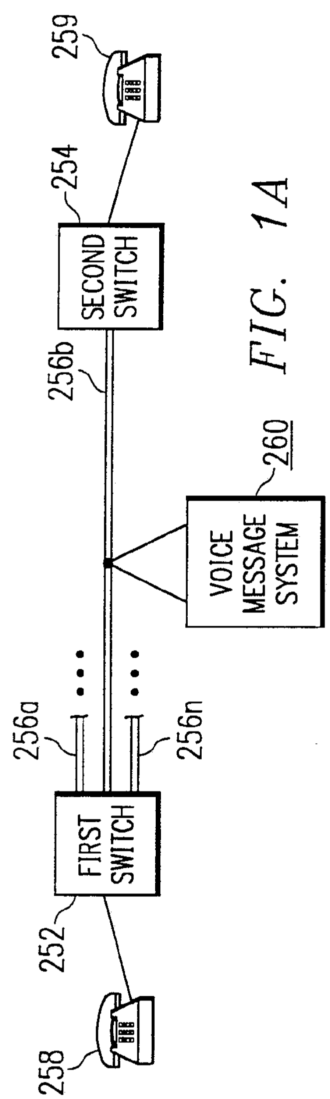


FIG. 1A

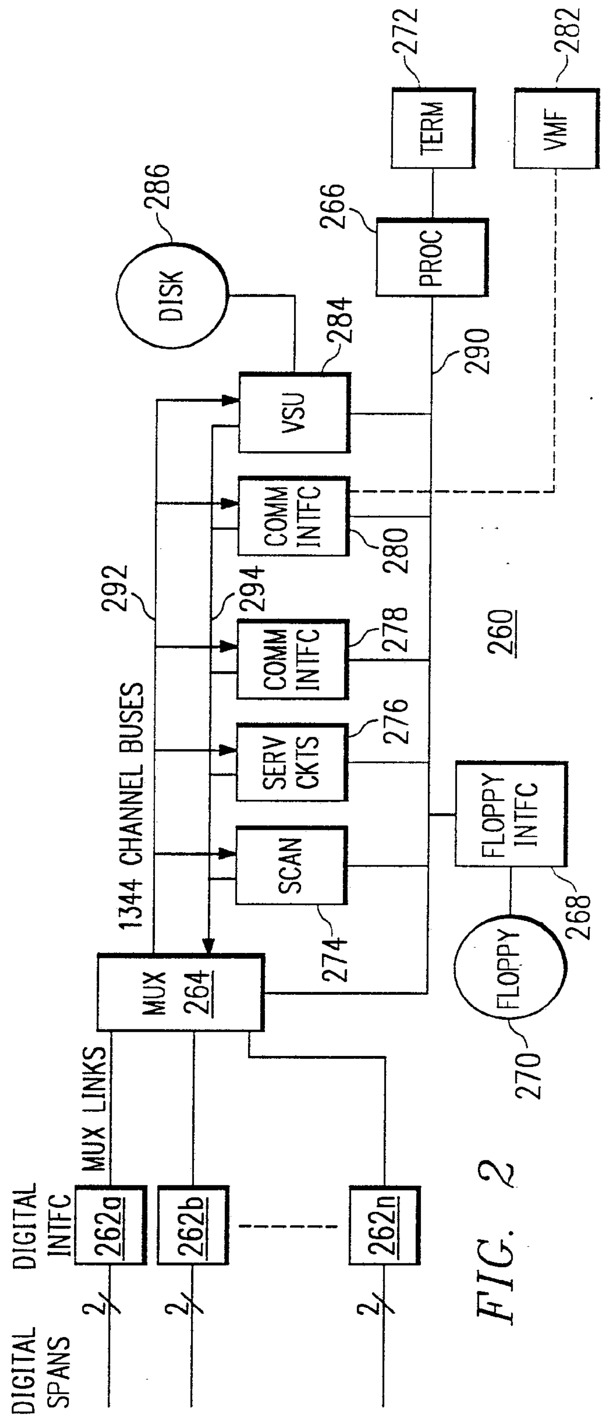


FIG. 2

SUBSTITUTE SHEET

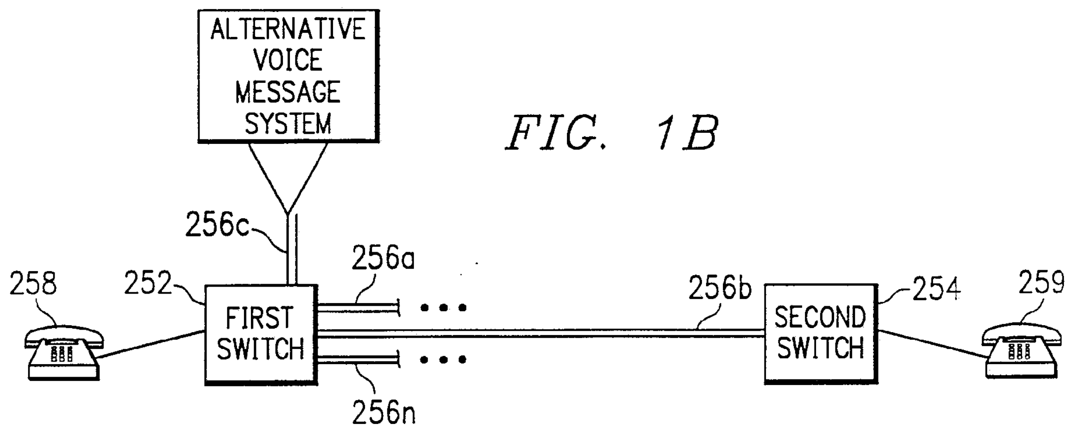
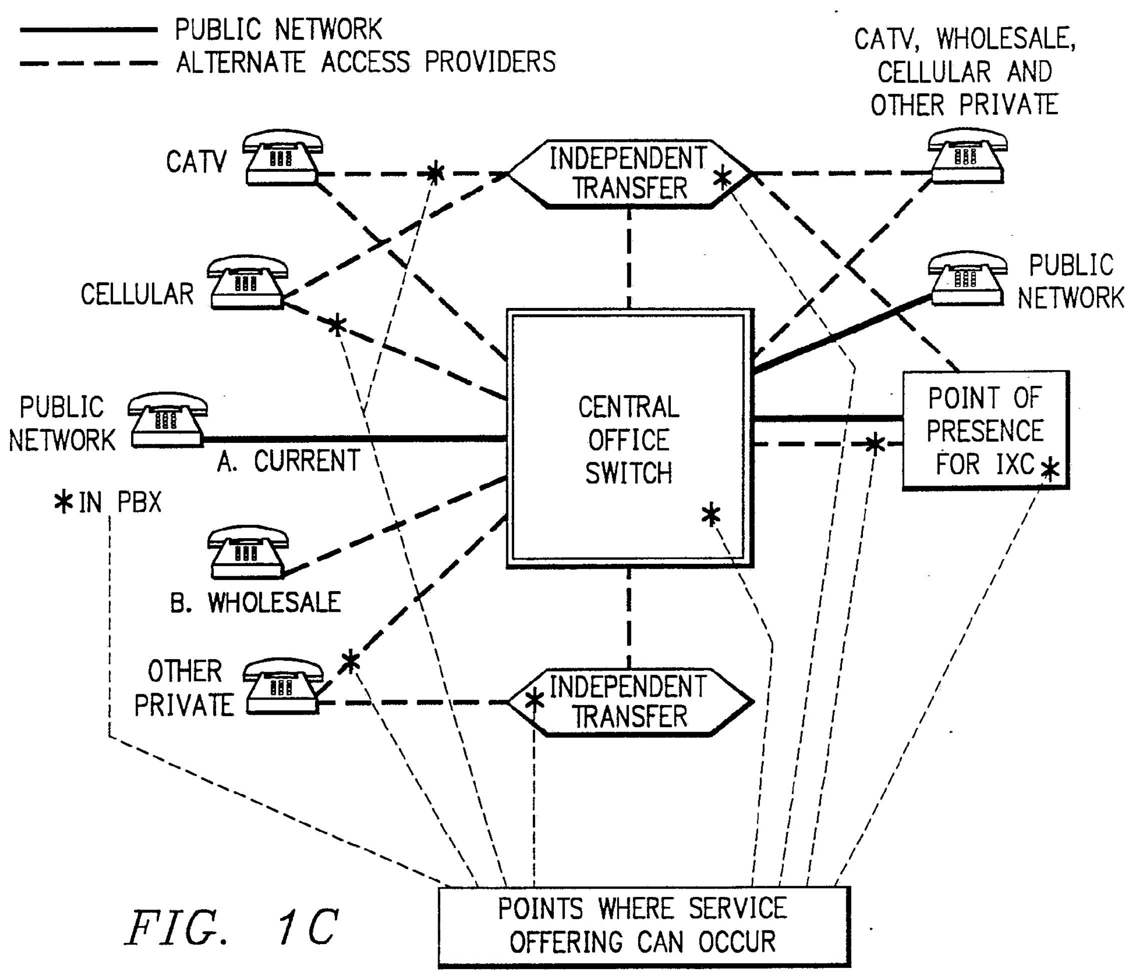

ALTERNATIVE VOICE MESSAGE SYSTEM
256c
256a
258
252
FIRST SWITCH
256b
SECOND SWITCH
254
259
256n

FIG. 1B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ALTERNATE ACCESS PROVIDERS
CATV, WHOLESALE, CELLULAR AND OTHER PRIVATE
CATV
INDEPENDENT TRANSFER
CELLULAR
PUBLIC NETWORK
PUBLIC NETWORK
A. CURRENT
CENTRAL OFFICE SWITCH
POINT OF PRESENCE FOR IXC
*IN PBX
B. WHOLESALE
OTHER PRIVATE
INDEPENDENT TRANSFER
POINTS WHERE SERVICE OFFERING CAN OCCUR

FIG. 1C

INTERNATIONAL SEARCH REPORT

International Application No. PCT/US91/03469

I. CLASSIFICATION OF SUBJECT MATTER (if several classification symbols apply, indicate all) [6]

According to International Patent Classification (IPC) or to both National Classification and IPC
US 379/67
IPC(5) H04M 3/50

II. FIELDS SEARCHED

Minimum Documentation Searched [7]

Classification System	Classification Symbols
U.S.	379/67,84,88,89,97,201,204,205,207,210,211,212,213,214

Documentation Searched other than Minimum Documentation to the Extent that such Documents are Included in the Fields Searched [8]

III. DOCUMENTS CONSIDERED TO BE RELEVANT [9]

Category *	Citation of Document, [11] with indication, where appropriate, of the relevant passages [12]	Relevant to Claim No. [13]
X,P	US, a, ,4959,855 (DAUDELIN) 25 SEPTEMBER 1990 See column 2 lines 60-68, column 3 lines 1-19, column 4 lines 22-65, column 5 lines 38-47, column 8 lines 54-57	1-7

* Special categories of cited documents: [10]

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IV. CERTIFICATION

Date of the Actual Completion of the International Search	Date of Mailing of this International Search Report
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International Searching Authority	**Signature of Authorized Officer**
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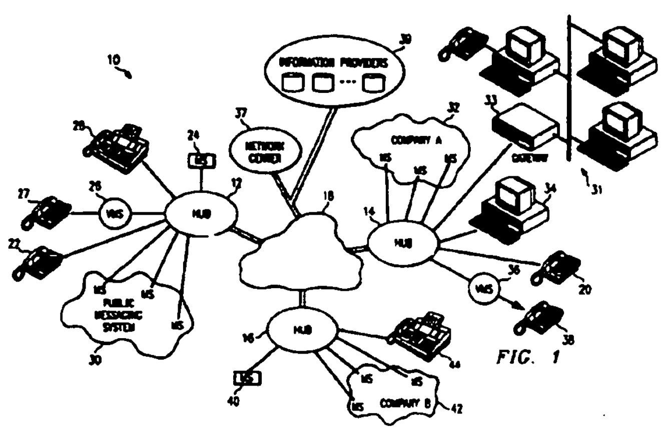

FIG. 1

(57) Abstract

A communications system (10) is provided which comprises a plurality of network hubs (12, 14 and 16). Network hubs (12, 14 and 16) are interconnected through a communications network (18). The system (10) interconnects messaging systems (24, 26, 28, 30, 32, 34, 36, 40, 42 and 44) having disparate capabilities and using disparate communications protocols. The network hubs use numbers of connection processors (52 and 54) to interact with the messaging systems. A hub database (68) and message store (58) are used to store control information and messaging information within the network hubs. A network processor (60) is used to interact with other hubs within the communications system (10). A message router (72), connection manager (74), data replicator (76), and an administrative event manager (78) are used to control the operations of the hub in processing a message.

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NETWORK-BASED MULTIMEDIA COMMUNICATIONS AND DIRECTORY SYSTEM AND METHOD OF OPERATION

TECHNICAL FIELD OF THE INVENTION

This invention relates in general to the field of communications and information management systems and more particularly to an improved network-based voice messaging and multimedia communications and directory system and method of operation for private addressing plans using community addressing.

BACKGROUND OF THE INVENTION

Currently available communications facilities include voice communication, electronic mail communication, facsimile communication and video communication. These communications facilities are augmented by storage and retrieval facilities such as voice mail facilities, bulletin board services and the like. These various communications facilities have largely been operated on independent platforms, interconnected into private networks, and through independent and disparate channels of communication.

While local area network (LAN) based mail systems such as cc: Mail or large private electronic mail providers such as MCI Mail have facilitated some networking capability in electronic mail content, other communications facilities such as voice messaging and facsimile transmissions are largely localized facilities. For example, typical messaging systems are constrained within a single organization such as a company or at the broadest within a single local exchange carrier facility. In light of the largely local nature of messaging facilities and the incompatibility of proprietary messaging protocols, there has been little effort to supply large scale integrated network functionality to these communications services. In addition, most of these facilities are limited to a single media such as only voice, only electronic mail, or only facsimile transmissions.

Additionally, in particular, voice messaging systems have not provided large scale integrated network functionality due to the following limitations:

1) Their terminal equipment is usually a telephone, which can only communicate with audio signaling such as Dual Tone Multi-Frequency (DTMF) signals.

2) The methods of addressing are frequently short, fixed length numerical addresses and currently deployed numbering schemes.

3) Messages are typically large, spanning several minutes of digitized analog audio signals.

4) Identity confirmation of the sender or recipient must be a spoken identification such as a mailbox number or a name.

5) Directory type functions such as lookup can not be done with ASCII type inputs but again are restricted to DTMF inputs.

6) Communications protocols associated with voice messaging systems do not provide the facilities necessary to request or specify special services such as media translation, subject matter identification and routing, and the like.

A further complication in the growth of existing messaging systems and networks is the parallel increase in the complexity of managing the directory and addressing information associated with the network. Existing directory facilities are usually limited to a single system or, at most, a single organization. It is difficult, if not impossible with current systems, to acquire and use effectively directory information from other facilities as the integrated system increases in complexity as other facilities are added to the network. These large scale directories are more complicated to deal with in voice messaging systems due to the fact that any functionality, such as retrieval or lookup, provided to the user is restricted to DTMF inputs.

The isolated nature of present messaging systems provides for little standardization that may be used to effect the communications between disparate systems that must occur for effective networking of systems. As such, even messaging systems that are working in the same media, for example, two voice messaging systems, may be incapable of transferring information and messages between the systems due to the differences in the protocols used by the systems to process and transfer messages.

The management of message traffic in a networked environment creates additional concerns. As a message passes out of the control of a local messaging system and into the network, the responsibility for routing and delivery of the message must also pass to the network. This responsibility creates a need for a network with significant message tracking and management capabilities. The complexity of this management task grows enormously as the size of the network increases. This complexity further increases with voice messaging systems due to the addressing being numerical, and limited in size most often to the sender/recipient's phone number or some other local private numbering plan, and to the size of the addressing fields in any of the local networking protocols.

Further complications result from the desire for parties owning individual messaging systems to network those systems. Current messaging systems can employ convenient short form addressing among the members of the entity that use the voice messaging system. However, the short form addresses are problematic in a large shared network such as that in this invention because they do not provide globally unique addressing for all members of the widely-networked community.

Accordingly, a need has arisen for an integrated communications system which supplies network-based voice and multimedia communication facilities, voice and multimedia directory services for communications including voice mail, electronic mail, facsimile transmissions, voice transmissions and video transmissions, and further supplies the ability to network messaging systems which use a variety of disparate private addressing schemes.

SUMMARY OF THE INVENTION

In accordance with the teachings of the present invention, a communications system is disclosed which substantially eliminates or reduces disadvantages associated with prior systems and solutions and provides new solutions never before available. In accordance with one embodiment of the present invention, a network-based voice and multimedia communications system is provided that comprises a plurality of network hubs operable to communicate with each other and with local messaging systems. The network hubs operate to receive messages in any media using any addressing convention from users either directly or through local messaging systems and to transmit the messages to other users whether located on local messaging systems or otherwise through the plurality of network hubs. This capability introduces a further complication for voice messaging systems that has yet to be encountered, in that voice messaging systems must be accessed and addressed utilizing standard DTMF signaling, numerical addressing and directory queries, and all within the native protocol of the user messaging system. According to one embodiment of the present invention, the network hubs are in constant communication with one another and constantly update a directory database which comprises user profiles for users of the communications system. The user profiles include identification confirmations associated with the users, such as a user's spoken name, as well as other user-specific information. The communications system is operable to transmit an identification confirmation for a given user when another user of the communications system attempts to send a message to that user. For large scale integrated network functionality, interfacing with voice messaging systems introduces a further complication for user profiles including identification confirmation in that large distributed network directories must be built and maintained based upon numerical addressing and accessed utilizing DTMF signaling and the native protocols of the user messaging system.

According to a further embodiment of the present invention, the network hubs are operable to maintain and store user profiles that include user preferences associated with each user. The communications system of the present invention is operable to perform media translations such that messages received in one media can be translated into another media based upon the preference of the destination user as stored in the user's profile information. For large scale integrated network functionality, interfacing with voice messaging systems introduces a further complication for media preference translation in that large distributed network directories must be built and maintained, based upon numerical addressing, that are accessed utilizing DTMF signaling and the native protocols of the user system.

According to a further embodiment of the present invention, the communications system supports mailing list distribution of messages. According to this embodiment of the present invention, the communications system receives a message which is addressed to a configurable mailing list comprising a plurality of destination addresses associated with users of the communications system. A mailing list agent may act as the sender of the message to distribute the message to each of the addresses in the mailing list. For large scale integrated network functionality, interfacing with voice messaging systems introduces a further complication for mailing list distribution, in that large distributed network directories must be built and maintained. These directories must be accessed and addressed utilizing standard DTMF signaling, numerical addressing and directory queries using the native protocol of the user system.

According to one embodiment of the present invention, the network hubs maintain constant communication paths to one another such that the current state of the entire system is always available to each of the network hubs. In this manner, the location and status of each message within the communications system can be tracked and altered at any time after the message is created and before the message is delivered. For large scale integrated network functionality, interfacing with voice messaging systems introduces a further complication for accessing and update of user profile information, in that large distributed network directories must be built and accessed by the numerical addresses of the users of the messaging system.

According to a further embodiment of the present invention, access is provided into the communications system for nonsubscribers of the communications system. In this manner, nonsubscribers can place messages or other communications directly onto the communications system for delivery. Further, facilities are also provided to deliver messages to nonsubscribers by the placement of conventional telephone calls, facsimile transmissions or E-mail transmissions to addresses which are provided by the message sender. A further complication to the communications system for providing nonsubscribers the capability to deposit and retrieve messages with voice messaging systems requires that the communications system support standard telephone interfaces, support numerical addressing, support only voice as the entry and exit media, supply identification confirmation in a spoken form such as the spoken name or spoken address, and provide directory services via DTMF signaling.

According to a further embodiment of the present invention, the network hubs may be used to distribute mass mailings of message traffic to selected users of the communications system. According to this embodiment, the user profiles associated with each user may contain information to allow each user to specify what types of information and from whom that user would like to receive and in what media and location the user would like to receive the information. For large scale integrated network functionality, interfacing with voice messaging systems introduces a further complication for distribution of mass mailings in that large distributed network directories must be built and accessed by the numerical addresses of the users of the communications system. According to one embodiment of the present invention, advertising and other commercial special interest information can be routed through the communications system for fees charged to the providers of the information. Users of the network are billed for ordinary messages routed through the network or for information services and may be given credit for each advertising message actually received to provide an incentive for users to receive the advertising message traffic. For large scale integrated network functionality, interfacing with voice messaging systems introduces a further complication for providing advertising and other commercial special interest information messaging services in that large distributed network directories must be built and maintained, based upon numerical addressing and that the messages are delivered utilizing DTMF signaling and the native protocols of the user's messaging system.

According to a further embodiment of the present invention, the communications system is operable to deliver compound messages that comprise information in more than one media. For example, a single message may comprise both a voice message and a facsimile transmission. The communications system of the present invention is operable to route each portion of the message to an appropriate receiving facility associated with the destination user. For example, if a compound message is sent that comprises both a voice message and a facsimile message, the communications system of the present invention can deliver both messages together or split the two messages and deliver the voice message to a voice mailbox associated with the intended recipient and deliver the facsimile message to a fax destination associated with the intended recipient. The communications system of the present invention is also operable to translate messages received in one media to a media associated with the facilities of the destination user if the destination user does not have or does not desire delivery in certain media. For example, if a compound message is sent that comprises a voice message and an electronic mail transmission but the destination user only has a voice mailbox and a fax machine, the communications system of the present invention will translate the electronic mail message to a facsimile image and transmit the voice portion of the compound message to the voice mailbox and the translated portion to the fax destination. For large scale integrated network functionality, interfacing with voice messaging systems introduces a further complication for the processing of multimedia messages and the split routing that has yet to be developed, in that large distributed network directories must be built containing the numerical addresses for the different media destinations and accessed by the numerical addresses of the users of the communications system, and delivered utilizing DTMF signaling and the native protocols of the user system. Current technology and implementations of message addressing using short, fixed length numerical addresses are inadequate when trying to implement global addressing. The communications system of the present invention must supply additional information and mechanisms to its addressing schemes to create globally unique addressing from such short, fixed length numerical addresses.

According to a further embodiment of the present invention, the communications system is operable to translate messages between different languages to accommodate foreign-speaking users. If the message is not already a text message, the communications system of the present invention will use its media translating capability to place a message in a textual format such that an automated language translation of the text message can take place. The language-translated message can then be translated into the appropriate media for the destination user. For large scale integrated network functionality, interfacing with voice messaging systems introduces further complications for providing language translation, in that large distributed network directories must be built and maintained containing language preference information which must be accessed based upon numerical addressing and that the messages are delivered utilizing DTMF signaling and the native protocols of the user system.

According to a further embodiment of the present invention, the communications system is operable to coordinate password based, public and private key or other forms of messaging security to allow for secure transmission and receipt of messages through the communications system. For large scale integrated network functionality, interfacing with voice messaging systems introduces further complications for providing security features, in that large distributed network directories must be built and maintained containing the security access keys and methods which must be accessed based upon numerical addressing and that the messages and security keys are delivered utilizing DTMF signaling and 'he native protocols of the user system.

According to a further embodiment of the present invention, the communications system is operable to receive an address or other information associated with an intended recipient and to access user profile information associated with the intended recipient and retrieve addressing information associated with the intended recipient. As such, a sender of a message can use whatever information is known about an intended recipient of a message to identify the recipient and the communications system of the present invention will access and use the required addressing information appropriate for the identified recipient and the message being sent. For large scale integrated network functionality, interfacing with voice messaging systems introduces further complications for providing custom address translation, in that large distributed network directories must be built and maintained containing numerical address translation information which must be accessed based upon numerical addressing and that the messages are delivered utilizing DTMF signaling and the native protocols of the user system.

According to a further embodiment of the present invention, the communications system is operable to store configuration and directory information for messaging systems that are connected to network hubs used in the communications system of the present invention. Accordingly, if a messaging system is damaged or destroyed, the stored information can be used to reconfigure the repaired system or to initially configure a replacement system. Further, messages to be delivered to the damaged messaging system may be stored in the network hubs of the communications system of the present invention until the damaged messaging system is returned to service so that no messaging traffic is lost during the time required to repair or replace the damaged messaging system.

According to a further embodiment of the present invention, the communications system is operable to provide directory information to messaging systems that are connected to network hubs of the communications system of the present invention but which do not need to use the communications system of the present invention to actually deliver messages. According to this embodiment, a messaging system can present the communications system of the present invention with a request for directory information and the communications system will respond with routing information to enable the attached messaging system to deliver the message without the further intervention of the communications system of the present invention. For large scale integrated network functionality, interfacing with voice messaging systems introduces further complications for providing directory addressing services, in that large distributed network directories must be built and maintained containing address and delivery information which must be accessed based upon numerical addressing and that this information is delivered to the requesting system utilizing DTMF signaling and the native protocols of the user system.

In accordance with one embodiment of the present invention, a communications system is provided that includes a network hub that is coupled to a messaging system through at least two communication paths. One of the communication paths comprises a public communication path for public messaging traffic to and from the messaging system. The second communication path comprises a private communication path for messaging traffic within a messaging community. The network hub is operable to treat the single messaging system as two virtual messaging systems with each communication path associated with a single virtual messaging system. The network hub comprises translation tables which associate users of the messaging systems with the communities to which they belong and the addresses associated with the users within those communities.

BRIEF DESCRIPTION OF THE DRAWINGS

A more complete understanding of teachings of the present invention may be acquired by referring to the accompanying drawings in which like reference numbers indicate like features and wherein:

FIGURE 1 is a block diagram of the multimedia communications system of the present invention;

FIGURE 2 is a block diagram of the modular software architecture which is used in the network hubs of the present invention;

FIGURE 3 is a data flow diagram illustrating the flow of messages and control information between the software modules used in the network hubs of the present invention;

FIGURE 4 is a block diagram of an analog connection processor used in the communications system of the present invention;

FIGURE 5 is a state diagram of the analog connection processor, the digital connection processor and the network processor used in the communications system of the present invention;

FIGURE 6 is a block diagram of the digital connection processor used in the communications system of the present invention;

FIGURE 7 is a block diagram of the network processor used in the communications system of the present invention;

FIGURE 8 is a block diagram of the event processor used in the communications system of the present invention;

FIGURE 9 is a state diagram of the event processor used in the communications system of the present invention;

FIGURE 10 is a block diagram of the control processors, management processor, event processor, and databases used in the communications system of the present invention;

FIGURE 11 is a block diagram of the management processor used in the communications system of the present invention;

FIGURE 12 is a data flow diagram of the media translator used in the communications system of the present invention;

FIGURE 13 is a block diagram of the network center used in the communications system of the present invention;

FIGURE 14 is an example of the address translation operation of the communications system of the present invention;

FIGURE 15 is a schematic illustration of the system and method used by the present invention to implement private addressing plans;

FIGURE 16 is an illustration of a translation table used by the present invention;

FIGUREs 17a through 17e are examples of the translations performed to accomplish private messaging; and

FIGURE 18 is an illustration of an example of messaging using the Internet.

DETAILED DESCRIPTION OF THE INVENTION

Network Structure

FIGURE 1 is a block diagram of a multimedia network-based communications system, indicated generally at 10, comprising a number of network hubs 12, 14 and 16. Network hubs 12, 14 and 16 are coupled to one another through a communications network 18 that may comprise, for example, high speed datalinks, frame relay links or other suitable high speed data communication facilities. Communications system 10 operates to process and route communications traffic from a wide variety of message sources and to a wide variety of message destinations. For example, hub system 12 is shown coupled to a telephone 22, a messaging system 24, a conventional voice mail system 26 which is coupled to a large number of telephone terminals represented by telephone 38, a facsimile transmission system 28 and a public messaging network 30. Public messaging system 30 may comprise, for example, messaging services provided to the public by local exchange carrier. In addition, network hub 14 is shown coupled to a private system 32 that itself contains a number of messaging systems and to an electronic mail facility 34. Network hub 14 is also shown coupled directly to a telephone 20 and to a conventional voice mail system 36 which is coupled to a large number of telephone terminals represented by telephone 38. It should be understood that telecommunications connections that are shown as direct connections may actually include intermediate switching facilities such as private branch exchanges or central office switches that are part of private and public telecommunications networks.

Network hub 14 is also shown coupled to a private local area network, indicated generally at 31, which communicates with network hub 14 using a communications gateway 33. Local area network 31 may be used to support a wide variety of messaging operations and may connect user stations having electronic mail capability, facsimile capability, voice capability or video capability. The communications system 10 may be used to connect all these systems with other messaging systems through gateway 33 and network hub 14.

Similarly, network hub 16 is shown coupled to a messaging system 40, a private system 42 comprising a number of messaging systems, and a facsimile transmission and receive facility 44. The network hub systems 12, 14 and 16 are also coupled through the communications network 18 to a network center 37. The network center 37 monitors the operation of the network 10 as will be discussed more fully herein. Information providers 39 are also provided a gateway into the communications system 10 for data and message traffic from information providers 39. Information providers 39 may provide, for example, bulletin board information or mass distributed information services or advertising messages that are distributed to users of the communications system 10 based on the preferences or demographics of the users and the content of the information.

As will be discussed more completely herein, communications system 10 operates to integrate and interconnect disparate sources and technologies of communication traffic and to translate messages between them. The communications system 10 maintains a universal database of all users of the communications system and their individual communications profiles including the various media in which the users can send and receive messages. For example, a single user may control and receive communications using an electronic mail facility, a voice mail facilitv, a facsimile facility and a video facility. All of these facilities are identified in a user profile record associated with that user within the network database associated with system 10. As will be discussed herein, a copy of that database is maintained in each network hub within system 10 exemplified by network hubs 12, 14 and 16 in FIGURE 1. For large scale integrated network functionality, interfacing with voice messaging systems introduces further complications for maintaining individual user profiles in that large distributed network directories must be built and maintained based upon numerical addressing and accessed utilizing DTMF signaling and the native protocols of the user system. The communications system 10 further includes media, protocol, and language translation capabilities such that, for example, messages sent in one media can be received in a different media. For example, an electronic mail message might be sent to a destination user that does not have an electronic mail facility but does have a facsimile facility or prefers the receipt of a facsimile transmission over an electronic mail transmission. Accordingly, the communications system 10 will translate the electronic mail message into a facsimile message and deliver the message to the designated facsimile facility. For large scale integrated network functionality, interfacing with voice messaging systems introduces a further complication for the processing of multimedia messages and the alternate routing in that large distributed network directories must be built containing the numerical addresses for the different media destinations and accessed by the numerical addresses of the users of the communications system, and delivered utilizing DTMF signaling and the native protocols of the user system. In addition, the communications protocols associated with voice messaging systems do not have the ability to request and specify special handling for multimedia messages.

For purposes of describing the advantages of the present invention, all the various sources of and destinations for data traffic coupled to and serviced by the communications system 10 are referred to as "messaging systems" whether they comprise voice mail systems, electronic mail systems, facsimile transmission facilities, video transmission facilities or other data transmission or receipt facilities. As such, for purposes of this description, the data received from such a messaging system is referred to herein as a "message" regardless of its composition. For example, a message received, processed and delivered by the communications system 10 may comprise a voice message, an electronic mail message, a facsimile or video transmission or any combination of medium to form a compound message. As used herein, the "media" of a message refers to the manner in which the message is received or delivered. For example, various message media may comprise voice, electronic mail, facsimile or other graphic images, or video. Further, the "protocol" of a message refers to the manner in which the data comprising the message is encoded by the messaging system from which the message originates to the communications system 10, the manner in which the data comprising a message is encoded as it passes through communications system 10, and the manner in which the data comprising a message is encoded prior to its delivery in order for a destination messaging system to comprehend the message. The term "user" will be used herein to refer to human beings interfacing to the communications system 10 either directly or through messaging systems coupled to communications system 10.

The network hubs such as network hubs 12, 14 and 16 shown in FIGURE 1 operate as protocol translation facilities to allow for the connection of the communications system 10 to any of a large number of disparate messaging systems employing differing protocols. Currently, there are a great number of communication protocols which are used by private and public telecommunications and data transmission facilities to interconnect messaging systems. The communications system 10 operates to receive messages and administrative information from messaging systems using the protocol native to that system. The messages and administrative information can then be transmitted to the destination facility using the protocol associated with the destination facility. Certain companies maintain proprietary information delivery protocols that can, if such protocols are made available, be supported by the communications system 10. Further, public domain protocols such as X.400 messaging, SS7 signalling and both digital and analog versions of the audio message interchange specification (AMIS) are also supported by communications system 10. For example, the X.400 protocol includes support for virtually all communications features currently in use. A particular feature set is ordinarily dependent on the features used by a particular messaging system and will usually comprise a subset of the features supported by the X.400 protocol. The communications system 10 is flexible enough to support whatever features are implemented by messaging systems connected to the communications system 10. Additionally, for large scale integrated network functionality, interfacing with voice messaging systems introduces a further complication for the providing of multi-protocol translation capabilities in that messages are delivered utilizing DTMF signaling and numerical addresses. In addition, the communications protocols associated with voice messaging systems do not have the ability to request or specify a translation to a disparate protocol.

The communications system of the present invention also works with a variety of public and proprietary protocols for directory information. As will be discussed more fully herein, some messaging systems may only use the communications system of the present invention as a source of addressing and routing information. In these cases, the messaging system provides the communications system 10 with some information about the intended destination. The communications system 10 then returns specific routing information to the messaging system so that the messaging system can independently contact the destination and deliver the message. In these contexts, the directory information passed between the communications system 10 and the messaging system may use any number of public or proprietary directory information protocols understood by the communications system 10. For large scale integrated network functionality, interfacing with voice messaging systems introduces further complications for providing directory addressing services in that large distributed network directories must be built and maintained containing address and delivery information which must be accessed based upon numerical addressing and that this information is delivered to the requesting system utilizing DTMF signaling and the native protocols of the user's messaging system.

In operation, a network hub such as network hub 12 will receive a message through, for example, messaging system 24 shown in FIGURE 1. By way of example, suppose messaging system 24 utilizes the analog AMIS protocol form of DTMF signaling. Further, assume that the message from messaging system 24 is intended for a party serviced by private system 42 which utilizes a proprietary digital communication protocol. Accordingly, network hub 12 would receive the message from messaging system 24 in the analog AMIS protocol. Network hub 12 then transforms the information in the message to conform to a network transmission format and transmits the transformed message through the communications network 18 to hub 16. Network hub 16 then uses the proprietary communication protocol understood by private system 42 to transmit the information to private system 42. In this manner, communications system 10 not only acts to translate the media in which the messages are sent to the destination messaging system, but also acts to provide messaging between dissimilar, proprietary messaging systems by supporting disparate communication protocols. The communications system of the present invention uses a shared internal protocol for all routing and processing of messages. As such, the communications system of the present invention can easily be adapted as new communications protocols, such as Multipurpose Internet Mail Extension (MIME) or the like, become popular. The network hubs that are required to interface with systems utilizing these new protocols need only translate the new protocol to the internal protocol. As such, the operation of the communications system as a whole can support the addition of unlimited new protocols.

The network hub systems within communications system 10 are in constant communication with one another and with the network center 37 to provide updates as to the status of messages within the communications system 10 and further updates as to the user profile information stored in the user database in each network hub. The network center 37 receives these database and status updates and transmits those updates to the remaining hubs in the communications system 10. Due to the constant communication between the hubs, these updates provide for a universal directory of user profiles and a constantly changing body of information as to the status of all messages within the communications system 10. For large scale integrated network functionality, interfacing with voice messaging systems introduces further complications for access and update of user profile information and message tracking in that large distributed network directories must be built and accessed by the numerical addresses of the users of the communications system.

As will be discussed herein, the interconnected network hubs within the communications system 10 also provide for a large amount of virtual storage that is available to the messaging systems which are attached to the communications system 10. In this manner, large bulletin boards or other bodies of shared information can be stored on any of the network hubs and be instantaneously available to any messaging system connected to the communications system 10. For large scale integrated network functionality, interfacing with voice messaging systems introduces further complications for providing bulletin boards or other bodies of shared information services in that large distributed network directories must be built and maintained containing numerical address and access/delivery information for the requesting system utilizing DTMF signaling, numerical addressing and directory information and delivered in the native protocols of the user system.

Network Hub Architecture

Figure 2 is a block diagram of the interrelationship of the various software modules used in the network hubs of the present invention. FIGURE 2 illustrates the various modules functioning within a particular network hub such as network hubs 12, 14 and 16 described with reference to FIGURE 1 previously. Referring to FIGURE 2, connection processors 52 and 54 interact with messaging systems connected to the particular network hub. For example, an analog connection processor 52 communicates with external messaging systems that use analog communication protocols such as an analog communication protocol utilized by a voice messaging system that uses DTMF signaling. Similarly, a digital connection processor 54 communicates with external messaging systems that use digital communication protocols. Although only connection processors 52 and 54 are illustrated in FIGURE 2, it should be understood that there are a sufficient number of each of these connection processors for the number of messaging systems coupled to a particular network hub. Connection processors 52 and 54 communicate with the remainder of the software system using, for example, the Transport Control Protocol/Internet Protocol (TCP/IP) through internal interface 56 shown in FIGURE 2. Internal interface 56 serves as the main communication link between all of the modules within the software system operating within a particular network hub. Internal interface 56 is also coupled to a file server 59 that provides access to a message store facility 58 which may comprise, for example, a large scale digital storage media such as a hard disk drive. Message store 58 houses the messages received from and to be sent to messaging systems coupled to the network hub. When the media or format of a message must be converted, a media translator 69 is used. Media translator 69 performs media and other forms of translation on messages stored in message store 58. For large scale integrated network functionality, interfacing with voice messaging systems introduces a further complication for providing media and other translation services, in that translation parametrics must be accessed from distributed network directories utilizing numerical addressing methods and that communications protocols associated with voice messaging systems are not able to request or specify media translation services.

A network processor 60 is also coupled to internal interface 56. Network processor 60 is also coupled to a external interface 62 which couples a particular network hub to other network hubs and to the network center 37. Network processor 60 is responsible for collecting and distributing messages to other network hubs. In order to communicate with the other network hubs, the network processor 60 may use. for example, the simple message transport protocol (SMTP) and the MIME protocols.

A management processor 64 is also coupled to both internal interface 56 and external interface 62. Management processor 64 communicates with the network center 37 and the particular network hub and operates to monitor and manage message traffic within the particular network hub. For large scale integrated network functionality, interfacing with voice messaging systems introduces a further complication for tracking of user messages and information, in that messages must be accessed and tracked by the numerical addresses of the users of the voice messaging system.

A group of control processors 66 is coupled between the external interface 62 and a hub database 68. As will be described more completely with reference to FIGURE 3, the control processors include a message router 72, a connection manager 74, a data replicator 76 and an administrative event manager 78. In general, the control processors 66 operate to control the operation of the network hub and to manage and manipulate the information stored in the hub database 68. The hub database 68 is also manipulated and coupled to the remainder of the communications system including the internal interface 56 through an event processor 70. The event processor 70 provides the real time control of the network hub components. Event processor 70 responds to directory service requests, identification confirmation requests, analog, digital and network connection requests, message delivery events and administration event queues. For large scale integrated network functionality, interfacing with voice messaging systems introduces further complications for the operations of the control processors, such as message routing, and the operations of the event processor, such as directory service requests, in that large distributed network directories must be built and maintained containing numerical address of the users of the voice messaging system.

Network hubs may be implemented using a variety of hardware platforms depending on the quantity of traffic to be serviced by a particular network hub. In general, the connection processors reside in personal computer platforms serviced by telecommunications peripheral cards. The control processors and database facilities may be implemented on a suitable workstation platform. All of these various discrete hardware platforms may communicate with one another using local area or wide area network systems.

FIGURE 3 is a dataflow diagram which illustrates the routing and exchange of various types of data within the various software modules that operate within a network hub. Many of the facilities described with reference to FIGURE 2 are shown in more detail in the dataflow diagram of FIGURE 3. For example, the control processors 66 are broken out into constituent components. FIGURE 3 illustrates the message router 72, the connection manager 74, the data replicator 76 and the administrative event manager 78. As shown in FIGURE 3, each of these control processors 66 interacts with the hub database 68 through database access procedures which may comprise, for example, SQL/stored procedures. Essentially, the various software modules may interface with the control processors 66 through communications mechanisms that may comprise, for example, TCP/IP sockets or remote procedure calls.

Event processor 70 interacts with network processor 60, analog connection processor 52, media translator 69, and digital connection processor 54 using a suitable hub control protocol. The hub control protocol may comprise a suitable interprocess communication mechanism that provides client-server, request-response communications. The hub control protocol may be based, for example, on remote procedure calls or TCP/IP sockets. The media translator 69 accesses messages in message store 58 through file server 59 to perform media and format conversions as discussed previously.

Management processor 64 interacts with network processor 60, analog connection processor 52, digital connection processor 54 and event processor 70 using a suitable management protocol which may comprise, for example, SNMP proxy or SNMP sub-agent communications. Network processor 60, analog connection processor 52, and digital connection processor 54 interact with message store 58 through file server 59 through the storage and retrieval of messages that may utilize, for example, MIME. Finally, network processor 60, analog connection processor 52, digital connection processor 54, management processor 64 and data replicator 76 communicate with other network hubs, messaging systems, or the network center 37 using messaging protocols or directory protocols appropriate to the destinations connected to these facilities. For large scale integrated network functionality, interfacing with voice messaging systems introduces a further complication for the network hub operations, in that large distributed network directories must be built and maintained containing numerical addresses of the users of the messaging systems.

<u>Network Hub Storage Facilities</u>

Each network hub comprises two data storage facilities described previously. Each network hub comprises a hub database 68 and a message store 58. As described previously, the message store 58 serves to store inbound and outbound messages and is accessed through a file server 59. The hub database 68 comprises a high performance database for the storage of accounting information, directory service requests, identification confirmations, routing information and queue services information. For large scale integrated network functionality, interfacing with voice messaging systems introduces a further complication for the network hub database in that large distributed network directories must be built and maintained containing numerical addresses of the users of the communications system. As will be discussed herein, the hub database 68 includes a variety of administrative queues including inbound and outbound message queues that do not contain actual messages but contain delivery information for the messages stored in message store 58. Software modules that are distributed in nature will access the message store 58 through file server 59 by use of a network file system or other suitable data transport protocols.

The hub database 68 comprises a high performance database. In order to provide flexibility to the operation of the network hub, access to the hub database 68 is provided by a query requestor. A query responder receives abstracted queries from the various software modules through the network remote procedure calls. These calls are translated and answered by the query responder. By using this architecture, the hub database 68 can be altered by simply changing the query responder.

The analog connection processor 52, the digital connection processor 54 and the network processor 60 access the messages stored in message store 58 using file server 59 as necessary. The messages in message store 58 are also accessed by the media translator 69 through file server 59. The media translator 69 performs the translations of messages from one media to another such as, for example, the translation of an electronic mail message into a voice message using a text to speech system. The operation of media translator 69 will be discussed more fully with reference to FIGURE 11.

Analog Connection Processor

FIGURE 4 is a block diagram of the operation of the analog connection processor 52 discussed with reference to FIGURES 2 and 3 previously. According to one embodiment of the present invention, analog connection processor 52 is responsible for collecting and distributing various forms of messages from analog connected messaging systems using analog networking protocols such as analog AMIS. For large scale integrated network functionality, interfacing with voice messaging systems introduces a further complication to the analog connection processor's functionality, in that the communications system must support standard telephone interfaces, and support numerical addressing and access/deliver information to the requesting system utilizing DTMF signaling, and further support the native protocols of the attached voice messaging system. The analog connection processor 52 runs on a hardware platform that may comprise, for example, a 486-ISA bus personal computer system running the Unix, Windows NT, or SOLARIS operating systems. This hardware platform may contain a plurality of voice, fax, and modem processing cards as well as other forms of interface cards to process other forms of analog data transport. The hardware platform may also contain a high performance network adaptor card such as an Ethernet adaptor for connecting to the remainder of the facilities within a particular network hub. The hardware platform for analog connection processor 52 also comprises disk drive storage for storage of operating system files such as the UNIX operating system plus sufficient temporary storage space to hold incoming and outgoing messages.

Referring to FIGURE 4, the analog connection processor 52 is shown to comprise three internal modules. A message module 80 allows the analog connection processor 52 to communicate with the message store 58 described previously through file server 59. An analog interface module 82 provides a communication link between the analog connection processor 52 and messaging systems using analog messaging protocols. A control module 84 allows the analog connection processor 52 to communicate with the management processor 64 using a management protocol. Control module 84 also communicates with event processor 70 as described previously using the hub control protocol.

FIGURE 5 is a state diagram illustrating the operation of analog connection processor 52. Processing begins at a Boot state 86 which is automatically entered upon power-up of the analog connection processor 52 or upon either a hardware or a software reset condition. After the operating system and application software are successfully loaded in the analog connection processor 52, the analog connection processor 52 exits the Boot state 86 and passes to the event processor poll state 88. The event processor poll state is an idle state where the analog connection processor 52 waits for direction from the event processor 70 described previously as to which task the analog connection processor should perform. If while in the event processor poll state 88 the event processor 70 instructs the analog connection processor 52 to create a connection with a messaging system, the analog connection processor 52 leaves state 88 and passes to the make connection state 90 shown in FIGURE 5. In state 90, the analog connection processor 52 establishes a connection and confirms that the connection has been made.

The event processor 70 may also direct the analog connection processor 52 while in the event processor poll state 88 to wait for inquiries from messaging systems. If this is the case, the analog connection processor 52 passes to the wait for connection state 92 shown in FIGURE 5 where the analog connection processor 52 will remain until a predetermined amount of time has expired or a connection is made with a calling messaging system or network facility. The event processor 70 can also cause the analog connection processor 52 to pass from the event processor poll state 88 to an idle state 94. The idle state 94 is used by the control processors 66 and event processor 70 to hold a communication port in reserve for outgoing calls or otherwise prevent new calls from being received by the analog connection processor 52. In general, a predetermined percentage of the communication ports available to the network hub are maintained in the idle state 54 to provide a pool from which outgoing connections are made in order to deliver messages or directory information. The analog connection processor 52 remains in the idle state 94 until a predetermined amount of time has expired and then the analog connection processor 52 returns to the event processor poll state 88.

After a connection has been completed and authenticated in the make connection state 90, the analog connection processor 52 passes to the protocol master state 96. The protocol master state 96 is the message sending state where messages or directory information are delivered to a messaging system. Once in the protocol master state 96, the analog connection processc 52 continues to deliver all messages and directory information provided by the control processors 66 to the connected messaging system. At certain times during the connection, the analog connection processor 52 offers to pass to a protocol slave state 98, where it may receive messages, directory information, or other data from the messaging system. If no messages, directory information or data need to be received at that time, the analog connection processor 52 passes from the protocol master state 96 to a close connection state 99 where the connection with messaging system is terminated. Processing then returns to the event processor poll state 88.

The protocol slave state 98 is the message receiving state for the analog connection processor 52. When in the protocol slave state 98, the analog connection processor 52 receives and validates messages, directory information and data transmissions from a messaging system. The analog connection processor 52 remains in the protocol slave state 98 until no more messages, directory information, or other data are provided by the messaging system. At that time, the analog connection processor 52 returns to the event processor poll state 88 through the close connection state 99 described previously.

Digital Connection Processor

The digital connection processor 54 operates similarly to the analog connection processor 52 previously described. FIGURE 6 is a block diagram which illustrates the functional modules contained within digital connection processor 54. Digital connection processor 54 comprises a message module 100 which facilitates communication between the digital connection processor 54 and the message store 58 through file server 59. Digital connection processor 54 also contains a digital interface module 102 which facilitates communication between digital connection processor 54 and messaging systems that utilize digital protocols such as digital AMIS, SMTP-MIME, cc: Mail and external service providers such as Internet and MCI MAIL.

Digital connection processor 54 further comprises a control module 104 that facilitates communication using hub control protocol with the event processor 70. The control module 104 also communicates with the management processor 64 using the management protocol.

The digital connection processor 54 is responsible for collecting and distributing voice, fax, E-mail and other media messages from digitally-connected messaging systems using digital network protocols. According to one embodiment of the present invention, the digital connection processor 54 also provides and receives directory update services using digital directory protocols such as X.500 and other digital directory protocols. The digital connection processor 54 may be implemented on a personal computer platform which may comprise, for example, a 486-ISA bus personal computer or SPARC system running the Unix, Windows NT, or SOLARIS operating systems. The digital connection processor 54 platform supports a variety of interface systems such as conventional modems for dial-in directory service and mail submission, and directory information queries and Ethernet connections to provide for wide area network Ethernet connections and local area network connection 50 described previously with reference to FIGURE 2. Similar to the analog connection processor 52, the digital connection processor 54 also contains a disk drive facility for storage of an operating system such as the UNIX operating system plus sufficient storage to temporarily hold incoming and outgoing messages.

In operation, the digital connection processor 54 uses the identical states described with reference to

FIGURE 6 and the operation of the analog connection processor 52.

Message Format

Both the analog connection processor 52 and the digital connection processor 54 are responsible for accepting and validating incoming messages. For large scale integrated network functionality, interfacing with voice messaging systems introduces a further complication to the internal message format in that the communications system must support standard telephone interfaces and support messages that are typically large, spanning several minutes of digitized analog audio signals. The delivery of these messages involve a translation or conversion of the message. For example, the message may need to be translated into a different media. Data transmissions that are received may need to be converted from one format to another. As will be discussed more completely with reference to FIGURE 11, these conversions and translations are performed by media translator 69. In part, these functions are accomplished by encapsulating all received message data with a standard message wrapper to form a message for transport and storage within the communications system 10. This wrapper may be based, for example, on MIME encapsulation protocol. Many messages received by the communications system 10 already include a message wrapper. These messages are also converted to the standard internal message format. This format tags and labels each message media within a message with the addressing information provided by the sender. Each message contains a date field which comprises the time the message was sent as provided by the message submission protocol or, in the alternative, as provided by the communications system of the present invention. Each message also contains a "from" field where both the network identification of the message sender and the message sender's messaging system identifier are combined into an Internet style address. Each message also contains a "to" field where the network identification of the intended recipient and the messaging facility of the intended destination facility are combined into an Internet style address. Each message also receives a unique message identification field for use in administrative tracking of messages and other administrative concerns.

Various delivery features which will be discussed herein, such as privacy and urgency are usually not identified in the message itself. Rather the presence of these requirements are sent to the control processors 66 when a received message is placed in a message queue within the hub database 68.

<u>Network Processor Operation</u>

FIGURE 7 is a block diagram which illustrates the functional modules used by the network processor 60. The network processor 60 contains a control module 120 which allows the network processor module to communicate with the event processor 70, the management processor 60, and other network processors 122 shown in FIGURE 9. The network processor 60 also includes a message module 124 which facilitates communication with the message store 58 through file server 59. While the network processor 60 can be a stand-alone hardware platform similar to the analog connection processor 52 and the digital connection processor 54, under one embodiment of the present invention, the network processor 60 is implemented as a task on a central server that connects to the internal interface 56 and that contains the remaining storage facilities such as message store 58, file server 59, and hub database 68. The network processor 60 communicates with other resources in the network hub using three data paths. Control and query information exchanges between the network processor 60 and the event processor 70 are exchanged using the hub control protocol. Operational monitoring statistics and state information are exchanged with the management processor 64 using the management protocol. Messages are passed into and out of message store 58 using file server 59.

The network processor 60 is responsible for collecting and delivering messages to other network hubs within the communications system 10. In operation, the network processor 60 uses the identical states described with reference to FIGURE 6 and the operation of the analog connection processor 52. For large scale integrated network functionality, interfacing with voice messaging systems introduces further complications for the network processor operation in that large distributed network directories must be built and maintained containing numerical addresses of the users of the communications system.

Event Processor Operation

FIGURE 8 is a block diagram which illustrates the functional components of the event processor 70. Event processor 70 comprises a hub control protocol server 126 which receives remote procedure calls from one or more analog connection processors 52, network processors 60 and digital connection processors 54. Event processor 70 also comprises a database query engine 128 which operates to interrogate and manipulate the hub database 68 to service remote procedure calls from one or more analog connection processors 52, network processors 60 and digital connection processors 54. The event processor 70 also interfaces with the management processor 64 using the management protocol. The event processor 70 also interfaces with the media translator 69 to direct the conversion of messages accessed by the media translator 69 from the message store 58 using file server 59. For large scale integrated network functionality, interfacing with voice messaging systems introduces further complications for the event processor operation in that large distributed network directories must be built and maintained containing numerical addresses of the users of the communications system.

FIGURE 9 is a state diagram illustrating the operation of the event processor 70. Processing within event processor 70 begins in a boot state 130 which is entered upon power-up or reset of the communications system. When the appropriate applications, programs and operating systems are loaded, the operation of event processor 70 passes from boot state 130 to a get query state 132 where the event processor 70 remains waiting for remote procedure calls from analog connection processor 52, network processor 60 or digital connection processor 54, or management protocol queries from management processor 64. Upon receipt of such a query, the event processor 70 passes to a process query state 134 where the database query engine 128 accesses the hub database 68 using database access procedures. The response is received from the hub database 68 in process query state 134 and the event processor 70 passes to a send response state 136 where the appropriate response is forwarded to one of the processors 52, 54, 60, or 64 described previously. The event processor 70 then returns to the get query state 132 where it remains until an additional request is forwarded to the event processor 70.

Control Processor Architecture

As shown in FIGURE 10, each of the control processors 66 illustrated in FIGURE 3 shares a common architecture. The message router 72, the connection manager 74, the data replicator 76, and the administrative event manager 78 each control a collection of tasks responsible for the actions of the network hub. Overall control of the network hub is achieved through control of the hub database 68 and specifically the message queues and administrative event queues contained within the hub database 68. In this manner, control is achieved by prioritizing events stored in the message and administrative event queues and creating and manipulating entries in these queues to be read by the event processor 70. The event processor 70 is the actual real time engine of the network hub.

FIGURE 10 is a block diagram which illustrates the architecture of each of the control processors 66, the management processor 64 and the event processor 70, and the manner in which these entities interact with the various portions of the hub database 68. Each of the control processors comprises a hub controller module 138 and a database query engine 140. The hub controller 138 administers the tasks to be stored in the hub database 68 and communicates with the hub database 68 through the database query engine 140 using database access procedures.

Message Router Operation

The message router 72 comprises a hub controller 138 and a database query engine 140. The message router 72 interacts with an inbound message queue 142, an outbound message queue 146, and a message system database 147. In general, the message router 72 uses information from the message system database 147 to determine the next destination for each message passing through the network hub. As discussed previously, the message itself is actually stored in the message store 58 and is accessed using the file server 59. The inbound message queue 142 and the outbound message queue 146 store message records which indicate the location of the message within message store 58 as well as the source and destination of each message. The message router 72 reads message records from inbound message queue 142 after they have been placed there by event processor 70. The message router then determines the next destination for the messages associated with the message record and creates a new message record and places that message in outbound message queue 146.

The message router 72 is also responsible for prioritizing messages and, in the case of future delivery, delaying messages. In general, messages are ranked and delivered in order of priority. Efficient use of the communication system 10 of the present invention requires the ability to shape the messaging traffic into a somewhat constant volume. This traffic shaping is achieved by having the message router 72 selectively delay messages as needed until the network is less loaded. To accomplish this, the message router 72 attaches an internal priority for each message. The internal priority for the communications system is set by both the urgency specified by the submitted messages and how long it has been since the message was received for delivery. For large scale integrated network functionality, interfacing with voice messaging systems introduces a further complication for the message router operation, in that large distributed network directories for the hub database 68 must be built and maintained containing numerical addresses of the users of the communications system.

Connection Manager Operation

The connection manager 74 also contains a hub controller 138 and a database query engine 140. The connection manager 74 reads the message records within outbound message queue 146 and determines the nature of the connection that must be made to service each message record within outbound message queue 146. Connection queue records associated with these connections are formatted and placed in a connection queue 141 within hub database 168. The connection manager 74 also reads administrative event records from an outbound administrative event queue 143 and determines if connections within the network are required to service the administrative event records within outbound administrative event queue 143. If network connections are required, a record is similarly formatted and placed in connection queue 141. For large scale integrated network functionality, interfacing with voice messaging systems introduces further complications for the connection manager operation in that the message records within the inbound and outbound message queues, the administrative event records within the administrative event queue, and the connection queue records within the connection queue must be built, maintained, and accessed using the numerical addresses of the users of the communications system.

The connection manager 74 controls the connections between each network hub, the network center 37, and each messaging system. The connection manager 74 creates connection queue records for each message and administrative event within the outbound message queue 146 and outbound administrative event queue 143 of the hub database 68. The connection manager 74 manages conventional connections and also determines when multiple connections are needed and when failed connections should be retried. The connection manager 74 is triggered by any record within the outbound message queue 146 or the outbound administrative event queue 143. The connection manager 74 may also alternatively engage in periodic scans of all the queues within the network hub to reprioritize the connection queue 141 based on the number of events and the priority of events waiting to occur.

As discussed previously, the connection manager 74 maintains the connection queue 141 which comprises a list of destinations for which a connection needs to be made. These connections may be scheduled for a particular time or as necessary according to a priority based on the number of messages, the oldest message in a particular queue, or the last time the connection was made. The connection queue 141 is accessed by the event processor 70 to determine the next operation for a newly available communication port. The connection queue 141 is periodically updated by the connection manager 74 with a connection priority indication. Each connection queue record is marked with an overall priority indication based on the highest priority message in the connection queue 141 and a count of the number of messages in the connection queue 141.

These priority indications are Immediate, High, Normal, or Low. Immediate indicates that one or more events scheduled in the connection queue 141 for a given destination are overdue. High indicates that one or more events scheduled for a given destination have a high internal priority and that a connection to a destination should be made as soon as a connection becomes available. Once a connection is established and the High priority messages are sent, other messages may be sent if the load on the connection permits. The Normal priority indicates that no events scheduled with High priority are resident in the connection queue 141 and at least one Normal priority event is resident in the connection queue 141. The connection manager will open a connection if a connection is free and there are no pending High priority events in the connection queue 141. Once the connection is established and the Normal priority message has been sent, the Low priority events can be processed if load permits. The Low priority indication occurs when only scheduled events of Low priority are resident in the connection queue 141. The connection manager will open a connection is a connection is free and there are no remaining Normal, High, or Immediate priority connections required. The Low priority messages will escalate to a Normal priority if they remain in the connection queue 141 for a predetermined amount of time, for example, two hours.

Efficient operation of the communications system 10 relies on efficient queue management by the connection manager 74. Because a connection processor such as analog connection processor 52, digital connection processor 54 or network processor 60 can change roles on the same connection, the connections between network hubs are prioritized by the total volume of traffic to be exchanged. In addition, in the event that connections are scarce, interacting network hubs attempt to contact each other, increasing the likelihood that a successful connection will be made. In order to facilitate this interaction, a reservation methodology is implemented. When the connection manager 74 creates a connection record within connection queue 141, in addition to adding the delivery information record to the connection queue 141, a similar entry is sent to the remote network hub that is connected to the destination messaging system. In this manner, both the local network hub and the remote network hub know that a connection must be made and contact attempts can be initiated by either network hub. Once a message connection is established, either through the normal processing of the message queues or by the attempts of the network hubs to contact each other, both network hubs can use the connection to deliver all messages to each other.

Administrative Event Manager Operation

The administrative event manager 78 handles administrative events in the same manner that the message router 72 handles message records. Administrative event manager 78 comprises a hub controller 138 and a database query engine 140. Administrative event manager 78 reads entries from the inbound administrative event queue 145 which have been placed there by event processor 70. Administrative event manager 78 then determines if an administrative event message must be sent to another network hub within the communications system 10 or if a message must be sent to the network center 37. If an administrative event message must be sent to any of these locations, an entry is made in the outbound administrative event queue 143. For large scale integrated network functionality, interfacing with voice messaging systems introduces further complications for the administrative event manager operation in that the administrative event records within the administrative event queue must be built, maintained, and accessed using the numerical addresses of the users of the communications system.

Data Replicator Operation

As discussed previously, the data replicator 76 is responsible for the queuing and transport of database change events. The data replicator 76 manages the user information updates for the hub database 68 and processes database change events when they are received from the master database stored in the network center 37. The data replicator 76 insures that the message system database 147 and user profile database 149 within hub database 68 remain synchronized with the master database in the network center 37.

The data replicator 76 also comprises a hub controller 138 and a database query engine 140. The data replicator 76 operates to input new information into a user profile database 149 and the message system database 147. The data replicator 76 receives the updated information from the network center 37 and downloads the information to the user profile database 149 if the information involves new user profile information or downloads the information to the message system database 147 if the new information comprises configuration information for message systems attached to the communications system 10. The data replicator 76 processes information which is transmitted to all network hubs for updates to the user profile database 149 and the message system database 147. In contrast, the administrative event manager 78 accepts updated information from the network center 37 which comprises information which is specific to the particular network hub. In addition, the administrative event manager 78 accepts requests for information from the network center 37 for information which is stored on messaging systems connected to the network hub housing the administrative event manager 78.

Management Processor Operation

Although the architecture of the management processor 64 will be discussed in detail in FIGURE 11, its interaction with hub database 68 is shown in FIGURE 10. The management processor 64 contains a control module 150 and a database query engine 140. The management processor 64 interacts with an alarms database 131 within hub database 68. The alarms database 131 comprises a list of events that occur within the network hub processing and in some cases the time at which these events occurred. The list of events stored in the alarms database 131 can be used by systems within the network center 37 to track down errors in the processing of messages and to maintain control over when events occur or when events have occurred within the processing of messages within a particular hub.

The management processor 64 also interacts with the inbound message queue 142, the outbound message queue 146, the connection queue 141, the outbound administrative event queue 143, and the inbound administrative event queue 145 as necessary to monitor these queues. The management processor 64 is operable to reorder and delete entries in the queues discussed previously as necessary to repair errors and expedite messages. For large scale integrated network functionality, interfacing with voice messaging systems introduces further complications for the management processor operation in that the accessing, updating, deleting, and reordering of the inbound and outbound message queues, the inbound and outbound administrative event queues, and hub alarm event tables must be performed using the numerical addresses of the users of the communications system.

Event Processor Operation

As discussed previously with reference to FIGURE 8, the event processor 70 comprises an HCP server 126 and database query engine 128. The event processor 70 places message records within inbound message queue 142 as they are received into the network hub. Similarly, the event processor 70 places administrative event records into the inbound administrative event queue 145 as they are received into the network hub. The event processor 70 consumes the connection queue 141 as it makes connections to enable the passing of messages out of the particular network hub. For large scale integrated network functionality, interfacing with voice messaging systems introduces further complications for the event processor operation in that inputting message records into the inbound message queue and administrative event records in the inbound administrative event queue, and retrieving records from the connection queue must be performed using the numerical addresses of the users of the communications system.

<u>Management Processor Architecture</u>

FIGURE 11 is a block diagram which illustrates the functional components and interactions of the modules comprising the management processor 64. Management processor 64 comprises a control module 150 which interacts using the management protocol with the analog connection processor 52, the digital connection processor 54, the event processor 70 and the network processor 60. The control module 150 also interacts with the hub database 68 and, specifically the alarms database 131, using database access procedures. The management processor 64 acts as the chief liaison between the network hubs and the network center 37. Management processor 64 also receives Customer Service, HelpLine inquiries and directives from the network center 37 using the management protocol. As will be discussed more completely herein, the communications system 10 has the capability of being able to continually track and manage messages within communications system 10. The HelpLine operation allows for messages to be rerouted or terminated after they have been placed in communications system 10 and before they have been delivered. These directives are transmitted to the management processor 64 using the management protocol from the network center 37.

Media Translator

FIGURE 12 is a block diagram which illustrates the functional components and interactions of the modules comprising the media translator 69. Referring to FIGURE 12, media translator 69 comprises a control module 133 and a translator bank 144. The control module 133 interacts with the event processor 70 using the hub control protocol. The translator bank 144 receives messages from the message store 58 by making requests to file server 59. The translator bank 144 operates to convert the format of data and to translate the media of messages. The messages processed by translator bank 144 are then returned to message store 58 using file server 59. Translator bank 144 comprises a plurality of translators indicated as 144a-n in FIGURE 12. Each of the translators 144a-n within translator bank 144 accomplish a stage of the conversion to be performed by translator bank 144. Each translation or conversion is done by first converting the message or data into a high quality internal format that contains sufficient data representation to insure that no data is lost during the conversion process. Translations of messages from one media to another are performed while the message is represented in the internal format. In this manner, the translator bank 144 need only contain a minimum number of media to media translators. In summary, the control module 133 can cause the translator bank 144 to use many different filters in sequence to accomplish any permutation of media-to-media translation, message-to-message transformation or data-to-data conversion.

The communications system of the present invention offers a ubiquitous messaging service which provides for the delivery of messages to messaging systems of various technologies using their native protocols and allows a recipient to select, where practical, the preferred media for receiving messages. In order to offer this facility, the communications system of the present invention is able to communicate with messaging systems using various messaging protocols used by current messaging systems. For large scale integrated network functionality, interfacing with voice messaging systems introduces further complications for the communications system of the present invention, and especially the media transl-tor in that the user profile records must be accessed to perform media translation selection because the communications protocols associated with voice messaging systems are not able to indicate the necessity or to specify the nature of media translation services required.

Each network hub also includes the ability to translate messages into various forms of message media. These media translation features include the ability to translate text to speech, facsimile images to text and text images, text and text images to facsimile images, and speech to text. These media translation capabilities can also be combined for a single message. For example, to the extent that speech is required to be translated into a facsimile image, the data is first translated into text and then into a facsimile image of the text.

As discussed previously, media translation may be accomplished according to the desires of the message recipient. In addition, media translation can occur automatically responsive to the needs of a destination messaging system. For example, if a destination messaging system comprises a simple voice message system with no data, facsimile or capability other than voice, all messages to such voice message system can be translated to voice messages regardless of the media in which the message was sent. In this particular example, the communications system of the present invention would translate all non-voice messages to a textual format and utilize a text to speech converter to create the final, deliverable voice message.

Network Center

FIGURE 13 is a schematic block diagram of the components of the network center 37 used in the communications system of the present invention. The network center 37 communicates with the remainder of the communications system 10 through external interface 62 which is connected to or a part of communications network 18. The network center 37 further comprises an operations center management system 153 and an central access and control system 155 coupled to external interface 62. The central access and control system 155 is coupled to a HelpLine system 157, a customer service system 161, a billing system 159, a message tracking system 163, a customer computer interface system 167, a master database 151, and an interactive voice response system 169.

As discussed previously, master database 151 operates as a central repository for all user information, message system information, and message status information that is used by the network hubs to manage messages within communications system 10.

The central access and control system 155 comprises the interface to the remainder of the communications system for the master database 151, the HelpLine system 157, the billing system 159, the message tracking system 163, the customer service system 161, the customer computer interface system 167 and the interactive voice response system 169. The customer service system 161 supplies information as to additions and changes to the user databases and message system databases. Customer service system 161 can also serve as a gateway for questions requiring the use of the billing system 159. The HelpLine system 157 provides access to the central access and control system 155 for purposes of providing information to the message tracking system 163. In addition, the HelpLine system 157 can create trouble tickets identifying problems within the communications system 10 which are passed by the central access and control system 155 to the operations center management system 153. The trouble tickets are then investigated and corrected by the operations center management system 153.

The customer computer interface system 167 is coupled to a plurality of user terminals indicated generally at 171. Similarly, the interactive voice response system 169 is coupled to a plurality of user touchtone compatible telephones indicated generally at 173 in FIGURE 13. The customer computer interface system 167 and the interactive voice response system 169 provide gateways into the central access and control system 155 for users to directly interact with the central access and control system 155. Using the customer computer interface system 167, the users can digitally interface with central access and control system 155 to accomplish the same functions performed by HelpLine system 157, customer service system 161 and message tracking system 163. Similarly, users using touchtone telephones 173 can interact with the central access and control system 155 through interactive voice response system 169 to migrate through various choices presented to the users to accomplish the same functions accomplished by HelpLine system 157, customer service system 161, and message tracking system 163.

For large scale integrated network functionality, interfacing with voice messaging systems introduces further complications for the network center 37. First, the accessing of records in the master database files must be performed using the numerical addresses of the users of the communications system. Secondly, the providing of an interactive voice response capability must be performed using standard telephone equipment using DTMF signaling, addressing must be numerical, voice must be the only media, identification confirmation must be spoken, and directory access/inquiry must be performed using DTMF signaling.

Figure 13 also illustrates that information provider databases 39 are also coupled to external interface 62 either directly or through a bulk mailing list agent 171. The information provider databases 39 are accessed and updated by information providers 165 shown in FIGURE 13. As will be discussed more completely herein, the information providers 165 can supply information for the information provider databases 39 that can be accessed by users of the communications system 10. This information may comprise, for example, advertising messages or other information associated with products or services provided by the information providers 165.

User Profiles and Address Translation Operation

The communications system of the present invention contains a directory with which it offers a range of addressing and directory services. A copy of the directory is stored in each of the network hubs and includes the network identification of a number of addresses associated with each user. The directory further includes the routing information and feature parameters associated with each user as well as identification confirmations which may comprise, for example, name records for spelled and spoken name confirmation and graphic or video images of each user for visual recognition.

Internally, the communications system of the present invention uses a globally unique number to route and deliver messages which will be referred to as the user ID. The communications system of the present invention supports multiple numbers and non-numeric addresses for each identified network user. This facilitates support for a single user mailbox to be coupled to multiple phone numbers, facsimile numbers, voice mail system numbers, E-mail identifiers, external network addresses and the like. The external addresses associated with each of these facilities may comprise arbitrary character strings with arbitrary internal structure. Because of the directory system used by the communications system of the present invention, external addresses need not be globally unique. The internal directory specifies what type of messages are to be routed to each destination associated with a particular user. Using its internal directory, the communications system of the present invention transparently converts between any address type including proprietary voice mail system addressing, North American Numbering Plan identifications, international telephone numbers, X.400 addressing, Internet addressing, and a wide variety of E-mail addressing schemes.

In order to process messages to and from external messaging systems using closed subscriber communities or private numbering plans, the communications system of the present invention must be able to interface with systems that require four digit addresses, ten digit addresses and other formats for both the sender and recipient addresses. For purposes of describing the unique address translation and message routing capability of the communications system of the present invention, the term "address" is the media specific address used to describe a subscriber. As discussed previously, a "user ID" is the unique global internal identifier of a particular user. A "community" is associated with a closed system or private numbering plan. All addresses must be unique within a community. "Scope" indicates the context in which the address should be used for the purpose of representing a particular user ID upon final delivery of a message. Scope enables the use of multiple addresses per user ID of the same address type. For example, an address with local scope is used when sending a message to the messaging system on which the address resides. Each user ID may have only one local scope address for each messaging facility. In comparison, an address identified as having global scope is used when interacting with other messaging systems.

As discussed previously, the media of a message, messaging features, and message subject matter are also used for routing purposes. The media field in the address table is used to perform per-media message routing and filtering by indicating which address and message system is to be used for final delivery of the message. The feature field is used to perform per-feature routing and filtering, such as sending urgent messages to a call message delivery system rather than a conventional voice mail facility. Messages can also be routed and filtered based on the subject matter of the message using the message subject matter field in the address table.

FIGURE 14 is an example of a use of the address translation functions performed within the directory previously described. The sender in FIGURE 14 is named Arnie and Arnie supplies his home messaging system which is known as CASJ01 with a message and the telephone number of his friend, Greg. Greg is a subscriber to the communications system of the present invention and resides on a messaging system known as TXDL03. Arnie supplies Greg's telephone number as 214-555-2722 when Arnie records the message to Greg. The CASJ01 system supplies the message and Greg's telephone number to the communications system of the present invention. The contacted network hub can identify the source messaging system by using either the source or destination network address such as the ANI or DNIS codes of the incoming call from the CASJ01 messaging system. As shown in Figure 14, the communications system of the present invention accesses the copy of the address translation stored in the inbound network hub coupled to messaging system CASJ01 using the destination address, message media, the source address, and the source messaging system to retrieve the source user ID equal to 3, the community identification equal to 0, the destination user ID equal to 1 and the destination message system identified as TXDL03. This information is transmitted with the message to the outbound network hub which is associated with the TXDL03 messaging system.

The outbound network hub accesses its copy of the address translation table using the received information and retrieves Greg's destination address equal to 214-555-2722 and the appropriate source address for Arnie equal to 408-555-4437. The outbound network hub used the entry for Arnie corresponding to a "global" scope because the destination messaging system TXDL03 was not the same as the source messaging system VMX-ONEL. The use of the scope field allows the communications system of the present invention to interconnect users on different messaging systems within the same organization using short form addresses or users on different messaging systems using full length addresses.

Just as Arnie has different Local and Global scope entries in the address translation table, multiple table entries may also be used to route on the other variables discussed previously such as media, subject matter or messaging features like privacy or urgency.

The user profile directory stored within the communications system of the present invention is updated using a variety of mechanisms. For example, the customer service system 161 within the network center 37 is used to directly update the database. In addition, bulk changes in batch mode can be made to the master database 151 during the enlistment of large new groups of subscribers by downloading user profiles from new messaging systems.

In addition, various message protocols include fields within the protocols for basic database information such as spoken and ASCII name information. When this information is provided as part of the message, it is retrieved from the message to update the user profile information within the master database 151 used by the communications system of the present invention. As shown in FIGURE 13, users and messaging system administrators can also directly affect the information in the master database 151 by using the interactive voice response system 169 or the customer computer interface system 167 operating in the network center 37. These systems allow users to alter the user profiles associated with their addresses and, as such, alter the routing and filtering of messages passing within the communications system of the present invention to them.

Some messaging systems are already network-compatible and are able to directly share database information with the communications system of the present invention. Access to the database stored within a messaging system is achieved through analog and digital connection processors 52 and 54 described previously.

Compound Messaging

Communications system 10 is operable to receive, transport, and deliver compound messages comprising information in more than one media. These messages can either be delivered as compound messages (if supported by the messaging system of the recipient), be disintegrated into their respective media parts and delivered to their various terminals for particular media type, or have one or more media parts converted into an alternative media for delivery as either a compound or disintegrated message.

For example, a single message received by communications system 10 may comprise both a voice message and a facsimile transmission. The communications system 10 is operable to route the voice portion to a voice messaging system and the facsimile to a facsimile messaging system both of which are associated with the message recipient. The communications system 10 is also operable to translate messages received in one media to a media associated with the facilities of the message recipient if the message recipient does not have the appropriate facilities. For example, if a compound message is received by communications system 10 comprised of a voice message and an electronic mail transmission but the message recipient only has a voice messaging facility and a facsimile messaging facility, communications system 10 will transmit the voice portion of the compound message to the message recipient's voice messaging facility and translate the electronic mail message to a facsimile image for delivery to the message recipient's facsimile messaging facility. For large scale integrated network functionality, interfacing with voice messaging systems introduces further complications for the processing of compound messages in that the accessing of the user profile records to perform compound message separation and delivery must be performed using the short, fixed length numerical addresses of the users of the communications system. Additionally, the receiving of the destination addresses for the received message must be performed with DTMF signaling and the native protocol of the user's messaging system.

Messaging Features

The communications system of the present invention supports the sophisticated messaging features currently in use by various messaging systems such as privacy, urgency, delivery confirmation, return receipt, deferred delivery and the like. The role of the communications system of the present invention as a translator between messaging systems of disparate protocols and capabilities results in a unique problem for the communications system of the present invention. A user of a messaging system that has a great many sophisticated features may desire to send a message to a recipient who is only accessible through a messaging protocol or messaging system that does not support some or all of those features. The user sending his message using the communication system of the present invention is allowed to select whether or not failed support of that feature causes the message to be returned or to be delivered anyway. Accordingly, the communications system of the present invention implements the "if possible" feature to provide for flexibility in communication between messaging systems of disparate capability. For example, messages within the communications system 10 may be labeled as "private", "private if possible", or "non-private". Typically, "private" messages are interpreted by a destination messaging system as a message that cannot be forwarded and so, if "privacy" were not supported, the message would be returned to the sender with the appropriate error message. The "private if possible" designation on a message causes the communications system of the present invention to implement privacy if the destination messaging system supports that feature; if not, the message is delivered anyway.

Similarly, the communications system of the present invention supports "urgency". Messages can be labeled on submission as High, Normal, or Low urgency which determines the maximum amount of time for delivery of the message to achieve the urgency requested by the sender. As with "privacy", messages can be labeled "urgent if possible", so if the destination messaging system supports urgency or a prioritization of messages, the message is sent as urgent; if not, the message is delivered anyway.

The unique capability of the communications system of the present invention to access a variety of messaging media allows for greater flexibility in the handling of urgent messages. For example, a user may select in his parameter list that when a message is received as urgent, the message is to be delivered to a selected messaging facility and, in addition, a phone call is to be placed to his pager or other telephone number, for example.

The ability of the communications system of the present invention to hold a variety of access points for a particular user within a directory table allows for the splitting of messages as well. For example, an electronic mail message may also include a portion of voice data. If a recipient's electronic mail facility is unable to support voice (for example, the recipient is using a personal computer that does not have any sort of sound capability), the communications system of the present invention can route the voice portion of the message to the voice mailbox of the recipient and the text portion of the message to the electronic mail facility of the recipient or, alternatively, the voice portion of the message could be translated to text using the media translation facility of the communications system of the present invention and delivered with the E-mail message. On the other hand, if the recipient only has a voice messaging system, the text portion of the E-mail could be translated into voice and delivered with the voice message.

The communications system of the present invention also supports the messaging features of delivery confirmation and return receipt. For delivery confirmation, a message is sent to the message sender with confirmation of delivery of the message to the recipient's mailbox. For read receipt, a message is sent to the message sender with the time the message was heard or accessed by the recipient.

The communications system of the present invention also supports deferred delivery where a message is held for delivery at a later time. In this manner, the communications system of the present invention provides a reminder network for a reminder of important events, such as meetings, birthdays, etc. Due to the storage capabilities inherent in the communications system of the present invention, this feature can be supported without being supported by the recipient messaging system. For large scale integrated network functionality, interfacing with voice messaging systems introduces further complications for the providing of messaging features, such as privacy, urgency, delivery confirmation, etc. In the context of interaction with voice messaging systems, these features must be created, maintained, and accessed in the user profile records using the short, fixed length numerical addresses of the users of the communications system. Additionally, the receiving of these messaging features must be performed with DTMF signaling and the native protocol of the user's messaging system.

Mailing List Distribution

The communications system of the present invention supports mailing list distribution of messages using two methods. The first method uses the facility of a local alias, such as a group code which references a number of addresses. According to this facility, when the sender of the message accesses the group code, the message is replicated and a separate message is sent to each address referenced by the group code from the sender. The second method of sending a message to a large number of recipients is through a mailing agent such as, for example, bulk mailing list agent 171 discussed with reference to FIGURE 13 previously and used in the case of mass distributions of information from the information providers databases 39. According to this method, the sender sends the message to the mailing agent with the list of intended recipients. The mailing agent is an address that acts as an alias for the sender to send the message to the message recipients.

The distinction between the two methods arises in the case of errors, returned messages, and message replies. When the alias feature is used, errors, returned messages, and message replies are forwarded to the actual sender of the message. This is useful if it is essential for the sender to know the ultimate disposition of all of the messages sent. In contrast, the mailing agent is the recipient of all error messages or receipt notifications when the mailing agent is used as the sender. One embodiment of the present invention utilizes the bulk mailing list agent 171 to distribute advertising or other information to a large number of recipients. This feature may be used, for example to distribute content specific advertising to selected members of the user population based upon their user profile. In this particular embodiment, the sender of the message merely wants to know that the message was at least attempted to be sent to the entire class of intended recipients. In this application, the use of the bulk mailing list agent 171 is essential to the efficient distribution of these mass messages. For large scale integrated network functionality, interfacing with voice messaging systems introduces further complications for the providing of mailing list distribution, such as group codes and bulk mailing lists in that these features must be created, maintained, and accessed in the user profile records using the short, fixed length numerical addresses of the users of the communications system. Additionally, the receiving of these messaging features must be performed with DTMF signaling and the native protocol of the user system.

Message Routing

The distribution of messages to a large number of recipients is aided by the fact that the communications system of the present invention includes a message subject matter field with each message upon which the communications system can filter and route messages given the preferences of each user. The subject matter indication allows a sender of a message of the communications system of the present invention to designate the type of information within the message. Possible identifications include personal information, unsolicited advertising, commercial messages, or other such classes of messages. Using this feature, advertising information can be further subdivided using a message content type field. When a user is enrolled into the communications system, he can specify the types of messages he wishes to receive that reflect the user's particular interests or hobbies and to what messaging system those messages are to be routed. As discussed previously, users can alter the parameters of their user profiles including this information stored in the master database 151 through interactive voice response system

169 or customer computer interface system 167 contained in the network center 37.

The message subject matter information can also be used to route messages based on their content. For example, personal messages from a particular person might be routed directly to a special recipient facility whereas advertising material could be routed to a different facility. Further, messages of a certain content type or messages from certain senders amy be blocked based upon user preferences.

Users of the communications system of the present invention may be encouraged to receive advertising messages by crediting an account associated with the user for each advertising message received and accessed. Accordingly, a single user may maintain several mailboxes or recipient facilities expressly for the purpose of categorizing and managing incoming messages. Messages passing through the communications system of the present invention can be routed on the basis of the message subject matter, message content type, and the source address. Any permutation of these three factors can be used to route the message to any of the recipient's various messaging systems.

For example, a user can specify that advertising material from a particular source, even though it comes in the form of a facsimile transmission, should be translated and routed to a particular E-mail address which the user has dedicated to advertising material to prevent the advertising material from cluttering up the user's other messaging facilities. Further, a user may require that any electronic mail message of a personal nature from a particular sender is to be immediately translated into a voice message and placed in a mailbox and, further, that a pager call is to be instituted to the user to alert him that the message is present in the voice mailbox.

For large scale integrated network functionality, interfacing with voice messaging systems introduces further complications for the providing of message routing services in that the message subject matter field which is used for routing and filtering must be created, maintained, and accessed in the user profile records using the short, fixed length numerical addresses of the users of the communications system. Additionally, the routing and filtering of messages must be performed on messages that have been received with DTMF signaling and the native protocol of the user's messaging system.

Message Tracking and Revocation

As described previously, the network hubs within the communications system of the present invention maintain constant contact with one another through the communication network 18 to constantly update both the master database 151 of user profile information and messaging tracking system 163 with administrative information regarding the status of each message that is present within the communications system 10.

In this manner, the network center 37 holds a record of the presence and status of every message that is being routed within the communications system 10 before the message has been delivered. To perform this function, message tracking system 163 is accessed through HelpLine system 157, customer computer interface system 167 or interactive voice response system 169 to determine the current location of a given message and if necessary, to cancel delivery and return it to the sender.

Message Submission from and Delivery to Nonsubscribers

The communications system of the present invention allows for the use of the communications system by nonsubscribers through public access facilities into the communications system 10. For example, a conventional telephone line may route nonsubscribers into interactive voice response system 169 or customer computer interface system 167 which collects information on the nonsubscribers such as their identification confirmation, the address for the message delivery features desired, and if necessary, billing information. Message submission from nonsubscribers can be billed on a per-message basis using 900 or 976 telephone numbers, credit cards, or on a collect basis to those subscribers who agree to pay the cost of a collect message.

Message delivery to nonsubscribers is also supported. In this manner, whenever a subscriber or a nonsubscriber sends a message using communications system 10 to a recipient who also a nonsubscriber to communications system 10, the message will be delivered to the recipient using whatever means are available to communications system 10 for the particular media type.

For voice messages or messages of other media which must be converted to voice to be delivered, communications system 10 will repeatedly attempt to call the recipient to deliver the message. Once the connection is made to the destination telephone number, the called party may use interactive voice response system 169 or similar interactive voice response systems within each network hub to facilitate delivery of the message. Secure messages are received by using a password to access the message. Interactive voice response system 169 asks the answering party to enter the appropriate password and will only deliver the message if the password is entered. Further options available to the answering party include redirecting the message to a messaging system or to another party in the case of misdelivery. Further, the answering party is also given the option to terminate delivery of the message if the intended recipient is not available and will not be available at any time. The sender of the message will then be notified using a message generated by the communications system that the message was undeliverable. Additionally, the answering party may also enter a time when the intended recipient of the message would like redelivery of the message. For large scale integrated network functionality, interfacing with voice messaging systems introduces further complications for the providing of interactive voice response services. For example, the providing of this service must be performed using standard telephone equipment using DTMF signaling, addressing must be numerical, voice must be the only media, identification confirmation must be spoken, and directory access/inquiry must be performed using DTMF signaling.

Disaster Recovery

The master database 151 includes, as described previously, constantly updated status information on messaging traffic within communications system 10 which is constantly updated for each subscriber. In addition, master database 151 contains configuration information for each messaging system coupled to communications system 10. In this manner, a large scale failure in a messaging system can be quickly restored by reconfiguring a new system using the configuration information within the master database 151. For example, if a disaster such as a fire were to strike a messaging system coupled to the communications system 10, a new messaging system could be loaded with all of the configuration data stored in the master database 151 to greatly speed the process of returning messaging capability to the subscribers.

In addition, the large storage capability within the communications system of the present invention can be used to provide storage of all messages directed to the failed messaging system until the new system is in place. The stored messages then can be downloaded in batch mode to the newly configured replacement system when it is brought on-line. In this manner, the communications system 10 provides redundant security to all messaging systems connected to the communications system 10 and provides for intermediate storage of messages during disaster recovery.

Secure Messaging Traffic

The communications system of the present invention utilizes Internet Privacy-Enhanced Mail (PEM) technology and the master database 151 to implement a public key cryptographic system to support the scrambling of messages for secure communications. For example, a message sent to a mailbox regularly read by several people, such as one read by a secretary and a supervisor, can be encrypted such that only one individual can decode the message using a secret key or double password. Other uses include private non-subscriber delivery whereby a message delivery can only be made to a specific non-subscriber if the person answering the call has the key to the encrypted message.

While PEM is a public standard and can be implemented in other contexts, the key management infrastructure to make a public key cryptographic work is difficult without a centralized authority. The communications system of the present invention uses master database 151 to provide an immediate advantage over a peer-to-peer networking system. For large scale integrated network functionality, interfacing with voice messaging systems introduces a further complication for the providing of secure messaging services, in that the public key which is used for encrypting and decrypting must be created, maintained, and accessed in the user profile records using the short, fixed length numerical addresses of the users of the communications system.

Identification Confirmation

A key aspect of a socialization of users of a networked messaging facility is the ability to confirm the identification of a recipient of a message. In other words, people are much more willing to leave a message or transmit information to a person when they receive confirmation from the communications system that the communications system is aware of the identity of the intended recipient and can present the sender with some information that creates a sense of security that it is the proper recipient. This information may be a still portrait or full motion video of the intended recipient in the video context, a rendition of the person's spoken or spelled name in the audio context, or a textual or graphic representation of the person's name either as text data or as an image of the person's signature. The user profiles contained in master database 151 and in each network hub include identification confirmations. It should be understood that the identification confirmations associated with the system of the present invention comprise more information than has been provided by prior directory based systems. For example, some prior systems have provided for directory confirmation as a user inputs the letters of an intended recipients name. These systems do not provide information other than the names from the directory and do not help to resolve the dilemmas presented by persons having the same name or directory entry. According to the teachings of the present invention, however, a subscriber who has the full capabilities of communications available to the subscriber may have data stored in the subscriber's user profile that includes the user's portrait or video of the subscriber. The same user may also have the ASCII spelling of the subscriber's name, a digitally encoded recording of the subscriber speaking his or her own name, and a graphic image of the subscriber's signature. In this manner, no matter the media of the sender, be it text, voice or video, the network can respond with information from the user's profile which will allow the sender to feel secure that the communications system is aware of the intended recipient and the appropriate recipient has been identified. The user profiles contained on master database 151 are continuously updated to allow for changes to identification confirmations of each user.

In addition, user profile information for non-subscribers is also accumulated by master database 151 by requesting identification confirmations and addressing information when nonsubscribers send messages on communications system 10. For example, when a nonsubscriber sends a message for another nonsubscriber or for a subscriber using interactive voice response system 169, interactive voice response system 169 will ask the sender to also transmit their identification confirmations. These identification confirmations are stored with the address of the sender as a new entry in master database 151. Accordingly, this user profile information is then available to future senders of messages to this nonsubscriber. Likewise, when a message is received by a nonsubscriber, the nonsubscriber may be asked to provide identification confirmations and other information for delivery of messages in the future.

For large scale integrated network functionality, interfacing with voice messaging systems introduces a further complication for the providing of identification confirmation services because this service is usually a spoken name and must be replayed to the user via standard telephone equipment. The spoken name is a large, digitized segment of audio that must be stored and replayed to message senders in real time as messages are sent.

Extended Absence Processing

Many messaging systems allow for the extended absence notification to message senders. In essence, if a user will not be checking for messages due to being away from his office, place of business or home for an extended period, or for another reason, a notification can be provided to message senders to apprise them of that fact. This notification can be uploaded to the communications system and recorded in the user's profile in master database 151. Accordingly, instead of or in addition to identification confirmation being provided to a sender attempting to send a message to a recipient with an extended notification in place, the sender of the message will receive the extended absence notification. In this manner, before or after the message is sent, the sender of the message will be notified that the intended recipient of the message will not be checking messages for an extended period of time. The sender may then decide to send the message anyway or pursue other options to contact the intended recipient. For large scale integrated network functionality, interfacing with voice messaging systems introduces a further complication for the providing of extended absence services because this service is a spoken greeting and must be replayed to the user via standard telephone equipment. The extended absence greeting is a large, digitized segment of audio that must be stored and replayed to a sender in real time as messages are sent.

Directory, Addressing, and Routing Services

The communications system of the present invention can also provide messaging systems with directory addressing services. Messaging systems can connect to communications system 10 to utilize the directory and addressing information stored in the communications system 10 to determine how to deliver a message to another messaging system. For example, a messaging system attempting to deliver a message to another messaging system can contact communications system 10 to obtain all the necessary information required to deliver the message based upon whatever addressing information was originally entered by the sender of the message. Communications system 10 then performs the address translation function described previously to retrieve the information needed by the sending messaging system to enable it to make its own connection to the other messaging system to deliver the message. In this manner, messaging systems can access the entire database of user addressing information even if they do not use the communications system 10 to transport and deliver the message.

Billing and Accounting Operations

The billing system 159 within the network center 37 interacts with the central access and control system 155 which in turn stores and processes all of the billing events that occur on communications system 10 to create billing records associated with services performed by the communications system 10. An account is maintained for each user and the costs of services provided by communications system 10 are determined by each user's class of service. Services for delivery of messages may be billed in a similar fashion to that of conventional telephone calls using V&H coordinates with time of day discounts, day of week discounts and other selectable pricing structures.

Additional fees may be charged for other administrative services such as directory and address services, message tracking and revocation, media and protocol conversions, language translation, or the generation of message traffic reports to enable users of the communications system to monitor and account for message expenses.

The message priority system used in the communications system of the present invention and discussed previously allows for the communications system to offer different quality of service levels. For example, while some users may require virtually instantaneous message delivery, many users are willing to settle for longer message delivery times in exchange for discounted pricing structures. The message priority system used by communications system 10 allows the communications system to guarantee maximum message delivery times for the various classes of service.

As discussed previously, the communications system 10 provides an efficient gateway for information providers to provide services to advertisers to access a large number of customers. The communications system allows for mass distribution of information services or advertising materials to selected users who have expressed an interest in particular classes of information using their user profiles. In the case of information services, users may be billed based upon the quantity and quality of information requested.

As an incentive to get users to accept advertising material, the billing system 159 may credit the account associated with a user a small amount when the user accepts and reviews an advertising message. In addition, communications system 10 can interact with users after receipt of an advertising message through interactive voice response system 169, customer computer interface system 167, or directly through the external interface 62 using a real time communication protocol to input orders for these advertised goods. These orders may be optionally billed to the user's account within the communications system 10. In such a case, the billing system 159 will add the charge to the user's account and will also charge the information provider or advertiser a fee for processing the order.

Directory Services

Communications system 10 can provide directory services to both message senders, subscribers and non-subscribers to allow them to locate a particular person to whom they might wish to send a message. As such, communications system 10 can guide a message sender through an automated directory access to the appropriate record in the database of directory information in order to locate the particular message recipient. A variety of systems may be presented to the user depending upon the particular messaging system and its interface with the message sender. For example, if the interface with the user is through a conventional analog line, directory access can be through interactive voice response system 169 using DTMF signals from the user's touchtone phone. The interactive voice response system 169 can first ask the message sender for categories of known information about the message recipient; for example, prompting the message sender to enter on the telephone whatever information is known about the message recipient such as the message recipient's: name, telephone number, city, company, the products and services offered, or other information. Each successive response by the message sender provides additional information which narrows the search through the master database 151 until finally the particular message recipient is identified. At such time, the identification confirmation for the identified message recipient is presented to the message sender to confirm proper message recipient has been identified.

For more complex interfaces with message senders, the automated directory can be accessed using customer computer interface system 167. In this environment, the customer computer interface system 167 can present the message sender with an automated telephone directory in graphical form with entries based on geography, company names or the products and services offered by the message recipient. In this scenario, the location of the message recipient can result in the presentation of the identification confirmation for the identified recipient as well as information identifying the messaging facilities available for the identified recipient. In this manner, the message sender can select the media of delivery of the message, if supported, as determined by the user profile information of the message recipient.

For large scale integrated network functionality, interfacing with voice messaging systems introduces further complications for the providing of directory and addressing operations services in that the address which is used for delivery and routing must be created, maintained, and accessed in the user profile records. Additionally, textual search criteria must be entered using a ten-digit numeric keypad. Further, the providing of an interactive voice response capability must be performed using standard telephone equipment using DTMF signaling, addressing must be numerical, voice or spoken, spelled text must be the only media, identification confirmation must be spoken, and directory access/inquiry must be performed using DTMF signaling.

Language Translation

According to a further aspect of the present invention, the communications system 10 is operable to translate messages between different languages to accommodate messaging between users, both subscriber and non-subscriber. An entry in each user profile indicates the user's language of choice. This indication is used by the communications system to determine the language used by the user as a sender of a message and the preferred language for receipt of messages. The language of choice indications of the sender and the recipient are used by the communications system to select the appropriate translator to use on the message. Regardless of the media of the incoming message, be it a compound message or otherwise, communications system 10 translates all parts of the message into a textual format to allow for the automated translation of the text message to take place.

The translated message may then be converted into the media supported by the messaging system of the message sender.

Private Addressing Plans

Many existing private voice messaging systems allow access to voice mailboxes within the system using a variety of entry methods. Parties using the existing voice messaging system can use short form addresses or private addressing methodologies to access the voice mailboxes of other persons using the same voice messaging system. Routing and delivering messages from the general public via a connection to a network such as the communications system 10 of the present invention is not a problem because the public address of these voice mailboxes is ordinarily a unique number, such as an international or ten digit telephone number. However, the private numbering plans and private addressing methodologies are problematic if independent private voice messaging systems are to be networked using shared facilities because different parties on different private messaging systems could conceivably have the same private mailbox number. In this sense, private numbering schemes create problems because the private numbers are not globally unique numbers. The system of the present invention allows the networking of multiple systems that use private addressing plans by implementing a community addressing scheme and by treating a single actual messaging system that needs both public and private messaging services as two virtual messaging systems, a public messaging system and a private messaging system.

FIGURE 15 is a block diagram which illustrates the implementation of the present invention that provides for networking of messaging systems using private addressing plans through the implementation of community addressing. Referring to FIGURE 15, a hub 200 is connected to the communication cloud 18 described with reference to FIGURE 1 previously. Hub 200 is constructed identically as hubs 12, 14 and 16 described previously. Hub 200 is connected to messaging systems 202, 204, 206, and 208. Messaging system 208 is a conventional messaging system which desires only public messaging traffic. As such, messaging system 208 accesses hub 200 through a single port 210. Messaging systems 202, 204 and 206 need both public and private messaging services. For example, messaging systems 202, 204 and 206 could comprise three different messaging systems all owned by the same company. This company may desire to allow employees of the company to send messages to one another using short form addressing or some other private addressing plan. These short form addresses may comprise globally non-unique numberes and, as such, may conflict with other short form addresses of other parties using the communications system 10 of the present invention. As such, the hub 200 must have a way of identifying whether or not private or public addressing is desired by a particular messaging system. This is accomplished by providing two ports into the hub 200 for the messaging systems 202, 204 and 206. For example, messaging system 202 accesses hub 200 through a public port 212 and a private port 214. From the perspective of the hub 200, messaging system 202 comprises two virtual messaging systems 202a and 202b. Messaging system 202a belongs to the public community and is allowed to message with other public traffic. In comparison, virtual messaging system 202b is a member of a private community and is only allowed to message with other members of that private community.

Similarly, messaging system 204 accesses hub 200 through a public port 216 and a private port 218. As such, messaging system 204 appears as a public virtual messaging system 204a and a private virtual messaging system 204b.

Finally, messaging system 206 accesses hub 200 through a public port 220 and a private port 222. As such, message system 206 appears to hub 200 as a public virtual messaging system 206a and a private virtual messaging system 206b.

Messaging systems 202, 204 and 206 are programmed to call a different number associated with hub 200 depending upon whether or not the message to be transferred to the communications system 10 is a message intended for another member of a private community or a message intended for a recipient outside of the private community associated with systems 202, 204 and 206. The two different numbers associated with hub 200 correspond to the public and private ports described previously. Although FIGURE 15 shows each of systems 202, 204 and 206 having two different access ports, it should be understood that the telephone numbers to be dialed and, as such, the access ports into hub 200 can be shared by many external messaging systems. The only requirement is that each messaging system contact a different number for private and public message traffic. For example, virtual messaging system 202a, 204a and 206a could all use public access port 212. In addition, virtual messaging system 202b, 204b and 206b could all use private access port 214. This is true because the hub 200 can use the call line identification information associated with the call into hub 200 to identify which messaging system is contacting the particular port 212 or 214. Even if messaging systems 202, 204 and 206 were all connected to access ports 212 and 214, the hub 200 could use the call line identification information and the information as to which of the ports 212 or 214 was being accessed to identify a single virtual messaging system associated with the call.

FIGURE 16 is a representation of a translation table that may be used by a hub such as hub 200 to route messages in public and private messaging contexts. FIGURE 16 shows eight table entries associated with three different users of communications system 10 and particularly hub 200. The three users are Arnie, Jane and Frank. Arnie is a public user of messaging system 208 shown in FIGURE 15. He is not a member of any private messaging community. As such, the community variable is set to zero in the two table entries associated with Arnie which indicates a member of the public community. Arnie's user i.d. is "3" and his local mailbox address is "4437", whereas his global mailbox address is "408-555-4437".

Jane is a user of messaging system 202 which resides in Dallas, Texas. Messaging system 202 as discussed previously is divided into a public messaging system 202a which is denoted in FIGURE 16 as "PbDalTx". In addition, messaging system 202 comprises a private messaging system 202b which is designated in FIGURE 16 as "PrDalTx". Jane has a user i.d. of "6" and is a member of both the public community through the PbDalTx virtual messaging system and a private community through the PrDalTx virtual messaging system. The two entries for Jane associated with the PrDalTx virtual messaging system include a community designation of "101" which designates Jane as a member of the 101 private community. To illustrate the potential problems associated with non-unique mailbox identifications, Jane also has a short form mailbox identifier of "4437". Jane's global mailbox identifiers are "374437" for the private messaging network and "214-555-4437" for the global identifier for the public messaging network.

The last user illustrated is Frank who has a user i.d. of "8" and uses messaging system 204. Frank is only a member of the private virtual messaging system 204b which is designated as virtual messaging system "PrAusTx". Frank's global mailbox address for community 101 is "274437". Frank's local mailbox address on PrAusTx is also "4437", the same as Jane's and Arnie's. As will be seen in the example in FIGURE 17, because of the use of virtual messaging systems and community identifiers, the fact that all three participants have the same local mailbox number does not create any problems.

FIGUREs 17a through 17e show how hub 200 uses the information from the virtual information systems as well as the translation table shown in FIGURE 16 to receive a message, deliver a message and to include with the delivered message the appropriate sender addressing required by the messaging protocol. In the example in FIGURE 17, a message will be sent from Jane to Frank. As discussed previously, Jane and Frank are both employees of the same company and, as such, are accustomed to using six digit identifications in order to message with the company. Because of this, when Jane sends her message to Frank, she inputs a recipient address of "274437". Jane utilizes messaging system 202 which will contact hub 200 through private access port 214. Messaging system 202 is programmed to route messaging traffic to the private access port 214 when the messaging traffic is within the private community.

Referring to FIGURE 17a, the hub 200 uses the ANI and DNIS call line identification information to identify the source voice messaging system i.d. and the community. In the example shown in FIGURE 17a, the source voice messaging system i.d. is PrDalTx and the community identifier is 101. The hub 200 was able to obtain this information by noting the telephone number of the call being placed and the fact that the call was placed to access port 214.

Referring now to FIGURE 17b, the hub 200 uses the sender mailbox number, namely Jane's mailbox number of 4437, and the source voice message system i.d. found previously, and determines the sender user i.d. Here, the hub 200 accesses the translation table shown in FIGURE 16 and particularly the fourth column of the access of the translation table which is the first column associated with Jane to find Jane's sender user i.d., which is "6".

In FIGURE 17c, the hub 200 uses the recipient address input by Jane and the community identification calculated earlier to find the recipient user i.d. and the destination virtual messaging system i.d. Here, the hub 200 accesses the last column in FIGURE 16 to identify the destination virtual messaging system as PrAusTx and the recipient user i.d. as "8".

Referring to FIGURE 17d, the hub 200 uses the sender user i.d. and the community to identify the appropriate sender address to include with the message when it is delivered to messaging system 204. Here, the hub 200 accesses the fifth column of FIGURE 16, which is the second column associated with Jane, to locate the sender address of "374437". The hub 200 uses the column associated with Jane which includes a "global" scope for return addresses.

Referring to FIGURE 17e, the hub 200 utilizes the recipient user i.d. and the destination virtual messaging system identification to access the recipient mailbox identifier. In this case, the hub 200 uses the first column associated with Frank to retrieve the recipient mailbox identification of "4437". The hub 200 uses the column associated with Frank which includes a "local" scope to retrieve the appropriate mailbox identifier for messaging traffic being sent to Frank.

Accordingly, the virtual messaging system identification, user identification, scope, and community variables are all used to allow for private addressing plans to be implemented in a network communications system that includes a wide variety of messaging systems and messaging traffic. The members of the company associated with messaging systems 202, 204 and 206 can utilize their private addressing schemes and through the appropriate structure of the address translation tables, messaging traffic will be routed in a virtual private network.

FIGURE 18 illustrates an example of a message being sent from Arnie to a new user, David, through the Internet. The access to the Internet is no different than access to any other virtual messaging system where the Internet comprises a single virtual messaging system identification. As shown in FIGURE 18, the sender mailbox identifier is found in the second column associated with Arnie in FIGURE 16. The recipient address input by Arnie is 214-555-1234. Arnie's source virtual messaging system identifier is "CASJ01". The translation tables are used to find that public addressing to David is to be routed to the destination virtual messaging system of the Internet. The community identifier is zero because the messaging traffic is public. David's recipient user i.d. is "9". The recipient mailbox is then retrieved from David's translation table entry and comprises "David@destination.comm." Arnie's return address is altered slightly to comply with Internet addressing by concatenating an Internet-style address to the end of the sender mailbox identification. This results in a return address of "4085554437@SourceSys.net." An appropriate domain within the Internet is reserved for messaging traffic.

Accordingly, by defining the Internet as a single virtual messaging system within the community and private addressing schemes, messaging traffic can also be routed through the Internet using the same address translation tables.

Summary

Accordingly, a communications system is provided that allows for the addressing, transport, and management of multi-media messages between various messaging facilities with disparate capabilities, media and protocols. The network provides the capability to route messages to users of the network facilities based on any address that is known to a sender. Further, the communications system of the present invention provides translation services such that messages in one media can be delivered to a recipient in another media. The provision of a universal constantly-updated user-profile database containing information about each user provides the ability to support many user specific capabilities to be invoked for various users at various times including messaging system choice; messaging features desired; preferred protocol, media, and language; service messaging; information services protocol desired; advertising content screening; and identification confirmation. Non-subscriber sending and receiving of messages is also supported. Operational capabilities include disaster recover, billing, secure messaging, bulletin boards, message tracking and revocation, as well as the creation of virtual private networks by allowing separate messaging systems to be interconnected through network hub systems. The messaging systems are programmed to contact different access ports within the network hubs to allow the network hubs to determine whether or not private or public messaging traffic is being sent. Address translation tables are then used by the hubs to route the messaging traffic appropriately through either public or private communities and to

provide the expected return address information to the destination messaging system.

Although the present invention has been described in detail, it should be understood that various changes, alterations, and modifications may be made to the systems and methods described herein without departing from the spirit and scope of the present invention which is solely defined by the appended claims.

WHAT IS CLAIMED IS:

1. A communications system, comprising:

a network system operable to be connected to external voice messaging systems;

interface systems resident in the network system and operable to be coupled to the external voice messaging systems, the network system operable to receive communications traffic through the interface systems from the external voice messaging systems, at least two of the external voice messaging systems communicating with the network system using disparate communications protocols; and

database storage resident within the network systems operable to store user profile information associated with users of at least two of the external voice messaging systems, the user profile information comprising routing information for at least some of the users of the system specifying the routing of messages according to the contents of the messages and the messaging facilities available to the users.

2. The s''stem of Claim 1 and further comprising: a network center coupled to the network system through an internal data communication path, the network center containing network support systems for the administration, tracking and operation of the communications system.

3. The system of Claim 2 where the network center comprises a message tracking system operable to store status information associated with messages being processed by the system, the message tracking system operable to report the status of messages being processed by the system and further operable to stop the delivery of messages at the request of the sender of the message.

4. The system of Claim 2 where the network center comprises an central access and control system operable to manage the interaction of the systems within the network center.

5. The system of Claim 4 where the network center comprises an interface system operable to allow users to access the central access and control system to interact with the system.

6. The system of Claim 5 where the interface system comprises a customer computer interface system operable to allow a particular user operating a computer terminal in communication with the customer computer interface system to alter user profile information within the directory information associated with the particular user.

7. The system of Claim 4 where the network center comprises an interactive voice response system operable to allow users to access the central access and control system to interact with the system using a telephone to alter user profile information within the directory information associated with the particular user.

8. The system of Claim 2 where the network center comprises:

a message tracking system operable to store status information associated with messages being processed by the system, the message tracking system operable to report the status of messages being processed by the system and further operable to stop the delivery of messages at the request of the sender of the message;

a central access and control system operable to manage the interaction of the systems within the network center and the interaction of the network operations system with the network hub systems; and

a customer computer interface system operable to allow users operating computer terminals to access the central access and control system to interact with the system, the customer computer interface system operable to allow a particular user operating a computer terminal in communication with the customer computer interface system to receive reports on the status of messages being processed by the system and to stop the delivery of messages sent by the particular user and not yet delivered by the system.

9. The system of Claim 2 where the network center comprises:

a message tracking system operable to store status information associated with messages being processed by the system, the message tracking system operable to report the status of messages being processed by the system and further operable to stop the delivery of messages at the request of the sender of the message;

a central access and control system operable to manage the interaction of the systems within the network center and the interaction of the network center with the network system; and

an interactive voice response system operable to allow users to access the central access and control system to interact with the system using telephones, the interactive voice response system operable to allow a particular user using a telephone in communication with the interactive voice response system to receive reports on the status of messages being processed by the system and to stop the delivery of messages sent by the particular user and not yet delivered by the system.

10. The system of Claim 2 where the network center comprises:

a billing system operable to store and accumulate accounts associated with users of the system and to report the status of the accounts;

an central access and control system operable to manage the interaction of the systems within the network center and the interaction of the network operations system with the network hub systems; and

an interactive voice response system operable to allow users to access the billing system through the central access and control system to interact with the billing system using telephones, the interactive voice response system operable to allow a particular user using a telephone in communication with the interactive voice response system to receive reports on the status of the account associated with the particular user.

11. The system of Claim 2 where the network center comprises:

a billing system operable to store and accumulate accounts associated with users of the system and to report the status of the accounts;

an central access and control system operable to manage the interaction of the systems within the network center and the interaction of the network center with the network system; and

a customer computer interface system operable to allow users operating computer terminals to access the central access and control system to interact with the system, the customer computer interface system operable to allow a particular user operating a computer terminal in communication with the customer computer interface system to receive reports on the status of the account associated with the particular user.

12. The system of Claim 1 where the network system comprises a plurality of network hubs interconnected by internal data communications paths and where each of the external voice messaging systems is coupled to one of the network hubs.

13. The system of Claim 12 where each of the network hubs comprises an analog connection processor operable to connect the system to external messaging systems which utilize analog communications protocols.

14. The system of Claim 12 where each of the network hubs comprises a digital connection processor operable to connect the system to external messaging systems which utilize digital communications protocols.

15. The system of Claim 12 where each of the network hubs comprises a network processor operable to communicate to other network hubs through the internal data communication paths and where the network processor is operable to receive updated information regarding users of the system and updated information regarding status of messages being processed by the system and is operable to update a database within the network hub system to reflect the updated information received from other network hub systems.

16. The system of Claim 12 where each network hub comprises a message storage facility operable to store incoming messages received from external messaging facilities and to store outgoing messages received from other external messaging facilities or other network hub systems.

17. The system of Claim 12 where each network hub comprises:

a message system database operable to store routing information used to route messages through the system;

an inbound message queue operable to store message records associated with messages received by a particular network hub;

an outbound message queue operable to store message records associated with messages to be transmitted from a particular network hub to another network hub or to an external messaging system; and

a message router operable to access the message system database and the inbound message queue in order to construct the message records placed in the outbound message queue.

18. The system of Claim 12 where each network hub system comprises:

an outbound message queue operable to store message records associated with messages to be transmitted from a particular network hub to another network hub or to an external messaging system; and

a connection manager operable to read the message records in the outbound message queue, to determine the connections required to service the outbound message queue message records, and to construct a connection queue comprising connection records ordering the required connections according to the priority of the messages associated with the connections.

19. The system of Claim 12 where each network hub further comprises an event processor operable to read the connection from the connection queues and to create the connections specified by the connection records.

20. The system of Claim 12 where the system further comprises a network center coupled to the network system through an internal data communication path, the network center containing network support systems for the administration, tracking and operation of the communications system and wherein each network hub comprises:

a message system database operable to store routing information used to route messages through the system;

an inbound administrative message queue operable to store message records associated with administrative messages received by a particular hub from another hub, from the network center system, or from an external messaging system;

an outbound administrative message queue operable to store message records associated with administrative messages to be transmitted from a particular hub to another hub, to the network center, or to an external messaging system; and

an administrative event manager operable to access the message system database and the inbound administrative message queue in order to construct the message records placed in the outbound administrative message queue.

21. The system of Claim 20 where each network hub further comprises a connection manager operable to read the message records in the outbound administrative message queue, to determine the connections required to service the outbound administrative message queue message records, and to construct a connection queue comprising connection records ordering the required connections according to the priority of the administrative messages associated with the connections.

22. The system of Claim 20 where the network center comprises a master database comprising directory information for users of the system and where each network hub comprises:

a data replicator operable to communicate with the master database and to receive database changes in order to update databases of user directory information and message system information stored within the network hub.

23. The system of Claim 20 where each network hub comprises a management processor operable to allow the network hub to interact with the network center and to process requests from the network center for status information and for information associated with external messaging systems connected to the network hub.

24. The system of Claim 23 where each network hub comprises an alarms database comprising a list of records specifying past events in the operation of the network hub and where the management processor is operable to manage the alarms database.

25. The system of Claim 12 where each network hub comprises a media translator comprising a plurality of transformation and translation modules, the media translator operable to sequentially use selected ones of the transformation and translation modules to selectively change messages processed by the network hub.

26. The system of Claim 25 where the media translator is operable to selectively translate the media of a message, transform the protocol of a message and translate the language of a message.

27. The system of Claim 1 where the system is operable to receive messages that are encrypted and deliver the messages in encrypted form to provide secure message delivery.

28. A communications system, comprising:
a network system operable to be connected to external messaging systems;

interface systems resident in the network system and operable to be coupled to the external messaging systems, the network system operable to receive communications traffic through the interface systems from the external messaging systems;

database storage resident within the network system operable to store directory information associated with users of the external messaging systems and the communications system; and

information provider databases coupled to the network system and operable to store advertising messages for distribution to selected users of the communications system based on information within user profiles of the selected users stored in the database storage and information associated with information services available to users of the communications system.

29. The system of Claim 28 and further comprising a network center coupled to the network system through an internal data communication path, the network center containing network support systems for the administration, tracking and operation of the communications system.

30. The system of Claim 29 wherein the network center further comprises:
a central access and control system operable to manage the interaction of the systems within the network center

and the interaction of the network operations system with the network hubs; and

an interface system operable to allow users to access the central access and control system to interact with the system, the interface system operable to allow a particular user in communication with the interface system to alter user profile information associated with the particular user to select types of advertising messages to be received by the particular user.

31. The system of Claim 30 where the interface system comprises a customer computer interface system operable to allow users operating computer terminals to access the central access and control system to interact with the system.

32. The system of Claim 30 where the interface system comprises an interactive voice response system operable to allow users to access the central access and control system using telephones.

33. The system of Claim 29 where the network center further comprises:

a central access and control system operable to manage the interaction of the systems within the network center and the interaction of the network operations system with the network hubs; and

an interface system operable to allow users to access the central access and control system to interact with the system, the interface system operable to allow a particular user in communication with the interface system to alter user profile information associated with the particular user to select types of information services to be utilized by the particular user.

34. The system of Claim 33 where the interface system comprises a customer computer interface system operable to allow users operating computer terminals to access the central access and control system to interact with the system.

35. The system of Claim 33 where the interface system comprises an interactive voice response system operable to allow users to access the central access and control system using telephones.

36. The system of Claim 28 wherein the network system comprises a plurality of network hubs interconnected by internal data communications paths and where each of the external messaging systems is coupled to one of the network hubs.

37. A communications system, comprising:

a network system operable to be connected to external messaging systems;

interface systems resident in the network system and operable to be coupled to the external messaging systems, the network system operable to receive communications traffic through the interface systems from the external messaging systems;

database storage resident within the network system operable to store directory information associated with users of the external messaging facilities and the communications system including media translation preference information; and

a media translation system resident in the network system and operable to receive data associated with a message in a first media, access the media translation information, and translate the data such that the message may be delivered in a second media.

38. The system of Claim 37 where each one of at least some of the users of the communications system have media translation information stored in user profiles within the database storage and wherein the media translation system is operable to access the media translation information associated with a destination user such that messages are translated into the second media according to the preferences of the destination user.

39. The system of Claim 37 where at least some of the messaging systems coupled to the communications system have media translation information stored in the

database storage and wherein the media translation system is operable to access the media translation information associated with a destination messaging system such that messages are translated into the second media according to the requirements of the destination messaging system.

40. The system of Claim 37 and further comprising a network center coupled to the network system through an internal data communication path, the network center containing network support systems for the administration, tracking and operation of the communications system.

41. The system of Claim 37 wherein the network system comprises a plurality of network hubs interconnected by internal data communications paths and where each of the external messaging systems is coupled to one of the network hubs.

42. A communications system, comprising:

a network system operable to be connected to external messaging systems;

interface systems resident in the network system and operable to be coupled to the external messaging systems, the network system operable to receive communications traffic through the interface systems from the external messaging systems; and

database storage resident within at least one of the network hubs operable to store directory information associated with users of the external messaging systems and the communications system, the directory information including user profile information including default language information associated with at least some of the users specifying a particular language of choice for the users; and

language translation systems resident in the network system and operable to receive messages in a first language and further operable to translate the message into a second language.

43. The system of Claim 42 where the language translation systems each comprise a media translator operable to translate the received messages into a text format, if necessary; and

the media translator further operable to translate the text format message into the second language and to translate the text format message in the second language to a media other than text for delivery.

44. A communications system, comprising:

a network system operable to be connected to external voice messaging systems;

interface systems resident in the network system and operable to be coupled to the external voice messaging systems, the network system operable to receive communications traffic through the interface systems from the external voice messaging systems, at least two of the external voice messaging systems communicating with the network system using disparate communication protocol; and

database storage resident within the network systems operable to store user profile information associated with users of at least two of the external voice messaging systems, the user profile information comprising delivery preference information associated with at least some of the users of the external messaging systems,some of the messages routed to particular users comprising advertising messages identifiable by subject matter information and where a particular user may elect to receive particular advertising messages by electing to receive messages identified by selected subject matter information, th· user profile of the particular user reflecting the particular user's preferences with respect to the advertising messages.

45. The system of Claim 39 and further comprising: a network center coupled to the network system through an internal data communication path, the network center containing network support systems for the administration, tracking and operation of the communications system.

46. The system of Claim 39 where an account is maintained for each user of the system within the network center and where the account of a particular user is credited a predetermined amount for each advertising message delivered to the particular user.

47. The system of Claim 39 where the delivery preference information comprises information specified by the user as to the media in which at least some messages are to be delivered to the user.

48. The system of Claim 39 where at least some of the messages processed by the communications system contain information identifying the subject matter of the message and where the delivery preference information associated with at least some of the users specifies whether delivery is allowed and, if so, the desired delivery location of certain specified classes of message content.

49. The system of Claim 39 where the delivery preference information comprises information specified by the user directing the system on the mode of delivery desired by the user if a messaging feature is not available on the destination external messaging facility.

50. The system of Claim 39 where the directory information further comprises the messaging features that are available in each external messaging system.

51. A communications system, comprising:

a network system operable to be connected to external messaging systems;

interface systems resident in the network system and operable to be coupled to the external messaging systems, the network system operable to receive communications traffic through the interface systems from the external messaging systems; and

database storage resident within the network center operable to store directory information associated with users of the external messaging facilities where the directory information for at least some of the users of the system comprises identification confirmations associated with the user and where an identification confirmation is supplied to at least some of the users of the system when the users attempt to send a message to a user of the system associated with the identification confirmation information.

52. The system of Claim 51 where the identification confirmations comprise data encoding the spoken name of the user associated with the identification confirmations.

53. The system of Claim 51 where the identification confirmations comprise data encoding the picture of the user associated with the identification confirmations.

54. The system of Claim 51 where the identification confirmations comprise data encoding the visual image of the signature of the user associated with the identification confirmations.

55. The system of Claim 51 where the database storage is further operable to store an extended absence greeting comprising the recorded voice of the user associated with the identification confirmations informing would-be senders of messages of the extended absence of the user associated with the identification confirmations.

56. A communications system, comprising:

a network system operable to be connected to external messaging systems;

interface systems resident in the network system and operable to be coupled to the external messaging systems, the network system operable to receive communications traffic through the interface systems from the external messaging systems;

database storage resident within the network system operable to store directory information associated with users of the external messaging systems and the communications system; and

an interface system associated with the network system, the interface system operable to interact with nonsubscribers to obtain messages and delivery information from the nonsubscribers, the network center then operable to route the message received from the nonsubscriber within the system as necessary to deliver the message to the intended recipient.

57. The system of Claim 56 and further comprising a network center coupled to the network system through an internal data communication path, the network center containing network support systems for the

administration, tracking and operation of the communications system, the interface system resident in the network center.

58. The system of Claim 56 where the interface system comprises a customer computer interface system operable to allow users operating computer terminals to access and interact with the system.

59. The system of Claim 56 where the interface system comprises an interactive voice response system operable to allow users to access the system using telephones.

60. A communications system, comprising:

a network system operable to be connected to external messaging systems;

interface systems resident in the network system and operable to be coupled to the external messaging systems, the network system operable to receive communications traffic through the interface systems from the external messaging systems;

database storage resident within the network system operable to store directory information associated with users of the external messaging facilities and the communications system; and

an interface system associated with the network center, the interface system operable to place outgoing calls in order to deliver messages to non-subscribers to the system, the interface system operable to interact with the non-subscribing intended recipient to deliver the message by prompting the person answering the outgoing call to confirm that they are the non-subscribing intended recipient.

61. The system of Claim 60 and further comprising a network center coupled to the network system through an internal data communication path, the network center containing network support systems for the administration, tracking and operation of the communications system, the interface system resident in the network center.

62. The system of Claim 60 where the interface system is further operable to ask the person answering the outgoing call to enter an alternate destination telephone number where the non-subscribing intended recipient can be reached if the person answering the outgoing call indicates that they are not the non-subscribing intended recipient.

63. The system of Claim 62 where the interface system is further operable to repeatedly attempt to place the outgoing call a predetermined number of times at predetermined intervals if the outgoing call is not answered or if the outgoing call is answered but the non-subscribing intended recipient is not there to accept delivery of the message.

64. A communications system, comprising:

a network system operable to be connected to external messaging systems;

interface systems resident in the network system and operable to be coupled to the external messaging systems, the network system operable to receive communications traffic through the interface systems from the external messaging systems;

database storage resident within the network system operable to store directory information associated with users of the external messaging facilities and the communications system, the directory information comprising a user record for each user comprising a unique internal identifier, at least one external address associated with the user, and routing information for messaging facilities available to the user; and

the network system operable to receive messages from external messaging systems with an external address of an intended recipient having a user record, the network system further operable to use the external address to access the user record of the intended recipient to retrieve the internal identifier of the intended recipient and the routing information for the messaging facility associated with the intended recipient appropriate for the message to be delivered.

65. The system of Claim 66 where the external address comprises an international telephone number.

66. The system of Claim 64 where the external address comprises a ten digit number based on the North American Numbering Plan.

67. The system of Claim 64 where the external address comprises a number associated with the intended recipient in a private numbering plan.

68. The system of Claim 64 where the external address comprises an electronic mail address associated with the intended recipient.

69. The system of Claim 64 where the external address comprises an Internet address associated with the intended recipient.

70. The system of Claim 64 where the external address comprises an X.400 address associated with the intended recipient.

71. The system of Claim 64 where the network system is operable to deliver messages with information identifying the source of the message and wherein the format of the source address and the destination address are different.

72. The system of Claim 64 where the network system is operable to access media identification information associated with the intended recipient of a message and to retrieve routing information associated with a messaging facility of the intended recipient appropriate for the media of the message to be delivered.

73. The system of Claim 64 where the network system is operable to access subject matter delivery preference information associated with the intended recipient of a message and to retrieve routing information associated with a messaging facility of the intended recipient appropriate for the subject matter of the message to be delivered.

74. A communications system, comprising:

a plurality of network hubs interconnected by internal data communication paths;

interface systems resident in the network hubs and operable to be coupled to external messaging facilities and the internal data communication paths, the network hubs operable to receive communications traffic through the interface systems from the external messaging facilities and the internal data communication paths; and

database storage resident within at least one of the network hubs operable to store directory information associated with users of the external messaging facilities and the communications system; and

a message system database within each particular network hub operable to store configuration information detailing the configuration of each external messaging system connected to the particular network hub such that the messaging system database may be used to reconfigure an external messaging system should the external messaging system be damaged or destroyed.

75. The system of Claim 74 where each network hub comprises an emergency message storage system operable to temporarily store messages intended for users of a damaged or destroyed external messaging system such that the temporarily stored messages can be delivered to the users of the damaged or destroyed external messaging system after the external messaging system is brought back into operation.

76. A communications system, comprising:

a network system operable to be connected to external messaging systems;

interface systems resident in the network system and operable to be coupled to the external messaging systems, the network system operable to receive communications traffic through the interface systems from the external messaging systems;

database storage resident within the network system operable to store directory information associated with users of the external messaging facilities and the communications system; and

a mailing list distribution system operable to accept a message and deliver the message to a pl··-ality of destinations, where the mailing list distribution system is recognized by the system as the sender of the message to the plurality of destinations such that the mailing list distribution system receives all replies and error messages resulting from the delivery of the message to the plurality of destinations.

77. A communications system, comprising:

a network system operable to be connected to external messaging systems;

interface systems resident in the network system and operable to be coupled to the external messaging systems, the network system operable to receive communications traffic through the interface systems from the external messaging systems;

database storage resident within the network system operable to store directory information associated with users of the external messaging facilities and the communications system; and

a mailing list distribution system operable to accept a message and deliver the message to a plurality of destinations, where the mailing list distribution system identifies the user requesting delivery of the message as the sender of the message such that the user requesting delivery of the message receives all replies and error messages resulting from the delivery of the message to the plurality of destinations.

78. A communications system, comprising:

a plurality of network hubs interconnected by internal data communication paths;

interface systems resident in the network hubs and operable to be coupled to external messaging facilities and the internal data communication paths, the network hubs operable to receive communications traffic through the interface systems from the external messaging facilities and the internal data communication paths;

a network center coupled to at least one of the network hubs through an internal data communication path, the network center containing network support systems for the administration, tracking and operation of the communications system; and

database storage resident within at least one of the network hubs operable to store directory information associated with users of the external messaging facilities and the communications system; and

a message tracking system resident in the network center and operable to store status information associated with messages being processed by the system, the message tracking system operable to report the status of messages being processed by the system and further operable to stop the delivery of messages at the request of the sender of the message.

79. The system of Claim 78 and further comprising an interface system operable to allow users to access the message tracking system, the interface system operable to allow a particular user to receive reports on the status of messages being processed by the system and to stop the delivery of messages sent by the particular user and not yet delivered by the system.

80. The system of Claim 79 where the interface system comprises a customer computer interface system operable to allow users operating computer terminals to access and interact with the message tracking system.

81. The system of Claim 79 where the interface system comprises an interactive voice response system operable to allow users to access and interact with the message tracking system using telephones.

82. The system of Claim 78 where an account is maintained for each user of the system within the network center and where the account of the user accessing the message tracking system is charged an amount dependent on the service requested of the message tracking system.

83. A communications system, comprising:

a plurality of network hubs interconnected by internal data communication paths;

interface systems resident in the network hubs and operable to be coupled to external messaging facilities and the internal data communication paths, the network hubs operable to receive communications traffic through the interface systems from the external messaging facilities and the internal data communication paths;

a network center coupled to at least one of the network hubs through an internal data communication path, the network center containing network support systems for the administration, tracking and operation of the communications system; and

database storage resident within at least one of the network hubs operable to store directory information associated with users of the external messaging facilities and the communications system comprising at least one external address and a unique internal identifier for each user of the system; and

an interface system resident in the network center operable to interact with would-be users of the system to identify particular users of the system using information supplied by the would-be users.

84. The system of Claim 83 where the interface system is operable to perform searches of the directory information responsive to parameters supplied by the would-be users of the system and to report the results of the searches to the would-be users by presenting the would-be users with identification confirmations from the users of the system identified by the search.

85. The system of Claim 83 where the interface system may be directly accessed by nonsubscriber users of the system in order to identify users of the system and to send messages to the identified users.

86. The system of Claim 83 where the interface system comprises a customer computer interface system operable to allow users operating computer terminals to access and interact with the system.

87. The system of Claim 83 where the interface system comprises an interactive voice response system operable to allow users to access and interact with system using telephones.

88. A communications system, comprising:

a network system operable to be connected to external messaging systems;

interface systems resident in the network system and operable to be coupled to the external messaging systems, the network system operable to receive communications traffic through the interface systems from the external messaging systems;

database storage resident within the network system operable to store directory information associated with users of the external messaging facilities and the communications system, the external messaging systems operable to request message routing information for messages received from users of the external messaging systems, the network hubs operable to access the directory information and retrieve the appropriate routing information and further operable to transmit the retrieved routing information to the external messaging system such that the external messaging system can deliver the message independent of the communications system.

89. A network hub of a communications system operable to interact within that communications system with other network hubs through internal data lines, the network hub comprising:

a plurality of computer systems, each comprising a local area network adaptor system;

a local area network coupled to each of the local area network adaptor systems within the plurality of computer systems;

a first one of the plurality of computer systems comprising certain digital connection processor systems each coupled to at least one external messaging facility through communication paths utilizing digital communications protocols;

a second one of the plurality of computer systems comprising certain analog connection processor systems each coupled to at least one external messaging system through communication paths utilizing analog communications protocols;

a third one of the plurality of computer systems comprising a database server system operable to access and manage a database, the database comprising directory information comprising records containing profile information and operational preference information for users of the communications network, the network hubs operable to receive updates to the directory information from other network hubs and alter the information stored in the database; and

a fourth one of the plurality of computer systems comprising a communications system operable to manage communications between the network hub and other network hubs through the high speed communication lines coupled to the fourth computer system.

90. A network hub of a communications system operable to interact within that communications system with other network hubs through internal data lines, the network hub comprising:

a management processor operable to monitor the operation of the network hub;

a plurality of control processors;

an event processor operable to schedule events in the network hub;

a hub database operable to store user profile information associated with users of messaging systems coupled to the network hub;

a connection processor operable to create and manage connections between the network hub and other systems; and

a file server providing access to a message storage facility.

91. The network hub of Claim 90 wherein the connection processor comprises an analog connection processor coupled to at least one external messaging system through communication paths utilizing analog communications protocols.

92. The network hub of Claim 90 wherein the connection processor comprises a digital connection processor coupled to at least one external messaging system through communication paths utilizing digital communications protocols.

93. The network hub of Claim 90 wherein the connection processor comprises a network connection processor operable to create and manage connections with other network hubs.

94. The network hub of Claim 90 and further comprising a media translator operable to receive a message in a first media and to translate the message into a second, different media prior to delivery of the message to an intended recipient.

95. The network hub of Claim 90 wherein the plurality of control processors comprise:

a connection manager operable to determine the necessity of future connections and to build a connection queue comprising records specifying the necessary future connections;

a message router operable to interact with message system information in the hub database to determine the next destination for each message received by the network hub;

a data replicator operable to interact with other systems to keep the hub database synchronized with the information contained in other network hubs; and

an administrative event manager operable to schedule and manage administrative events within the network hub.

96. The network hub of Claim 90 wherein the connection processor comprises a client process to request services from the event processor, management processor and file server.

97. A communications system, comprising:

a plurality of network hubs interconnected by internal data communication paths;

interface systems resident in the network hubs and operable to be coupled to external messaging facilities and the internal data communication paths, the network hubs operable to receive communications traffic through the interface systems from the external messaging facilities and the internal data communication paths;

a network center coupled to at least one of the network hubs through an internal data communication path, the network center containing network support systems for the administration, tracking and operation of the communications system; and

database storage resident within at least one of the network hubs operable to store directory information associated with users of the external messaging facilities and the communications system, the directory information comprising a user record for each user comprising information associated with at least one delivery feature, the user record further comprising information indicating the actions to be taken by the system in the event that the delivery feature is not supported by the messaging system associated with an intended message recipient.

98. The system of Claim 97 where the delivery feature comprises the ability to designate a selected message as urgent.

99. The system of Claim 98 where the information indicating the actions to be taken may be configured to cause the message to be returned if the urgency feature is not supported.

100. The system of Claim 98 where the information indicating the actions to be taken may be configured to cause the message to be delivered even if the urgency feature is not supported.

101. The system of Claim 97 where the delivery feature comprises the ability to designate a selected message as private.

102. The system of Claim 101 where the information indicating the actions to be taken may be configured to cause the message to be returned if the privacy feature is not supported.

103. The system of Claim 101 where the information indicating the actions to be taken may be configured to cause the message to be delivered even if the privacy feature is not supported.

104. A method of processing a message using a communications network, the message comprising a compound message having a plurality of message portions having different message medias requiring different destination messaging facilities, them method comprising the steps of:

receiving the message from an external messaging system in a source message format associated with a source protocol used by that messaging system;

translating the message into a network format in a first network hub;

transmitting the message in the network format over internal data communications lines connecting the first network hub to a second network hub;

translating the message from the network format to a destination message format associated with a destination communications protocol in the second network hub, the destination protocol associated with a destination messaging system coupled to the second network hub;

separating the message portions;

identifying the messaging facilities available to the intended recipient;

translating the message portions that use medias associated with messaging facilities not available to the intended recipient into media associated with messaging facilities that are available to the intended recipient;

transmitting the message in the destination message format to the destination messaging system such that the message can be delivered to an intended recipient associated with the destination messaging system; and delivering the translated message portions and the untranslated message portions to the appropriate messaging facilities available to the intended recipient.

105. A method of processing a message using a communications network comprising:

receiving the message from an external messaging system in a source message format associated with a source protocol used by that messaging system;

retrieving identification confirmations associated with the intended recipient of the message from a database of user profile information; and

transmitting the retrieved identification confirmations to the source messaging facility such that the identification confirmations can be provided to the sender of the message to confirm the intended recipient of the message;

translating the message into a network format in a first network hub;

transmitting the message in the network format over internal data communications lines connecting the first network hub to a second network hub;

translating the message from the network format to a destination message format associated with a destination communications protocol in the second network hub, the destination protocol associated with a destination messaging system coupled to the second network hub; and

transmitting the message in the destination message format to the destination messaging system such that the message can be delivered to an intended recipient associated with the destination messaging system.

106. A method of processing a message using a communications network comprising:

receiving initial address information as to the identity of a destination user from a source user of the system;

accessing a user profile database using the initial address information to retrieve a unique internal identifier and identification confirmation for the destination user;

transmitting the retrieved identification confirmations to the source user to confirm the intended recipient of the message;

receiving a message from the source user; and routing the message to the destination user using the unique internal identifier.

107. The method of Claim 106 and further comprising the steps of:

accessing delivery preference information for the destination user; and

routing the message to a particular messaging facility associated with the destination user responsive to the delivery preference information.

108. The method of Claim 106 where the media of the message corresponds to a messaging facility that is not supported by the destination user, the method further comprising the steps of:

translating the message to an internal format; routing the message to a network hub connected to the destination user in the internal format;

translating the message from the internal format into a media supported by a messaging facility associated with the destination user; and

delivering the translated message to the messaging facility associated with the destination user.

109. A communications system, comprising:

a hub system comprising at least one public access port and at least one private access port;

a messaging system operable to contact and connect to both the private and public access ports, the messaging system accessible to users and operable to receive and deliver messages from and to the users of the messaging system where at least some of the users are able to use public addressing and private addressing forms to address messages; and

the network hub system comprising stored user tables comprising community information identifying particular users who are able to use private addressing forms to route messages to each other such that such users may use globally non-unique private addressing forms to address messages.

110. The communications system of Claim 1 wherein the messaging system comprises a first messaging system and wherein the network hub system comprises a plurality of both private and public access ports, the system further comprising a second messaging system operable to contact and connect to both a private and a public access port, the user tables including entries including community information for at least some users of both the

first and second messaging systems belonging to the same private messaging community.

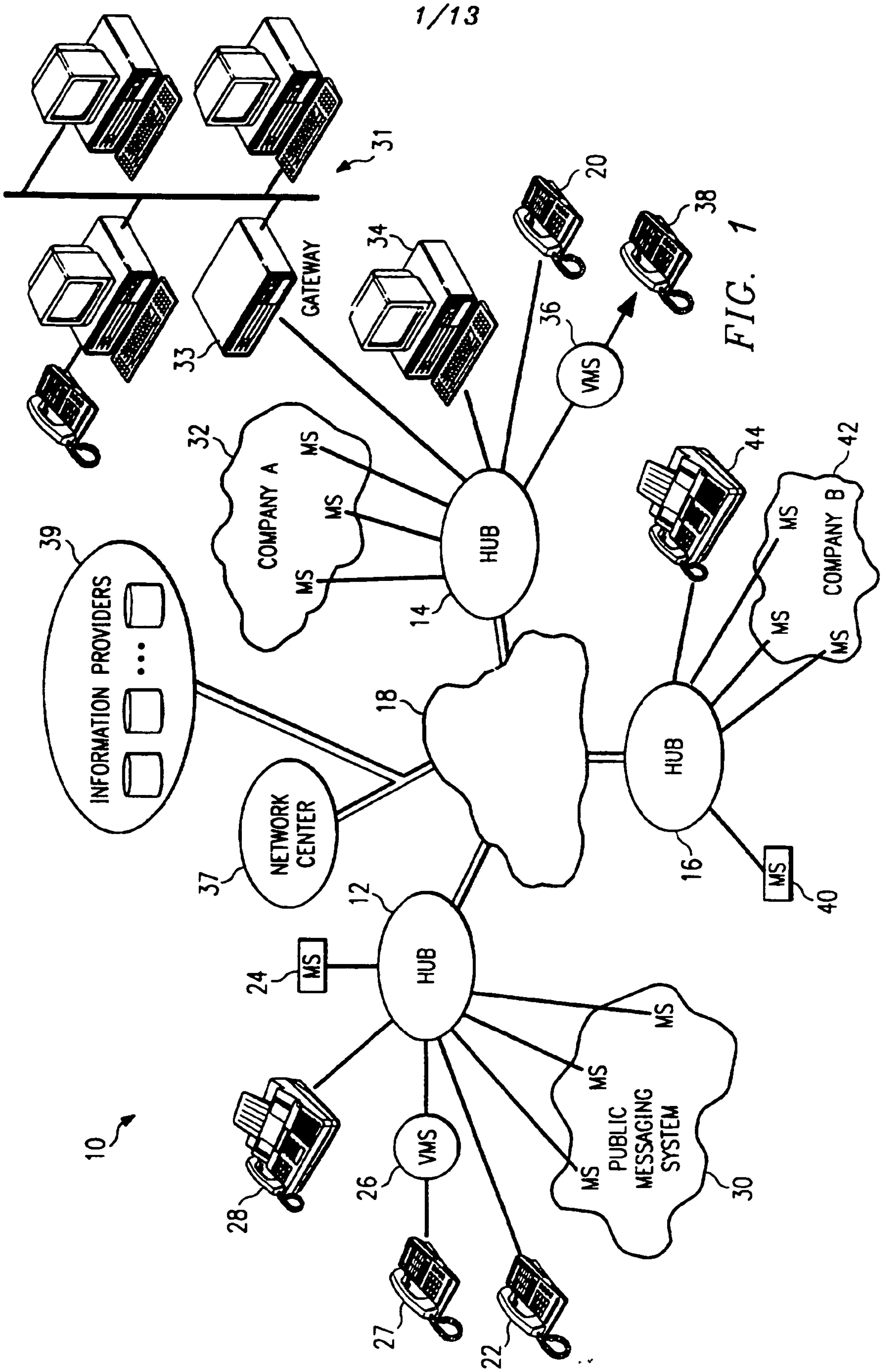

INFORMATION PROVIDERS
NETWORK CENTER
COMPANY A
COMPANY B
HUB
HUB
HUB
MS
VMS
GATEWAY
PUBLIC MESSAGING SYSTEM
10
12
14
16
18
20
22
24
26
27
28
30
31
32
33
34
36
37
38
39
40
42
44

FIG. 1

FIG. 2

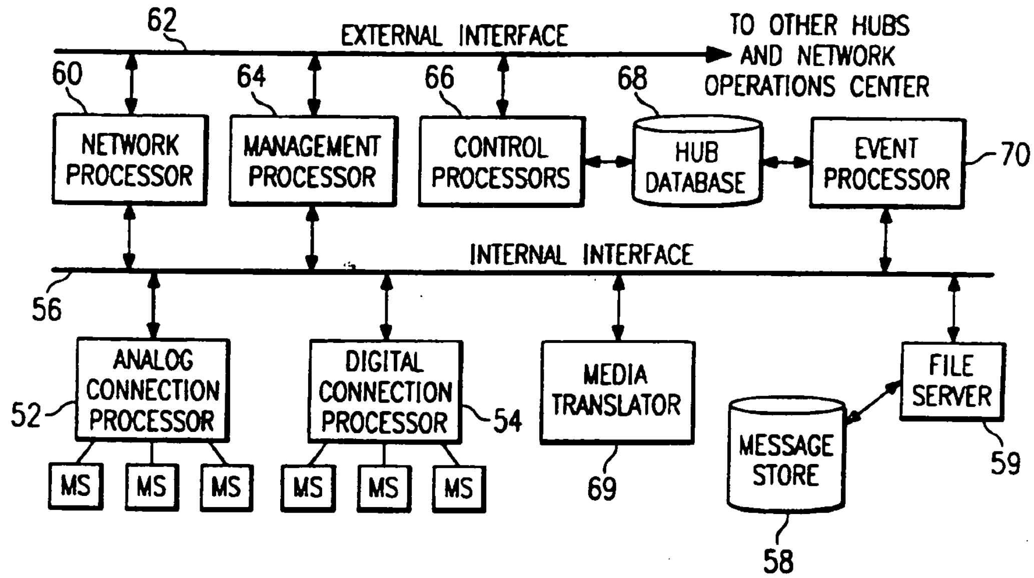

62
EXTERNAL INTERFACE
TO OTHER HUBS AND NETWORK OPERATIONS CENTER
60
64
66
68
NETWORK PROCESSOR
MANAGEMENT PROCESSOR
CONTROL PROCESSORS
HUB DATABASE
EVENT PROCESSOR
70
INTERNAL INTERFACE
56
ANALOG CONNECTION PROCESSOR
52
DIGITAL CONNECTION PROCESSOR
54
MEDIA TRANSLATOR
69
FILE SERVER
59
MESSAGE STORE
58
MS
MS
MS
MS
MS
MS

FIG. 4

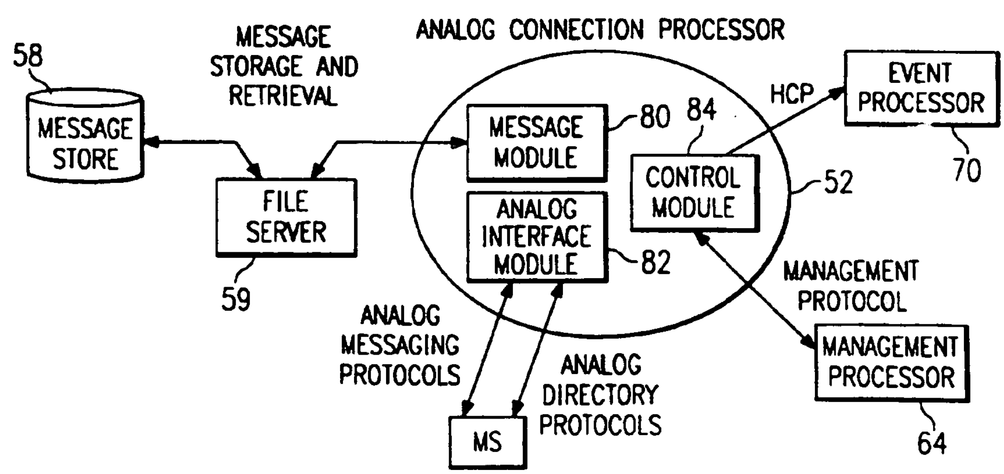

ANALOG CONNECTION PROCESSOR
58
MESSAGE STORE
MESSAGE STORAGE AND RETRIEVAL
FILE SERVER
59
MESSAGE MODULE
80
84
ANALOG INTERFACE MODULE
82
CONTROL MODULE
52
HCP
EVENT PROCESSOR
70
MANAGEMENT PROTOCOL
MANAGEMENT PROCESSOR
64
ANALOG MESSAGING PROTOCOLS
ANALOG DIRECTORY PROTOCOLS
MS

FIG. 3

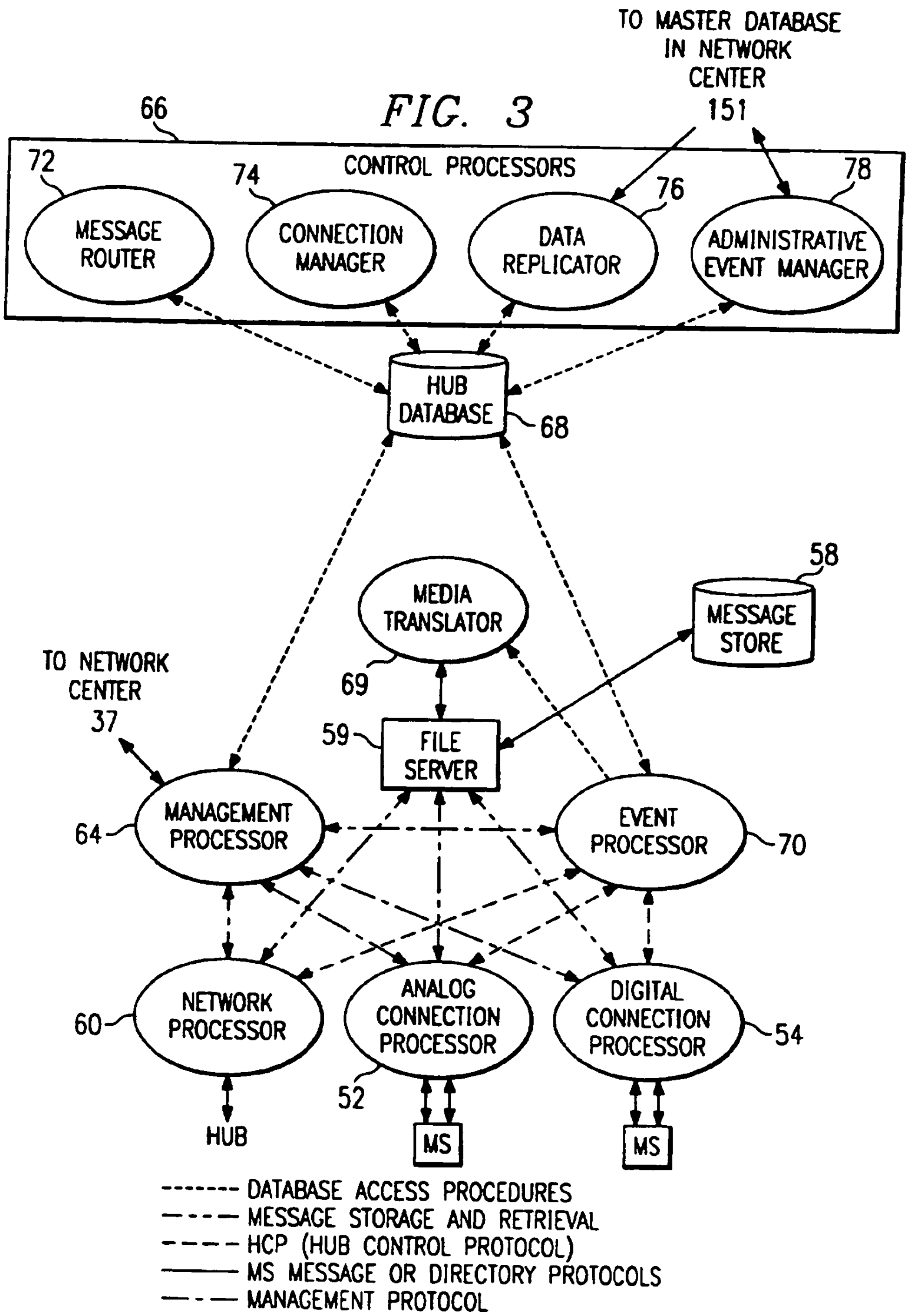

TO MASTER DATABASE IN NETWORK CENTER
151
66
CONTROL PROCESSORS
72
MESSAGE ROUTER
74
CONNECTION MANAGER
76
DATA REPLICATOR
78
ADMINISTRATIVE EVENT MANAGER
HUB DATABASE
68
58
MEDIA TRANSLATOR
MESSAGE STORE
69
TO NETWORK CENTER
37
59
FILE SERVER
MANAGEMENT PROCESSOR
64
EVENT PROCESSOR
70
NETWORK PROCESSOR
60
ANALOG CONNECTION PROCESSOR
DIGITAL CONNECTION PROCESSOR
54
52
HUB
MS
MS
DATABASE ACCESS PROCEDURES
MESSAGE STORAGE AND RETRIEVAL
HCP (HUB CONTROL PROTOCOL)
MS MESSAGE OR DIRECTORY PROTOCOLS
MANAGEMENT PROTOCOL

FIG. 5

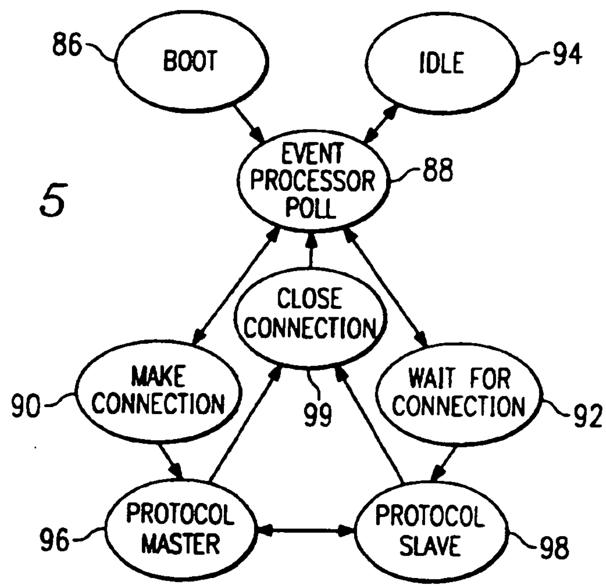
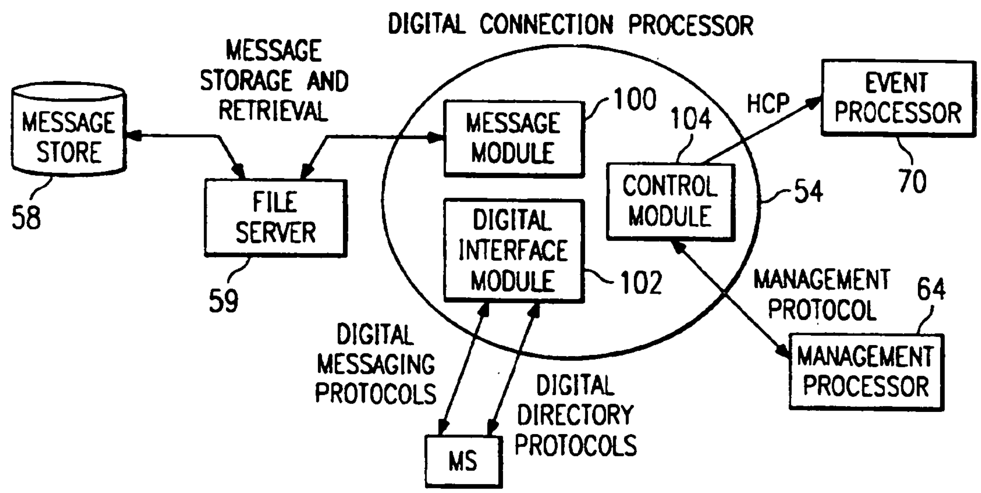

86
BOOT
IDLE
94
EVENT PROCESSOR POLL
88
CLOSE CONNECTION
MAKE CONNECTION
90
99
WAIT FOR CONNECTION
92
PROTOCOL MASTER
96
PROTOCOL SLAVE
98

FIG. 6


DIGITAL CONNECTION PROCESSOR
MESSAGE STORAGE AND RETRIEVAL
MESSAGE STORE
58
FILE SERVER
59
MESSAGE MODULE
100
104
HCP
EVENT PROCESSOR
70
CONTROL MODULE
54
DIGITAL INTERFACE MODULE
102
MANAGEMENT PROTOCOL
64
MANAGEMENT PROCESSOR
DIGITAL MESSAGING PROTOCOLS
DIGITAL DIRECTORY PROTOCOLS
MS

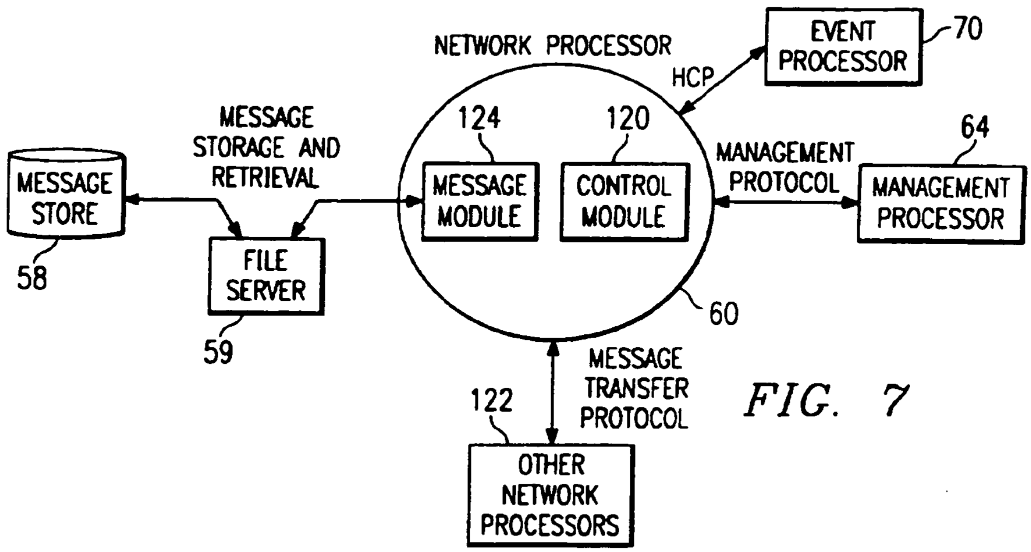


FIG. 7

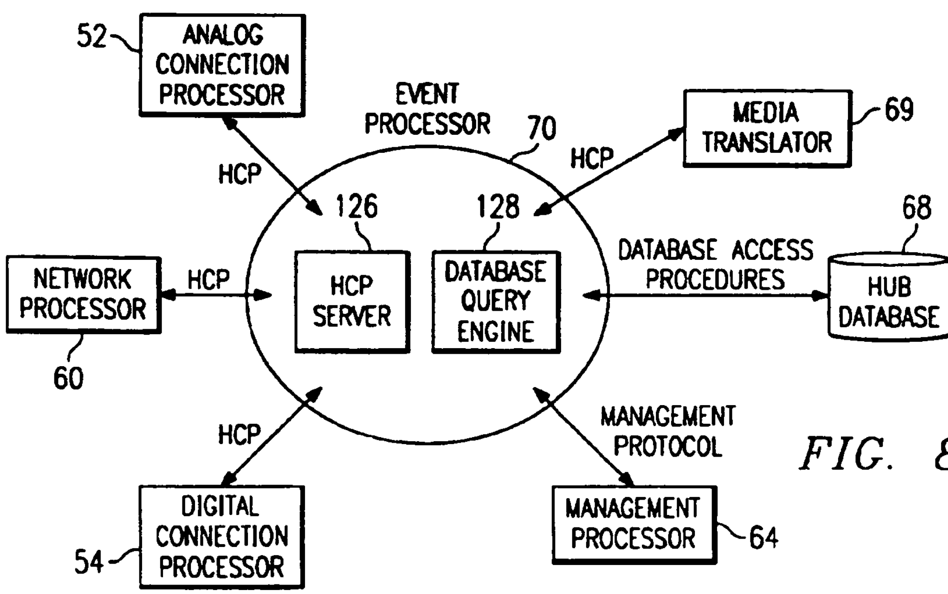


FIG. 8

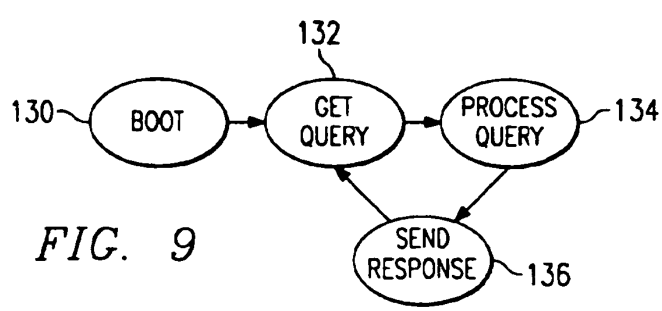
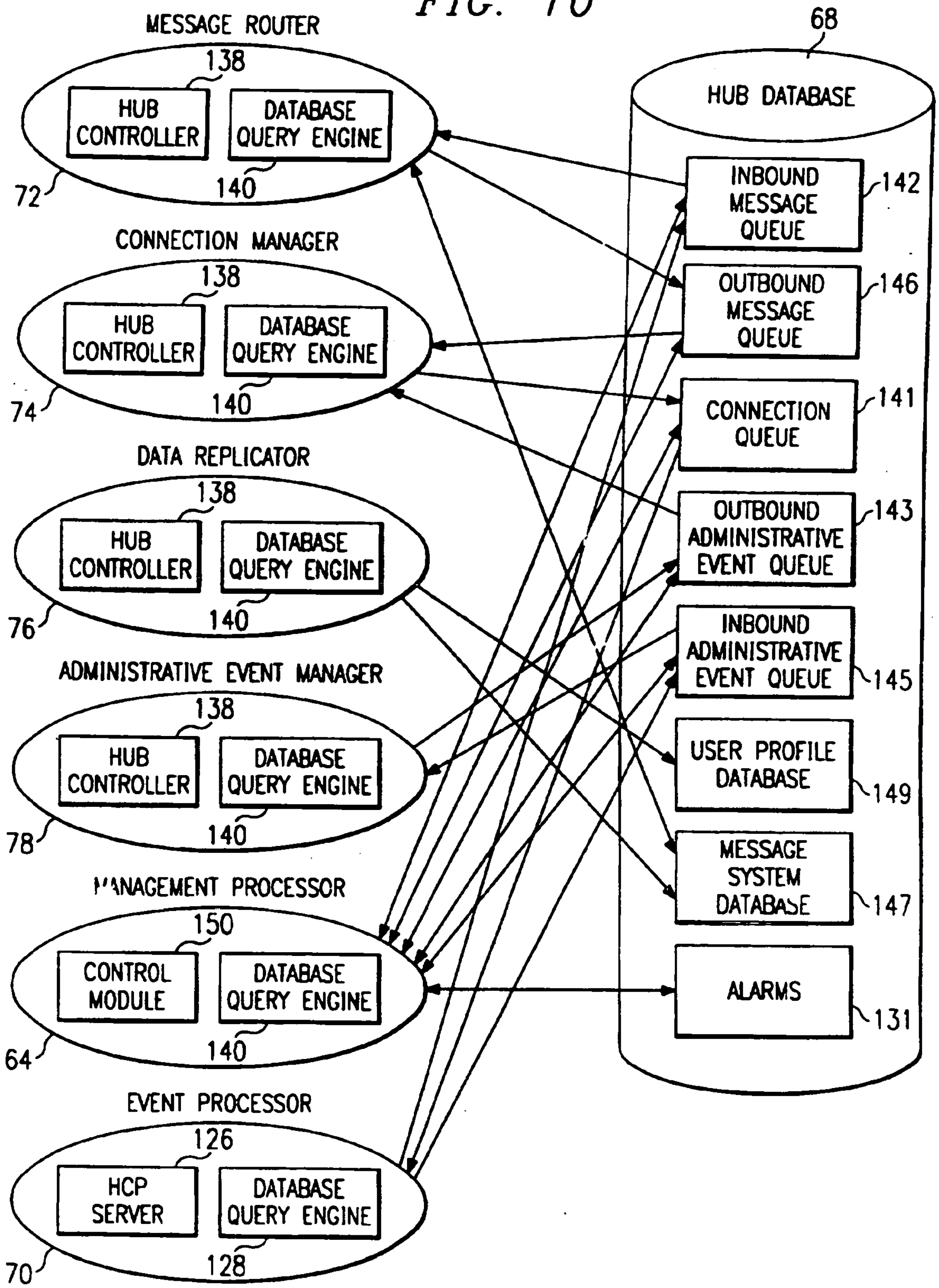


FIG. 9

FIG. 10


MESSAGE ROUTER
138
HUB CONTROLLER
DATABASE QUERY ENGINE
140
72
CONNECTION MANAGER
138
HUB CONTROLLER
DATABASE QUERY ENGINE
140
74
DATA REPLICATOR
138
HUB CONTROLLER
DATABASE QUERY ENGINE
140
76
ADMINISTRATIVE EVENT MANAGER
138
HUB CONTROLLER
DATABASE QUERY ENGINE
140
78
MANAGEMENT PROCESSOR
150
CONTROL MODULE
DATABASE QUERY ENGINE
140
64
EVENT PROCESSOR
126
HCP SERVER
DATABASE QUERY ENGINE
128
70
68
HUB DATABASE
INBOUND MESSAGE QUEUE
142
OUTBOUND MESSAGE QUEUE
146
CONNECTION QUEUE
141
OUTBOUND ADMINISTRATIVE EVENT QUEUE
143
INBOUND ADMINISTRATIVE EVENT QUEUE
145
USER PROFILE DATABASE
149
MESSAGE SYSTEM DATABASE
147
ALARMS
131

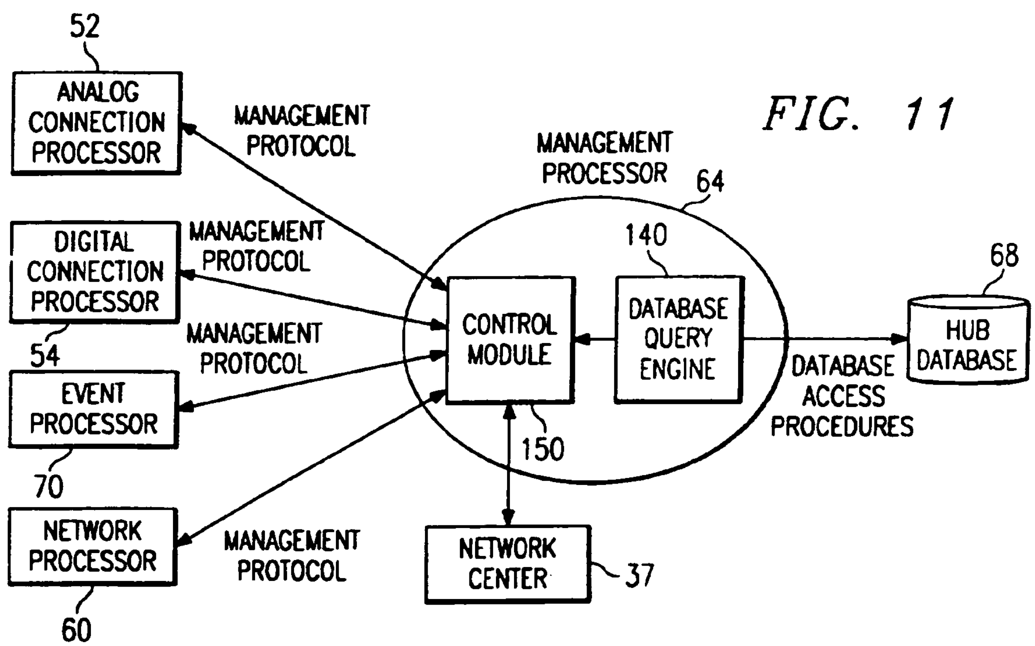

52
ANALOG CONNECTION PROCESSOR
MANAGEMENT PROTOCOL
MANAGEMENT PROCESSOR
64
DIGITAL CONNECTION PROCESSOR
MANAGEMENT PROTOCOL
140
68
54
MANAGEMENT PROTOCOL
CONTROL MODULE
DATABASE QUERY ENGINE
HUB DATABASE
EVENT PROCESSOR
DATABASE ACCESS PROCEDURES
150
70
NETWORK PROCESSOR
MANAGEMENT PROTOCOL
NETWORK CENTER
37
60

FIG. 11

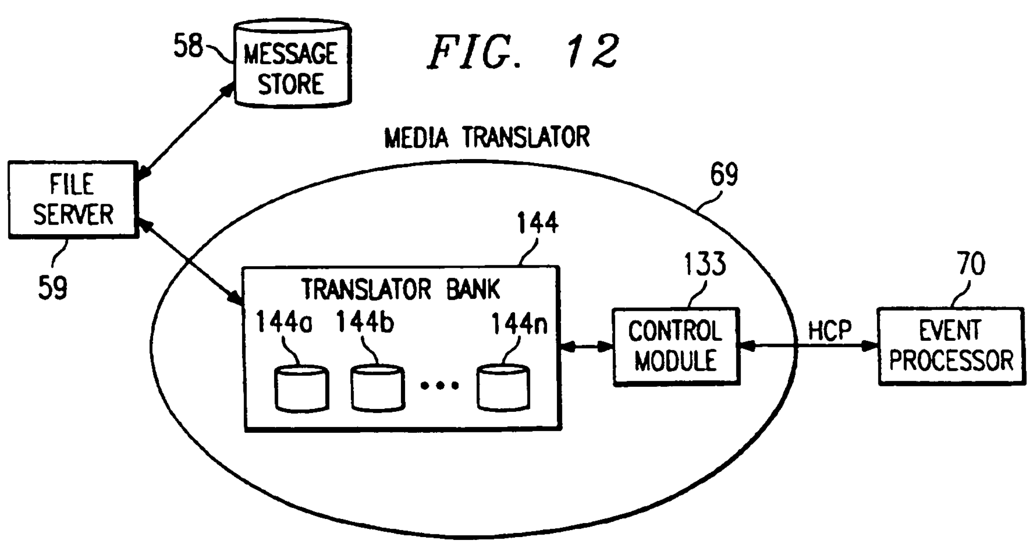

58
MESSAGE STORE
MEDIA TRANSLATOR
69
FILE SERVER
144
59
133
70
TRANSLATOR BANK
144a 144b 144n
CONTROL MODULE
HCP
EVENT PROCESSOR

FIG. 12

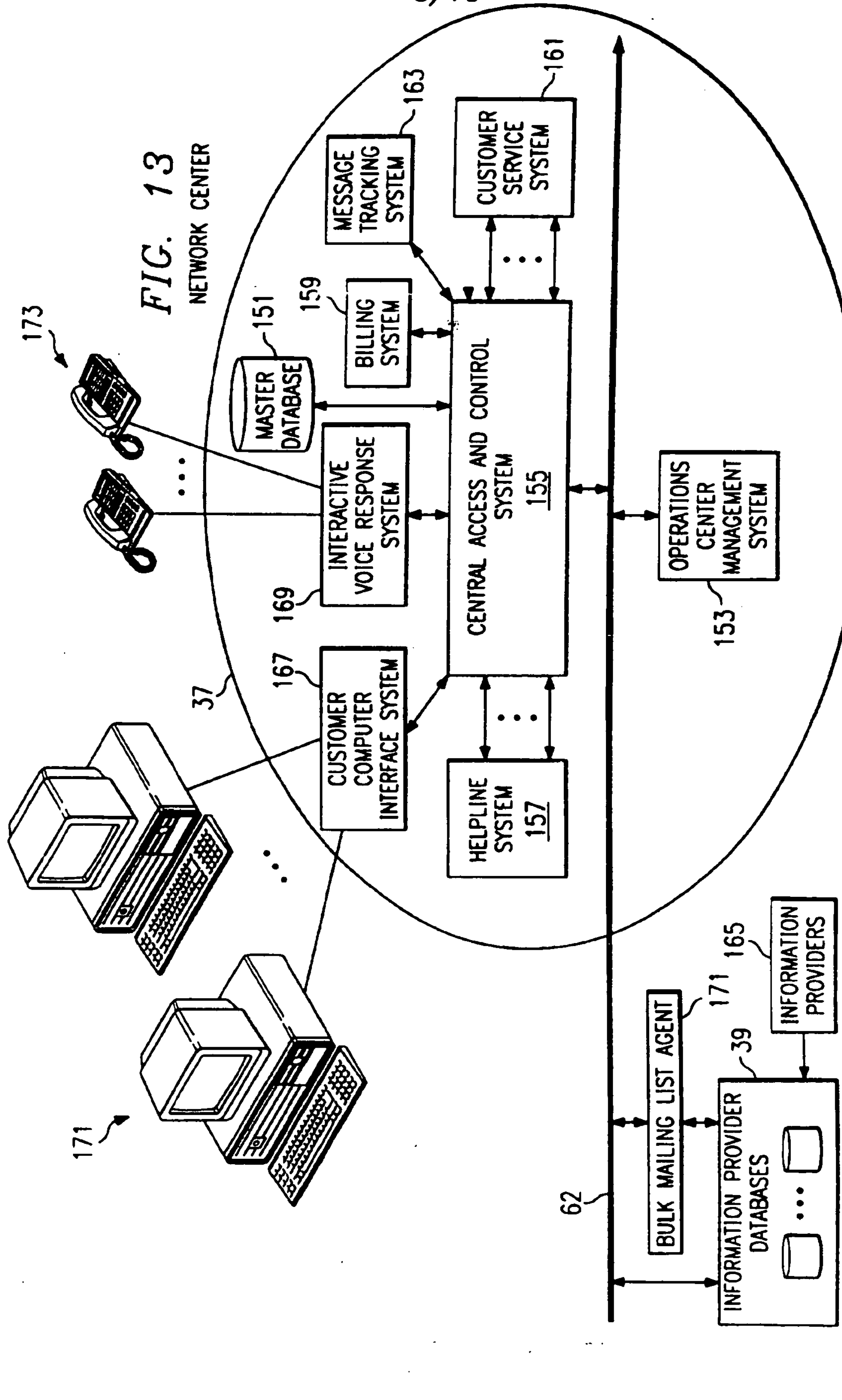

NETWORK CENTER
MESSAGE TRACKING SYSTEM
163
CUSTOMER SERVICE SYSTEM
161
BILLING SYSTEM
159
MASTER DATABASE
151
INTERACTIVE VOICE RESPONSE SYSTEM
169
CENTRAL ACCESS AND CONTROL SYSTEM 155
OPERATIONS CENTER MANAGEMENT SYSTEM
153
CUSTOMER COMPUTER INTERFACE SYSTEM
167
HELPLINE SYSTEM 157
37
173
171
BULK MAILING LIST AGENT
171
INFORMATION PROVIDERS
165
INFORMATION PROVIDER DATABASES
39
62

FIG. 13

FIG. 14

DESTINATION ADDRESS ⟹ 214-555-2722
MESSAGE MEDIA ⟹ VOICE
SOURCE ADDRESS ⟹ 4437
SOURCE MESSAGING SYSTEM ⟹ CASJ01

INBOUND NETWORK HUB TABLE

	ARNIE	ARNIE	GREG
ADDRESS	4437	408-555-4437	214-555-2722
MS	CASJ01	CASJ01	TXDL03
USER ID	3	3	1
COMMUNITY	0	0	0
MEDIA	VOICE	VOICE	VOICE, FAX
SCOPE	LOCAL	GLOBAL	LOCAL, GLOBAL
FEATURES	ALL	ALL	ALL
SUBJECT MATTER	ALL	ALL	ALL

SOURCE USER ID ⟹ 3
COMMUNITY ⟹ 0
DESTINATION USER ID ⟹ 1
DESTINATION MESSAGING SYSTEM ⟹ TXDL03

OUTBOUND NETWORK HUB TABLE

	ARNIE	ARNIE	GREG
ADDRESS	4437	408-555-4437	214-555-2722
MS	CASJ01	CASJ01	TXDL03
USER ID	3	3	1
COMMUNITY	0	0	0
MEDIA	VOICE	VOICE	VOICE, FAX
SCOPE	LOCAL	GLOBAL	LOCAL, GLOBAL
FEATURES	ALL	ALL	ALL
SUBJECT MATTER	ALL	ALL	ALL

DESTINATION ADDRESS ⟹ 214-555-2722
SOURCE ADDRESS ⟹ 408-555-4437

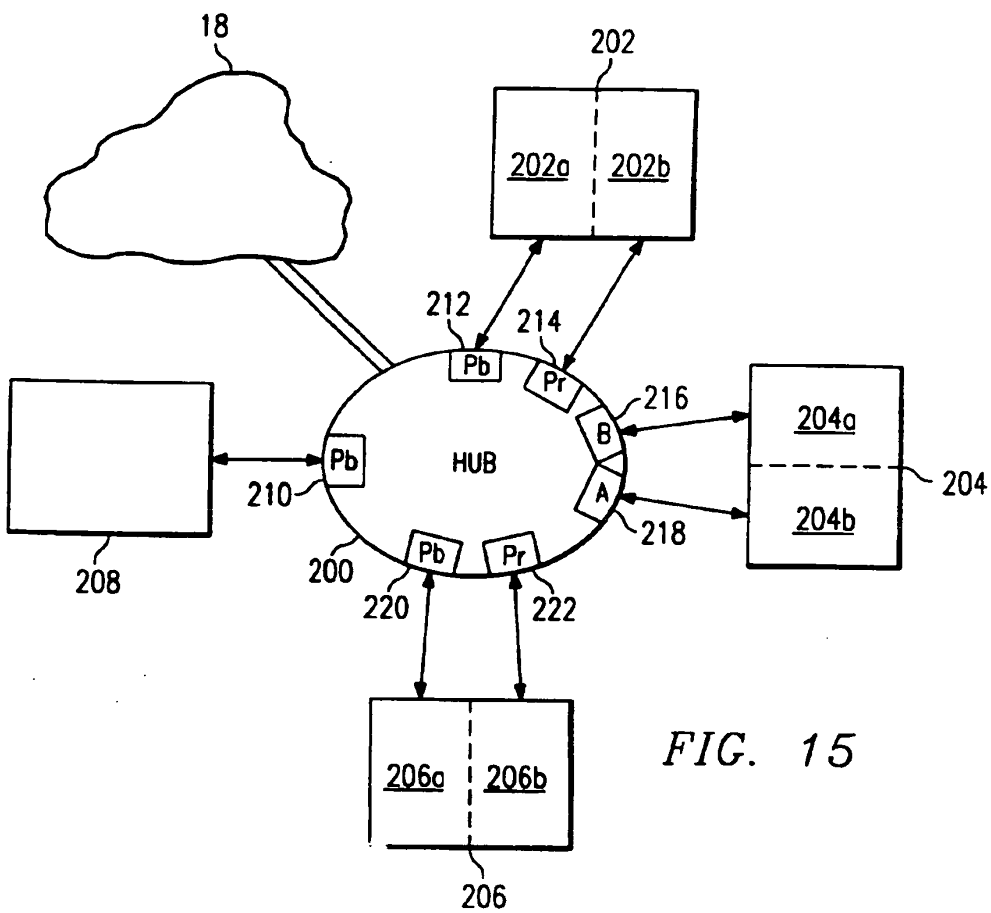

18
202
202a
202b
212
214
Pb
Pr
216
B
204a
204
A
HUB
Pb
210
218
204b
208
200
Pb
Pr
220
222
206a
206b
206

FIG. 15

NAME	ARNIE	ARNIE	JANE	JANE	JANE	JANE	FRANK	FRANK
MBX ADDR	4437	408-555-4437	4437	374437	4437	214-555-4437	4437	274437
VMS	CASJOI	CASJOI	PrDalTx	PrDalTx	PbDalTx	PbDalTx	PrAusTx	PrAusTx
USER ID	3	3	6	6	6	6	8	8
SCOPE	LOCAL	GLOBAL	LOCAL	GLOBAL	LOCAL	GLOBAL	LOCAL	GLOBAL
COMMUNITY	0	0	101	101	0	0	101	101

FIG. 16

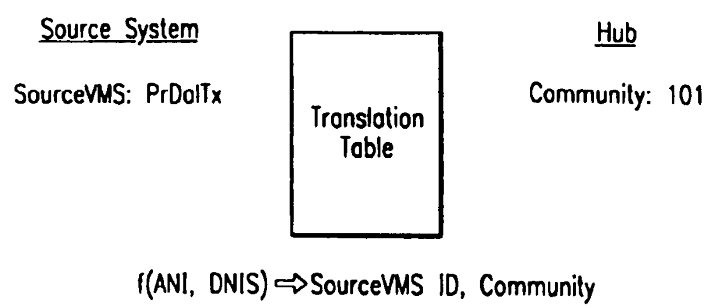


FIG. 17A

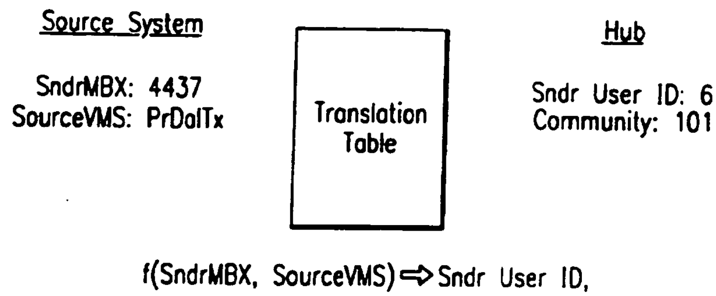


FIG. 17B

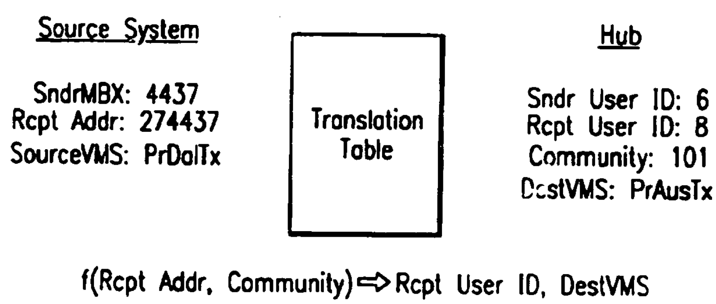


FIG. 17C

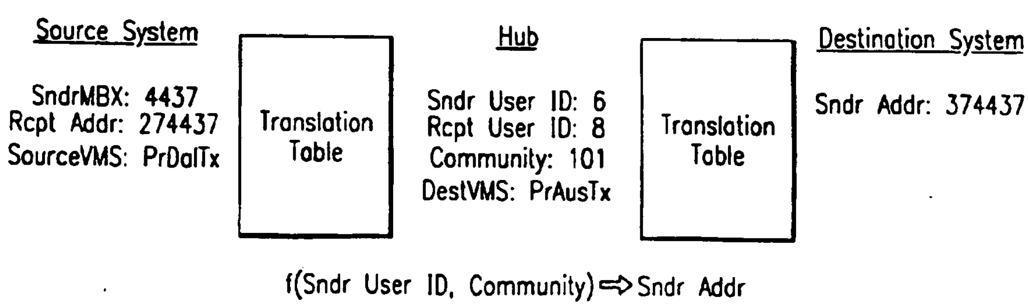


FIG. 17D

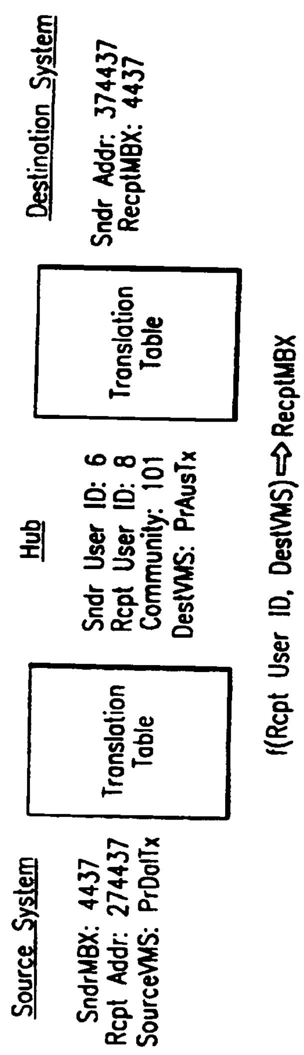


FIG. 17E

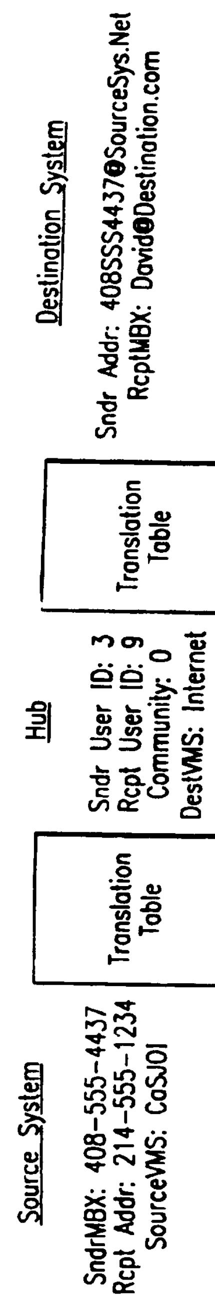


FIG. 18

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US95/11772

A. CLASSIFICATION OF SUBJECT MATTER

IPC(6) :H04M 1/64, 11/00, 3/42; H04Q 11/04; H04J 3/02, 3/12
US CL : 379/67, 93, 95, 201, 207, 220; 370/60, 85.13, 110.1; 395/200
According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

U.S. : 379/67, 93, 201, 207; 370/60, 85.13, 110.1; 395/200

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

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)
APS
search terms: hubs,messaging systems, network center, database,

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X ---- Y	US, A, 5,113,430 (RICHARDSON, JR. ET AL) 12 MAY 1992, see cols. 3, 14 and fig. 1.	56--61, 64-67, 83, 85, 87 ---- 62, 63, 76, 77
Y	US, A, 4,939,771 (BROWN ET AL) 03 JULY 1990, see cols. 12-15.	62, 63, 76, 77, 106, 107
X,P	WO, A, 94/24803 (AHUJA ET AL) 27 OCTOBER 1994, see entire document.	106-110
A	US, A, 5,333,266 (BOAZ ET AL) 26 JULY 1994, see entire document.	106-110

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

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"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
02 FEBRUARY 1996	04 MAR 1996

Name and mailing address of the ISA/US	Authorized officer
Commissioner of Patents and Trademarks Box PCT Washington, D.C. 20231	JEFFREY HOFSASS
Facsimile No. (703) 305-3230	Telephone No. (703) 305-5403

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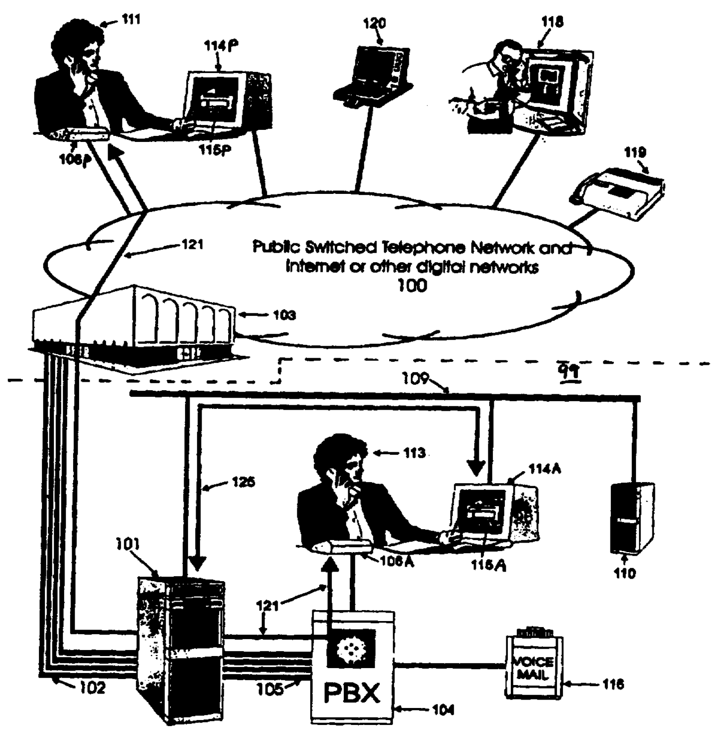
(54) Title: CALL MANAGEMENT SYSTEM WITH CALL CONTROL FROM USER WORKSTATION COMPUTERS

(57) Abstract

A Call Management System (99) provides for management of calls directly by system users (106A or 106P) at their workstation computers (114A or 114P) via a digital data network (125) such as a digital network not controlled via the user's telephone instruments (106A or 106P) as in prior systems. A call management computer (101) interccepts incoming calls and controls the handling of such calls according to instructions received from the users' workstations, which are accessed via the digital data network (125).



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CALL MANAGEMENT SYSTEM WITH CALL CONTROL FROM USER WORKSTATION COMPUTERS

BACKGROUND OF THE INVENTION

This invention pertains to telephone switching systems in general.

Business communication has taken two separate paths. One involves telephone conversations and the other involving computer communication.

Until now, business telephone communications have been based upon the approach that each individual controls his own call traffic through multiple buttons on proprietary telephone instruments and/or simple commands entered through "hookflash" or the telephone keypad. Further, the architecture and philosophy applied to business PBXs or other telephone switches is limited to the "switching" of calls, such as incoming calls, to internal stations or internal stations to internal stations. This approach strictly avoids operation based upon "call content" such as the type of call, from whom it originates, etc. The limited capabilities of the multi-button telephone instruments and the lack of awareness of call content severely restrict the capabilities and features available and thus reduce the overall effectiveness of the business telephone systems of the past.

The focus of computer technology has become the desktop workstation computer attached to one or more business enterprise-wide, high-speed digital networks which interconnect the workstation computers of business enterprise's employees with a variety of information servers, communications and computing devices. The business enterprise's digital network may be a combination of Local Area Networks LANs and Wide Area Networks WANs attached together via a variety of transmission media augmented by the Internet. These corporate communications worlds, i.e., business enterprise's digital networks and the public switched telephone network PSTN remain separate and distinct until now.

SUMMARY OF THE INVENTION

In accordance with the invention, a Call Management System is provided for handling business communications. The system alters the architecture and philosophy of the past, providing the users an array of new features and functions and expanded existing features.

A Call Management System provides for the real-time management of incoming voice calls by called parties. Real-time call management enables the called party to know who is calling before the call is accepted and, thus, to establish the likely priority of the call and decide how best to handle the call before his telephone rings. This method of call handling is intended to improve significantly the efficiency of the called party's interactions with customers, vendors, coworkers, and others. Each called party is notified via his/her computer terminal of each incoming call and the caller's identity, even when the called party's extension is already busy, allowing the called party to choose the appropriate handling of each incoming call before ringing the extension (hold, transfer, put through, send to voice mail, etc.

In accordance with the invention, calls to an organization are directly controlled through networked user workstation computers, not telephone instruments, a call management computer intercepts telephone and data trunks which link the business to the telephone provider's central office. The call management computer interacts with and controls telephone and data trunks connecting with the telephone service provider.

The call management computer receives and answers all calls from the telephone provider's central office, determines the type of call voice, fax, data, determines for whom the call is intended the called party, and proactively determines who is the calling party. This information determines how the call will be handled. Proactive caller identification is used to identify the caller by interacting directly with the caller to obtain an identifiable telephone number or the caller's spoken voice which are then identified through specialized primary or secondary Caller ID databases or a voice name database.

Call alert information is transferred via operator or digital network interconnecting workstation computer to the called party's workstation, even when the called party's telephone extension is busy. The called party instructs the call management computer via the digital network what to do with the calls in progress .

The call management computer also provides for call handling rules to be defined by the business organization or by the system users. These rules, called "VIP rules", are an adjunct to the called party's direct control and provide for special handling of important individuals, groups or even for all callers.

The call management computer either receives control commands from the called party or operates in accordance with an appropriate VIP rule and responds to the calling party accordingly by, for example, playing out recorded voice messages, receiving additional information from the caller, transferring the call to the called party, to voice mail or elsewhere either within or outside the organization.

One significant advantage of the Call Management System is that it provides system users with the many unique features and functions while requiring nothing more than simple "POTS" (plain old telephone service) telephones or headsets instead of expensive multi-button proprietary business telephone instruments.

The Call Management System also functions as an outbound call processor, working in conjunction with software in each user's workstation to provide outbound call processing services. The personal call logs can be reviewed by a user and used to return missed phone calls through a point-and-click interface. A database containing caller identification information may also be used on-line for outdialing calls to selected people, all without the need for manual dialing.

The Call Management System creates reusable "voice pathways" from the call management computer to the called party when it is appropriate to put a call through to a destination because of user selection or VIP rule processing. Voice pathways, once created, are reused repeatedly so long as the destination has calls in process. This enables rapid switching between calls with only the click of a workstation mouse and avoids the typical operation of establishing and tearing down entire calls in order to switch between them.

Real-time protocol conversion is provided between central office trunks and PBX trunks of the Call Management System. This allows the system to receive new or different types of services from the telephone provider while still connecting to and utilizing existing telephone systems which cannot otherwise accept the new capabilities directly. It also permits the Call Management System to utilize directly the user's telephone instruments or headsets, removing the need for a separate PBX or other switch. "Conversion" between different trunk circuits allows the system's many new features and functions to be implemented without upgrade of the organization's legacy PBX or other switch or alternatively as a replacement for an existing PBX.

The system monitors and controls the individual trunk circuits obtaining call content information and directly interacts with the caller to handle voice, Fax and data calls automatically in any combination.

A Call Management System in accordance with the invention treats all calls external and internal in the same way, allowing the transferring and conferencing of calls from inside to outside, outside to outside, or in any other combination. This removes the historical limitations on the handling of calls depending upon their source.

The Call Management System provides for the use of a single unique telephone number for each user. This "one number" is used to receive, identify and automatically handle all the user's voice, fax and data calls, one or several at a time using multiple trunk circuits. The use of only one number per user significantly reduces the costs, complexity, inefficiency and confusion of having multiple different telephone numbers for different functions.

Proactive caller identification is provided by using direct system interactions with the calling party. Predetermined messages and acquired responses are used to identify the caller for the called party. This provides the system user and call management system with knowledge of who each caller is so that appropriate priority and special handling can be applied to each call.

Specialized databases containing caller information are used to identify callers as part of Proactive Caller Identification.

The call management computer automatically answers each call, identifies the called party, determines the call type and identifies the calling party. The call management computer alerts the called party system users through the organization's local area or wide area networks or via the Internet, providing the called party direct call control via their workstation.

Users may utilize the call management system from remote locations having the same features and functions as though they were onsite.

A system user can handle multiple calls at the same time, knowing who each caller is and applying appropriate priority to each call, eliminating "voice-mail-jail" since only humans, not machines, send callers to voice mail and reducing the incidence of "telephone tag". This capability improves the user's ability to service multiple customers at the same time, as well as saves the time and costs of the otherwise inevitable "telephone tag".

In accordance with the invention, certain callers may be identified as "VIP" callers. When an incoming call is identified as originating from a VIP caller, special handling of the call is initiated. The special handling may include personalized voice messages in the user's own voice, user-generated voice "menus" for the caller, receipt of and routing based on caller-entered information, special call rerouting inside or outside, including "follow me" and "find-me" rerouting of the call to the user, even when the user is out of the office, and/or a distinctive ringing sound to alert the user. "Page me" or "Beep me" features are also included, even for calls which are routed to voice mail or elsewhere.

The system routes or conferences calls from any station directly connected to the system, connected to the system via a PBX or other switch, or connected to the system via the public service telephone network or via Internet. The Call Management System views all sources and destinations as equivalent, only differing in their technological access requirements.

"Call tags" which may be digital, text or voice messages, may be attached to voice calls by any system user to provide additional information to other system users to whom the call may be transferred or conferenced.

The Call Management System identifies and automatically receives Fax and data calls directed to each individual system user. Once received and stored in the call management database, the user is alerted to the presence of the file and allowed to transfer it to his/her workstation for further examination or use. Likewise, Faxes or data files can be sent to selected or specified receivers through the Call Management System with individualized Fax banner information for each user.

Summaries are maintained of all incoming and outgoing calls in an interactive, real-time, user-accessible "call log" portion of the Call Management Databases, allowing the called party to know who called, when they called, which calls were missed even if no voice mail or other form of message was left, and to return calls automatically through simple mouse clicks.

A single telephone call can retrieve voice messages, e-mail, Fax and data messages even though in different forms and stored in different places. Users are provided a variety of retrieval mechanisms including: sending to a remote Fax machine anywhere in the world, or sending to a remote computer. This improves the ability for traveling and at-home users to stay in touch with all their information and thus their customers and prospects.

Voice mail-jail is prevented because only humans, not machines, send callers to voice mail. The Call Management System transfers callers to voice mail when system users request or because of their VIP rules, it alerts users when voice mail exists for them to hear and it utilizes voice mail access to provide "One-Call Message Retrieval" of user's e-mail, voice mail, Fax, and data.

The Call Management System monitors the current status of all system users and makes status information available to all other system users on demand.

BRIEF DESCRIPTION OF THE DRAWINGS

The invention will be better understood from a reading of the following detailed description in which like reference numerals designate like elements and to which:

Fig. 1 is a diagram of a Call Management System;

Fig. 2 is a block diagram of a call management computer;

Fig. 3 is a block diagram of a digital signal processor;

Fig. 4 is a detailed block diagram of a portion of the digital signal processor of Fig. 3;

Fig. 5 is a diagram showing how callers are identified;

Figs. 6a through 6e show components of the call management window as they appear at a workstation display;

Fig. 7 shows components of call management windows for VIP rule creation and management;

Fig. 8 shows a Fax handling display;

Figs. 9a and 9b show components of call management windows;

Figs. 10a and 10b show "copper bypass" configurations for Call Management System fault tolerance.

DETAILED DESCRIPTION

1. Overview

1.1 General.

Fig. 1 is an overall block diagram of one embodiment of the improved Call Management System, in which call control is provided by the user through a networked workstation computer, not a conventional telephone instrument. Fig. 1 shows the organization's environment with its Local Area Network and/or Wide Area Network (LAN/WAN), an on-site system user with a LAN/WAN based workstation, a PBX or similar switch, voice mail and a call management computer. In Fig. 1, an organization utilizing the Call Management System is clustered at the bottom of the figure and the outside world of callers and system users is clustered at the top. The organization's calls are handled using a call management computer which is placed so as to intercept telephone and data trunks between the telephone provider's central office and the organization's PBX or other switch (or as its replacement). Also shown are a work-at-home system user with workstation connected via the Internet, a voice caller at a pay phone as well as Fax and data callers all connecting through the public switched telephone network.

All central office calls ring directly into the Call Management System, the system has direct access to all information being provided by the central office. Since the system gets the "first look" at all incoming calls, it can direct and process calls according to user requirements.

In the Call Management System of Fig. 1, call control is provided through a user workstation 114 to provide new and improved capabilities for the user and substantially eliminating the shortcomings and disadvantages of past systems.

Fig. 1 shows pictorially the public switched telephone network (PSTN) with voice 118, Fax 119, a call management system 99 coupled to the public switched telephone network 100 through a telephone central office (CO) 103. The call management system 99 includes a PBX or similar switch 104 and connections to user telephone instruments 106. A digital data network 109 attaches to user workstation computers 114a - 114n. The digital data network of the illustrative embodiment is a conventional Local Area Network (LAN) or Wide Area Network (WAN). In Fig. 1, two different types of system users are shown, the first is an in-house user 113 associated with workstation 114a and telephone instrument 106a while a second user is a work-at-home or traveling employee with a workstation or laptop computer 114p attached remotely via the Internet or WAN extension of the LAN 109, through ISDN or otherwise. The PSTN allows access to/from the call management system 101 via voice communication device 118, fax device 119, data 120.

A call management computer 101 is placed so as to attach to five 5 separate interfaces described below.

1.2 CO Trunks.

Call Management System 99 is coupled to central office 103 via Central Office trunks 102 for both voice and data connections.

CO trunks 102 includes a variety of trunks, including analog, DID, ISDN, T-1, DID over T-1, 800/900 T-1 services, data, and Internet. The central office 103 is interconnected within the public switched telephone network 100 via Local Exchange Carriers, Inter-exchange long-distance Carriers, Cable companies, RF or satellite carriers, digital Internet providers or any other types of voice or data carriers. CO trunks 102 may include multiple individual "circuits" ISDN, T-1, etc. which carry voice and/or data for individual calls.

Voice calls over Internet and similar means are processed utilizing conventional digital techniques but are then fed into the system as though they were voice trunks. For the purposes of this description, CO trunks include Internet connections.

Call management computer 101 is configured and programmed to appear to telephone service providers 103 as though it is a business PBX or other business telephone switch and/or an Internet or other data server or node.

1.3 PBX Trunks.

Within the call management system 99, PBX trunks 105 are the means by which the Call Management System 101 provides voice or data connections to system users or workstations or other devices within the business 99. For traveling or work-at-home users 111, access to the outside using CO trunks is considered just a part of PBX trunk access, *e.g.*, voice calls to an at-home system user's 111 telephone 106p or Internet voice or data connections.

"PBX" includes a variety of different telephone switches including classical private branch exchanges PBXs, automatic call directors ACDs, key telephone sets, or integrated switches within the call management computer 101. Telephone instrument 106a - 106n, as shown as physical telephones but may also include headsets, earpieces, computer sound systems, isochronous network technology such as isoEthernet and ATM, and other means of providing a voice or data connection to a user.

The call management computer 101 may attach to the organization's PBX or other telephone switch 104 through PBX trunks 105. The call management computer can sit in front of virtually any type of switch. No switch-specific hardware or software is required for integration. PBX trunks 105 may be analog, DID, DISA, ISDN, T-1, DID over T-1, 800/900 T-1 services or other available types. in all available variations and combinations. In addition, the call management computer 101 may be connected directly to the organization's telephone instruments 106a - 106n or directly to the user's workstation 114a - 114n for voice or data connections in place of a switch 104.

PBX trunks may or may not be of the same kind and/or number as the CO trunks 102. The Call Management System may provide a one-to-one direct relationship between CO trunks 102 and PBX trunks 105 or it may provide protocol "conversion" between differing CO and PBX trunk types and/or numbers. PBX trunks also include direct connections to the user's telephone instruments 106.

The call management computer 101 is so configured and programmed that it appears to the business PBX or other switch as though it is a central office and/or it appears to the direct telephone instruments 106a - 106n as though it is a business PBX switch such as 104.

1.4 DSP Processing & Switching.

Trunk interfaces 203, 206, circuit switches 204 and DSP digital signal processors 208 interact with and control the CO and PBX trunks under the overall control of the call management computer 101.

All CO trunks 102 and PBX trunks 105 are attached to the call management system through appropriate trunk interfaces 203 and PBX trunk interfaces 206. The interfaced trunk signals are further attached through circuit switches 204 and high-speed telephony buses 210 to each other and to special DSP's 208.

The configuration of the call management computer 101 with individual interface boards 203, circuit switches 204, DSP processors 208 and the high-speed buses 210 provide means for real-time sensing, switching and management of calls and the means for the call management computer 101 to appear to the central office 103, through the CO trunks 102, 202, as a business PBX 104 or other switch and/or a server computer for data functions. This configuration further permits computer 101 to appear to the business PBX or other switch 104 or to the user's telephone instrument through the PBX trunks 105, 205 as a central office switch such as 103 and/or a data server.

1.5 Digital Data Networks.

Notification of events and control over multiple calls is accomplished independently of the organization's PBX or other switch system even if the user's telephone instrument 106a - 106n is currently busy because the digital networks 109 are separate from and independent of the user's telephone instrument 106 or telephone system 104.

The call management computer 101 attaches to the organization's digital data network including a LAN as well as Internet and/or other external WAN networks such as Internet via interfaces 209, 213 and 214, through which it has immediate access to the user workstations, whether in-the-office 113 or at the site of a remote user 111. These networks operate independent of whether the user's telephone instrument 106 is busy or not. The digital networks 109 are used by the call management computer 101 to alert called users such as users 111 and 113 to incoming voice calls and newly received Fax, voice or data messages and for receiving back user controls of all types from the user's workstation 114. In addition the digital networks 109 provides access to the organization's LAN server computers 110 for e-mail, voice mail, database, Internet access and other services.

1.6 Call Management Databases.

The call management system 101 utilizes a variety of interactive call management databases 215 for functions including: system and user configurations, primary and secondary caller identifications and voice caller identification, VIP rules, phone directories, Fax and data file storage, voice message storage, user-accessible call logs and many other functions. These on-line, real-time databases 215 may reside on the call management computer 101 itself or elsewhere on the digital network, *e.g.*, on a LAN-based database server 110.

The Call Management System structure of the embodiment includes call management computer 101, the CO Trunks 102, the PBX Trunks 105, the trunk interfaces 203, circuit switching 204 and DSP processing 208, the organization's digital network(s) 109, 209 and the call management databases 215. However, implementation may be in various combined or extended ways, such as when the call management computer 101 is built into the central office 103 or into a PBX or similar switch 104. The call management computer 101 may replace the PBX and control the organization's telephone instruments 106 directly.

1.7 Call Management Computer.

The call management computer 101 is configured as shown in Fig. 2. It is based on an industry-standard computer with processors, memory, power supply and cabinetry. The computer 101 is coupled to a data bus 211. The data bus has connections to LAN interface 209 and disk memory which stores Call Management Databases 215. The databases may alternatively reside on the digital network system 109. The data bus 211 is connected to interfaces for Digital Internet connections 213 and bulk calling line identification BCLID data link 214 to the central office.

The telephony subsystem ties together the CO trunk 102 and PBX trunks 105 through their specific trunk interfaces 203, 206 and circuit switches 204 to the telephony signal buses 210. Each of the trunk interfaces 203, 206 is also coupled to the computer data bus 211, through which the computer processors 201 receive information and provide control and data to both the CO trunk interface 203 and its circuit switches 204. The DSP digital signal processors 208 include multiple DSPs. The multiple DSPs as needed are attached to the telephony signal buses 210 through switches 204 and to the computer data bus 211 through which they provide information to the computer processor 201 and receive back control and data *e.g.* voice messages to play out to the caller. For voice-over-Internet or similar digital connections, voice interface board 207 is connected to the computer data bus 211 and through its own circuit switches 204 to the telephony signal bus 210, through which the voice connections can be made to/from any telephone instrument 106.

1.8 Call Reception.

Typical call paths 121, 221 are shown on Figs. 1 and 2. The in-band call information from central office 103 is sent through a CO trunk 102 to the call management computer 101 where it attaches through an appropriate CO trunk interface 203 and circuit switch 204 to the telephony signal buses 210. For each trunk/circuit, the call management system assigns one or more DSP processors 208 connected to the telephony signal bus 210, to provide a monitoring and control link 219 for that call.

An incoming call such as from the payphone caller 118 or the Fax caller 119 or data callers 120 is routed through the PSTN 100 to the central office 103 and then to the call management system 99 through a CO trunk 102. The assigned DSP 208 and/or CO interfaces 203 monitor for an incoming call analog or digital signals in any available form, appropriate to the type of trunk and/or circuit.

When the call is presented by the CO 103, the call setup commands are recognized through the trunk interface 203 or through the associated DSP 208 and the call management computer processors 201 receives this information via the trunk interface 203 or DSP 208 connections to the computer signal buses 211. Control signals from the call management computer 101 then cause the call to be answered via the same route according to the trunk and circuit type.

At this point in the process, a first call path segment 221a to a DSP monitoring and control link 219 for the trunk and circuit terminates at the assigned DSP 208.

Connections to a system user 113 are created by the call management computer 101 selecting an available, appropriate CO trunk inbound 105 and establishing a call to the PBX 104 or to remote system users over additional trunk 102 to the central office 103. The PBX 104 or CO 103 responds to the call setup commands depending upon the type of trunk and circuit (including voice-over-Internet and other digital services). These are sensed by the trunk interface 203 or 206 or the assigned DSP 208 and passed to the call management computer 101. Call management computer 101 then controls the appropriate circuit switches 204 to connect the voice pathway from the calling party 118 to the voice pathway to the called party's 113 telephone instrument 106 or via a second call path segment which includes segment 218 and call path segment 221b, leaving the assigned DSP 208 attached to continue providing the DSP monitoring and control link 219.

The call is then put through in a conventional manner by the PBX 104 or CO 103 to the called party's 113 or 111 telephone instrument 106a or 106p where it rings and is answered by the called party 113 or 111 completing the connection between the caller 118 and the called party 113, 111. The typical call path 221 with the associated DSP monitoring and control link 219 is then completed as shown. The voice pathway so created may be reused as described below.

The CO trunk interfaces 203 and PBX trunk interfaces 200 and assigned DSP 208 remain active throughout a call or series of calls to a destination, watching for either end to terminate the call, by hanging up the telephone instrument, or otherwise changing the call state while the call management computer 101 watches for the system users 111, 113 to select a command changing the call's state.

1.9 Call Origination.

Calls are originated by system users through their workstation 114. Such originated calls may be destined to any system user 113, 111, a non-system user, or anywhere else in the PSTN 100. Depending upon the destination, the call management computer 101 selects an available, appropriate CO trunk 102 or PBX trunk 105 and establishes the call to the CO 103 or to the PBX 104 or the telephone instrument 106 using appropriate signaling techniques for that trunk or circuit. The call management computer 101 then instructs the circuit switches 204 to connect the call circuits together, "bridging" the originator to the destination and creating the typical call path 118 in Fig. 1 and DSP monitoring and control link 119 as shown in Fig. 2.

The CO trunk interface 203, the PBX trunk interface 206 and assigned DSP 208 remain active throughout each call, monitoring for either end to terminate the call, by hanging up the telephone instrument, or otherwise to change the call state while the call management computer 101 monitors for the system user 111, 113 to select a command changing the call's state.

The call management computer 101 manages the available, appropriate CO and PBX trunks 102, 202, 105, 205 so as to share between non-system user calls established from the PBX or telephone instruments and system user 113, 111 calls established by itself.

The call management computer software runs under Microsoft's Windows NT operating system, a multi-threading, multi-tasking operating system required by the real-time call management aspects of the system. The Call Management System may be configured as a "client" to the organization's existing digital networks or as a "server" when incorporating Internet or other server functions for the digital network(s).

1.10 "One Number" Processing.

For system users 111, 113, the Call Management System uses only a single DID (direct in dial) or extension number to receive all calls, voice, fax or data in any mixture and number within the limit of the number of available trunks and circuits at any time or all at the same time. The call management computer 101 is programmed to sort them out and handle each appropriately.

1.11 Called Party Identification.

After a call is received, the call management computer 101 determines a destination party for the call, either automatically through reception of DID, DNIS (dialed number identification service), ISDN or other signals or messages from the central office 103 as detected by the trunk interface 203 or the assigned DSP 208 or, alternatively, a call attendant feature of the Call Management System.

In Call Attendant mode, the call management computer 101 instructs the DSP 208 to play out one or more voice messages from the call management database 215 asking the caller 118 to identify the destination party by name, spelling, extension number or otherwise. The DSP 208 receives the information from the calling party 118 and passes it to the call management computer 101 where the called party 111 or 113 is identified through the digits entered, through voice recognition or otherwise. The called party's extension number is found or verified using the call management database 215.

For voice over Internet or similar techniques, the caller is provided and fills in a "form" which includes: the name of the caller, the name of the called party and other appropriate information.

1.12 Call Type.

The incoming call type voice, Fax or data is also determined using the DSP 208, which searches for appropriate signaling from the call source 118 *e.g.* none for voice, or CNG for Fax, specified DTMF, carrier or other signals for data whether files, video data, video conferencing, etc. in analog or digital form.

For non-voice calls to system users, the attached DSP 208 is instructed to switch to the appropriate Fax or data mode and to receive the transmission automatically for storage in the call management database 215 and later use by the called party 111, 113 or to transfer the call automatically to an appropriate extension, *e.g.*, for video conferencing.

1.13 Proactive Caller Identification.

The identity of the voice caller 118 is determined either automatically through reception of Caller ID, ISDN, ANI, BCLID or other information from the central office 103 as received by the trunk interface 203, the assigned DSP

208, the BCLID data link 214 or otherwise. Depending upon what information was received, if any, Proactive Caller Identification may then use direct interaction with the caller 118 and the Caller ID databases 215 for additional information. Proactive Caller Identification is described in Section 6 and the Caller ID databases are described in Section 7.

1.14 VIP Rules.

Specific rules, called "VIP rules", are created to specify special handling for important callers, sets of callers or even for all callers. These VIP rules precede and augment direct user controls and are described in Section 10.

1.15 Notification and Control via the Called Party's Workstation Computer.

Voice calls destined for system users either working in the office 113 or outside the office 111 are handled by alerting the called party at his workstation 114 through sending a message from the call management computer 101 across the digital network 109 to the user's workstation 114 and thus to the user's call management window 115 which "pops up" onto the user's workstation screen. The user then controls the call through selections made using his call management window 115 and its subscreens. This procedure, the windows and the user's functions are described in Section 8.

1.16 Call Management.

Once the call type, called party and caller are identified, the call management computer 101, 201 handles these calls based first on any applicable VIP rules and then on commands from the user, via their workstation call management window 115. A call is "held" and manipulated by the call management computer 101 throughout answering of a call, identification of the call type, identification of the called party, proactive identification of a calling party, playing out messages to the caller, receiving information received from the caller, VIP rule handling and/or notification of the called party 113. The typical call pathway 221, 219 is terminated at the assigned DSP 208 and is not passed beyond.

Some of the call functions which may be exercised by a system user are described in the following paragraphs insofar as what happens with the call management computer 101. Detailed descriptions of the call management window 115 and the user's functions, controls and management using their workstation 114 are provided in Section 8.

1.17 "Answer" a Call.

A system user such as user 111 or 113, may select the "Answer" function for a call announcement. If the user has no calls currently active, the call management computer 101 selects an available CO trunk (inbound trunk 105 for user 113 or outbound trunk 102, for user 111) and establishes a call to the PBX 104 or the central office 103. The PBX 104 or central office 103 responds to call setup commands depending upon the type of trunk and circuit. Call set up signaling is sensed by the trunk interface 203 or 206 or an assigned DSP 208 and passed to the call management computer 101 which then controls the appropriate circuit switches 204 to connect the voice pathway from the calling party 118 to the voice pathway 121 to the called party's 113 or 111 telephone instrument 106a or 106p, leaving the assigned DSP 208 attached as well to continue providing the DSP monitoring and control link 219.

The call is then processed conventionally by the PBX or CO to connect to the called party's 113 or 111 telephone instrument 106a or 106b where it rings and is answered by the called party 113 or 111 to complete the connection between the caller 118 and the called party 113 or 111. The typical call path 221 with the associated DSP monitoring and control link 219 is then completed as shown. The voice pathways so created may be reused as described in Section 2.

For cases where the Call Management System is connected in the central office 103 and replaces the PBX and controls the telephone instruments directly, the process is basically the same.

If a system user such as user 113 or user 111 currently has an active call, the voice pathway 121 already exists and the user's current call will be moved to "hold" or "hang Up" mode as defined by the user and the voice pathway switched immediately to the new caller (see "Transfer" below).

The CO trunk interfaces 203 and PBX trunk interfaces 206 and assigned DSP 208 remain active throughout a call or series of calls to a destination, monitoring for termination of the call by either end through hanging up the telephone instrument or otherwise changing the call state. Call management computer 101 monitors for the system users 111 or 113 to identify selection of a command from the user's call management window 115 to change the call's state.

1.18 "Transfer".

The "Transfer" function is used to cause a received call to be transferred to another destination either inside the business 99 or to any destination coupled to the PSTN 100. Transferring a call requires the user to select a "Speed-Transfer" button or a "Transfer" screen from which he may select a destination from a directory or the user may type in the phone number to use for the destination.

To transfer a call to another destination, the call management computer 101 receives a transfer message from a user's call management window 115 via the digital networks 109, the call management computer 101 instructs switches 204 to disconnect the voice path 121, instructs the appropriate trunk interface 206 to "hang up" the call to the user if appropriate and instructs the DSP 208 to return to call monitoring.

If the new destination is a system user, call management computer 101 checks for any appropriate VIP rules for this calling party and the called party and processes the call as specified by the VP rules. Otherwise, call management computer 101 alerts the called party and awaits user control.

If the destination is not to another system user, the call management computer 101 establishes a new voice pathway to the destination wherever it may be, instructs the appropriate switches 204 to connect the voice pathways together 221 and controls the trunk interface 103, 106 and the DSP 108 to monitor the progress of the call, searching for hang up or change of state at either end.

1.19 "Send to Voice Mail".

One special function provided by the call management window 115 is to send a call to voice mail whether the call is active or not. This is a special variant of the "Transfer" function with a pre-specified destination.

1.20 "Conference".

The Call Management System can conference calls independent of whether the parties are directly coupled to the PBX or are accessed via the central office 103. When a system user 113 selects the "Conference" function for an existing active call or with no active call, he then selects one or more destination parties from the call management window 115 directory and/or types in one or more telephone numbers. The call management window 115 sends a "Conference" message down the organization's digital network(s) 109 to the call management computer 101, 201, whereupon the call management computer 101 alerts the new system users.

As each called system user "Answers" the Conference call, the call management computer 101, 201 creates a new voice path 121, as appropriate, instructs the appropriate circuit switches 204 to connect the voice paths together to the assigned DSP 208, and instructs the DSPs to combine the signals appropriately to create a conference call.

For "Conference" destinations which are not system users, the call management computer 101, 201 immediately establishes a call to the destination and "bridges" the circuits, as with "Outdial" calls described below.

1.21 "Hold".

A system user can select the "Hold" function for any call currently active. The "Hold" messages are sent by the call management window 115 via the organization's LAN or WAN 109 to the call management computer 101, call management computer 101 instructs the appropriate circuit switches 204 to break both the inbound and outbound portions of the voice path 121 effectively placing a caller on "hold" but providing the user rapid access to the call through simple mouse clicks. During unused time when the calling party would otherwise be waiting, the Call Management System provides "Music on Hold" and/or corporate messages (sales, information, etc.). The caller can terminate any messages at any time by entering a "#".

1.22 "Mute".

A system user can select the "Mute" function for any call currently active. "Mute" messages are relayed via the call management window 115 and sent down the organization's LAN or WAN 109 to the call management computer 101 which instructs the appropriate circuit switches 204 to break just the outbound portion of the voice path from the system user to the caller leaving the inbound portion active, thus muting the call.

1.23 "Record" and "Playback".

A system user can select the "Record" function for any call currently active. The "Record" messages are relayed via the call management window 115 and sent via the organization's digital network(s) 109 to the call management computer 101 which instructs the assigned DSP 208 to record the call content, sending it to the call management computer 101 which saves it to the Call Management Databases 215 for future replay.

1.24 "Hang Up".

When either calling or called party terminates a call by hanging up or by selecting the "Hang Up" function from his call management window 215, the call path 121 is dismantled by the call management computer 101 instructing the switches 204 to disconnect the voice path, instructing the appropriate trunk interface 203 or 206 and/or DSP 208 to "hang up" the call as appropriate and instructing the DSP to return to searching for a new call. However, either end currently with new calls waiting is kept active as a re-usable voice pathway.

1.25 "Outdial".

The users of the system 99 may originate calls to other users located at business 99 or to any numbers outside the business 99. The user utilizes his/her call control window 115 to identify or type in a destination for the call internal or external and then instructs the Call Management System to dial to that destination.

Workstation 114 sends the dialing control messages via the digital network(s) 109 to the call management computer 101, 201 which, in turn, causes the call to be placed using an available CO or PBX trunk/circuit 102, 105 of the appropriate type through the circuit switches 204. When a call is in process or completed to the destination, a voice path 121 is established to the calling party's telephone instrument 106.

Once established, the system user 11, 113 has available all of the system features for originated calls, as for inbound calls.

1.26 Calls Received for Non-System Users.

Calls received for business 99 employees who do not have appropriate workstations or who do not choose to be system users, or for the organization's voice operator and for dedicated Fax machines and other hardware devices are immediately switched by the call management computer 101 to the PBX 104 or telephone number dialed, and are handled conventionally. For such calls, the call management computer 101, 201 selects an available and appropriate PBX trunk 105, establishes a call to the desired extension controls and the circuit switches 204 to connect the voice paths together. DSP 208 monitors progress of the call and monitors for either end to hang up. This process includes protocol conversion features for CO trunk/circuit 102, which are of different type than the PBX trunk/circuit 105.

1.27 Predefined Call Routing.

Fax or data calls received for specified numbers, such as the business's base number, are accepted as though directed to a specified user, *e.g.*, the organization's operator, who is then expected to sort out and the Faxes send them to the appropriate users or print them.

1.28 Calls Originated by Non-System Users.

Calls originated by non-system users, via their PBX or through their telephone instruments controlled directly by the Call Management System, are handled conventionally except that, the call management computer 101 receives the call establishment from the PBX 104 through the PBX trunk/circuit 105 and passes it on to the central office 99 via an available CO trunk circuit 102, connecting the circuits together via appropriate circuit switches 204. The user is unaware of this process and sees no difference from conventional telephone usage.

1.29 Telephone Requirements.

One significant advantage of the Call Management System is that it provides system users the many unique features and functions described here while requiring nothing more than plain old telephone services (POTS) telephones or headsets instead of expensive multi-button proprietary business telephone instruments. None of these features require the telephone instrument to be anything more than just a voice path to the user's ear and mouth.

1.30 International CallBack.

The Call Management System provides a "CallBack" subsystem with which a calling party places a call from a foreign telephone, the system receives the call and telephone number, then terminates the call and immediately dials the caller back at the number received. This process saves significant telephone expenses compared with the costs of calls from many foreign countries.

2. VOICE PATHWAYS

When a call is put through to a system user, the call management computer 101 creates a reusable "voice pathway" 121 to the called party either an in-house user 113 or an external user 111. Voice pathway switching, rather than establishing and tearing down multiple separate calls, provides the ability to switch rapidly between multiple calls, on demand, based on the user's changing priorities through simple point-and-click mouse, keyboard or menu operations.

Fig. 1 shows two re-usable voice pathways 121, one to the "in-house" system user 113 via PBX trunks 105 and the organization's PBX 104 to the user's telephone instrument 106 and another to the "at-home or traveling" system user 111 via a CO trunk/circuit 102 to the central office 103, and then via the PSTN 100 ultimately to the user's telephone instrument 106. Both of these are valid cases, even though they utilize different CO and PBX trunks of any appropriate type. Note that the Call Management System <u>does not care</u> whether the called party is at an in-house extension or is not directly connected to the call management system but is anywhere reachable via the PSTN or through voice over Internet or some other network.

A voice pathway 121 is created by the call management computer 101 in any of a variety of ways:

1. Placing a call through the PBX trunks 105 and the organization's PBX 104 or other switch to a called party's extension 106;

2. Activating directly a called party's telephone in the case where the call management computer 101 directly controls the telephone instruments 106;

3. Dialing out through the PSTN to a remote called party 111;

4. Connecting to a called party using an external network such as the Internet;

5. Connecting through remote links to another of the organization's PBX or other switches;

6. Passing voice over the organization's communications infrastructure within the site or external to it;

7. Playback of recorded voice messages may be done via a "Voice Pathway" created to the user's telephone instrument 106 or via the "Data Path" transferring the voice message to the user's workstation to be played out via the workstation's sound capability.

8. Any other such means which can establish a voice connection to a user's telephone instrument as defined.

Once established, the voice pathway 121 is used for the <u>entire</u> duration of that call and all other calls dialed or transferred to that same destination, until all such calls have been processed and the voice pathway is no longer needed.

Once a voice pathway is established to a system user such as user 113 or user 111, the calls being held in the call management computer 101 for that user may be rapidly "switched" 204 to a destination voice pathway 121 on demand as controlled by the user 113 or via the user workstation 114 through the call management window 115.

The central office trunk interfaces 203, PBX trunk interfaces 206 and assigned DSP 208 remain active throughout a call or series of calls to a destination, monitoring for either calling or called party to terminate the call or otherwise change the call state. Likewise the call management computer 101 monitors for the system user 113 to request change of the call state, e.g., "hangup", "Hold", etc.. When appropriate, the call path is dismantled by instructions to the circuit switches 204 and DSP 208 from the call management computer 101.

If either calling or called party has other calls waiting, the call management computer 101 does not "hang up" that end of the call path 121. Instead it is kept so that it may be re-used as a voice pathway for the waiting calls. Otherwise, the call management computer 101 instructs the trunk interface 203, 206 and/or DSP 208 to "hang up" the call and return to waiting for another call to be presented.

Voice pathway switching, rather than requiring the establishment and tearing down of multiple separate calls as conventionally done, provides the ability to switch rapidly between multiple calls, on demand, based on the user's changing priorities.

3. REAL-TIME PROTOCOL AND SIGNAL CONVERSION

The Call Management System provides for real-time protocol conversion between central office trunk type 102 and number and PBX trunk/circuit 105 types and number. The number and type of central office trunk/circuits need bear no direct relationship with the number and type of PBX trunk/circuits. This conversion allows the Call Management System's new and expanded features and functions to be implemented using existing telephone systems which cannot otherwise accept new telephone capabilities directly or in a cost-effective manner.

As shown in Figs. 1 and 2, the call management computer 101 attaches to central office trunk/circuits 102 from one or more central offices 103 on the one side and through PBX trunk circuits 105 to the PBX 104 and/or directly to telephone instruments 106. Each type of CO trunk and/or PBX trunk analog, analog DID, T-1, DID over T-1, ISDN, Internet or others is attached through its own appropriate type of interface board 203, 206 which converts the trunk signals to standardized bus signals for the circuit switches 204 and telephony signal buses 210 and it also monitors various aspects of the call.

To identify both the calling party 118, 119, 120 and called party 111, 113 as automatically as possible, it would be logical for the organization to elect to subscribe to modern telephone provider services even though the organization's existing PBX cannot reasonably or cost effectively be upgraded to support such new services. One example of such new services would be ISDN PRI, where the digital "D" channel identifies the telephone number of both the calling party and the called party for each of up to 23 or 24 circuits. ISDN PRI provides a good start toward automating proactive caller identification.

While most central offices 103 can provide ISDN PRI or other new CO trunk/circuit 102, 202 services, existing PBXs 104 are customarily outfitted with old-style analog DID or even simple analog 1FB or other trunks having entirely different signaling and control requirements from the new ISDN PRI. Rather than requiring an expensive or even impossible upgrade of the existing PBX 104, the call management computer 101 provides "Conversion" of the new types of CO trunk/circuits 102 with their new signaling requirements to older PBX trunk/circuits 105 with their older signaling requirements.

New ISDN PRI services are provided by the central office 103 through the CO trunks 102 to the call management computer 101, 201 where the circuits are "converted" to older analog DID services for the PBX trunks 105 to the organization's existing PBX adding all of the Call Management System features and functions without changing the organization's existing PBX investment.

"Conversion" is made possible by the structure of the call management computer 101 with the logical and physical separation between trunks, where each different CO trunk 102 and PBX trunk 105 has its own unique trunk interface 203 and 206 which converts the unique trunk signals to standardized circuit signals for the circuit switches 204 and telephony signal bus 210. The only thing that is common is that their voice paths can be connected together, when appropriate, using the circuit switches 204 and telephony signal buses 210. The handling of trunks/circuits from one side is independent of and separate from the handling of trunks/circuits from the other, yet their voice paths can be connected together to complete the circuit whenever appropriate, coming in via one type of trunk interface, converting to standard signals and going out via an entirely different trunk interface being "Converted" en route.

Because of this "Conversion" ability, no direct one-to-one relationship need exist between the CO trunk/circuits 102 and the PBX trunk/circuits 105. A special case occurs where the call management computer 101 directly connects to and controls the organization's telephone instruments 206 with no PBX switch 204 at all.

Real-time protocol and signal conversion thus provides a crucial enabling mechanism for all of the new features and functions of the Call Management System using information available through new types of telephone provider services, while still using an existing PBX which cannot otherwise support such services. In effect, the Call Management System only requires the legacy PBX to provide voice paths to user telephone instruments, to voice mail and other devices ignoring its other existing features.

4. INTELLIGENT CALL MANAGEMENT THROUGH REAL-TIME DSP VOICE AND DATA PROCESSING AND CIRCUIT SWITCHING

The Call Management System's unique "Intelligent" call management capabilities are based upon the real-time sensing, control, voice and data processing provided by the call management computer's 101 configuration of DSP processors 208, central office trunk interfaces 203, PBX trunk interfaces 206 and circuit switches 204 through which circuits calls are assigned to one or more DSP processors 208 at all times. The DSPs provide real-time DSP voice and data processing which is the essential means by which the content of each call is monitored and known. The Call Management System monitors call content and uses that information to provide "intelligent" call management, unlike conventional PBXs or other telephone switches, which strictly avoid knowledge of the call content and are conceptually limited to "call switching".

The amount of DSP processing power applied up to at least 800 MIPS per system coupled with the hardware sensors of the trunk interfaces 203 allowing the Call Management System to apply a broad range of voice and data processing tasks to each and every call, processing capabilities and power not available in conventional business PBXs or similar switches.

4.1 DSP Subsystem.

Each DSP subsystem consists of a DSP motherboard which attaches to the call management computer 101, 201 via the computer signal buses 211 and to the telephony signal buses 210 through circuit switches 204. In addition, the motherboard supports one to four DSP daughterboards of three DSP processors each 3, 6, 9 or 12 DSPs per subsystem with 100, 200, 300 or 400 MIPS of processing power.

Each call management computer 101, 201 utilizes one or more such DSP subsystems.

The commercially available Analog Devices DSP ADSP-2181 is used for the DSPs, operating at 16.67 Mhz 33 MIPS with 32K bytes of program RAM and 32K bytes of data RAM. DSP daughterboards are populated on the DSP motherboard as required to provide system support for broad categories of services:

"Call Monitoring" DSPs are those assigned to support the actual number of existing CO and PBX trunks, monitoring and controlling individual calls *e.g.* call progress monitoring, voice playback, etc. Each call monitoring DSP in this implementation typically can handle four through circuits/calls both CO and PBX sides for a total of 12 to 48 pairs per DSP subsystem.

"Special function" DSPs provide special functions as required for each specific system *e.g.* voice recognition, text-to-speech, etc.

Because of the anything-to-anything circuit switching capabilities of the Call Management System, DSPs can be assigned to any circuit/call as needed to support any particular function or set of functions.

4.2 Computer Signal Bus Interface.

The DSP motherboard connects to the call management computer 101 through its interfaces with the computer signal buses 211 through which the call management computer's software drivers control, monitor and pass information between the call management computer processors and memory 201, the call management databases 215, the DSP motherboards and the DSP processors 208 on daughterboards.

The DSP motherboard uses an industry standard PCI bus interface with industry-standard "plug-and-play" support:

1. Assignable interrupt level
2. Assignable shared memory addressing
3. Assignable control addressing
4. Other fixed and/or dynamically changeable parameters.

4.3 Dual-Port RAM.

Each DSP has its own 32K bytes of dual-port shared RAM memory which it shares with the call management computer processors 201. Thus, each daughterboard uses 96K bytes of shared memory and a fully-populated DSP subsystem uses 384K bytes of shared memory. The shared memory provides communications between the DSP and the call management computer's 101, 201 software drivers and minimizes the overhead of system interruptions external memory delay states. Shared memory is used for:

1. Passing voice buffers for voice playback and record

2. Passing Fax and data buffers

3. Providing "mailboxes" for control and status information.

4.4 DSP Signal Processing Task.

Each DSP operates with its own Multi-tasking software environment, sharing the available time and MIPS among a series of tasks. The number and choice of which tasks are active at any moment depends upon the state of each of several calls the DSP is handling. Each call is itself considered a "state" machine. These tasks include:

1. DTMF decoding Dual Tone Multi Frequency, with talk-off play-off prevention

2. DTMF generation

3. MF decoding and generation

4. Call Progress decoding Dial Tone, Ring back, Busy, Fast Busy based on cadence, etc.

5. Precision Call Progress Decoding based on frequency

6. Call Progress tone generation Dial tone, Ring back, Busy

7. Analog signaling control FXO LS/GS, FXS LS/GS, E&M, E&M Wink, DPO, DPT

8. Caller ID decoding FSK modem signaling

9. Caller ID generation FSK modem signaling

10. Voice Playback messages played out to the caller

11. Voice Recording voice name for caller identification, recording of voice messages and recording of calls

12. Voice compression and decompression PCM, ADPCM, etc.

13. Conferencing of calls chimes and voice

14. CNG Fax tone detection

15. Fax Group 3 reception and transmission including CNG tone generation and special Fax identification for reception and special Fax banner for transmission for each system user

16. X.36, X.34 and other modem protocols

17. BCLID data reception

18. Text-to-speech conversion

19. Text-to-Fax conversion

20. Voice recognition

21. Name recognition voiceprint

22. Many other tasks.

Most of these DSP signal processing tasks are not typically available in conventional business PBXs or similar switches but may available in interactive voice response and similar equipment. In the future, many more voice and data processing tasks will be added to the Call Management System's DSP processors 208, significantly expanding the functions and features of the Call Management System.

4.5 External Connectivity.

Each DSP motherboard also supports the following external connections:

1. External audio connection for "music-on-hold"

2. External audio microphone input which can be used for voice recording if desired

3. External headphone connection so that calls in progress can be monitored for debug purposes

4. Status LEDs red and green for self-test, board diagnostics and operational external notification

5. "Stay-alive" signal for the "copper bypass" unit.

4.6 DSP Motherboard.

Fig. 3 shows the DSP motherboard block diagram. The DSP motherboard functions primarily as an interface between the computer signal buses 211, the telephony signal buses 210, the DSP daughter boards and their DSPs 308 and internal decoders, sequencers and logic. This architecture utilizes common internal address and controls 315 and data 316 signals from the PCI interface 301 connecting the call management computer's 201 PCI bus 211 to essentially everything on the motherboard 208 and through it to the DSP daughterboards. Each motherboard 208 includes:

1. The FMIC telephony bus circuit switches 204 which provide signal paths to/from the telephony signal buses 210 and the DSP processors 208

2. The address counters 317, chip select circuits 319 and address and memory sequencer 318 which manages the on-board controls

3. DSP and FMIC controls are handled via DSP selects 320, interrupt control register 321 and the mailbox and error register controls 322

4. External audio input for music on hold and headset for debugging are provided 225 by a separate audio circuit.

Each daughterboard interface consists of:

1. Address bus 315
2. Data bus 316
3. Telephony voice circuits 313
4. Status and error signals to the DSP motherboard buffers and logic 314.

4.7 DSP Daughterboard Block Diagram.

Fig. 4 shows the DSP daughterboard. Each daughterboard 2081, 2082, 2083, and 2084 attaches to the DSP motherboard 208 through the daughterboard interfaces address lines 315, data lines 316, telephony circuit buses 313 and status/error lines 314. Each DSP 208, 308, 408 has its own dual-port RAM data memory 410 accessible either from the DSP or from the call management computer 101, 201 via the PCI motherboard interface 301.

4.8 Trunk Interfaces.

The call management computer's 101, central office trunk interfaces 203 support one or more trunks per board, depending upon the type of trunks, with varying number of circuits per trunk typically 1 or 2 T-1 or ISDN PRI to 24 analog, DID or stations. The trunk interfaces 203, 206 provide a variety of different connections, interfaces and features, as is appropriate for each different type, including the following central office and PBX connections:

A. Central Office Connections

1. Loopstart - Ringing detection

2. Loopstart - Loop current detection
3. Loopstart - Loop current direction
4. Loopstart - On/Off hook control
5. Loopstart - Hookflash generation
6. Groundstart - Ringing detection
7. Groundstart - Loop current detection
8. Groundstart - Loop current direction
9. Groundstart - On/Off hook Control
10. Groundstart - Hookflash generation
11. Groundstart - Ring ground seizing trunk for outdialing
12. Groundstart - Tip ground detection of seized trunk

13. AnalogDID - Battery power to trunks
14. AnalogDID - Loop current detection
15. AnalogDID - Battery reversal wink generation
16. AnalogDID - Reverse battery answer supervision identifying when the call is answered and when the station side disconnects
17. AnalogDID - DID decoding of DTMF signals
18. T-1 Trunks - Signaling type E&M Wink and Immediate, loopstart, groundstart
19. T-1 Trunks - Wink generation
20. T-1 Trunks - "robbed-bit" signal decoding
21. ISDN PRI - "D" channel digital support

4.9 PBX Connections

1. Loopstart - Battery power to trunk
2. Loopstart - Ringing generation

3. Loopstart - Loop current detection
4. Groundstart - Battery power to trunk
5. Groundstart - Ringing generation
6. Groundstart - Loop current detection
7. Groundstart - Loop "open" for idle state
8. Groundstart - Tip Ground seizing trunk for outdialing
9. Groundstart - Ring Ground Detection of PBX initiating call
10. AnalogDID - Loop current detection
11. AnalogDID - Loop current direction detection for wink and answer supervision
12. AnalogDID - On/Off hook control
13. AnalogDID - DID transmission via DTMF
14. T-1 Trunks - Signaling type E&M Wink and Immediate, loopstart, groundstart
15. T-1 Trunks - Wink generation
16. T-1 Trunks - "robbed-bit" signal decoding
17. ISDN PRI - "D" channel digital message support

4.10 Telephony Signal Buses.

In this implementation, the Telephony signal buses are based on the industry standard Multi-Vendor Integration Protocol (MVIP) which provide for 256 bi-directional voice/data channels, divided into 16 uni-directional or 8 bi-directional "streams" of 32 time slots each, operating at 2.048 Mhz. However, the buses could just as well have been based on SCSA, PEB or other types of available standards or conventions — so long as "clear", full-bandwidth circuit paths are provided among the trunk interfaces 203, 206, 207, the DSPs 208, and their circuit switches 204.

4.11 Circuit Switches.

Trunk interfaces 203, 206, the Internet voice interface 207, DSP 208 and others boards connect to the telephony signal buses 211 through their own circuit switches 214. The circuit switches are the FMIC Flexible MVIP Interface Circuit. The FMIC connects the specified time slots of the telephony signal buses 210 to/from the "on-board" internal circuitry.

At any time, the call management computer's software drivers can change the circuit switch 204 settings using commands through the computer signal buses 211. This is the means through which calls/circuits/voice paths are dynamically "switched" from one point to another, placed on "hold", or otherwise as described elsewhere.

5. "ONE NUMBER" RECEPTION OF VOICE, FAX, DATA CALLS

Without changing existing information and telephone systems, the Call Management System provides a single, unique "One Number" for each system user which is his "personal point-of-contact" and "never busy" telephone extension (his Telco DID or equivalent number or his call attendant or DISA extension number) for all voice, FAX and data communications. This "one number" is used to receive, identify and automatically handle all the user's voice, fax and data calls, one or several at a time using multiple trunk circuits, even when the user is on his/her telephone. Thus, user telephone extensions are converted into a "never busy" extension for voice, e-mail, FAX and data with direct control from the desktop computer where users exercise direct, real-time control of all calls including call queuing with multiple calls on hold, call transfers, call forwarding and other forms of real-time call processing not currently available. The use of only one number per user significantly reduces the costs, complexity, inefficiency and confusion of having multiple different telephone numbers for different functions.

The Call Management System uses only a single DID or extension number for each user such as user 111 or user 113 to receive all their direct calls, voice, fax or data in any mixture and number within the limit of the number of available trunks and circuits. Various calls to a system user may occur at any time or several may occur at the same time.

The Call Management System uses only a single DID or extension number for each user 111 or 113 to receive all direct calls, voice, fax or data in any mixture and number within the limit of the number of available trunks and circuits. Various calls to a system user may occur at any time or several may occur at the same time. The call management computer 101 is programmed to sort them out handle each appropriately, all at the same time. This "One Number" feature is a significant improvement over the conventional use of multiple numbers for different functions.

Fig. 1 shows four different types of callers: an outside voice caller 118, an inside voice caller 113, an outside Fax caller 119 and an outside data caller 120. These callers all use the same "One Number" to call the same system user 111 and they may do so all at the same time.

So long as a sufficient number of trunks or trunk circuits are available, all of the outside calls will be routed through the PSTN 100 to the CO 103 where they will be sent to the organization via the CO trunks 102, 202 and presented to the call management computer 101. When the calls arrive even if all arrive at the same time, their trunk interfaces 203, 206 and the assigned DSPs 108 receive and answer ALL the calls, determine the called party DID, ISDN or otherwise and determine the call type, voice, Fax or data.

For Fax or data calls, the call management computer instructs the attached DSP 208 to establish a FAX or data call session and to receive the transmission. When complete and stored on the Call Management Databases 215, the call management computer 101, 201 alerts the called party to the new Fax or data files.

For each voice call received from outside, the call management computer 101 with assigned DSP 208 proceeds with proactive caller identification, checks for applicable VIP rules and alerts the called party to the call even if other calls are currently active or waiting.

Thus, for each system user only "One Number" is needed to receive all voice, Fax and data calls from outside or inside with the Call Management System able to identify and handle each type appropriately, even if multiple calls for the same party arrive concurrently.

This "One Number" ability of the Call Management System removes the typical requirement for each user to have expensive, separate telephone lines and equipment for each different type of call and it also prevents the conventional "busy" signal being received by callers, improving efficiency and obviating starting telephone tag.

6. PROACTIVE CALLER IDENTIFICATION

Proactive Caller Identification is the means whereby the Call Management System augments and improves central office-delivered calling party identification. Even with no central office-delivered calling party identification, Proactive Caller Identification can identify the called party. Call Management System configurations are provided for organizations which can include one or more forms of called party identification from their telephone providers and those which cannot. However, even for those which can, the correct caller identification using central office-delivered information occurs for only a modest fraction of all calls, not from blocked lines, pay phones, cellular phones, etc. thus, proactive caller identification is required in any case.

If no or unusable calling party identification is provided by the telephone service provider, A Proactive Caller Identification of the Call Management System requests the calling party to provide identification of the caller to the called party. Call management computer 101 utilizes a DSP 108 to access a call management database 215 message to be provided to the calling party 118. A typical message is:

"So that I may inform Mr. Johnson of your call, please enter your home of office telephone number or state your name."

The calling party identifies himself through one of several means:

telephone keypad entry of an identifiable telephone number;

telephone keypad entry of an identifiable business number with attached extension number;

telephone keypad entry of a unique assigned PIN number; or

speaking his/her name with subsequent voice recognition by the call management computer.

Proactive Caller Identification is described using Figs. 1 and 2.

A typical voice call to a system user might well originate from an outside caller at a payphone 118. The PSTN 100 routes that call to the central office 103 which, in turn, presents the call to the Call Management System via the CO trunks 102 and through the trunk interface 203, circuit switches 204, and telephony signal buses 210 to the DSP 208.

For businesses, the forms which CO-provided calling party identification take are limited, although growing. These include:

1. Caller ID containing the telephone number and/or name of the billed party in FSK;

2. ISDN providing both the calling and the called party's numbers as part of the "D" channel communications;

3. ANI DTMF or FSK provided by inter-exchange carriers with some 800/900 services;

4. BCLID Bulk Calling Line Identification, in which a BCLID data line from the central office is used by the central office to provide the calling number;

5. Transmitting the calling number along with the DID number in either DTMF or FSK form.

When a call is received 501, two different and parallel functions are started, identification of the called party 502 and identification of the call type 503. The calling party will be identified. Identification occurs through receipt of the calling party's extension DID after "wink" or T-1 in-band signaling or telephone number ISDN "D" channel signal or via other signaling means.

When the auto attendant mode is used, the DSP 108 accesses the call management databases 215 to play a message to the caller 118. Entry is made of the called party's extension number or name encoded from the telephone keypad or the name of the called party may be spoken and then recognized utilizing conventional auto attendant steps.

If the called party is not a system user 504, the PBX 104 is used to process the call in the form required by the PBX trunks 105 and the call is switched over switches 204 via the telephony signal bus 210 to the specified number for conventional treatment.

If the called party is a system user 505, the calling party's telephone number is determined 506 automatically through the receipt of a name or number from the list above.

If a caller identification is received 507 the identification number is compared with the Primary Caller Identification Database.

If no automatic detection of caller identification occurs 508, the system provides a pre-recorded message to the caller 509 such as the one above. The requested information is received by the attached DSP 208 and used in subsequent identification.

If no response 511 is received the caller is considered "unknown" 521.

If the name was spoken 512, the call management computer 101 compares the name with entries in a voice identification database 214. If the name corresponds to one in the voice identification database 214, the calling party is identified 521. If the corresponding name is identified, the caller is "unknown", the recorded name is retained during the call, in case the user wishes to have the name added to the voice identification database 521 for future calls.

If a telephone keypad entry was made 513, the entry is used as an index into the Primary Caller ID Database 515.

If a match is found in the Primary Caller ID Database 516, the caller is identified as the party in the matching database record.

If a match is not found in the Primary Caller ID Database 517, the entry of a telephone number is used as an index into the Secondary Caller ID Database 518.

If a match is not found in the Secondary Caller ID Database 519, logic goes to step 521 with the caller considered "unknown". If a match is found in the Secondary Caller ID Database 520 the caller is identified as the party or business matching the Secondary Caller ID Database entry.

Step 521 represents the end of proactive caller identification. At that point, the call is handled according to any appropriate VIP rules 521 and/or the called party is alerted to the presence of the call 522.

This basic procedure can be accomplished with many variations which provide the same results but may add, move or remove various steps to accomplish it, *e.g.*, obtaining information from the caller can be done through a series of different requests and responses, not just the efficient single one described above, or auto attendant identification of the called party can be done following caller identification, instead of before.

Additional Proactive Caller Identification capability is provided to a called party once a call is received whether identified or unknown, by "double-clicking" a toolbar button requesting "More Identification", *e.g.*, a caller identified only as from "General Motors" may give too little information to the called party. Selecting the button to request "More Identification" causes Proactive Caller Identification to request a different number from the caller so that he may be more closely identified as "Mr. Jones" calling from "General Motors".

Voice calls arriving through Internet or other similar digital networks are identified using a "form" presented to the calling party, which he fills out providing the needed calling party information.

7. CONTINUOUSLY-IMPROVING CALLER IDENTIFICATION DATABASES

7.1 Calling Number Databases.

Identification of a calling party, *e.g.*, party 118, from the calling party's telephone or PIN number or other entries is accomplished with two Caller ID Databases which are part of the overall Call Management Databases 215. A Primary Caller ID Database is dynamic and continuously updating. It includes names, telephone numbers and/or affiliations of callers to the organization, including employees, and is automatically searched by the call management computer 101 first as soon as a number is known or through Proactive Caller Identification. If a match is found, the name from that entry is used by the Call Management System in subsequent VIP processing or to alert the called party.

The Primary Caller ID database contains specific entries relevant to individual system users, *e.g.* for system user John Adams, the automatically identified business number from one site of General Motors is assigned to Mr. Jones, but for a different system user, Sam Archer, that same General Motors site number is assigned to Sarah Smith. Since both entries will match the automatically identified number, the one matching the called party will be chosen. A further extension of this process includes numbers not assigned to an individual system user, but to their group or even to the entire organization. All of these are matched to the calling party and the one most appropriate for the called party is chosen. One reason for this process is that all calls from an entire business site are customarily identified by the billing number for the site's PBX, not by individual. Thus, this procedure increases the probability of correct identification of callers based upon who they are calling.

A second variation includes alerting the called party using all of the matching names and allowing the called party to select which one actually is calling, removing the others.

If no match is found in the Primary Caller ID Database, the Secondary Caller ID Database is searched. It contains a commercially available list of individual and business names and telephone numbers or an extraction from such a list. If a match is found to the Secondary Caller ID Database, the name and/or affiliation and number is transferred to the Primary Caller ID database to be used for subsequent calls. If no match is found, the caller is finally considered "unknown".

With or without a call present, the called party, such as user 111 or user 113, may use his workstation's call control window 115 to update or correct the Primary Caller ID Database for a caller or add a new caller to the database using any of the following indexes:

1. Home telephone number;
2. Business telephone number;
3. Business telephone number plus the person's extension;
4. Special PIN numbers assigned to or selected by callers;
5. Voice mailboxes for users;
6. Special coded entries including the * and # keys; and
7. Other indexes.

With or without a call present, the called party 111 or 113 may use the workstation call control window 115 to update or correct the Primary Caller ID Database for a caller, or add a new caller to the database.

7.2 Voice Name Identification.

Voice name identification is accomplished through a comparison of pre-recorded spoken names in a Voice Name Database which cross-references the person's name, affiliation and phone number similar to the Primary Caller ID Database.

The user can have a current caller speak their name, which is then recorded by the DSP 108 and stored by the call management computer 101, 201 in the Voice Name Database for future use. Alternatively, if the caller had spoken their name at the beginning of the call when requested, that saved recording may be used for the Voice Name Database.

Thus, the Primary Caller ID Database and the Voice Name Database are continuously updated both automatically and through user action, becoming ever more effective in identifying callers.

8. CALL NOTIFICATION & CONTROL VIA THE DIGITAL NETWORK WORKSTATION COMPUTER

The Call Management System provides intelligent call management through which calls are handled by called parties using their workstation computer, not the telephone instrument as with conventional business PBX or other telephone systems. A called party, such as user 111 or user 113, controls one or many concurrent calls directly through a call control window 115 displayed on a workstation 114. Section 1 describes the actions taken by the call management computer 101 and the relationships of calling parties and called parties for many of the specific user control examples. This section describes the user interactions at their workstations 114 using the many aspects of their graphical user interface call management window 115 and its subsidiary screens. Figs. 6a-6e show the call management window and selected subsidiary screens. It is understood that different combinations and organizations of screens, layouts, buttons, etc. can be configured to provide the user his many Call Management System features and functions.

Thus, the implementation shown and described is but one of many potential graphical user interface layouts which could be used to implement the user aspects of the Call Management System.

8.1 Call Notification of the Called Party.

When a call arrives for a system user, the call management computer 101, in concert with its DSP processors 208, identifies the called party, the call type and the calling party as described in Sections 1, 6 and 7. For voice calls, the call management computer 101 reviews any applicable VIP rules from the Call Management Databases 215 and, if none apply, to divert or affect the call, and the called party is in an appropriate "available" status, it notifies the called party 111, 113. Notification messages are sent through the digital network(s) 109 to the user's workstation 114. Notification of the user is accomplished through a call control window 115 on the user's workstation 114 which "pops up" when a call arrives; a flashing icon which, when double-clicked, launches the popup call control window; a special sound from the workstation alerting the user to activate the call control window; any combination of these or other alerting mechanisms.

This notification is not the conventional telephone "ring" typically used by existing telephone systems. Instead, it uses the separate, independent and high-speed information path of the digital network(s) 109 from the call management computer 101 to a called party. In addition, notification conveys significantly greater information to the user and enables an entire array of new features and capabilities not previously available.

Fig. 6a shows a basic call management window 115 user screen, as seen when it pops to the front on the user's computer display due to any of the alert reasons listed above, or because the user activated it directly. Fig. 6b is a fully expanded user screen as might be configured by a "power" user providing more readily available functions, but in a more "busy" environment. The call management window Figs. 6a and 6b supports a number of subsidiary windows through 6c-6e designed for specific purposes and described elsewhere.

The computer program behind the call management window is kept active at all times when the user is in any of the "available" states. It is designed to be consistent and compatible with Microsoft Windows and other appropriate conventions for its overall look and feel, menu conventions, buttons, borders, help, etc. A user can activate different features in a variety of ways including the following, all of which are referred to as "selecting":

1. clicking the primary mouse button on the selected button or item;

2. clicking on the associated menu, such as "Calls" and then clicking on the selected item;

3. double-clicking call notifications to transfer to that call;

4. clicking the secondary mouse button on a call notification to bring up a short menu of available options and then clicking on the selected one ; or

5. typing in the associated keystrokes such as "Alt-A" for the "answer" button.

The main call management window 115 screen 6a is logically broken into the following areas:

8.2 Customer Logo.

On the top left of the screen is the customer's or vendor's logo 601 as shown. The logo can be changed at any time by providing a new graphic file to replace the existing one.

8.3 User Status.

The elongated button just below the logo is the user's status button 602 which the user may select to change his status to the system. User status includes:

1. "available to receive all calls";
2. "available only for VIP calls";
3. "unavailable - transfer calls to another location;
4. "unavailable - until 4:00".
5. "unavailable for all calls" .

For the fourth listed status, when the time occurs, the status would automatically change back to "available".

The last listed status provides a list of options such as "Out to Lunch"; "In a meeting"; "On vacation"; "On a sales call"; "With a customer"; and others. Alternatively the user may type in or modify one of the standard options to provide more useful information for other users (see

Section 17). Examples include "Out to lunch til 2:00"; "Giving a demo til 4:00";
"Out of the country til July 21, send everything to Judy"; and others.

8.4 The Message Board.

Voice mail 606, along with Fax and data messages 603, e-mail 605 and "Flash" Notes 604 are "historical" messages, and are treated differently than "real-time" calls. For these historical messages, the user's call management window contains a "message board" area just below the user status button with an appropriate control button for each type of message 603-606. Whenever a message is available to be reviewed, *e.g.*, a new Fax message, the Call Management System highlights the appropriate button name in flashing red and places a number adjacent to the button indicating the number of such new messages available to be reviewed. This procedure separates the various types of historical messages and substantially simplifies reviewing messages by users.

8.5 "FAX" Notification.

When a user has received a new Fax transmission, the "FAX" button is highlighted and the count of new Faxes is provided. Selecting the "FAX" button launches the FAX selection subscreen. That screen shows the user's list of Faxes and summarizes the total number of Faxes, and identifies the number which are new messages (see Section 13).

The user may select one or more of the Faxes to be viewed or handled in other ways. Also, the user may select a checkbox to limit the display to only new Faxes. If the user selects a Fax to be viewed, the computer operating system's FAX viewer is launched with the name of the selected Fax file, popping the selected Fax up in front of the other windows.

8.6 "Flash" Mail Notification.

Flash Notes sometimes called "Flash" mail, are quick, simple messages which may be passed among system users. They are the equivalent of an "electronic shout across the office" intended as an improved, electronic version of the classical "pink slip" notes on which messages used to be written then carried to the intended party and presented to him. The user is notified of his unread flash notes by a highlighted "Flash" Notes button and an associated count of such unread notes. Selecting the "Flash" button allows the user to review his Flash Notes as shown in Fig. 6c. The Flash Notes screen indicates the originator 660 and the note itself 661, and allows the user to close it 662, reply to it with another Flash Note 663, or to return the call 664 by making a voice call to the originator. Flash notes can be sent to one or to many system users at a time, selecting multiple users for the note.

8.7 "e-mail" Notification.

When a user receives a new e-mail message, the Call Management System is notified by e-mail services assuming the organization's digital network e-mail services provides this capability. The user is notified immediately of his unread e-mail by a highlighted "e-mail" button with associated count of unread e-mail messages. When the user selects his e-mail button, it launches the organization's e-mail client program, allowing the user to review and read his new e-mail messages. One special feature the Call Management System provides is the ability to record a voice message and attach that message to an e-mail message for later retrieval by the recipient. This feature is based on the capability of the organization's e-mail system.

8.8 "Voice-mail" Notification.

When a user receives a new voice-mail message, the Call Management System is notified by voice-mail services, assuming the organization's voice-mail services provides this capability. The user is notified immediately of his new voice mail messages by a highlighted "voice mail" button with associated count of new voice mail messages. If the Call Management System includes an integrated voice mail subsystem, the voice mail messages may be retrieved in any order from the list provided. See Section 16 for a further description. Otherwise, the user accesses his voice mail messages serially via his telephone instrument, as is conventionally done.

8.9 User's Call Status.

Near the top of the call management window 115, adjacent to the customer's logo, is provided the call status of the user 607, to whom he is currently connected, and the length of time the call has been connected 608 to the user.

The length of time the call has been connected to the user is different than the length of time the call has existed as shown in the "time" field of the call alert box below. When the user has selected an "unavailable" status, the message on the screen is that selected or created status message.

8.10 Call Alert Box.

The call alert box is immediately below the user's call status, and it displays all real-time call alert messages for the user to control his real-time calls. The components shown on the screen for each call include: "CALL STATUS ICON" 611 sometimes flashing or moving which indicates the status of each individual call; and a "CALL STATUS STATEMENT" 612 showing the individual status of each call. The call status statement may state that the status is "ringing", "connected", "holding" or "recalling". "Ringing" indicates the call has been received by the call management computer and processed, but has not yet been answered by the user. "Connected" indicates the call has been accepted by the user, put through by the call management computer 101, and is currently the "active" call. "Holding" indicates a call has been previously connected to the user and then placed on hold. "Recalling" indicates a call has been holding for longer than a predefined length of time, typically 60-120 seconds, but the call is still "holding'

Also displayed are the following: TIME 613 which is the time since the call was answered by the call management computer 101; CALLER 614 to identify the name and/or affiliation of the caller or "Unknown"; and NUMBER 615 to identify telephone number from which the call was made, when known, including: telephone numbers, "Internet" and other applicable identifiers.

"Source" identifies the source of the call as "outside", "inside", "Internet", or "other".

FOR "group" handling or employees who have temporarily moved to a "meeting" environment, described below, an additional field is displayed showing the initials of the called party.

This call information display occurs even while other calls for the same called party are in process, allowing the called party to know who is calling and to apply appropriate priority to each call.

When a call noted as "Unknown" is received, the called party may speak to the caller and determine who they are. The called party may choose to enter that person in the Primary Caller ID Database so that they are identified properly the next time they call; or have them speak their name for recording in the Voice Name Database for future calls; or

simply type the caller's name and/or affiliation as part of the call information for use as the call is transferred to others in the organization.

In any case, the call information is corrected in view of the known identity of the caller.

8.11 Workstation Real-Time Call Controls and Management.

A called party can control one call, or many concurrent calls, directly through a call control window 115 on a workstation 114. Section 1 describes the actions taken by the call management computer 101 and the relationships of the caller and called parties for many of the specific cases. This section describes the user's interactions, features and functions for call alert, control and management. A user can exercise call controls by any one or a combination of the following or other actions:

1. clicking the special call-control buttons shown just below the call alert window 616-620 ;

2. typing the letter underlined for each button or menu item, *e.g.*, Alt-A for "answer";

3. opening and selecting from the "Calls" menu using the mouse or typed control letters Alt-C;

4. double-clicking the call to be answered see below;

5. secondary clicking the call to be handled to bring up a list of call commands, then choosing the desired command;

6. dragging and dropping a call alert onto a call function button; and

7. using the cursor keys to move among call alerts.

User controls may be exercised via the specialized call control buttons placed just below the call alert box, or call controls may be selectively activated from screen menus, or from the special "Button Bar" shown on Figure 6b. These controls and management functions include:

1. ANSWER 616 answers the selected call. If another call was active at the time, it will be placed on "Hold" or "Hang Up" according to the user's selected options.

2. HOLD 617 places the call on hold, disabling both inbound and outbound voice paths, and starting the "recall" timer.

3. TRANSFER 618 transfers the caller to anywhere in the directory described below, inside or outside the organization, or to a typed in number anywhere in the organization or accessible via the PSTN (see Section 11). For calls transferred from secretaries or telephone receptionists providing "group" support, the default for transfer is the person to whom the call was originally intended. Also, when a call is active, the individual "Speed Dial" buttons or page of Speed Dial buttons (see "call origination" below) become different "Speed Transfer" buttons for single-click, rapid transfers.

Transfers can be directed to any entry in the directory (person, location, inside, outside, etc.). Transferred calls are identified to the transferred party with the initials of the transferring party as further information for call management. When transferring a call, the call can be "Tagged" with a digital message or it can be transferred with an appended voice message selected by the user from his drop-down list and played out at the time the call gets "Answered" by the transferee.

4. HANG UP 619 hangs up the call, releasing the caller and dismantling the caller's part of the call path.

5. VOICE MAIL 620 sends the caller to the user's voice mail box (see Section 16).

6. DIRECTORY 630 opens the directory for update or use (see below).

7. CALL LOG 631 opens the call log screen and displays the call log of the user's calls (see Section 14).

8. VIP RULES 632 opens the VIP rules screen for update (see Section 10).

9. GROUP HOLD 633 sends the selected call, not necessarily the active call, with any attached "Tag", to all members of the selected "specialty list" for servicing by the first available member, potentially including the user (see "specialty list" description below).

10. "FLASH" NOTE 634 creates a "Flash" note for sending to another user, as described above and in Fig. 6d. Flash notes can be sent to one or many system users at a time (selecting multiple users for the note).

11. TAG CALL 635 adds an electronic message to the selected call, not necessarily the active call, which will display on the user's call alert box and on any other system user's call alert box to whom the call is transferred or conferenced (see Section 12). An example of a "Tag" is shown in Figure 6b attached to the second call 645 in the call alert box.

12. "BOMB" SELECTED CALLER 636 creates a VIP rule, or applies to the caller an existing VIP rule, which handles the current and all future calls from the selected caller, *e.g.*, the "cold-calling broker", in a predetermined manner.

13. "VIP" SELECTED CALLER 637 creates a VIP rule, or applies to the caller an existing VIP rule, which elevates the selected caller to a specific VIP status, by providing distinctive ringing, follow-me routing, etc.(see Section 10).

14. "PIM" ACTIVATION 638 links to a selected Personal Information Manager or database with the information about the selected call, not necessarily the active call, in order to provide "Screen Pop" of the information associated with the caller. Note that if multiple PIMs or databases are used, a pull-down selection list is provided for the user to select the appropriate one to use for this specific call. This feature can be done either by clicking the "PIM" button or when the call is answered for automatic "Screen Pop" (a user preference).

15. MUTE 639 disables the outbound voice path, retaining the inbound voice path of the active call.

16. CONFERENCE CALLS conferences the selected call, not necessarily the active call, with the selected party or group of parties from the directory or as typed in by the user. This includes anyone internally, working at home, or external to the organization anywhere in the PSTN.

17. RECORD CALL records the active call for later playback and analysis.

18. PLAYBACK RECORDED CALL plays back a recorded call selected from the list of such calls in its own subscreen. Playback is done via a "Voice Pathway" created to the user's telephone system 106 or the "Data Path" transferring the

voice message to the user's workstation to be played out via the workstation's voice capability.

19. MORE ID causes the call management computer 101 to play out a voice message from the Call Management Databases 215 asking the caller for additional information with which to identify him, including: entering a different telephone number or stating his name. Once done, the call is returned to the called party.

20. STATE CALLER'S NAME plays out the caller's name for the selected call, not necessarily the active call, as spoken by the caller during proactive caller identification or More ID interaction phases. The call management computer 101, 201 uses or creates a voice path to the called party and states the caller's name as spoken and recorded or uses the "Data Path" to transfer the voice message to the user's workstation to be played out via the workstation's voice capability.

21. PLAY MESSAGE selects a pre-recorded message from the pull-down list of messages and possible actions to be played out to the caller, *e.g.*, "I am tied up at the moment and am transferring your to Sam", and an action to be taken after the message is played out, *e.g.*, "Transfer call to Sam or return call to me, etc.". In effect, this is the creation of a temporary VIP rule for use with the highlighted call.

22. VIP RULES selects a VIP rule from a drop-down list to be applied to the highlighted call.

21. TRANSFER USER transfers the user from the current "available" state on this computer to the computer associated with the selected other user, and changes the current computer to the "Unavailable" status allowing the user to type in a reason for other users (see "meeting" environments described below).

22. USER PREFERENCES. The user may modify his screen and operational preferences at any time by selecting items from the "User" menu including a "Preferences" subscreen, allowing the user to change any of a series of his own operating characteristics including displaying the "Button Bar" 630-639 of Fig. 6b; displaying the user's "Speed Dial" buttons 640; displaying the call management window's "Status Bar" 641 showing the user's name, the current date and time; and when double-clicking a call alert, place the original active call, if any, on "Hold" or "Hangup".

User control messages resulting from user interactions with the call management window 115 and its subsidiary screens are returned, via the digital network(s) 109, to the call management computer 101 where they result in the requested actions being performed, and/or appropriate interactions with the Call Management Databases 215.

8.12 "Directory" Support.

One important aspect of the Call Management System is support for the "Directory", both corporate and personal. The directory user access shown in Figure 6d contains:

1. ENTRY 681. The name/affiliation of a person or organization including employees, voice mail, places, *e.g.*, "conference room", outsider.

2. NUMBER/STATUS 682. The received telephone number or status of the entrant, including local numbers without area codes; long distance numbers with area codes; international numbers with country codes, city codes, etc.; Internet access; status for system users; and other.

3. RETURN TELEPHONE NUMBER The telephone number to call that person, if different from above.

4. VIP STATUS defines this person as having VIP status.

The user maintains the directory through subscreens to add "New" 683 entries, to "Delete" 684 an existing entry or to "Edit" 685 a selected entry. The user may define an entry as "Corporate", available to all users, "Group", available to users specified as part of a specified "group", and "Private", available only to the user himself.

When the directory is opened, the user may select how it is sorted, limited, and displayed, including internal entries only, external entries only, private entries only, sorted by name, sorted by affiliation, sorted by number, limited to area codes, or ranges of names, and others.

Access to entries is conveniently available by scrolling the sidebar 686 using the mouse, scrolling down or up the entries using the cursor keys, clicking letter buttons 687 to jump to the start of names beginning with that combination of letters - after a limited time, three seconds or so, the list is cleared and the user may start again. As the user types the name, each letter entered causes the list to jump forward to the first of the names beginning with the list of characters entered. Once a match has been made, the user can key "Enter" or click "Call" to originate a call to that number, or the user configuration can be set so that the number is automatically dialed as soon as a match is made. After a limited time, three seconds or so, the list is cleared and the user may start again; and others.

The directory is used throughout the call management window 115 functions for routing calls VIP rules, originating calls 688, reviewing user status, transferring and conferencing calls, sending "Flash" notes, and many other things.

While the directory is a part of the Call Management Databases 215, it is used in a wide variety of ways and with entry sorting and display as appropriate to each.

8.13 Call Origination.

The Call Management System provides flexibility and user convenience for "originating" calls, in addition to its features and functions for handling "incoming" calls. The user may originate a call at any time, even while other calls are active or waiting, by opening the "Directory" described above, selecting an entry and clicking "Call" or pressing "Enter" as described above; opening the directory described above, typing a name, and having the system automatically begin dialing when a unique match has been found or waiting until the user clicks the "Call" button as selected for each user; clicking "SpeedDial" buttons 640 defined uniquely for each individual from the "directory". Multiple pages of SpeedDial buttons are provided for users requiring rapid access to a large number of dialable numbers.; or activating the "Dialpad" button 642 to bring up the dialpad subscreen; utilize the Secondary Caller ID database to find a number and/or name, then outdial it.

The "Dialpad" subscreen, shown in Fig. 6e, includes a representation of the telephone dialpad with the numbers 0-9, # and * 690. Calls may be originated by any of the following:

1. Clicking on the dialpad buttons to enter a dialable number then clicking on the "Dial" button. Note that the "flip" button allows the user to "flip" the dialpad so that it matches an adding machine keypad rather than a telephone keypad. Once entered, numbers may be removed one by one by the "Back" button 691, or cleared altogether 692. "Pause" 693 codes may be entered for use with modems, dialers, etc.

2. Typing in a number from the keyboard and then the "Enter" keyboard key.

3. Clicking the "Redial" button 694 then selecting any one of the last five called numbers shown.

4. Activating the "Directory" 695 and choosing an entry to be called.

5. Typing in a name from the directory and having the system dial that person when a match is made.

When outdialing a number, the Call Management System provides the ability to "Redial" repeatedly until the call gets through an otherwise busy line.

When no call alerts are present and no user calls pending, the call management window 115 shows a "Recall" button and a "Directory" button in place of the specialized call control buttons below the call alert box. Clicking them brings up the "recall list" or the "Directory list" as described above, simplifying call origination.

8.14 "Outside" Employee Support.

The Call Management System supports "Outside" employees who are rarely or never in the office and who have no workstation. Such "Outside" employees are given "pseudo" numbers for which no actual workstation or telephone extension exists, which may be used by callers. The "Pseudo" user establishes his own VIP rules which automatically handle all his calls, including his own voice messages, for selected callers and re-routing of his calls to other users.

Thus, the "Outside" person is supported, just like other system users, with callers being unaware that person is not in the same location.

8.15 "Group Secretary" Support for Calls to Specified Groups of Employees.

The destination party for calls does not have to be the actual called party. The Call Management System provides for a specific user to handle calls for a select "group" of others, system users or not. This "group" handling mode is intended for a secretary or telephone receptionist handling a group of executives or others, and for an organization's telephone operator. In either case, the specified user is expected to receive calls, prioritize them appropriately, respond to each accordingly, attach "call tags" and transfer calls as needed, thus effectively screening calls for the specified group.

The basic difference between direct notification of a system user and notification of the "group secretary" receiving calls for the individual, is that the call information displayed on the call management window is expanded to include the initials or other identification of the actual called party, allowing the secretary to know to whom the call was destined, and handle each call as appropriate. This capability requires appropriate information in the call management databases, identifying the group of people, the secretary to answer the calls, the initials or other identification for the called parties and other control information. On a dynamic basis, system users can be added to or removed from "Group Secretary" mode by those assigned "Secretary" status, allowing for the cases where a user runs by the secretary on his way out saying "I have to go, please handle my calls".

8.16 "Meeting" Support for Users Away from their Workstation System.

Users typically use their own workstation computer for most of their activities, but occasionally users need to leave their workstation and move to another location for a meeting or otherwise. Users can move to another location and, using the call management window on the computer where they are currently active and logged in, they may log off and choose an appropriate "unavailable" state, or select the "TRANSFER USER" function and move to the new site. Once there, they may log on to the call control window of the computer at that site, along with others who may be sharing the meeting site, or the person whose computer is at that site. When this happens, a temporary "group" definition is created and the users who are at that site sharing the site's computer, are treated as described above, with their call notifications sent to the site's computer as though a secretary or operator were screening the calls even though the individuals may be managing their calls themselves.

This procedure allows users to hold meetings in offices or conference rooms while continuing to receive and handle their calls as appropriate.

Users who attend meetings outside the office may create the same capabilities by using their portable or other computer at the outside meeting site, calling into the Call Management System via Internet or otherwise, and providing the telephone number for the meeting site. With this, one or more users can continue to receive and handle callers as though they were in the office.

8.17 "Specialty-List" Support for Special User Groups.

The Call Management System provides the ability to define multiple "Specialty lists" of users who work together or have certain affinities, *e.g.*, sales, accounting, customer service, manufacturing, etc. These groups represent arrays of users who can receive and handle certain kinds of calls, such as sales calls, customer service calls, etc. Each such special group is assigned a pseudo extension number through which it may be reached. The purpose of these "specialty lists" is to provide a new means for rapidly getting calls to the first available member of the group able to take the call.

When a call is made directly to the pseudo number or, a call is transferred to the number, the call management computer 101 concurrently alerts all of the users in the group by sending alert messages down the digital network(s) 109 to their workstations 114 and thence to their call management windows 115. The first person in the group to "answer" the call causes the call management computer to connect the call to the answering party and to remove the call notification from the call management windows of the other members of the group.

Specialty List Handling is a means to provide rapid access to the desired function within an organization, without having to go through an operator as is otherwise customary.

8.18 Feature Activation.

These many features can be turned on individually for each user, maximizing the effectiveness of the organization's training and call management.

8.19 TAPI Client.

The workstation software also provides a TAPI client for other applications which need to outdial or receive calls, *e.g.*, this feature means a PIM can place a TAPI call to have a number automatically dialed.

8.20 Automatic Updating.

The Call Management System automatically updates the user's workstation software from the call management computer whenever changes are needed. This is invisible to the user and is handled automatically at user logon time.

9. MULTIPLE CALL HANDLING USING A SINGLE EXTENSION

The Call Management System allows each user to be aware of and to manage multiple calls at the same time, using a single extension number, and using a simple POTS telephone instead of the customary expensive, multi-button, proprietary telephone instruments 106. By using the digital network(s) 109, workstation computer 114 and call control window 115 for communications with the called party 111, 113, the Call Management System is freed from the conventional limits on when, how and how much information can be communicated to the called party and likewise, when how and how much control can be exercised by the called party.

The Call Management System presents multiple calls, as they arrive and are identified, (Sections 6 and 7) to the called party 111, 113 using the call management window 115. Multiple calls are identified to the user through call alert information which is displayed in the call alert box of the call management window (see Section 8). The called party, applying his own preference, may then deal with each call as most appropriate. Calls can be: answered, placed on hold, sent to voice mail, transferred to an individual or "group list", recorded, hung up or any of many other appropriate actions. All of these capabilities are under the control of the called party, and all can be exercised for any of the calls, thereby giving the user complete freedom to manage their calls as best fits their ever-changing priorities, *e.g.* switching from a cold-calling broker to an important customer.

The called party can effectively interact with multiple callers concurrently by rapidly "switching back-and-forth" as needed among calls. The called party simply double-clicks a call, other than the one that is currently active, and the call control window 115 causes a message to be sent down the high-speed digital network(s) 109 to the call management computer 101 which instructs the circuit switches 204 to transfer the voice path of the new call to the re-usable voice pathway 121, 221 to the called party and leave the previous connected call on hold or hang it up as defined by the called party for their own specific configuration. In this way, the called party can "bounce back-and-forth" rapidly among many calls, servicing them as priority dictates. This capability is in sharp contrast to the time-consuming creation and tearing down of entire connections each time, as required by conventional PBX and other switch systems 104.

Thus, handling multiple calls at the same is simply, logically and rapidly controlled by the called party, using the computer's mouse or keyboard while repeatedly reusing the existing voice path.

10. USER-DEFINED VIP CALL HANDLING

VIP call handling is an automated adjunct to or an alternative for the call control window 115 notification of the called party 111, 113 and corresponding workstation control commands. With VIP call handling, specific callers, groups of callers or all callers are given unique treatment based on "VIP rules" defined by the organization and/or individually by the user. Call management window 115 "VIP Rule" subscreens allow the user to manage their own VIP rules. The VIP rule subscreens of the current implementation are shown in Figures 7a-7c. As with the other aspects of the user interface, this is but one potential configuration of buttons, selections and controls which can be configured to allow the user to create and manage his VIP rules. VIP call handling rules for each user are maintained by the call management computer 101, 201 as part of the call management databases 215 and are dynamically changeable by and for each user, except for corporate-wide rules. Changes to the rules can be made through the user's VIP Rule subscreens of the call management window 115 on their workstation 114, through the user's laptop or other remote computer attaching through a digital network or through the user calling and changing characteristics through direct telephone entry.

VIP handling for special callers, customers, etc., is a major improvement in the efficiency and effectiveness of call handling for organizations. VIP rules allow each user to tailor how callers are handled and routed, and are especially effective for "follow-me" routing in which an important caller, or group of callers, can be assigned to a rule which specifies that: when you are out of the office to call automatically to your cellular, car or other phone. Later, the user can call and change the rule to route calls to a different phone number. Additional features include, "find-me" capability where the user can have several numbers automatically dialed, such as car, cellular, home, etc. to find the called party for direct connection to the caller.

The basic VIP Rule subscreens consist of a list of existing VIP rules which may be selected 701; the ability to activate an additional subscreen to "Add" 702 another rule; the ability to "Delete" 703 the selected rule; the ability to "Copy" 704 a rule for later change; and the ability to make each rule currently "Active" or "Not Active" 705.

Each VIP Rule contains three parts to define who the rule applies to, what action is to be taken, and when the rule applies. Fig. 7 illustrates the three displays for the who, what, when parts of the VIP rules.

Screen 7700 is used to define "<u>WHO</u>" does the rule apply to. The rule will apply to the caller that is highlighted on the display. The rule applies to specific individual callers 710 whether included in the "Directory" or just typed in; selected callers 711 from the directory and/or typed in to whom the rule will be applied; or everyone who calls 712, typically for use when the caller is "unavailable".

Individual callers or groups of callers can be included in multiple VIP Rules, allowing the user to specify different handling under differing conditions.

The Call Management System provides for multiple greetings messages to be played out to defined groups of callers based upon their calling numbers, *e.g.*, all callers from New York area codes get one message and then get transferred to Sam. This provides special treatment for callers from different locations.

Screen 7701 is used to define "WHAT" actions are to be taken. VIP rules can result in one or a series of actions to be taken:

1. Play out selected pre-recorded messages 715 to the caller, personalized for the caller and recorded in the called party's own voice, *e.g.*, "John, I'm out of the office today. If you need to speak with sales, press one, or to speak to Sam, press two. Otherwise, I will call you back tomorrow." Or "John, I'm on the phone right now, but don't hang up, I'll be right with you.". These messages can also be used for "Callback" responses to important persons, *e.g.*, "Mark, I got your FAX and have found the shipping date is December 12. Let me know if I can help you further."

Creating new messages is done by selecting "New" from the list of prerecorded messages, whereupon, the call management computer 101, 201 will establish a voice pathway

121, 221 to the user's telephone instrument 106 and the user will record the message. When completed, the user will name the message, edit it as needed and then have it available for the VIP rule being created or modified.

Special "Menu" messages can be created, *e.g.* "To speak with Tom press 1, to speak with Sam, press 2, to speak to my secretary press 3 or to speak to the operator press 0".

2. Receive information entered by the caller via the telephone keypad, spoken or by other means.

3. Process the entered information, re-routing the call to one of a series of other numbers based upon the entered information, *e.g.*, "2" to reach Sam.

4. Use the entered information as a "Tag" to the call (see Section 12).

5. Activate another VIP rule based on information input by the caller.

6. "Forward" 716 the call to another destination anywhere inside or outside the organization as typed in or selected from the directory.

7. Transfer the caller to voice mail 717 whenever he calls.

8. "Return to me" when the transferred call is complete.

9. "Follow-me" routing for certain callers allows the user to specify one or a series of numbers to be used to reach the user when the specified caller calls. The user switches from one number to the next by calling his own extension and entering telephone keypad numbers to change

from one number to another, *e.g.* car phone, home phone, cellular.

10. "Find-me", "Chase me" routing for certain callers allows the user to instruct the Call Management System to call a series of different numbers simultaneously, attempting to locate the user. When found through voice entry or information entered via the telephone keypad or otherwise, the caller is switched automatically to that number, *e.g.*, home, car phone, cellular, etc.

11. "Page me" or "Beep me" instructs the system to place a call to the specified beeper service and page the user. It also includes transmitting the caller's name and telephone number for display on the user's beeper, where that service is available. This is useful for both real-time calls which are not completed because the user is "unavailable" and for when calls are sent to voice mail. One variation of the "Page me" approach allows the caller to stay on the line while the page is made and then, when the caller calls his own "One Number" and identifies himself, to connect the two calls together.

12. Provide "Priority Ringing" 718, a special sound when this caller calls to alert the user to a VIP caller.

13. "Hang Up" 719 when this caller calls.

14. Place the caller on "Hold" 720 whenever he calls.

Screen 7702 is used to select <u>"WHEN"</u> the rule applies. The following selections may be made for "When":

1. For certain status of the called party:

always 730

unavailable for calls or not answering 731
available but on the phone 732
available for VIP calls only, etc.

2. Within defined days of the week 733.
3. Within specified times of the day.

Certain VIP rules are pre-defined and merely have callers added to them. These include:

1. "Bomb" the caller and send all his calls to user's voice mail.
2. "VIP" the caller, giving him special ringing and notices.
3. "Follow Me" having the system follow the user.
4. "Find Me" having the system find the user.
5. "Page Me" having the system page the user.
6. Others.

10.1 Temporary VIP Rule Usage.

The user may highlight a call in his call alert box and then select a rule from a drop-down list for immediate application to that call, even though the caller does not normally have that rule applied to him. The user may also create a temporary rule (play out a message and specify a following action) for handling a highlighted call.

The Call Management System further enhances the user's ability to handle his important callers by providing the ability for the user, when out of the office, to call to the system and record new voice messages and/or change selections for his VIP rules via entries from his telephone keypad voice recognition or otherwise.

10.2 Advanced Message Notification.

The user may establish a set of rules that specify how the user is to be notified of various types of messages. For example, the user could set up a rule like, "If I receive an e-mail message from Victor between 5 AM and 7 AM, call me at my car phone; if I don't answer, page me." Another example might be, "If I receive a Fax from anyone at Motorola, forward a copy to my home Fax machine." One of the Call Management System's primary capabilities is to function as a central repository for message notifications, where the user has complete control over the notification mechanisms.

11. ROUTING CALLS INSIDE OR OUTSIDE TO THE ORGANIZATION

Unlike existing PBX or other telephone switching systems, the Call Management System routes calls internally within the organization or externally anywhere in the PSTN. Thus, calls can be transferred by a called party, or via VIP rules, anywhere the telephone network reaches. This is a major departure from the conventional telephone switch capability, which is customarily limited to intra-organization routing only.

When a call is received by the call management computer 101, the calling party, called party, and call type are used to determine the needed action. For voice calls, the called party may be alerted, receive the call and transfer it to a specified number or VIP rules may specify transfer to another number. It matters not that the number specified is within the organization or external to it. In either case, the call management computer transfers the call as specified.

For internal calls, the call management computer establishes a voice pathway 121 to a destination extension, e.g., user 111, and alerts the called party if that destination applies to a system user. For destinations outside the organization, an available outgoing CO trunk circuit 102 is seized or a two-way circuit is negotiated through which a voice pathway is established and the call path transferred.

Calls utilizing Internet voice capabilities are placed through the Voice-over-Internet interface.

The ability to route calls anywhere, not just within the organization, is a major improvement in efficiency for the organization.

12. "CALL TAGS"

The Call Management System provides a mechanism whereby system users can "tag" calls with digital messages, which then remain with the call, but can be modified so long as the call exists, no matter to which system user it may be transferred. Call tags are a unique ability, available only because of call management via the user's workstation computer for which digital information is conventional, unlike telephone instruments with their lack of or minimal display capabilities. These digital call tags provide an advanced and convenient means for one user to provide useful information to other users, within the organization, about the needs or interests of the caller, or anything else that may be appropriate, *e.g.*, "Sam needs to know about system installation" or "John has a question re: pricing".

A system user may "tag" any call showing in the call alert box of his call management window 115 with or without his answering the call, thus creating a digital message which, along with the initials of the tagging party, immediately appears attached to the call alert line. As that call is transferred to or conferenced with one or more other system users, the notification and attached call tag moves with it, always displaying along with the call alert line. Any receiving system user can modify, expand or even erase the call tag message, as appropriate.

"Special Tags" can also be added to a call which provide for return of the call when completed by the transferee; return of the call if not taken by the transferee; transfer of the call to another destination when done; or others.

Call tagging is an electronic tool similar to the old-fashioned "pink slip" notes of paper which were used to pass messages within an organization. However, call tagging is a significant improvement, since it is digital, automatic and, specifically, ties the call tag message to the call itself. Call tagging is a powerful new means to improve efficiency and information flow within an organization.

13. FACSIMILE FAX AND DATA CALLS

The Call Management System converts every system user's telephone extension into a never busy FAX gateway, with automatic detection and reception of incoming Fax messages, and confidential delivery to the recipient's desktop computer. No Fax hardware need be added at the desktop, and no special Fax telephone lines or numbers are required to receive Faxes, Faxes or data transmissions are simply sent to the user's "One Number" unique personal point-of-contact extension. The system detects and automatically receives each Fax or data transmission and electronically delivered it to the user's desktop machine, without his extension ever ringing or interfering with normal voice traffic (private, paperless, immediate delivery of electronic transmissions.

Notification of received Faxes or data files is via the desktop computer, similar to e-mail, and is also given by the system (in voice) when the user calls in to retrieve his voice mail. Outbound Faxes may be composed in the desktop machine and passed to the system for private, paperless, immediate transmission to the recipient(s).

These features improve the efficiency, confidentiality, security and punctuality of Fax and data delivery and transmission within the organization.

13.1 Receiving Fax and Data Transmissions.

During all calls, assigned DSPs continue to search for Fax or data signals which, if encountered, immediately switch control to the appropriate Fax or data mode protocols. Thus, even during a voice call, the caller may switch to Fax "Send" mode, causing a CNG or other appropriate signal to be sent. The assigned DSP 208 receives the signal, alerts the call management computer 101 and the assigned DSP is then switched to a Fax protocol mode to complete the transmission.

The call management computer 101 identifies an incoming call as a Fax or data call type because the system received a CNG or other signal for Fax or an appropriate DTMF or other signal for data or a "reverse" modem courier signal. If the fax or data call was dialed directly to a system user's extension number or the caller used the call attendant feature to identify the called party, the call management computer 101 instructs the attached DSP processor 108 to receive the fax transmissions automatically using a standard Fax protocol with adjunct file transmission capabilities. For data transmissions, the DSP is instructed to use an appropriate data protocol Fax plus data, X.34, ISDN, TCP/IP or other. Even during a voice call, the caller may switch to Fax mode, causing a CNG or other appropriate signal to be sent, identified by the assigned DSP 208 and the DSPs to be switched by the call management computer 101, 201 to Fax protocol mode to receive the transmission.

Because the Call Management System knows who is receiving each Fax, it applies a unique Fax identification for each user uniquely identifying the receiver to the Fax sender.

When a transmission is received, Fax, file, video, etc., it is stored on the call management computer's database 215 or on the organization's digital network(s) message storage facility. The destination party 111, 113 is then notified of the new Fax or data through messages sent by the call management computer 101 down the digital network(s) 109 to the user's workstation 114 and finally to the user's call management window 115 (as described in Section 8).

When notified, the destination party 111, 113 may review the list of unread Fax or data messages and then may request that the Fax or data message be transported to their workstation 114 via the digital network(s) 109, from whence it can be viewed, printed, archived and treated as any other such file (see below).

The result is that each system user is provided with private, paperless, and immediate Fax and data send and receive capabilities without having to add any hardware to their desktop computer 114.

This use of a "One Number" direct user access with automated reception of Fax and data messages is a significant improvement over other approaches, which typically require: a dedicated Fax/data line and hardware for each person's desktop computer 114 doubling the number of CO circuits in use, or having all Fax and data messages routed to a designated person for later sorting out and distribution, thereby delaying delivery and losing privacy, or assigning a separate block of telephone numbers attached to a special fax/data transmission server.

13.2 "FAX" Notification.

When the user has received a new Fax transmission, the call management window's 115 "FAX" button is highlighted and the count of new Faxes is provided. Selecting the "FAX" button launches the FAX selection subscreen Fig. 8 of the current implementation. That screen shows the user's list of Faxes and summarizes the total number of Faxes and the number which are new 800. The list includes:

1. "Check" 801 if the Fax has been previously seen;
2. The date and time received 802;
3. From whom the Fax originated 803;
4. Subject of the Fax 804 for future reference;
5. "Print" the Fax document;
6. "Send" copy or original Fax documents to someone else from the directory or typed in with attached "Annotation" voice or digital message and name of the original recipient;
7. "Broadcast" Fax document to selected people/locations form the directory or typed in with attached "Annotation" voice or digital message and name of original recipient. Broadcast of Fax documents includes selection of dates, times, retries, etc.;
8. "Routing" one or more Fax documents to the system users assigned to the routing list with attached "Annotation" voice or digital message and name of recipient.

The user may select one or more of the Faxes to be viewed 806, deleted 805 or forwarded 807. Also, the user may select a checkbox to limit the display to only new Faxes 809. If the user selects a Fax to be viewed, the computer operating system's FAX viewer is launched with the name of the Fax file, popping up the selected Fax in front of the other windows. Voice or digital "Annotation" by the system user is provided in the same way that voice messages are recorded for VIP rules.

13.3 Unique Call Routing for Faxes or Data.

Fax or data transmissions, received for specified numbers such as the organization's base number, may be accepted for a specified user, *e.g.*, the organization's operator who sorts them out and sends them to the appropriate users, prints them and deals with them conventionally, or sends them directly to a Fax machine or computer for data prefacing.

13.4 Special Data Calls.

For special kinds of data calls, *e.g.*, video conferencing, the Call Management System transfers the call automatically to an appropriate extension for handling by a specialized device.

13.5 Laptop Data Calls.

The Call Management System provides for users to call into their "One Number" with their laptop computer and, after being identified to have access to their desktop computers for file transfers and maintenance as well as to have access to their e-mail and other electronic messages. The user also has access to their Call Management System functions (VIP rules, status, etc.) in order to change and update them as appropriate.

13.6 Outgoing Fax and Data Transmissions.

Fax and data transmission of documents/files created at the desktop computer 114 is also provided through conventional "printing" to the fax/data feature of the call management computer, from whence the document can be transmitted automatically immediately, or at specified times, to one or more recipients in the Call Management System directory or to numbers typed in by the user and with specified retry attempts.

Because the Call Management System knows who is sending each Fax, it applies a unique Fax banner for each user specifically identifying the sender to the Fax receiver.

13.7 Retrieving Fax or Data files via "One-Call" Message Retrieval.

The Call Management System also addresses the special needs of the traveling or at-home user. Using any touch-tone telephone, the user retrieves his voice mail messages from the enterprise's existing voice mail system and, in the same call, is notified by system of the existence of his unread Fax, data and e-mail messages. He can then instruct the system to Fax these messages to a convenient Fax machine near him, *e.g.*, a hotel, airport or home Fax, or to transmit them to his home or laptop computer (see Section 15).

14. USER-ACCESSIBLE CALL LOGS

All calls received by or sent from the Call Management System are summarized in a user-accessible call log as part of the overall call management database. This allows users dynamic access on demand to the call logs, and enables the ability to return missed calls with only the click of a mouse. The use of a user-accessible call log improves the ability for system users to be aware of who called, to get back to callers missed and to monitor their telephone usage. Management of the organization may also use the call log database to monitor responsiveness to returning calls and misuse of business telephones.

All calls received or placed by the Call Management System are summarized in the Call Log portion of the call management database. Each user has direct access to his own call log containing all his calls, even if the caller chooses not to leave a voice mail message. Using this call log, the user may simply double-click to return missed calls, with the system automatically outdialing them.

Each user is provided a complete log of his own voice calls, available on demand, through subscreens of his call management window 115, as shown in Figure 9a and with log sorting options in 9b. As with all other aspects of the user's graphical interface, different screens, buttons, wordings, etc. can be used as the means to accomplish the same ends. The current implementation as shown is but one of many possible arrangements.

Each log entry includes the following fields:

1. CALL INDICATOR 905

- "Blank" for incoming calls
- "Red Dot" for missed calls sent to voice mail without answering, transferred without answering, etc.
- "Out" for outdialed calls
- "VM" for calls sent to voice mail

- Others.

2. DATE 906 Date the call occurred

3. TIME 907 Time the call began

4. LENGTH 908 Duration of the call HH:MM

5. NUMBER 909 Telephone number from which the call originated

6. CALLER 910 Name and/or affiliation of the caller or called party.

7. "TAG" Associated "Tag" message and initials of the "Tagger".

8. TRANSFERRED TO Number or name of transferred party for transferred calls.

9. CALLBACKS Count of times a callback was attempted and the date/time when the callback was successful.

This directory is retained as a user-accessible part of the Call Management Databases 215 and can be accessed at any time through the user's call management window 115. When opened, the user can click the "Options" button 911 to bring up the "Options" subscreen 912 shown in Fig. 9b. The user can define how his log is to be sorted by picking and choosing among the various options as shown, including only unanswered missed calls; incoming calls, outgoing calls, only new calls since the user last was available, calls only within the last number of days, weeks, months, etc., and others.

If a call is active at the time the call log is accessed, the calls are sorted and limited to those to/from the caller of the active call, allowing the user to know when calls occurred between himself and the caller.

Whenever the call log is accessed, any call shown can be "double-clicked" by the user to tell the system to outdial the caller immediately. This allows users to keep track of their missed calls and to return calls quickly, *e.g.* after returning from lunch, even if the caller failed to leave a voice mail message.

Even though each user is limited to viewing only his own log entries, the system manager may have the logs of all employees sorted, combined, processed and printed in any way management requires using the industry-standard database tools as an added means for managing the business. For example, it may desirable to know how long on the average it takes for a customer to get a call back, or whether certain employees are misusing corporate telephone resources for personal purposes.

Additionally, these call logs provide the means for corporate management to do accounting comparisons with central office bills, verifying accuracy and completeness and often saving telephone costs. The logs can also be sorted and compiled to show traffic flow for workplace maximization.

15. "ONE-CALL" MESSAGE RETRIEVAL

The Call Management System permits for traveling or out-of-the-office employees to make a single telephone call to receive all of their electronic messages, whether voice mail, Fax messages, data files or e-mail. This feature is a significant improvement over the usual requirement to make several calls, using different technologies to retrieve messages in different forms. Retrieving all of the user's electronic messages during a single telephone call is a major improvement, especially from remote locations where telephone call connections are difficult, unreliable or can take a long time to establish, and re-establish.

One-Call Message retrieval is accomplished by the traveling system user placing a call to the organization's voice mail 116, as is done conventionally by calling the voice mail retrieval number. The call management computer 101 recognizes the destination number as that of voice mail retrieval and puts the call directly through but remains online with the attached DSP 208 assigned to identify the caller's mailbox number, as entered through the telephone keypad or by voice. With the voice mailbox number, the call management computer 101 identifies the caller 118 through correlating voice mailbox numbers in the call management database 215.

The Call Management System then identifies the caller as a system user and checks to determine if any new Fax or data messages are stored in the call management database, and determines if any new e-mail messages for the caller exist, assuming the e-mail system can report such information. If any of these electronic messages exist, the Call Management System plays an unobtrusive chime for the caller to hear saying, in effect, "you have electronic messages in other forms waiting for you, don't just hang up." The user then knows to logoff from the voice mail system 116 when he is finished.

When the assigned DSP 208 detects the user's entry of a voice mail's logoff sequence, it informs the call management computer 101 which then instructs the DSP to play out a message from the Call Management Database 215, *e.g.* "You have two new Faxes and three new e-mail messages, press one for immediate delivery." If the caller responds as requested, the caller may then be asked for and respond with an additional security code or his spoken voice which is verified.

The caller is then given a menu of delivery options for the electronic messages, including:

1. Dial out and send them to my home computer
2. Dial out and send them to my home Fax machine
3. Dial out and send them to a convenient Fax machine at this number
4. I will attach my laptop computer and then send them to me electronically now
5. Read part or all of the messages to me (via text-to-speech) before I decide which to send

Depending upon the option selected by the user, the call management computer 101 will respond accordingly including providing translations as needed, *e.g.*, e-mail sent to a Fax machine is converted to Fax format, etc.

The system user may also call his own "One Number", override the greeting message and identify himself using his voice mailbox number and appropriate password. Once identified, the user is provided the same retrieval capabilities as though he had called voice mail, without having to go through the voice mail process described above.

The Call Management System thus provides the means for a system user to retrieve all his electronic messages during a single telephone call to the organization.

16. VOICE MAIL HANDLING

The Call Management System utilizes voice mail in four broadly different ways: transferring callers to voice mail, alerting system users to the presence of voice mail messages, as a part of "One-call" message retrieval (see Section 15) and as an integrated subsystem of the overall Call Management System, thereby becoming the organization's voice mail, providing expanded voice mail capabilities to system users.

16.1 Transferring Callers to Voice Mail.

The Call Management System eliminates "voice-mail-jail", to which callers are so commonly subjected, because only system users send calls to voice mail, not automated machines as in the past. Callers to system users are transferred to voice mail only because the user makes that choice directly or because of predetermined VIP rules.

When a call is transferred to voice mail (see Sections 1 and 8), the call is transferred to the organization's internal voice mail extension, kept in the call management database 215, and the extension number of the called party, also from the call management database, is entered by the attached DSP 208 to tell the voice mail system, whether integrated with the PBX or not, which voice mailbox to use. The call management computer 101 records the length of time the caller uses the voice mail as part of the voice mail log, from which the called party can obtain some additional knowledge about the caller's message, or lack thereof. Even if no message is left, the call log reflects to the user that the call was received, allowing the called party to return the call using only the click of his mouse.

The Call Management System also provides a "fake" voice mail capability to which an annoying caller can be sent, appearing like normal voice mail but without recording any message.

16.2 Alerting System Users to New Voice mail Messages.

The Call Management System alerts each system user to the presence of new voice mail messages. The alerts are subscreens of the "Voice Mail" part of the "Message Board" of the call management window 115, as described in Section 8. Included are:

1. The name of the caller
2. The date and time of the call
3. The length of voice mail message left
4. The telephone number of the caller.

The user may go to the voice mail system to hear his messages and/or he may return the call directly from the voice mail subscreen by a click of his mouse, or he may delete the notification about the voice mail message.

For Call Management System integrated voice mail subsystems (see below) and other voice mail systems which can integrate with digital network(s) and computer-based systems, the Call Management System provides additional features of:

1. Alerting system users to the presence of all new voice mail messages, not just those received through the Call Management System, including those calls for which no message was left.

2. Retrieving the voice mail messages directly following mouse-click selection from the voice mail alert screen.

3. Message selection can be made by the user in any order.

4. Having the Call Management System establish a voice pathway to the user and then instructing the voice mail system to play out the messages selected through that voice pathway.

5. Returning calls by the click of the mouse.

16.3 Integrated Voice Mail Subsystem.

When the Call Management System provides the integrated voice mail for the organization, the voice mail is tightly integrated with the rest of the system, unlike other voice mail systems. Calls transferred to the integrated voice mail are provided the usual array of voice mail caller features, controlled by entry using the telephone keypad. Voice messages are stored either as part of the call management database 215 or as part of the organization's e-mail or other message storage capabilities 110.

System users are alerted to and may review and activate their voice mail messages from their voice mail alert screen in any order, knowing who each caller is and the length of their voice mail message. When a voice mail message is selected to be heard by the double-click of a mouse or otherwise, the call management computer 101 creates a voice pathway to the user's telephone instrument 106, if one is not already present, which it then uses to play back the selected voice mail messages from the call management database or other storage. During playback, many new capabilities are provided the user including:

1. Knowing the identity of the caller and the length of each voice mail message

2. Selecting voice mail messages to be heard in any order

3. Playback controls over speedup and slowdown, backup, fast forward, fast reverse, and many others which have limited availability with conventional voice mail systems

4. Returning the calls with the click of the mouse

5. Many others.

With integrated voice mail, the user can send a call to voice mail and, once free from other tasks, retrieve that call.

Integrated voice mail removes the limitations and barriers of existing voice mail systems, affording system users much more information and entirely new capabilities for its use.

17. USER STATUS

The current dynamic status of all system users is made available to other system users through their call management window. Knowing the current status of users improves the ability of members of an organization to communicate both within the organization and externally with their customers and prospects.

The user status of all system users 111, 113 is continuously maintained by the system through the user's call management window 115 on their workstation 114 and the call management database 215, as described in Section 8. The system user may change his status at any time, selecting among a variety of status conditions:

1. "Available to receive all calls",
2. "Available only for VIP calls",
3. "Unavailable - transferred to another workstation",
4. "Unavailable for all calls",
5. and others.

When the user selects the item 4 "unavailable" option, he is given a list of potential reasons from which he may choose, *e.g.*, "Out of the Office", "Out to lunch", "On vacation", etc.. He may choose one of these, choose one and modify it or type in anything else he wishes, *e.g.*, "Away from my desk til 2:00", "Out of the office back Friday", "Giving a demo til 10:00" etc.

When the user is in one of the "available" states, the system automatically applies appropriate status, *e.g.*, "On the phone", "Not responding to calls", etc.

Whenever a system user needs to transfer a call, conference a call, contact another user or any number of other reasons, the call management window's "Directory" as accessed in a variety of ways, (see Section 8) provides the means to determine the current status of other system users. Obviously, if another user is not available, there is no use trying to transfer or conference that person with the existing call.

18. FAULT TOLERANCE AND "COPPER BYPASS"

Call Management System fault tolerance is accomplished in two ways, "copper bypass" and "dual system" configurations.

18.1 "Copper Bypass" Fault Tolerance.

For Call Management Systems having the same kind and number of trunks on both the CO and PBX sides 102, 202, 105, 205 "copper bypass" fault tolerance is provided. "Copper bypass" is implemented through an external set of physical switches 1001, Fig. 10a, which are arranged so that, when deactivated, the CO trunks "bypass" the Call Management System altogether, connecting directly to the PBX trunks, removing the call management computer 101, 201 from the configuration, connecting the CO directly to the PBX.

The switches in the copper bypass box are normally energized by a signal from the DSP motherboard 1008, connecting the CO and PBX trunks to the call management computer 101, 201. The normal energized state continues so long as:

1. The power to the call management computer 101, 201 remains

2. The call management computer 101, 201 continues to operate in an appropriate manner and to refresh the switch control circuit on a regular basis, at least every few seconds

3. The DSP processors continue to operate as programmed.

If any of these conditions fail, the switches are deactivated and the system is instantly removed from connection to the CO and PBX trunks and the trunks are bridged together, attaching the CO 103 to the organization's PBX 104 as before, allowing telephone service to be restored, but without the new Call Management System features.

18.2 "Dual-System" Fault Tolerance.

Dual-system fault tolerance is used for configurations in which CO trunk to PBX trunk conversions are implemented, or for configurations requiring a very high degree of up-time reliability where essentially no down-time is acceptable.

Dual-system fault tolerance is provided through implementing dual call management computers 101 with their trunk interface boards 203, 206, 207, telephony buses 210, circuit switches 204, DSP processors 208, digital network connections 209 and databases 215. During normal operations, one of the computers is connected to the CO and PBX trunks 202, 205 and is providing the Call Management System features. The other backup system is also alive, attached to the data pathway 109 but not to the CO and PBX trunks. Both systems remain alive during normal operations in order to maintain equality of the two copies of the databases via the digital network. An alternative to this process is to keep the Call Management Databases 215 on the LAN server 110 or elsewhere instead of on the call management computer 101.

When the primary system fails, the backup system is switched in-line in its place allowing business operations to continue while the other system is repaired and placed back in service. This process requires the Call Management Databases 215 to be "equalized" prior to the repaired system being placed on "backup" status, assuming they are kept on the call management computer 101, 201. Switching the CO and PBX trunks from the "primary" system to the "Backup" system is done automatically using variations of the copper bypass boxes, as shown in Figure 10b.

WHAT IS CLAIMED IS:

1. A call management system comprising:

a. at least one user position, comprising a computer workstation and associated telephone apparatus;

b. a call management computer;

c. a digital data network connecting the workstation of said at least one user position with said call management computer;

d. said call management computer including means for intercepting an incoming call to said at least one user position;

e. means for determining that an intercepted call is for said at least one user position;

f. means for interacting with the workstation of said at least one user position to determine how the intercepted call is to be processed; and

g. means for processing the call according to instructions received from the workstation of the called user.

2. A call management system in accordance with claim 1 wherein said call management computer includes means for identifying the calling party.

3. A call management system in accordance with claim 1 wherein said call management computer includes means for identifying the type of call.

4. A call management system in accordance with claim 3 wherein said call types include voice calls and fax calls.

5. A call management system in accordance with claim 4 wherein said call types include data calls.

6. A call management system in accordance with claim 3 wherein said call types include voice calls and data calls.

7. A call management system in accordance with claim 6 wherein said call types include fax calls.

8. A call management system in accordance with claim 1 wherein said means for identifying which user was called uses direct inward dialing signals.

9. A call management system in accordance with claim 1 wherein said means for identifying which user was called uses DNIS signals.

10. A call management system in accordance with claim 1 wherein said means for identifying which user was called uses ISDN signals.

11. A call management system in accordance with claim 1 wherein said means for identifying which user was called uses voice recognition apparatus to identify the called party's name when spoken by the caller.

12. A call management system in accordance with claim 1 wherein said means for identifying which user was called uses Internet screen forms filled in by the caller.

13. A call management system in accordance with claim 1 wherein said means for identifying which user was called includes requesting that the caller enter information and using that information to identify the called user.

14. A call management system in accordance with claim 13 wherein said information includes the called party's extension number.

15. A call management system in accordance with claim 13 wherein said information includes spelling or partial spelling of the called party's name on the telephone keypad by the caller.

16. A call management system in accordance with claim 13 wherein said information includes the called party's name or a portion thereof as spoken by the caller.

17. A call management system in accordance with claim 4 wherein said call management computer includes means for receiving fax documents.

18. A call management system in accordance with claim 17 wherein said call management computer includes storage for received fax documents.

19. A call management system in accordance with claim 18 wherein said computer workstation selectively retrieves stored fax documents determined to be for its user position.

20. A call management system in accordance with claim 4 wherein said call management computer includes means for receiving data files.

21. A call management system in accordance with claim 20 wherein said call management computer includes storage for received data files.

22. A call management system in accordance with claim 21 wherein said computer workstation selectively retrieves stored data files determined to be for its user position.

23. A call management system in accordance with claim 4 further including means for identifying said fax calls by detecting CNG signals.

24. A call management system in accordance with claim 4 further including means for identifying said fax calls by detecting ISDN messages.

25. A call management system in accordance with claim 5 further including means for identifying said data calls by detecting DTMF signals.

26. A call management system in accordance with claim 5 further including means for identifying said data calls by detecting data carrier signals.

27. A call management system in accordance with claim 6 further including means for identifying said data calls by detecting DTMF signals.

28. A call management system in accordance with claim 6 further including means for identifying said data calls by detecting data carrier signals.

29. A call management system in accordance with claim 7 further including means for identifying said fax calls by detecting CNG signals.

30. A call management system in accordance with claim 7 further including means for identifying said fax calls by detecting ISDN messages.

31. A call management system in accordance with claim 1 wherein said digital network includes at least one local area network.

32. A call management system in accordance with claim 1 wherein said digital network includes at least one wide area network.

33. A call management system in accordance with claim 1 wherein said digital network includes the Internet.

34. A call management system in accordance with claim 1 wherein said digital network includes at least one ISDN network.

35. A call management system in accordance with claim 4 wherein an identifying message is returned to the calling fax machine which confirms the identify of the called party.

36. A call management system in accordance with claim 7 wherein an identifying message is returned to the calling fax machine which confirms the identify of the called party.

37. A call management system in accordance with claim 2 wherein said means for identifying the calling party uses ANI signals.

38. A call management system in accordance with claim 2 wherein said means for identifying the calling party uses BCLID signals.

39. A call management system in accordance with claim 2 wherein said means for identifying the calling party uses ISDN signals.

40. A call management system in accordance with claim 2 wherein said means for identifying the calling party uses Caller ID signals.

41. A call management system in accordance with claim 2 wherein said means for identifying the calling party uses DTMF signals.

42. A call management system in accordance with claim 2 wherein said means for identifying the calling party uses FSK signals.

43. A call management system in accordance with claim 2 wherein said means for identifying the calling party includes requesting that the caller enter information and using that information to identify the caller.

44. A call management system in accordance with claim 43 wherein said information includes entry of a telephone number.

45. A call management system in accordance with claim 44 wherein said telephone number is the caller's home telephone number.

46. A call management system in accordance with claim 44 wherein said telephone number is the caller's business telephone number.

47. A call management system in accordance with claim 44 wherein said telephone number is the caller's business telephone number plus the extension number.

48. A call management system in accordance with claim 44 wherein said information includes entry of a personal identification number.

49. A call management system in accordance with claim 43 wherein said information includes speaking an identifying word or phrase.

50. A call management system in accordance with claim 49 wherein said identifying word or phrase is the caller's name or a portion thereof.

51. A call management system in accordance with claim 2 wherein said means for identifying the calling party includes storage containing at least one caller ID database.

52. A call management system in accordance with claim 51 wherein said at least one caller ID database is updated with information whenever a call is received from a caller not in said caller ID database.

53. A call management system in accordance with claim 52 wherein said at least one caller ID database contains the telephone numbers and listed names of potential callers.

54. A call management system in accordance with claim 52 wherein said at least one caller ID database contains representations of the voices of callers.

55. A call management system in accordance with claim 52 wherein said at least one caller ID database contains Internet addresses of callers.

56. A call management system in accordance with claim 52 wherein said at least one caller ID database contains e-mail addresses of callers.

57. A call management system in accordance with claim 51 further including means for updating said at least one caller ID database with information entered from the computer workstation at said at least one user position.

58. A call management system in accordance with claim 1 wherein said means for processing calls includes apparatus for playing a pre-recorded message to the caller.

59. A call management system in accordance with claim 2 wherein said means for processing calls includes apparatus for playing a pre-recorded message to the caller.

60. A call management system in accordance with claim 59 wherein the identity of the caller determines at least in part which pre-recorded message is played.

61. A call management system in accordance with claim 59 wherein the identity of the called user determines at least in part which pre-recorded message is played.

62. A call management system in accordance with claim 59 wherein the identity of the caller and the identity of the called user determines at least in part which pre-recorded message is played.

63. A call management system in accordance with claim 1 wherein said means for processing the call includes apparatus for switching the call to a destination selected by the called user.

64. A call management system in accordance with claim 63 wherein said apparatus for switching the call includes at least one external telephone switch.

65. A call management system in accordance with claim 64 wherein said at least one external telephone switch includes a private branch exchange or PBX switch.

66. A call management system in accordance with claim 64 wherein said at least one external telephone switch includes a telephone key system.

67. A call management system in accordance with claim 64 wherein said at least one external telephone switch includes an automatic call distributor or ACD switch.

68. A call management system in accordance with claim 64 wherein said at least one external telephone switch includes a telephone central office.

69. A call management system in accordance with claim 63 wherein said apparatus for switching the call includes switching apparatus contained within said call management computer.

70. A call management system in accordance with claim 1 wherein said system includes one or more processing rules applicable to at least one user which determine at least in part how calls to that at least one user are processed.

71. A call management system in accordance with claim 70 wherein said call management system includes storage for said processing rules.

72. A call management system in accordance with claim 70 wherein which of said processing rules is applicable is determined at least in part by the identity of the called user.

73. A call management system in accordance with claim 70 wherein which of said processing rules is applicable is determined at least in part by the current status of the called user.

74. A call management system in accordance with claim 73 wherein the current status of the called user includes whether or not he or she is on the phone.

75. A call management system in accordance with claim 73 wherein the current status of the called user includes whether or not he or she is available to receive calls.

76. A call management system in accordance with claim 73 wherein the current status of the called user includes whether or not he or she is accepting only priority calls.

77. A call management system in accordance with claim 73 wherein the current status of the called user includes his or her current location.

78. A call management system in accordance with claim 70 wherein which of said processing rules is applicable is determined at least in part by the current date, day of the week and/or time of day.

79. A call management system in accordance with claim 70 wherein said processing rules include instructions for routing calls from at least one caller to a destination other than the user position.

80. A call management system in accordance with claim 79 wherein said other destination is a destination on the public switched telephone network.

81. A call management system in accordance with claim 79 wherein said other destination is another user position.

82. A call management system in accordance with claim 79 wherein said other destination is a destination on the Internet.

83. A call management system in accordance with claim 79 wherein said other destination is a voice mailbox.

84. A call management system in accordance with claim 70 wherein said processing rules include playing a pre-recorded message to the caller.

85. A call management system in accordance with claim 84 wherein the identity of the called user determines at least in part which pre-recorded message is played.

86. A call management system in accordance with claim 84 wherein said pre-recorded message requests the caller to enter information.

87. A call management system in accordance with claim 86 wherein said entered information is in the form of DTMF signals.

88. A call management system in accordance with claim 86 wherein said entered information is in the form of spoken words.

89. A call management system in accordance with claim 86 wherein said entered information determines at least in part the subsequent processing of the call.

90. A call management system in accordance with claim 70 wherein said call processing rule specifies at least in part that the call be transferred to the called user at a location other than the normal user position.

91. A call management system in accordance with claim 90 further including means for the user to change the location

to which the call is to be transferred by calling the call management system and entering appropriate instructions.

92. A call management system in accordance with claim 90 wherein said call processing rule specifies a series of alternate destinations which are to be called.

93. A call management system in accordance with claim 70 wherein said call processing rule specifies at least in part that the call be transferred to a paging service.

94. A call management system in accordance with claim 70 wherein said call processing rule specifies at least in part that the call be transferred to a voice mailbox.

95. A call management system in accordance with claim 70 wherein said call processing rule specifies at least in part that the call be transferred to an alternate location and subsequently transferred back to the called user.

96. A call management system in accordance with claim 70 wherein said call processing rule specifies at least in part that another call processing rule should be applied to the call.

97. A call management system in accordance with claim 70 wherein said call processing rule specifies at least in part that a special ringing sound should be used for the call.

98. A call management system in accordance with claim 70 wherein said call processing rule specifies at least in part that the call should be disconnected.

99. A call management system in accordance with claim 70 wherein said call processing rule specifies at least in part that the call should be placed on hold.

100. A call management system in accordance with claim 84 wherein the user may call the system and re-record the pre-recorded message(s), changing content as desired.

101. A call management system in accordance with claim 2 wherein said system includes one or more processing rules applicable to at least one caller which determine at least in part how calls from that at least one caller are processed.

102. A call management system in accordance with claim 101 wherein said call management system includes storage for said processing rules.

103. A call management system in accordance with claim 101 wherein which of said processing rules is applicable is determined at least in part by the identity of the called user.

104. A call management system in accordance with claim 101 wherein which of said processing rules is applicable is determined at least in part by the current status of the called user.

105. A call management system in accordance with claim 104 wherein the current status of the called user includes whether or not he or she is on the phone.

106. A call management system in accordance with claim 104 wherein the current status of the called user includes whether or not he or she is available to receive calls.

107. A call management system in accordance with claim 104 wherein the current status of the called user includes whether or not he or she is accepting only priority calls.

108. A call management system in accordance with claim 104 wherein the current status of the called user includes his or her current location.

109. A call management system in accordance with claim 104 wherein which of said processing rules is applicable is determined at least in part by the current date, day of the week and/or time of day.

110. A call management system in accordance with claim 101 wherein said processing rules include instructions for routing calls from at least one caller to a destination other than the user position.

111. A call management system in accordance with claim 110 wherein said other destination is a destination on the public switched telephone network.

112. A call management system in accordance with claim 110 wherein said other destination is another user position.

113. A call management system in accordance with claim 110 wherein said other destination is a destination on the Internet.

114. A call management system in accordance with claim 110 wherein said other destination is a voice mailbox.

115. A call management system in accordance with claim 101 wherein said processing rules include playing a pre-recorded message to the caller.

116. A call management system in accordance with claim 115 wherein the identity of the called user determines at least in part which pre-recorded message is played.

117. A call management system in accordance with claim 115 wherein said pre-recorded message requests the caller to enter information.

118. A call management system in accordance with claim 117 wherein said entered information is in the form of DTMF signals.

119. A call management system in accordance with claim 117 wherein said entered information is in the form of spoken words.

120. A call management system in accordance with claim 117 wherein said entered information determines at least in part the subsequent processing of the call.

121. A call management system in accordance with claim 101 wherein said call processing rule specifies at least in part that the call be transferred to the called user at a location other than the normal user position.

122. A call management system in accordance with claim 121 further including means for the user to change the location to which the call is to be transferred by calling the call management system and entering appropriate instructions.

123. A call management system in accordance with claim 121 wherein said call processing rule specifies a series of alternate destinations which are to be called.

124. A call management system in accordance with claim 101 wherein said call processing rule specifies at least in part that the call be transferred to a paging service.

125. A call management system in accordance with claim 101 wherein said call processing rule specifies at least in part that the call be transferred to a voice mailbox.

126. A call management system in accordance with claim 101 wherein said call processing rule specifies at least in part that the call be transferred to an alternate location and subsequently transferred back to the called user.

127. A call management system in accordance with claim 101 wherein said call processing rule specifies at least in part that another call processing rule should be applied to the call.

128. A call management system in accordance with claim 101 wherein said call processing rule specifies at least in part that a special ringing sound should be used for the call.

129. A call management system in accordance with claim 101 wherein said call processing rule specifies at least in part that the call should be disconnected.

130. A call management system in accordance with claim 101 wherein said call processing rule specifies at least in part that the call should be placed on hold.

131. A call management system in accordance with claim 115 wherein the user may call the system and re-record the pre-recorded message(s), changing content as desired.

132. A call management system in accordance with claim 101 wherein which of said processing rules is applicable is

determined at least in part by the identity of the calling party.

133. A call management system in accordance with claim 115 wherein the identity of the calling party determines at least in part which pre-recorded message is played.

134. A call management system in accordance with claim 115 wherein the telephone number of the calling party or a portion thereof determines at least in part which pre-recorded message is played.

135. A call management system in accordance with claim 79 wherein said other destination is a plurality of user positions.

136. A call management system in accordance with claim 79 wherein said other destination is an extension number.

137. A call management system in accordance with claim 110 wherein said other destination is a plurality of user positions.

138. A call management system in accordance with claim 110 wherein said other destination is an extension number.

139. A call management system in accordance with claim 3 wherein said system includes one or more processing rules wherein the type of call determines at least in part how calls of each type are processed.

140. A call management system in accordance with claim 139 wherein said call management system includes storage for said processing rules.

141. A call management system in accordance with claim 139 wherein which of said processing rules is applicable is determined at least in part by the identity of the called user.

142. A call management system in accordance with claim 139 wherein which of said processing rules is applicable is determined at least in part by the current status of the called user.

143. A call management system in accordance with claim 142 wherein the current status of the called user includes whether or not he or she is on the phone.

144. A call management system in accordance with claim 142 wherein the current status of the called user includes whether or not he or she is available to receive calls.

145. A call management system in accordance with claim 142 wherein the current status of the called user includes whether or not he or she is accepting only priority calls.

146. A call management system in accordance with claim 142 wherein the current status of the called user includes his or her current location.

147. A call management system in accordance with claim 142 wherein which of said processing rules is applicable is determined at least in part by the current date, day of the week and/or time of day.

148. A call management system in accordance with claim 139 wherein said processing rules include instructions for routing calls from at least one caller to a destination other than the user position.

149. A call management system in accordance with claim 148 wherein said other destination is a destination on the public switched telephone network.

150. A call management system in accordance with claim 148 wherein said other destination is another user position.

151. A call management system in accordance with claim 148 wherein said other destination is a destination on the Internet.

152. A call management system in accordance with claim 139 wherein said call processing rule specifies at least in part that the call be transferred to the called user at a location other than the normal user position.

153. A call management system in accordance with claim 152 further including means for the user to change the location to which the call is to be transferred by calling the call management system and entering appropriate instructions.

154. A call management system in accordance with claim 152 wherein said call processing rule specifies a series of alternate destinations which are to be called.

155. A call management system in accordance with claim 139 wherein said call processing rule specifies at least in part that the user be paged upon receipt of certain calls.

156. A call management system in accordance with claim 139 wherein said call processing rule specifies at least in part that another call processing rule should be applied to the call.

157. A call management system in accordance with claim 139 wherein said call processing rule specifies at least in part that a special ringing sound should be used for the call.

158. A call management system in accordance with claim 139 wherein which of said processing rules is applicable is determined at least in part by the identity of the calling party.

159. A call management system in accordance with claim 1 wherein said interaction with said workstation of said at least one user position includes a command to put the call through to said at least one user position.

160. A call management system in accordance with claim 1 wherein said interaction with said workstation of said at least one user position includes a command to disconnect the call.

161. A call management system in accordance with claim 1 wherein said interaction with said workstation of said at least one user position includes a command to mute the call.

162. A call management system in accordance with claim 1 wherein said interaction with said workstation of said at least one user position includes a command to transfer the call to another destination.

163. A call management system in accordance with claim 162 wherein said another destination is a voice mailbox.

164. A call management system in accordance with claim 1 wherein said interaction with said workstation of said at least one user position includes a command to route the call

to a plurality of user positions for answer by the first available user from that plurality.

165. A call management system in accordance with claim 1 wherein said interaction with said workstation of said at least one user position includes a command to put the call on hold.

166. A call management system in accordance with claim 101 wherein said interaction with said workstation of said at least one user position includes a command to add the caller to a specific rule.

167. A call management system in accordance with claim 70 wherein said interaction with said workstation of said at least one user position includes a command to apply a selected rule to the call.

168. A call management system in accordance with claim 101 wherein said interaction with said workstation of said at least one user position includes a command to apply a selected rule to the call.

169. A call management system in accordance with claim 1 wherein said interaction with said workstation of said at least one user position includes a command to record a call.

170. A call management system in accordance with claim 1 wherein said interaction with said workstation of said at least one user position includes a command to playback a selected recorded call or a portion thereof.

171. A call management system in accordance with claim 1 wherein said interaction with said workstation of said at

least one user position includes a command to conference at least one other party in on the call.

172. A call management system in accordance with claim 1 wherein said interaction with said workstation of said at least one user position includes a command to associate an electronic message with the call.

173. A call management system in accordance with claim 172 wherein said associated electronic message is displayed on the workstation along with the call.

174. A call management system in accordance with claim 173 wherein said displayed associated electronic message is displayed on the associated workstation of another user whenever the call is transferred to or conferenced with that user.

175. A call management system in accordance with claim 1 wherein said interaction with said workstation of said at least one user position includes a command to ask the caller for additional information.

176. A call management system in accordance with claim 171 wherein said additional information is requested to be entered in voice.

177. A call management system in accordance with claim 171 wherein said additional information is requested to be entered using DTMF signals.

178. A call management system in accordance with claim 171 wherein said additional entered information is transmitted to the user position.

179. A call management system in accordance with claim 58 wherein said interaction with said workstation of said at least one user position includes a command to play a pre-recorded message to the caller.

180. A call management system in accordance with claim 179 wherein the pre-recorded message is selected from a list by the user.

181. A call management system in accordance with claim 179 wherein the user may specify actions to be taken following the playing of the pre-recorded message.

182. A call management system in accordance with claim 2 wherein said interaction with said workstation of said at least one user position includes a command to play a pre-recorded message to the caller which is personalized for the specific caller.

183. A call management system in accordance with claim 1 wherein a log of call information is maintained for calls placed to and/or from system users.

184. A call management system in accordance with claim 183 wherein said interaction with said workstation of said at least one user position includes a command to display some or all of that user's calls as recorded in the call log on said workstation.

185. A call management system in accordance with claim 2 wherein a log of call information is maintained for calls placed to and/or from system users.

186. A call management system in accordance with claim 185 wherein said interaction with said workstation of said at least one user position

least one user position includes a command to display some or all of that user's calls as recorded in the call log on said workstation.

187. A call management system in accordance with claim 186 wherein said interaction with said workstation of said at least one user position includes a command to display call information from the call log relating to a particular caller.

188. A call management system in accordance with claim 186 wherein said displayed information from said call log may be restricted to calls which did not reach the user.

189. A call management system in accordance with claim 186 wherein said displayed information from said call log may be restricted to incoming calls to the user.

190. A call management system in accordance with claim 186 wherein said displayed information from said call log may be restricted to outbound calls from the user.

191. A call management system in accordance with claim 186 wherein said displayed information from said call log may be restricted to calls received and/or placed after a certain time or event.

192. A call management system in accordance with claim 186 wherein the user may select a particular call from the call log and command that a call be placed to that caller.

193. A call management system in accordance with claim 1 wherein said interaction with said workstation of said at least one user position includes a facility for composing an

electronic message for at least one other user and a command for delivering said message to said at least one other user.

194. A call management system in accordance with claim 193 wherein each of said workstations includes means to receive electronic messages from other users, means to alert the user of receipt of such electronic messages, and means to display such electronic messages when desired.

195. A call management system in accordance with claim 194 wherein each of said user workstations includes means to reply to received electronic messages.

196. A call management system in accordance with claim 194 wherein each of said user workstations includes a command to initiate a call to the sender of received electronic messages.

197. A call management system in accordance with claim 1 wherein the present status of users of the system is maintained.

198. A call management system in accordance with claim 197 wherein said present status of users includes whether or not such users are currently available to receive calls.

199. A call management system in accordance with claim 197 wherein said present status of users includes whether or not such users are currently handling a telephone call.

200. A call management system in accordance with claim 197 wherein said present status of users includes whether or not such users are currently available only to receive calls from selected callers.

201. A call management system in accordance with claim 197 wherein said present status of users includes when said users are expected to be available to receive calls.

202. A call management system in accordance with claim 197 wherein said present status of users includes at least one optional electronic message entered by each user.

203. A call management system in accordance with claim 197 wherein the present status of other users may be viewed at any time on a user's workstation.

204. A call management system in accordance with claim 1 wherein said means for interacting with the workstation of said at least one user position includes popping up a window on the computer workstation associated with said at least one user position.

205. A call management system in accordance with claim 1 wherein said means for interacting with the workstation of said at least one user position includes flashing an icon on the computer workstation associated with said at least one user position.

206. A call management system in accordance with claim 1 wherein said means for interacting with the workstation of said at least one user position includes playing an audible signal on the computer workstation associated with said at least one user position.

207. A call management system in accordance with claim 1 wherein said means for interacting with the workstation of said at least one user position includes flashing a window

on the computer workstation associated with said at least one user position.

208. A call management system in accordance with claim 1 wherein said means for interacting with the workstation of said at least one user position includes notifying the user when a call is received for that user.

209. A call management system in accordance with claim 1 wherein said means for interacting with the workstation of said at least one user position includes notifying the user when a fax is received for that user.

210. A call management system in accordance with claim 1 wherein said means for interacting with the workstation of said at least one user position includes notifying the user when an e-mail message is received for that user.

211. A call management system in accordance with claim 1 wherein said means for interacting with the workstation of said at least one user position includes notifying the user when a call is transferred to the voice mailbox associated with that user.

212. A call management system in accordance with claim 2 wherein said means for interacting with the workstation of said at least one user position includes notifying the user when a call is transferred to the voice mailbox associated with that user.

213. A call management system in accordance with claim 212 wherein said notification includes the identity of the caller.

214. A call management system in accordance with claim 1 wherein said means for interacting with the workstation of said at least one user position includes a command to activate the e-mail software system for that user.

215. A call management system in accordance with claim 19 wherein said means for interacting with the workstation of said at least one user position includes a command to activate the fax viewing software system for that user.

216. A call management system in accordance with claim 1 further including a directory database with entries of at least names and associated telephone addresses.

217. A call management system in accordance with claim 216 wherein said directory database further includes entries which are private to each user of the system.

218. A call management system in accordance with claim 216 wherein said directory database further includes entries which are associated with groups of users.

219. A call management system in accordance with claim 216 wherein said means for interacting with the workstation of said at least one user position includes a command to view the directory database at least in part on the workstation.

220. A call management system in accordance with claim 219 wherein said command further includes an instruction to sort the directory database or a portion thereof according to selected criteria.

221. A call management system in accordance with claim 219 wherein said command further includes instruction to limit the displayed entries according to selected criteria.

222. A call management system in accordance with claim 216 wherein an entry may be selected by the user.

223. A call management system in accordance with claim 222 wherein the selection is made by finding a match in the database with data entered on the workstation keyboard.

224. A call management system in accordance with claim 222 wherein the selection is made by using a pointing device associated with the workstation.

225. A call management system in accordance with claim 222 wherein the selected entry is used to transfer calls to a corresponding destination.

226. A call management system in accordance with claim 222 wherein the selected entry is used to originate calls to a corresponding destination.

227. A call management system in accordance with claim 222 wherein the selected entry is used to conference calls with a corresponding destination.

228. A call management system in accordance with claim 222 wherein the selected entry is used to originate fax calls to a corresponding destination.

229. A call management system in accordance with claim 222 wherein the selected entry is used to originate data calls to a corresponding destination.

230. A call management system in accordance with claim 222 wherein the selected entry is used to review the current status of the selected party.

231. A call management system in accordance with claim 222 wherein the selected entry is used to route electronic messages to the selected party.

232. A call management system in accordance with claim 222 wherein the selected entry is used to access the voice mailbox of the selected party.

233. A call management system in accordance with claim 216 further including means for updating the directory database from user workstations.

234. A call management system in accordance with claim 1 further including means for originating calls in response to commands from user workstations.

235. A call management system in accordance with claim 234 wherein calls are originated in response to user selection from speed dial buttons displayed on the user workstation screen.

236. A call management system in accordance with claim 234 wherein calls are originated in response to user commands using a dial pad displayed on the user workstation screen.

237. A call management system in accordance with claim 234 wherein calls are originated in response to user typed commands from the workstation keyboard.

238. A call management system in accordance with claim 237 wherein said typed commands include the name of the party to be called.

239. A call management system in accordance with claim 237 wherein said typed commands include the telephone number of the party to be called.

240. A call management system in accordance with claim 234 further including a log of recent calls from each user position.

241. A call management system in accordance with claim 240 wherein calls may be re-dialed in response to user commands from the log of recent calls from that user.

242. A call management system in accordance with claim 2 further including means to activate a personal information manager or other software program in the user workstation in response to the receipt of a call from one of a group of specified callers.

243. A call management system in accordance with claim 1 wherein callers who are on hold hear music while on hold.

244. A call management system in accordance with claim 1 further including one or more speed transfer buttons displayed on user workstations to command that a selected call be transferred to a destination associated with each button.

245. A call management system in accordance with claim 244 wherein speed dial buttons which originate calls replace the speed transfer buttons when no calls are active at the user position.

246. A call management system in accordance with claim 1 further including means for receiving multiple calls for a single user at the same time.

247. A call management system in accordance with claim 246 further including means for displaying the multiple calls on the called party's computer workstation.

248. A call management system in accordance with claim 246 further including:

means for establishing a bi-directional voice path from each caller to the call management computer;

means for establishing a bi-directional voice path to the associated telephone apparatus of the called party; and

means for connecting the bi-directional voice path from a selected caller to the bi-directional voice path of the called party.

249. A call management system in accordance with claim 248 further including means responsive to user commands to switch the bi-directional voice path to the called party from one caller to another.

250. A call management system in accordance with claim 249 further including means to put the first caller on hold when the user commands that the bi-directional voice path be switched to a second caller.

251. A call management system in accordance with claim 249 further including means to hang up the call to the first caller when the user commands that the voice path be switched to a second caller.

252. A call management system in accordance with claim 248 further including means for preserving the bi-directional call path to the called party after all calls have been completed for use in handling subsequent calls.

253. A call management system in accordance with claim 1 wherein said associated telephone apparatus is integrated with said computer workstation.

254. A call management system in accordance with claim 1 wherein said means for intercepting a call for at least one user position further includes telephone trunk lines to a telephone service provider's central office.

255. A call management system in accordance with claim 254 wherein said means for intercepting a call for at least one user position further includes telephone trunk lines to on-premises telephone equipment.

256. A call management system in accordance with claim 255 wherein said on-premises telephone equipment includes a private branch exchange or PBX telephone system.

257. A call management system in accordance with claim 255 wherein said on-premises telephone equipment includes a telephone key system.

258. A call management system in accordance with claim 255 wherein said on-premises telephone equipment includes user telephone apparatus directly controlled by said call management computer.

259. A call management system in accordance with claim 255 wherein said on-premises telephone equipment includes telephone apparatus integrated with user workstations.

260. A call management system in accordance with claim 1 wherein said means for intercepting a call for at least one user position further includes:

telephone trunk lines to a telephone service provider and;

telephone trunk lines to on-premises telephone equipment.

261. A call management system in accordance with claim 260 wherein said trunk lines to a telephone service provider and said telephone trunk lines to on-premises telephone equipment are of the same type.

262. A call management system in accordance with claim 261 further including switching means to bridge the telephone trunk lines to the telephone service provider to the telephone trunk lines to the on-premises telephone equipment, removing the call management computer.

263. A call management system in accordance with claim 262 wherein said switching means bridges automatically whenever the call management computer experiences a failure.

264. A call management system in accordance with claim 260 wherein said trunk lines to a telephone service provider and said telephone trunk lines to on-premises telephone equipment are of different types and/or numbers.

265. A call management system in accordance with claim 260 further including switching means to switch all telephone trunks to a second call management computer.

266. A call management system in accordance with claim 265 wherein said switching means switches automatically whenever the first call management computer experiences a failure.

267. A call management system in accordance with claim 1 further including at least another call management computer,

wherein each call management computer intercepts and processes a portion of the calls.

268. A call management system in accordance with claim 267 further including means to remove one call management computer from active call handling, with the remaining call management computer(s) assuming all call interception and processing.

269. A call management system in accordance with claim 268 wherein said means to remove one call management computer from active call handling operates automatically whenever a call management computer fails.

270. A call management system in accordance with claim 4 wherein each user has a single telephone number for both voice and fax calls.

271. A call management system in accordance with claim 5 wherein each user has a single telephone number for voice, fax and data calls.

272. A call management system in accordance with claim 6 wherein each user has a single telephone number for both voice and data calls.

273. A call management system in accordance with claim 7 wherein each user has a single telephone number for voice, fax and data calls.

274. A call management system in accordance with claim 1 further including:

means for each computer workstation to transmit a document to said call management computer and;

means for said call management computer to transmit said document to at least one destination via facsimile.

275. A call management system in accordance with claim 274 further including means for logging successful and unsuccessful fax transmissions.

276. A call management system in accordance with claim 275 further including means for viewing said log from said computer workstation.

277. A call management system in accordance with claim 274 further including a user-specified facsimile banner for each user.

278. A call management system in accordance with claim 1 further including:

means for each computer workstation to transmit a data file to said call management computer and;

means for said call management computer to transmit said data file to at least one destination.

279. A call management system in accordance with claim 278 further including means for logging successful and unsuccessful data file transmissions.

280. A call management system in accordance with claim 279 further including means for viewing said log from said computer workstation.

281. A call management system in accordance with claim 1 further including a voice mailbox for each user and means for storing voice messages in each voice mailbox and means for retrieving said voice messages.

282. A call management system in accordance with claim 1 wherein calls received for non-users are routed to appropriate destinations without further processing.

283. A call management system in accordance with claim 4 wherein fax calls received for specified numbers are accepted as though directed to a specified user.

284. A call management system in accordance with claim 7 wherein fax calls received for specified numbers are accepted as though directed to a specified user.

285. A call management system in accordance with claim 5 wherein data calls received for specified numbers are accepted as though directed to a specified user.

286. A call management system in accordance with claim 6 wherein data calls received for specified numbers are accepted as though directed to a specified user.

287. A call management system in accordance with claim 1 further including means to receive a call from a remote destination, determine the telephone number of that remote destination, and originate a call to the remote destination, whether or not the original call was answered.

288. A call management system in accordance with claim 1 further including:

means for intercepting a call placed by a system user to his or her voice mailbox;

means for identifying said system user;

means for determining if said system user has received new electronic messages;

means for notifying said system user of these new electronic messages;

means for accepting delivery instructions for some or all of said new electronic messages from said system user and;

means for delivering the specified new electronic messages according to said instructions.

289. A call management system in accordance with claim 288 wherein said new electronic messages include fax documents.

290. A call management system in accordance with claim 288 wherein said new electronic messages include e-mail messages.

291. A call management system in accordance with claim 288 wherein said new electronic messages include data files.

292. A call management system in accordance with claim 1 further including:

means for intercepting a call placed by a system user to a specified telephone number or group of numbers;

means for identifying said system user;

means for determining if said system user has received new electronic messages;

means for notifying said system user of these new electronic messages;

means for accepting delivery instructions for some or all of said new electronic messages from said system user and;

means for delivering the specified new electronic messages according to said instructions.

293. A call management system in accordance with claim 288 wherein said new electronic messages include fax documents.

294. A call management system in accordance with claim 288 wherein said new electronic messages include e-mail messages.

295. A call management system in accordance with claim 288 wherein said new electronic messages include data files.

296. A call management system in accordance with claim 288 wherein said new electronic messages include voice messages.

297. A call management system comprising:

a plurality of user positions, comprising a computer workstation and associated telephone apparatus;

a call management computer;

a digital data network connecting each workstation of said plurality of user positions with said call management computer;

said call management computer including means for intercepting an incoming call and routing it to a first user position;

means for accepting instructions from said user position to direct the call to a sub-set of said plurality of user positions;

means for presenting the call to said sub-set of user positions and;

means responsive to actions at one of said sub-set of user positions for removing the notification at the remaining positions within the sub-set.

298. A call management system in accordance with claim 297 further including means for associating an electronic message with said call at said first user position and presenting said associated electronic message along with the call notification on the workstations of said sub-set of said plurality of user positions.

299. A call management system in accordance with claim 298 wherein the content of the associated electronic message is determined at least in part by which number was dialed by the caller.

300. A call management system in accordance with claim 298 wherein the content of the electronic message is determined at least in part by the telephone number or a portion thereof of the caller.

301. A call management system comprising:

a plurality of user positions, comprising a computer workstation and associated telephone apparatus;

a call management computer;

a digital data network connecting each workstation of said plurality of user positions with said call management computer;

said call management computer including means responsive to receipt of a call dialed to a specified number or group of numbers for presenting the call to a specified sub-set of said user positions and;

means responsive to actions at one of said sub-set of user positions for removing the notification at the remaining positions within the sub-set.

302. A call management system in accordance with claim 301 further including means for associating an electronic message with said call and presenting said associated electronic message along with said call presentation on the workstations of said sub-set of said plurality of user positions.

303. A call management system in accordance with claim 302 wherein the content of the associated electronic message is determined at least in part by which number was dialed by the caller.

304. A call management system in accordance with claim 302 wherein the content of the electronic message is determined at least in part by the telephone number or a portion thereof of the caller.

305. A call management system in accordance with claim 292 wherein specified telephone number is the telephone number of the system user's user position.

306. A call management system in accordance with claim 301 wherein the sub-set of said user positions is determined at least in part by the telephone number which the caller dialed or a portion thereof.

307. A call management system in accordance with claim 301 wherein the sub-set of said user positions is determined at least in part by the telephone number from which the call originated or a portion thereof.

308. A call management system in accordance with claim 297 wherein the sub-set of said user positions is determined at

least in part by the telephone number which the caller dialed or a portion thereof.

309. A call management system in accordance with claim 297 wherein the sub-set of said user positions is determined at least in part by the telephone number from which the call originated or a portion thereof.

310. A call management system comprising:

a plurality of user positions, each with an associated telephone apparatus;

a call management computer;

said call management computer including means for receiving incoming calls;

means for identifying a called user position for one incoming call;

means for establishing a call path to said called user position associated telephone apparatus;

means for completing said one incoming call via said call path; and

means for preserving said call path for use with a second received call.

311. A call management system in accordance with claim 310, comprising:

means for notifying said called user position of calls available to be switched to said called user position telephone apparatus.

312. A call management system in accordance with claim 311, further comprising:

means for said called user position to instruct said call management computer to switch said call path to one of said calls available to be switched.

313. A call management system in accordance with claim 310; comprising:

means for releasing said preserved call path.

314. A call management system in accordance with claim 310, wherein:

said identifying means is responsive to DNIS signals.

315. A call management system in accordance with claim 310, wherein:

said identifying means is responsive to ISDN signals.

316. A call management system in accordance with claim 310, wherein:

said identifying means comprising voice recognition apparatus to identify said called user from a received spoken name.

317. A call management system in accordance with claim 310, wherein:

said identifying means uses Internet screen forms filled in by a calling party.

318. A call management system in accordance with claim 310, wherein:

said identifying means comprises means for requesting that a calling party enter information, and means for receiving and using said information to identify the calling party.

319. A call management system in accordance with claim 318, wherein:

said information includes said called party's extension number.

320. A call management system in accordance with claim 318, wherein:

said information comprises information generated by said calling party spelling or partial spelling of said called party's name on a telephone keypad.

321. A call management system in accordance with claim 318, wherein:

said information includes said called party's name or a portion thereof as spoken by said calling party.

322. A call management system in accordance with claim 310, wherein:

said call management computer includes means for identifying a calling party.

323. A call management system in accordance with claim 322, further comprising:

means for notifying said called party of the identity of said calling party.

324. A call management system in accordance with claim 322, wherein:

said identifying means is responsive to ANI signals.

325. A call management system in accordance with claim 322, wherein:

said identifying means is responsive to BCLID signals.

326. A call management system in accordance with claim 322, wherein:

said identifying means is responsive to ISDN signals.

327. A call management system in accordance with claim 322, wherein:

said identifying means is responsive to Caller ID signals.

328. A call management system in accordance with claim 322, wherein:

said identifying means is responsive to DTMF signals.

329. A call management system in accordance with claim 322, wherein:

said identifying means is responsive to FSK signals.

330. A call management system in accordance with claim 322, wherein:

said identifying means comprises means for requesting that said calling party enter information and means for using said information to identify said calling party.

331. A call management system in accordance with claim 330, wherein:

said information comprises a telephone number.

332. A call management system in accordance with claim 331, wherein:

said telephone number is a home telephone number of said calling party.

333. A call management system in accordance with claim 331, wherein:

said telephone number is a business telephone number of said calling party.

334. A call management system in accordance with claim 331, wherein:

said telephone number is a business telephone number and an extension number of said calling party.

335. A call management system in accordance with claim 331, wherein:

said information comprises a personal identification number.

336. A call management system in accordance with claim 330, wherein:

said information comprises one or more spoken words.

337. A call management system in accordance with claim 336, wherein:

said one or more spoken words comprises at least a portion of the name of said calling party.

338. A call management system in accordance with claim 322, wherein:

said identifying means comprises at least one caller ID database.

339. A call management system in accordance with claim 338, comprising:

means for updating said database with information received from a calling party not in said caller ID database.

340. A call management system in accordance with claim 339, wherein:

said database contains telephone numbers and listed names of calling parties.

341. A call management system in accordance with claim 339, wherein:

said database contains voice representations of the calling party.

342. A call management system in accordance with claim 339, wherein:

said database contains Internet addresses of calling parties.

343. A call management system in accordance with claim 339, wherein:

said database contains e-mail addresses of calling parties.

344. A call management system in accordance with claim 338, comprising:

means for updating said database with information entered from at least one user position.

345. A call management system in accordance with claim 310, comprising:

means for playing a pre-recorded message to said calling party.

346. A call management system in accordance with claim 322, comprising:

means for playing a pre-recorded message to said calling party.

347. A call management system in accordance with claim 346, comprising:

means responsive to said identifying means identifying a calling party to determine at least in part which pre-recorded message is played.

348. A call management system in accordance with claim 346, wherein:

said identifying means is responsive to the identity of a called party to determine at least in part which pre-recorded message is played.

349. A call management system in accordance with claim 346, wherein:

said identifying means is responsive to the identity of said calling party and the identity of the called party to select said pre-recorded message.

350. A call management system in accordance with claim 311, further comprising:

means at each said user position for notifying said call management computer to place a received call on hold and to switch said call path to another call.

351. A call management system in accordance with claim 311, comprising:

means at said user position for notifying said call management computer to disconnect a first calling party coupled to said call path and to couple said call path to a second party.

352. A call management system in accordance with claim 310, comprising:

means at each said user position for notifying said call management computer to release said preserved call path.

353. A call management system in accordance with claim 310, wherein:

said call path includes a destination on the public switched telephone network.

354. A call management system in accordance with claim 310, wherein:

said call path includes a destination on a private telephone network.

355. A call management system in accordance with claim 310, wherein:

said call path includes a destination on the Internet.

356. A call management system in accordance with claim 310, wherein:

said call path includes a destination on a private digital network.

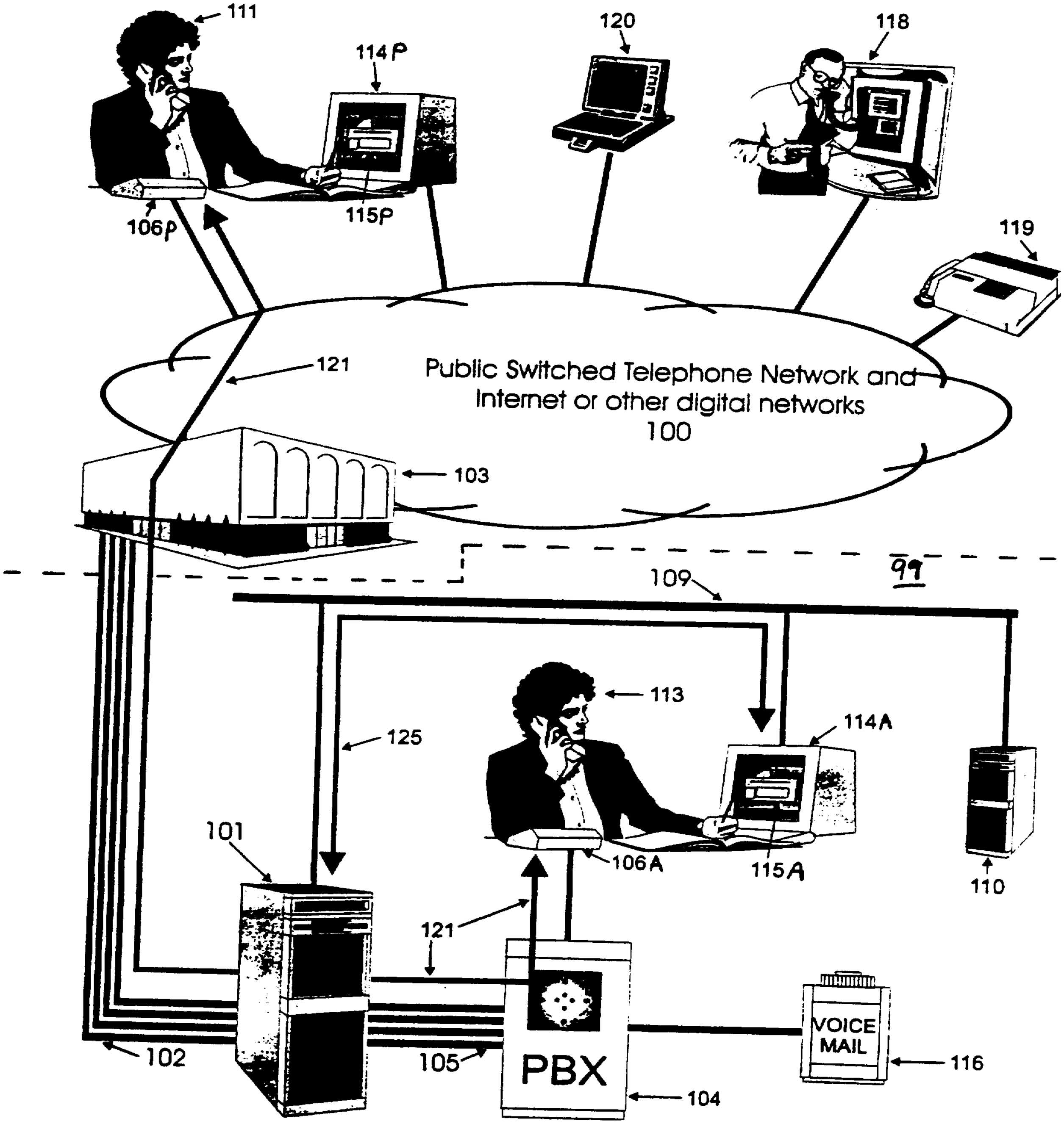

111
120
118
114P
115P
106P
119
121
Public Switched Telephone Network and
Internet or other digital networks
100
103
109
99
113
114A
125
101
106A
115A
110
121
VOICE
MAIL
102
105
PBX
116
104

Fig. 1.

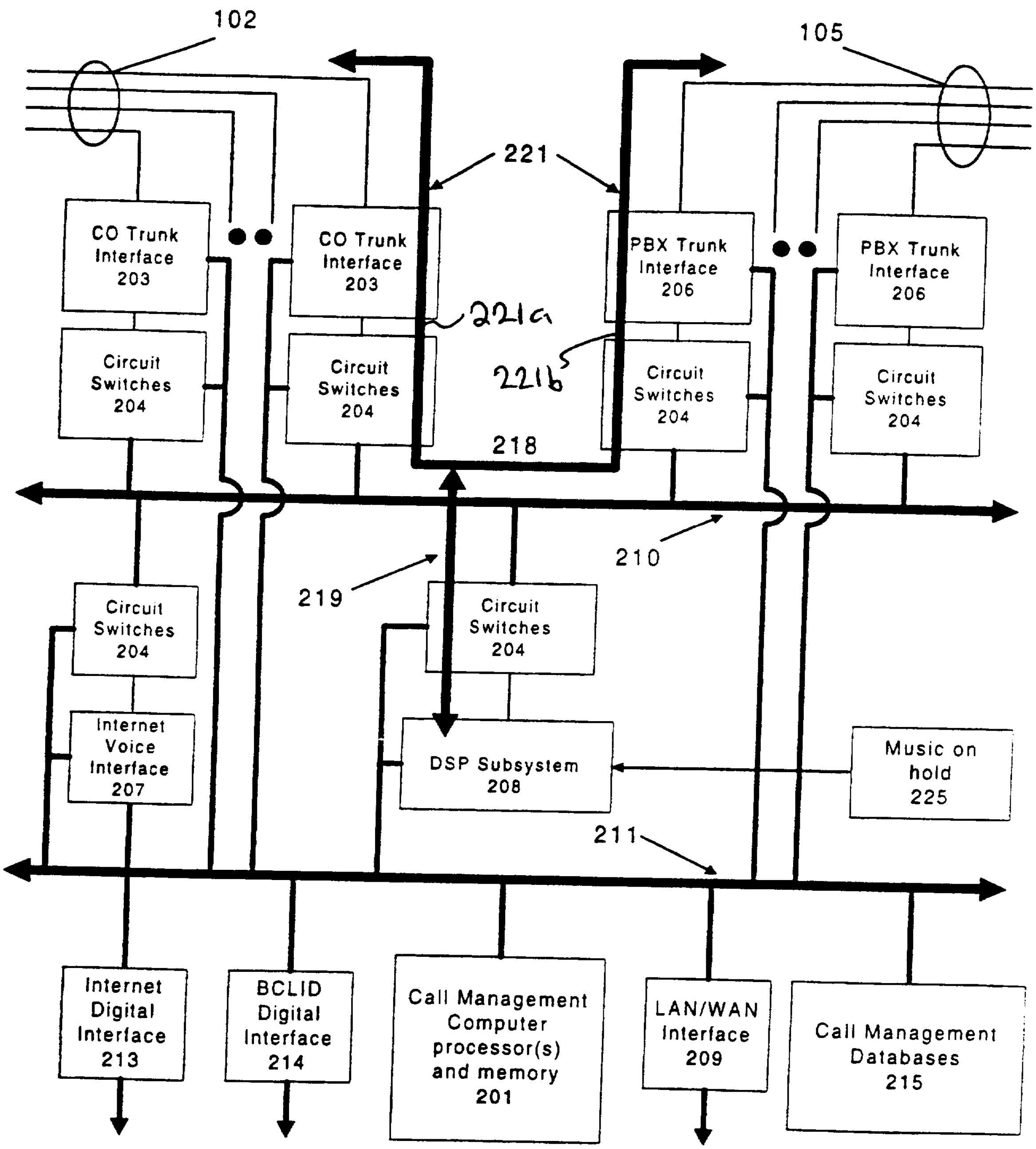

102
105
221
CO Trunk Interface 203
CO Trunk Interface 203
PBX Trunk Interface 206
PBX Trunk Interface 206
221a
221b
Circuit Switches 204
Circuit Switches 204
Circuit Switches 204
Circuit Switches 204
218
210
219
Circuit Switches 204
Circuit Switches 204
Internet Voice Interface 207
DSP Subsystem 208
Music on hold 225
211
Internet Digital Interface 213
BCLID Digital Interface 214
Call Management Computer processor(s) and memory 201
LAN/WAN Interface 209
Call Management Databases 215

Fig. 2.

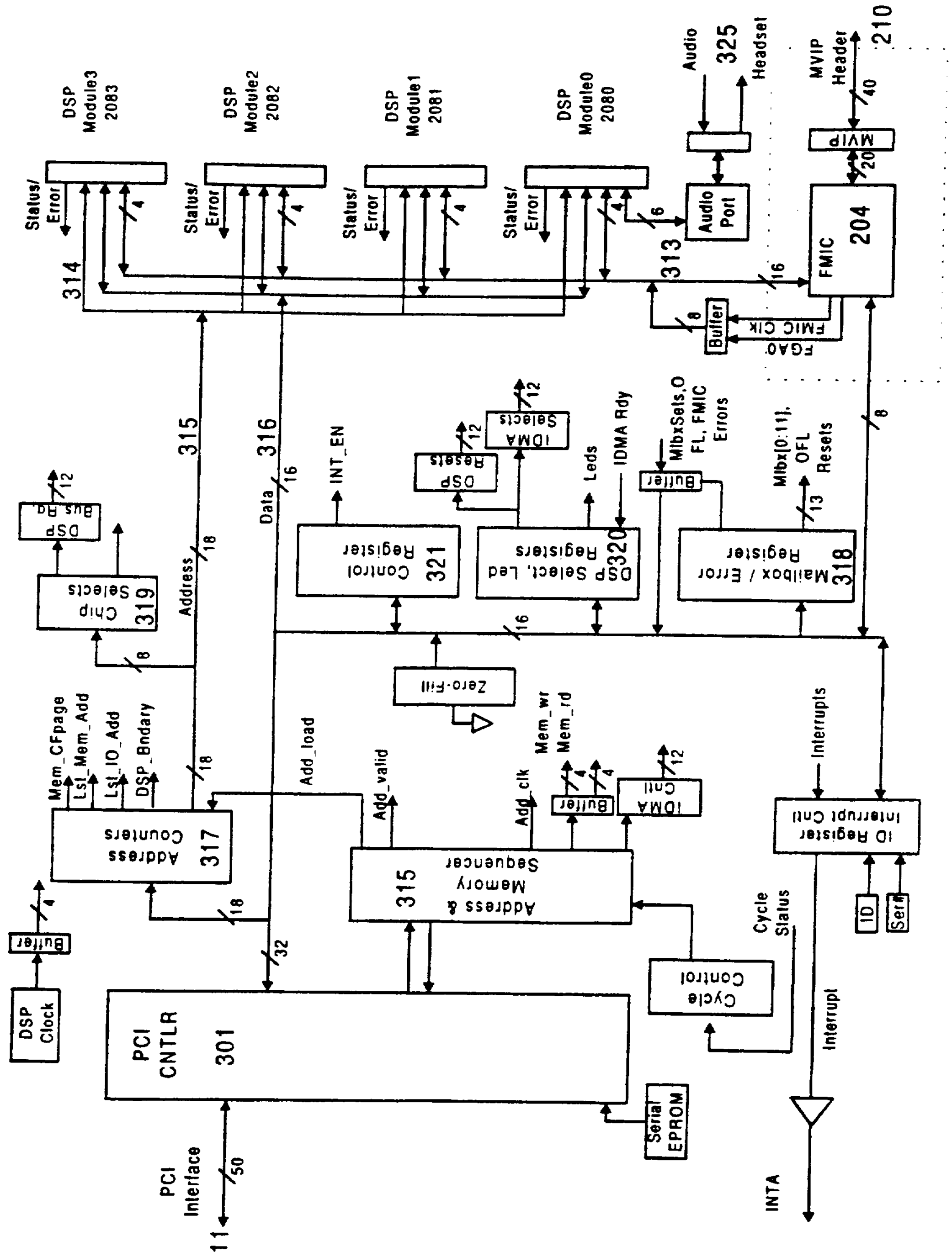

DSP Module3 2083
DSP Module2 2082
DSP Module1 2081
DSP Module0 2080
Audio
325
Headset
MVIP Header
210
MVIP
Audio Port
FMIC
204
313
314
Status/ Error
Buffer
FMIC Clk
FGA0
315
316
Data
Address
INT_EN
DSP Resets
IDMA Selects
Leds
IDMA Rdy
MlbxSels,O FL, FMIC Errors
Mlbx[0:11], OFL Resets
DSP Bus Rq.
Chip Selects
319
Control Register
321
DSP Select, Led Registers
320
Mailbox / Error Register
318
Zero-Fill
Mem_CFpage
Lst_Mem_Add
Lst_IO_Add
DSP_Bndary
Address Counters
317
Add_load
Add_valid
Add_clk
Mem_wr
Mem_rd
IDMA Cntl
Interrupts
ID Register Interrupt Cntl
Address & Memory Sequencer
315
ID
Serial
DSP Clock
Cycle Status
Cycle Control
PCI CNTLR
301
Serial EPROM
Interrupt
PCI Interface
211
INTA

Fig. 3.

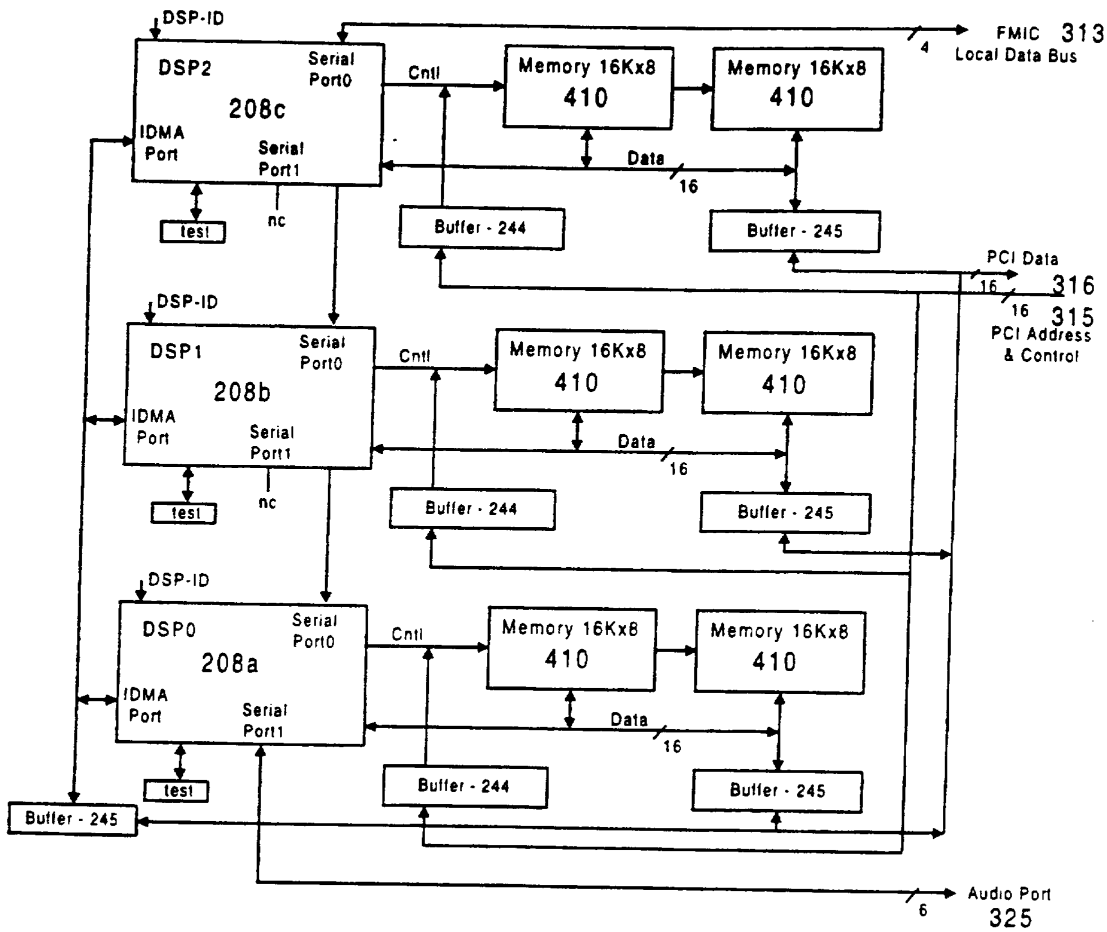
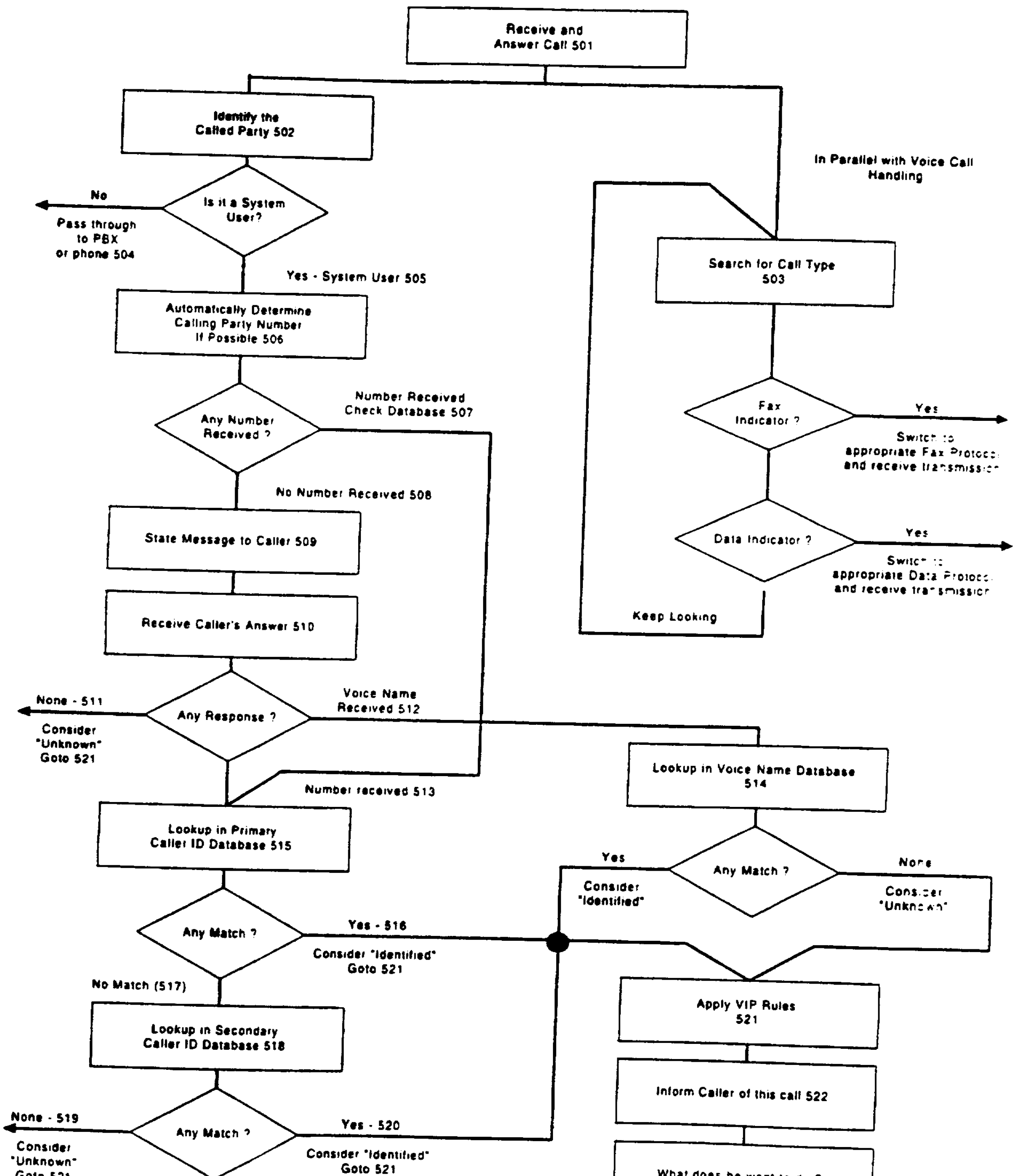

DSP-ID
DSP2
208c
Serial Port0
IDMA Port
Serial Port1
test
nc
Cntl
Memory 16Kx8 410
Memory 16Kx8 410
Data
16
Buffer - 244
Buffer - 245
4
FMIC 313
Local Data Bus
PCI Data
16
316
16
315
PCI Address & Control
DSP-ID
DSP1
208b
Serial Port0
IDMA Port
Serial Port1
test
nc
Cntl
Memory 16Kx8 410
Memory 16Kx8 410
Data
16
Buffer - 244
Buffer - 245
DSP-ID
DSP0
208a
Serial Port0
IDMA Port
Serial Port1
test
Cntl
Memory 16Kx8 410
Memory 16Kx8 410
Data
16
Buffer - 244
Buffer - 245
Buffer - 245
6
Audio Port
325

Fig. 4.



Receive and Answer Call 501
Identify the Called Party 502
In Parallel with Voice Call Handling
No
Is it a System User?
Pass through to PBX or phone 504
Search for Call Type 503
Yes - System User 505
Automatically Determine Calling Party Number If Possible 506
Number Received Check Database 507
Any Number Received ?
Fax Indicator ?
Yes
No Number Received 508
State Message to Caller 509
Data Indicator ?
Yes
Receive Caller's Answer 510
Keep Looking
None - 511
Consider "Unknown" Goto 521
Any Response ?
Voice Name Received 512
Lookup in Voice Name Database 514
Number received 513
Lookup in Primary Caller ID Database 515
Yes
Any Match ?
None
Consider "Identified"
Any Match ?
Yes - 516
Consider "Identified" Goto 521
No Match (517)
Apply VIP Rules 521
Lookup in Secondary Caller ID Database 518
Inform Caller of this call 522
None - 519
Any Match ?
Yes - 520
Consider "Unknown" Goto 521
Consider "Identified" Goto 521
What does he want to do ?

Fig. 5.

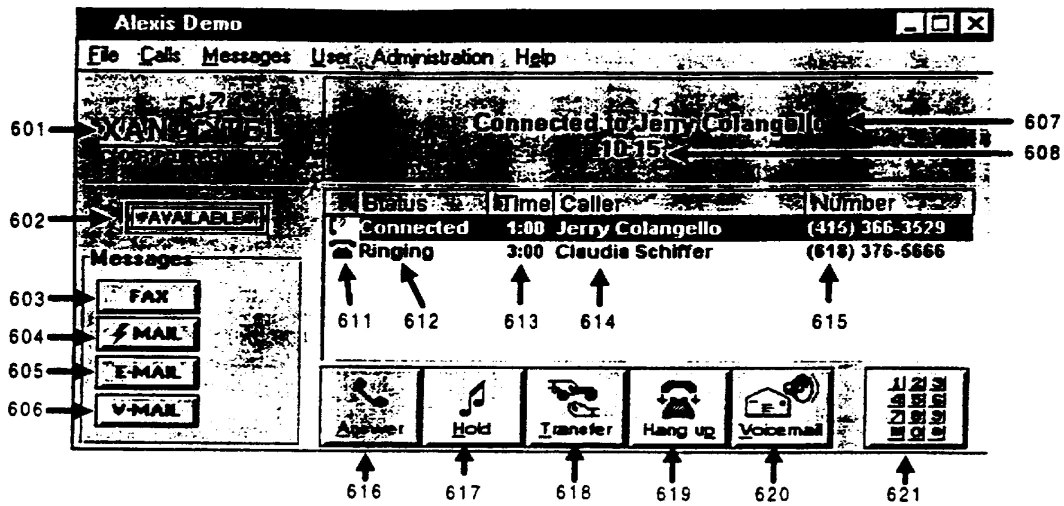

Alexis Demo
File Calls Messages User Administration Help
Connected to Jerry Colangelo
AVAILABLE
Messages
FAX
MAIL
E-MAIL
V-MAIL
Status Time Caller Number
Connected 1:00 Jerry Colangello (415) 366-3529
Ringing 3:00 Claudia Schiffer (618) 376-5666
Answer
Hold
Transfer
Hang up
Voicemail
601
602
603
604
605
606
607
608
611
612
613
614
615
616
617
618
619
620
621

Fig. 6a.

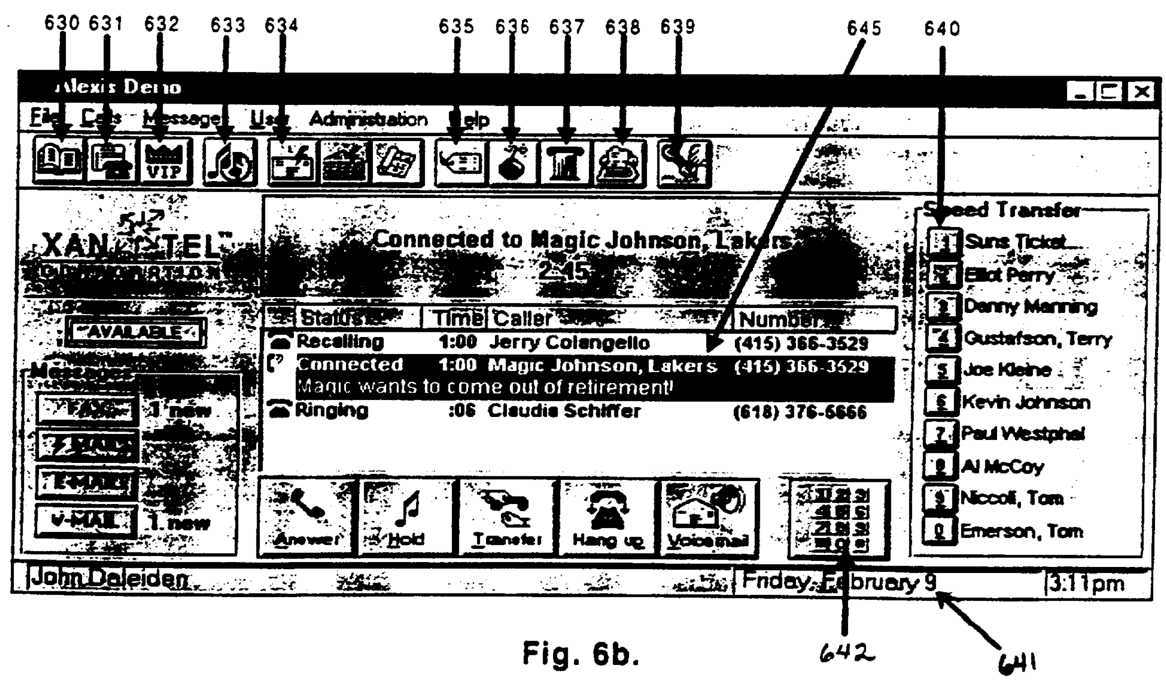

Alexis Demo
File Calls Messages User Administration Help
VIP
Connected to Magic Johnson, Lakers
2:15
AVAILABLE
Messages
1 new
1 new
Status Time Caller Number
Recalling 1:00 Jerry Colangello (415) 366-3529
Connected 1:00 Magic Johnson, Lakers (415) 366-3529
Magic wants to come out of retirement!
Ringing :06 Claudia Schiffer (618) 376-5666
Answer
Hold
Transfer
Hang up
Voicemail
Speed Transfer
Suns Ticket...
Elliot Perry
Danny Manning
Gustafson, Terry
Joe Kleine
Kevin Johnson
Paul Westphal
Al McCoy
Niccoli, Tom
Emerson, Tom
John Deleiden
Friday, February 9
3:11pm
630
631
632
633
634
635
636
637
638
639
645
640
642
641

Fig. 6b.

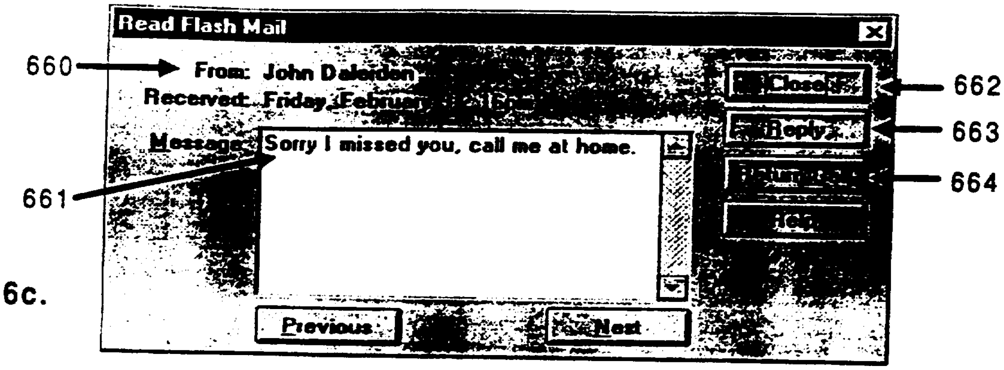
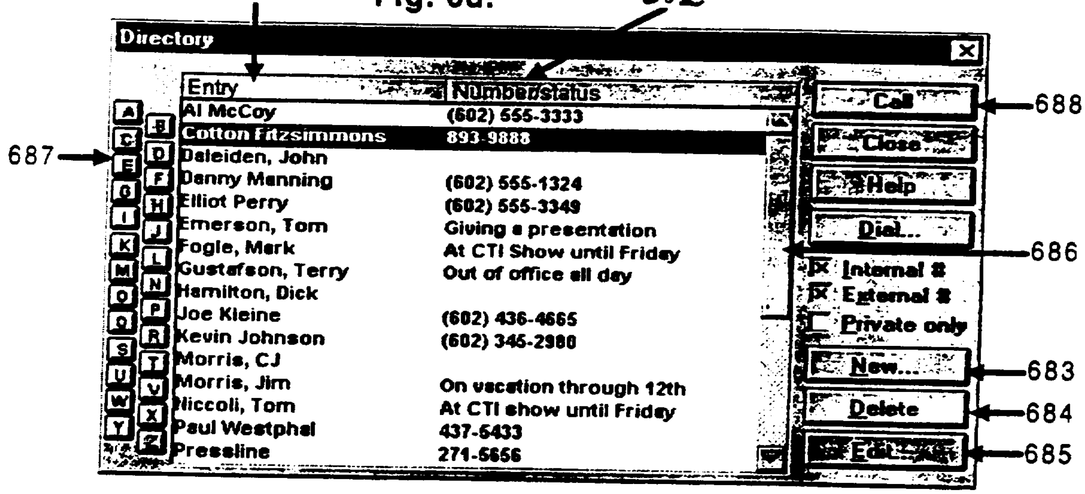


Fig. 6c.

Fig. 6d.



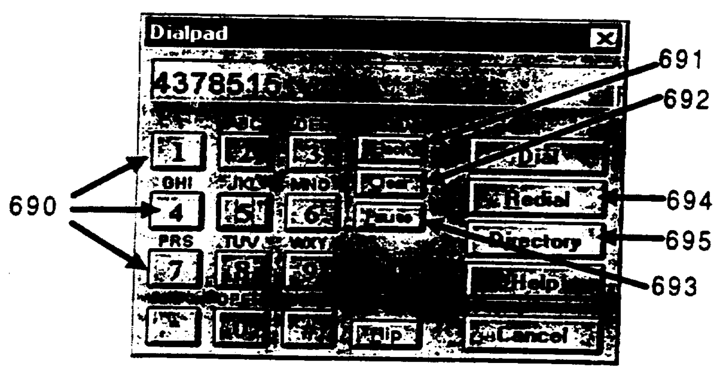


Fig. 6e.

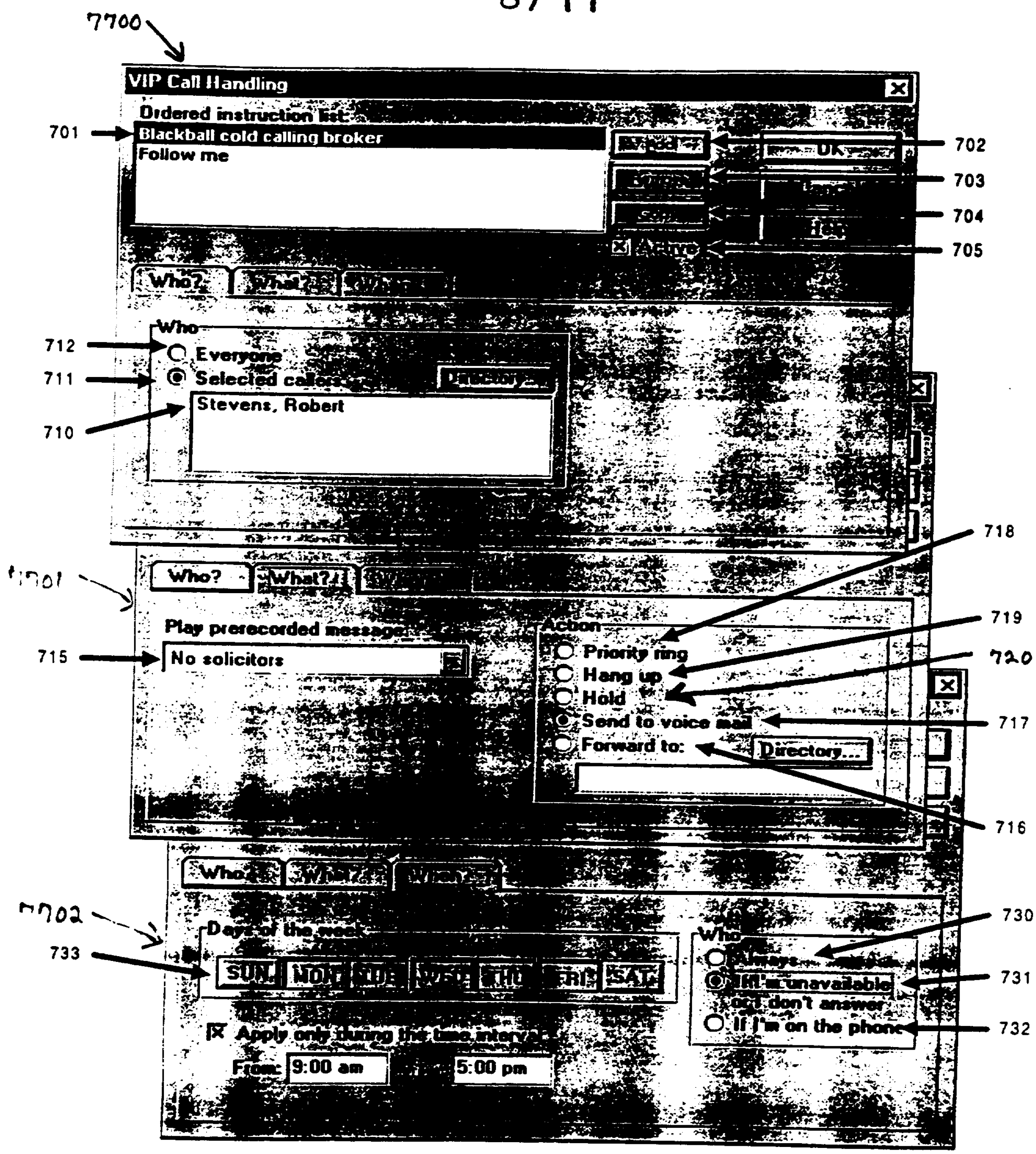
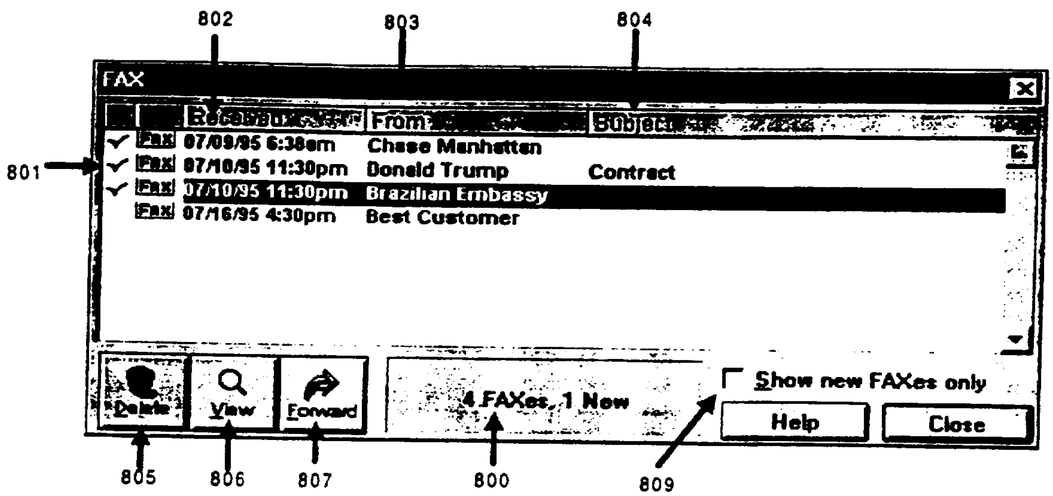

7700
VIP Call Handling
Ordered instruction list:
Blackball cold calling broker
Follow me
701
702
703
704
705
Who?
What?
Who
Everyone
Selected callers
Stevens, Robert
712
711
710
7701
Who?
What?
Play prerecorded message:
No solicitors
Action
Priority ring
Hang up
Hold
Send to voice mail
Forward to:
Directory...
715
718
719
720
717
716
7702
Who?
What?
When?
Days of the week
SUN
SAT
Apply only during the time interval
From: 9:00 am
5:00 pm
Who
Always
If I'm unavailable
If I'm on the phone
733
730
731
732

Fig. 7



802
803
804
FAX
Received
From
Subject
801
Fax 07/09/95 6:38am Chase Manhattan
Fax 07/10/95 11:30pm Donald Trump Contract
Fax 07/10/95 11:30pm Brazilian Embassy
Fax 07/16/95 4:30pm Best Customer
Delete
View
Forward
4 FAXes, 1 New
Show new FAXes only
Help
Close
805
806
807
800
809

Fig. 8.

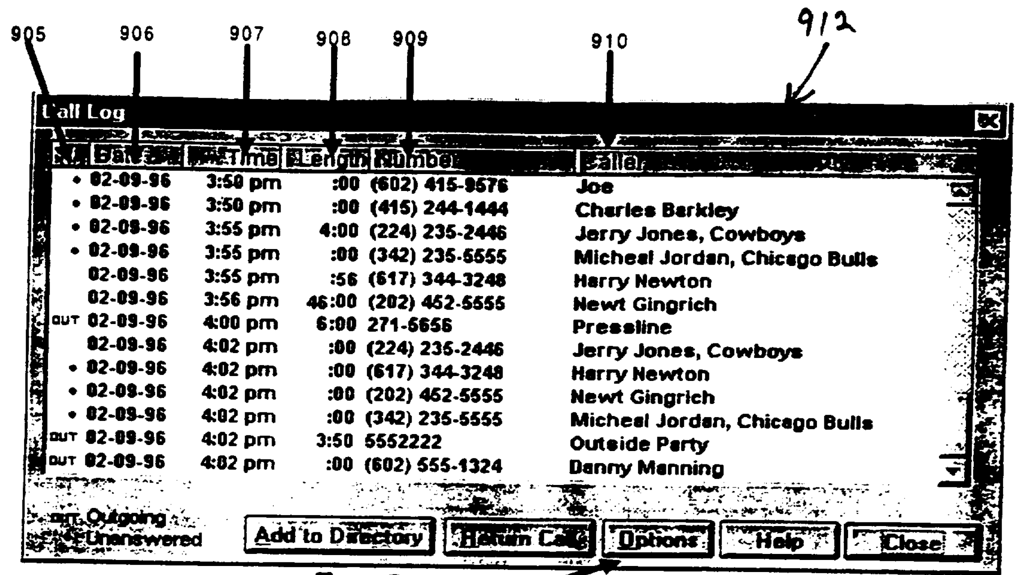

905
906
907
908
909
910
912
Call Log
Date
Time
Length
Number
Caller
• 02-09-96 3:50 pm :00 (602) 415-9576 Joe
• 02-09-96 3:50 pm :00 (415) 244-1444 Charles Barkley
• 02-09-96 3:55 pm 4:00 (224) 235-2446 Jerry Jones, Cowboys
• 02-09-96 3:55 pm :00 (342) 235-5555 Micheal Jordan, Chicago Bulls
02-09-96 3:55 pm :56 (617) 344-3248 Harry Newton
02-09-96 3:56 pm 46:00 (202) 452-5555 Newt Gingrich
OUT 02-09-96 4:00 pm 6:00 271-5656 Pressline
02-09-96 4:02 pm :00 (224) 235-2446 Jerry Jones, Cowboys
• 02-09-96 4:02 pm :00 (617) 344-3248 Harry Newton
• 02-09-96 4:02 pm :00 (202) 452-5555 Newt Gingrich
• 02-09-96 4:02 pm :00 (342) 235-5555 Micheal Jordan, Chicago Bulls
OUT 02-09-96 4:02 pm 3:50 5552222 Outside Party
OUT 02-09-96 4:02 pm :00 (602) 555-1324 Danny Manning
OUT Outgoing
• Unanswered
Add to Directory
Return Call
Options
Help
Close
911

Fig. 9b

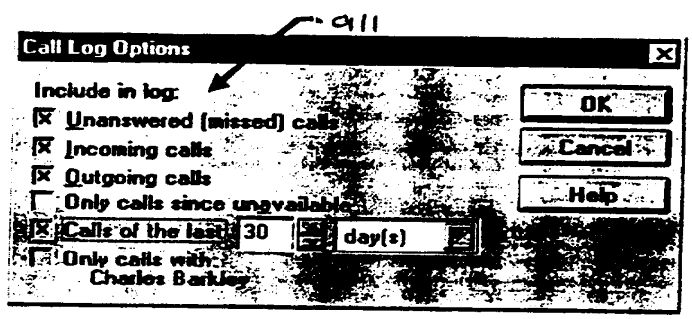

911
Call Log Options
Include in log:
Unanswered (missed) calls
Incoming calls
Outgoing calls
Only calls since unavailable
Calls of the last 30 day(s)
Only calls with:
Charles Barkley
OK
Cancel
Help

Fig. 9a .

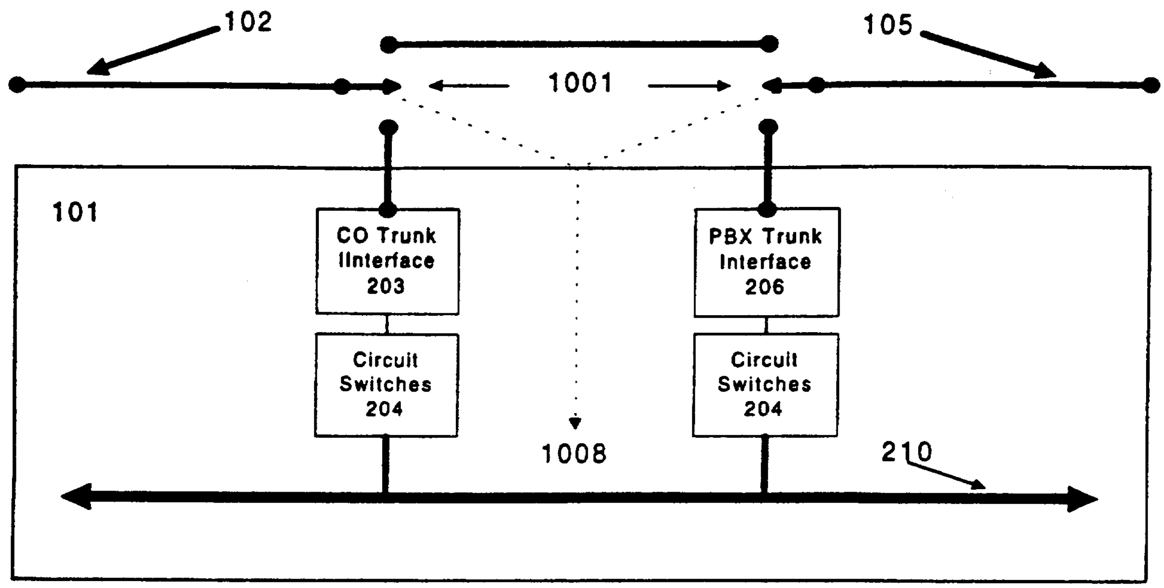


Fig. 10b

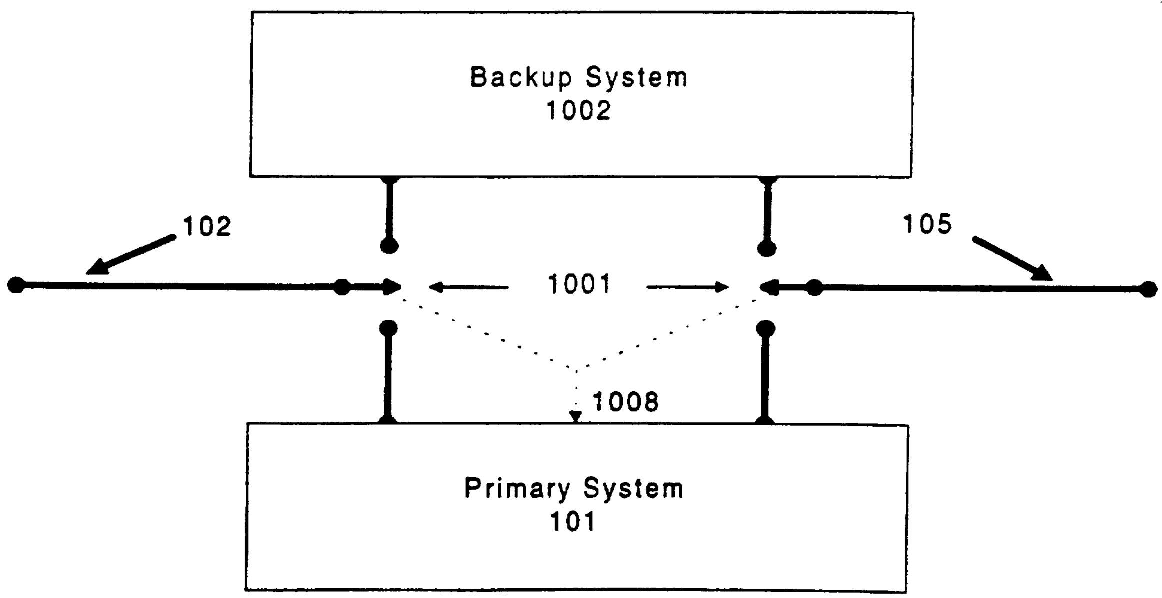


Fig. 10.a

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US97/04689

A. CLASSIFICATION OF SUBJECT MATTER
IPC(6) :H04M 1/64, 11/00, 3/00
US CL : 379/67, 100, 242
According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
U.S. : 379/67, 100, 242

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

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

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X --- Y	US, A, 4,797,911 (Szlam et al) 10 January 1989, Fig.1 and columns 12-14, 16.	1 ------- 2-296
X ---- Y	US, A, 5,097,528 (Gursahaney et al) 17 March 1992, Figs 2-5	1 ---- 2-296
X --- Y	US, A, 5,214,688 (Szlam et al) 25 May 1993, columns 2-3.	297,301 --------------- 298-300, 302-309
X	US, A, 4,313,035 (Jordan et al) 26 January 1982, columns 2-4	310-356

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

* Special categories of cited documents:

"A" document defining the general state of the art which is not considered to be part of particular relevance

"E" earlier document published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
25 JUNE 1997	11 AUG 1997

Name and mailing address of the ISA/US
Commissioner of Patents and Trademarks
Box PCT
Washington, D.C. 20231
Facsimile No. (703) 305-3230

Authorized officer
FAN TSANG
Telephone No. (703)3054700

Form PCT/ISA/210 (second sheet)(July 1992)★

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US97/04689

Box I Observations where certain claims were found unsearchable (Continuation of item 1 of first sheet)

This international report has not been established in respect of certain claims under Article 17(2)(a) for the following reasons:

1. ☐ Claims Nos.:
because they relate to subject matter not required to be searched by this Authority, namely:

2. ☐ Claims Nos.:
because they relate to parts of the international application that do not comply with the prescribed requirements to such an extent that no meaningful international search can be carried out, specifically:

3. ☐ Claims Nos.:
because they are dependent claims and are not drafted in accordance with the second and third sentences of Rule 6.4(a).

Box II Observations where unity of invention is lacking (Continuation of item 2 of first sheet)

This International Searching Authority found multiple inventions in this international application, as follows:

Please See Extra Sheet.

1. ☒ As all required additional search fees were timely paid by the applicant, this international search report covers all searchable claims.

2. ☐ As all searchable claims could be searched without effort justifying an additional fee, this Authority did not invite payment of any additional fee.

3. ☐ As only some of the required additional search fees were timely paid by the applicant, this international search report covers only those claims for which fees were paid, specifically claims Nos.:

4. ☐ No required additional search fees were timely paid by the applicant. Consequently, this international search report is restricted to the invention first mentioned in the claims; it is covered by claims Nos.:

Remark on Protest
☐ The additional search fees were accompanied by the applicant's protest.
☒ No protest accompanied the payment of additional search fees.

Form PCT/ISA/210 (continuation of first sheet(1))(July 1992)★

BOX II. OBSERVATIONS WHERE UNITY OF INVENTION WAS LACKING
This ISA found multiple inventions as follows:

This application contains the following inventions or groups of inventions which are not so linked as to form a single inventive concept under PCT Rule 13.1. In order for all inventions to be examined, the appropriate additional examination fees must be paid.

Group I, claim(s) 1-296 AND 305, drawn to VOICE MESSAGING SYSTEM, classified in U.S. class 379, subclass 67.

Group II, claim(s) 297-304, 306-309, drawn to ELECTRONIC MESSAGING SYSTEM, classified in U.S. class 379, subclass 100.

Group III, claim(s) 310-356, drawn to CENTRALIZED SWITCHING SYSTEM, classified in U.S.class 379, subclass 242.

The inventions listed as Groups I, II AND III do not relate to a single inventive concept under PCT Rule 13.1 because, under PCT Rule 13.2, they lack the same or corresponding special technical features for the following reasons:
A voice messaging system is used for storing, retrieving and processing voice signals. However, a electronic messaging system handles only non-voice electrical messages. It is clear that these two messaging systems are different from each other. Also, a centralized switching system deals with routing calls which is different from voice/electronic messaging systems for storing, processing or retrieving messages.

EP 22070 (3)

PCT

WORLD INTELLECTUAL PROPERTY ORGANIZATION
International Bureau

INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(51) International Patent Classification 6 : H04M 1/27, G10L 5/06 — A1

(11) International Publication Number: WO 97/37481

(43) International Publication Date: 9 October 1997 (09.10.97)

(21) International Application Number: PCT/CA97/00008

(22) International Filing Date: 9 January 1997 (09.01.97)

(30) Priority Data:
08/623,635 28 March 1996 (28.03.96) US

(71) Applicant: NORTHERN TELECOM LIMITED [CA/CA]; World Trade Center of Montreal, 8th floor, 380 St. Antoine Street West, Montreal, Quebec H2Y 3Y4 (CA).

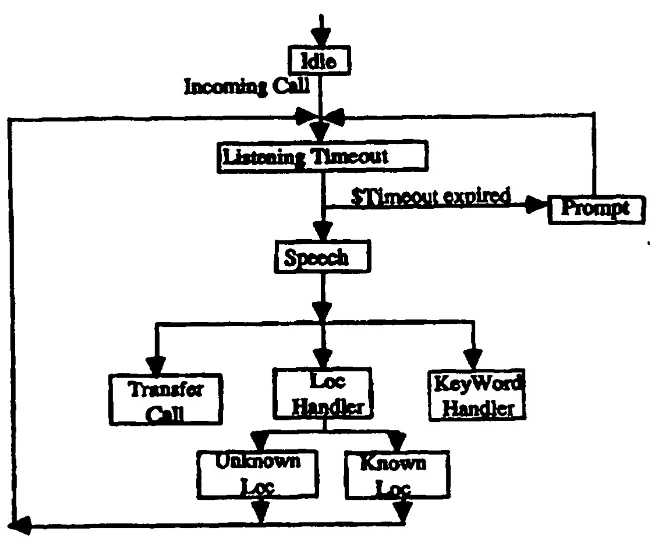
(72) Inventor: WONG, Chi; Apartment #3390, 2850 Middlefield Road, Palo Alto, CA 94306 (US).

(74) Agent: MacGREGOR, George; Northern Telecom Limited, Patent Dept., P.O. Box 3511, Station "C", Ottawa, Ontario K1Y 4H7 (CA).

(81) Designated States: CA, European patent (AT, BE, CH, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE).

Published
With international search report.

(54) Title: APPARATUS AND METHOD FOR REDUCING SPEECH RECOGNITION VOCABULARY PERPLEXITY AND DYNAMICALLY SELECTING ACOUSTIC MODELS



(57) Abstract

A method of reducing the perplexity of a speech recognition vocabulary and dynamically selecting speech recognition acoustic model sets used in a simulated telephone operator apparatus. The directory of users of the telephone network is subdivided into subsets wherein each subset contains the names of users within a certain location or exchange. A speech recognition vocabulary database is compiled for each subset and the appropriate database is loaded into the speech recognition apparatus in response to a requested call to the location covered by the subset. Furthermore, a site-specific acoustic model set is dynamically loaded according to the location of a calling party. An apparatus for carrying out the method is also discussed.

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CM	Cameroon	KR	Republic of Korea	PT	Portugal		
CN	China	KZ	Kazakstan	RO	Romania		
CU	Cuba	LC	Saint Lucia	RU	Russian Federation		
CZ	Czech Republic	LI	Liechtenstein	SD	Sudan		
DE	Germany	LK	Sri Lanka	SE	Sweden		
DK	Denmark	LR	Liberia	SG	Singapore		
EE	Estonia						

APPARATUS AND METHOD FOR REDUCING SPEECH RECOGNITION VOCABULARY PERPLEXITY AND DYNAMICALLY SELECTING ACOUSTIC MODELS

Field of the Invention

This invention relates to automatic speech recognition in telecommunication systems and to the use of such systems to provide large scale voice activated dialing and information retrieval services.

Background to the Invention

In the early development of telephone systems it was commonplace for a telephone subscriber to converse directly with a telephone operator at a telephone central office. The telephone subscriber would verbally request the telephone operator to complete a connection to a called party. As telephone exchanges were small the telephone operator was aware of virtually all of the subscribers by name and manually completed the requested connection. With the advent of dial telephone services, calls within an exchange could be completed automatically, and only certain toll calls required operator assistance. Today, operator assisted calls have become the exception and are usually comparatively expensive. Machine-simulated operator functions, including limited speech recognition services, have recently been available for expediting some typical operator-assisted functions. This includes "collect" long distance calls wherein completion of the connection is contingent upon the called party agreeing to pay for the service. However, these functions are limited to the simple recognition of "yes" or "no" so there is little room for non-functionality due to uncertainty as to which word was spoken. There have also been advancements in voice recognition services relating to directory assistance but these too are directed to a very limited vocabulary.

Prior Art

The prior art contains several recent developments pertaining to voice recognition in general, and to voice recognition applicable to telecommunication systems in particular.

U.S. Patent No. 5,091,947, which issued February 25, 1992 to Ariyoshi et al, entitled "Speech Recognition Method and Apparatus", discloses a voice recognition system for comparing both speaker dependent and speaker independent utterances against stored voice patterns within a coefficient memory. The voice identification comparator selects the one voice pattern having the highest degree of similarity with the utterance in question.

In U.S. Patent No. 5,165,095, which issued on November 17, 1992, Borcherding discloses a voice recognition system to initiate dialog to determine the correct telephone number. According to the '095 patent, the calling party is first identified so that a database containing speaker templates can be accessed. These templates are then used to compare the dial command so that the dialing instructions can be recognized and executed. An example of a dialing directive in the patent is "call home", with "call" being the dial command and "home" being the destination identifier.

Gupta et al, in U.S. Patent No. 5,390,278 issued February 14, 1995, disclose a flexible vocabulary speech recognition for recognizing speech transmitted via the public switched telephone network. This voice recognition technique is a phoneme based system wherein the phonemes are modeled as hidden Markov models.

In spite of these ongoing developments, the functionality of automatic recognition of human speech by machine has not advanced to a degree wherein a calling party can simply speak the called party's name and thereafter be connected as reliably as a human operator in situations where the database for a potential called party is very large (greater than 100 names).

Summary of the Invention

The present invention is in the field of human speech recognition performed by machines and more particularly relates to a reduction of the perplexity of the speech recognition task in the context of names spoken by a telephone user in a telephone system.

Individual users of telephone networks are divided into subsets to facilitate identification of the vast number of subscribers to the service. In the public network these subsets are local exchanges. Private switching networks such as the Nortel Electronic Switching Network (ESN) assigns individual ESN numbers to each location within the private network. The present invention relies on these subsets or location identifiers to reduce the perplexity of a speech recognition application.

Therefore in accordance with a first aspect of the present invention, there is provided a telephone network including a plurality of telephone exchanges, each for serving a plurality of telephone terminals and each being interconnected with at least one other of the telephone exchanges for providing telephone communications between users of the telephone terminals. The network function includes a simulated telephone operator apparatus for receiving a speech request from a user for connection to another telephone user and to translate this request into a directory number for use by the appropriate one of the telephone exchanges. The translation is in accordance with a speech recognition algorithm and an active speech recognition vocabulary selected in accordance with the origin of the request.

In an ESN application the active speech recognition vocabulary is limited to the names of the individuals serviced by the ESN number. In a preferred embodiment the ESN number, which is also a location code, is contained in the first two or three digits of the directory number.

In accordance with a second aspect of the invention there is provided a simulated telephone operator server for a telephone network. The server has means for storing voice utterances of a calling party telephone user and means responsive to location information in association with the telephone user for selecting an active speech recognition vocabulary. Speech detection means are provided for processing the stored voice utterance in accordance with a speech recognition algorithm and the active speech recognition vocabulary. Directory lookup means identify a directory listing of a called party corresponding to a result of the processing by the speech detection means. The server also includes means for transmitting the directory listing to a telephone exchange serving the called party.

In accordance with a further aspect of the invention there is provided a telephone exchange comprising: a plurality of ports for serving a plurality of telephone users' telephone instruments via telephone lines; a trunk facility for connection to another telephone exchange; a switching network for connecting and disconnecting the telephone instruments; a controller means for causing a newly OFF HOOK telephone instrument to be coupled via the switching network with a solicitation signal, and subsequently for being responsive to a telephone number received in association with the newly OFF HOOK telephone instrument for completing a telephone call via the switching network. The exchange also includes an originating register means for storing voice band signals received from the newly OFF HOOK telephone instrument via the switching network.

Means are provided for detecting digits represented by frequency signals, within the stored voice band signals, in accordance with a standard for key pad dialed telephone numbers and for transmitting detected digits to the call controller. A simulated telephone operator apparatus receives and translates voice band signals in accordance with a speech recognition algorithm and an active speech recognition vocabulary selected in accordance with the origin of the voice band signals into a directory number for use by the controller means. An interface facility is provided for transmitting the stored voice band signals via the switching network to the simulated telephone operator server apparatus in an event wherein the voice band signals did not include a key pad dialed digit.

In accordance with yet a further aspect of the present invention there is provided a method of detecting a voiced speech request of a calling party for connection to another user of an automatic telephone exchange. The method comprises storing a plurality of speech recognition vocabularies in association with geographic location of users; receiving the voiced request and information as to the geographic location of the user having voiced the request from the automatic telephone exchange; selecting an active speech recognition vocabulary in accordance with the information as to the geographic location of the user and, in accordance with a speech recognition algorithm and the selected active speech recognition vocabulary, translating the received request into a directory number for use by the automatic telephone exchange in setting up a telephone connection between the calling telephone user and the other telephone user.

Brief Description of the Drawings

The invention will now be described in greater detail with reference to the attached drawings wherein:

FIGURE 1 is a block diagram illustrating trunk connections between private switch locations;

FIGURE 2 is a block diagram of the system hardware architecture;

FIGURE 3 is an overall system state diagram; and

FIGURE 4 is a state diagram of the key word handler.

Detailed Description of the Invention

The following description relates to an enterprise-wide speech directory calling service within a company or corporation having a number of locations. Each location is assigned a unique electronic switching network (ESN) location code or ESN number. As shown in the block diagram of FIGURE 1, the on-site PBX 20 at each location is connected to each other location via trunk connectors 22. In this discussion the ESN comprises a three-digit code to identify the location. It is to be understood, however, that it is not essential to use all three digits to identify the location as it may be sufficient to use the first two for example.

FIGURE 2 illustrates the hardware architecture in accordance with a preferred embodiment of the invention. As shown, PBX 20 is connected to trunk 22 and to a plurality of on site telephone sets as known in the art. The speech recognition system 30 of the invention is connected to the PBX 20 via T1 line 32 via T1 card 34 and via signal link 36 and signal link card 38. Speech recognition system 30 includes a speech recognition processor operating on a speech recognition algorithm, central processor and control units as well as memory cards for storing active speech recognition vocabulary data bases.

Although FIGURE 1 refers to a private switching network using ESNs, it is to be understood that the invention is not limited to such networks but can also be adapted to use in public switching systems.

One objective metric used to measure the accuracy of a speech recognition system is the Word Error Rate (WER). The WER is defined as the total number of incorrectly recognized words made by a speech recognition system divided by the total number of words spoken by a user of the system.

$$WER = \frac{NumberofErrorsMadebyRecognizer}{NumberofWordsSpokenbyUser}$$

The present invention makes use of information as to the calling party's location for automatically assisting in improving the WER of a speech recognition system on a spoken called party's name for the purpose of connecting a telephone call.

It has been empirically shown that the WER of a speech recognition system will vary with the square root of the perplexity of the vocabulary of words being recognized. [Kimbal, O., et al., "Recognition Performance and Grammatical Constraints", Proceedings of a Workshop on Speech Recognition, Report Number SAIC-86/1546, Defense Advanced Research Projects Agency, Palo Alto, February 19-20, 1986.]

$$WER \propto \sqrt{Perplexity}$$

The perplexity of a vocabulary is defined as the measure of the constraint imposed by a grammar, or the level of uncertainty given the grammar of a population of users. Perplexity is mathematically modeled and quantified in the following way:

$$H = -\frac{1}{|V|}\sum_{w \in V} P(w) \bullet \log P(w)$$

$$B = 2^H$$

where: H is entropy
$P(w)$ is the probability of w being spoken
B is the perplexity of the application

The vocabulary of words in this implementation consists entirely of proper names; location names, and a small number of key words for command and control. For large corporations with a large number of employees, the proper names become the determining factor in measuring the perplexity since the number of employees will overwhelm the number of location names and key words. Thus location names and key words can be ignored in this calculation. If we make a simplifying assumption that every name is spoken with equal probability, then the equation above can be simplified to the following:

$$Perplexity = \sqrt[L]{|S|}$$

where: L is the average number of words in a sentence
S is the number of sentences in the vocabulary V

If we further make the simplification that proper names contain two words -- first and last name -- and the number of sentences in the vocabulary is equivalent to the number of employee names, then we can further reduce the equation to the following:

$$Perplexity = \sqrt{|S|}$$

If we make the assumption that the amount of confusability between names within a large database will be similar between large databases, the accuracy of a speech recognition system has the following relationship with the number of names in the vocabulary:

$$WER \propto \sqrt[4]{NumberofActiveDirectoryNames}$$

We can observe from the above equations that the WER increases with the perplexity and thus increases with the number of proper names in the vocabulary.

In the past, speech recognition scientists have used various methods to reduce the perplexity in an effort to improve the WER of a speech recognition system. To achieve this result, most of these efforts have been focused at the linguistic level. For example, scientists have used statistical language models and linguistics rules of phonology to reduce perplexity or uncertainty in recognizing a spoken word or phrase.

In this implementation the list of employee names for each location is stored in a separate speech recognition vocabulary. The employee name will normally be associated with the four digits of the telephone number following the three-digit ESN or location code. According to the system of the present invention a calling party wishing to speak to another employee at the same location will simply announce the first and last name of the employee to whom a connection is desired. The speech recognition system will assume that calling party and called party are at the same location and load the active speech recognition vocabulary database containing the names of all employees at that location. Using a conventional speech recognition algorithm the name spoken by the calling party is compared by the system against the names of all employees in the database and the closest match is selected. The name selected is announced to the calling party and the call is automatically connected to the line associated with the telephone number assigned to the called party unless the calling party interrupts the process by saying, "No." Thereafter the voice recognition system releases from the call.

If the called party is at a different location than the calling party, the calling party will first announce the location of the called party followed by the called party's name. The voice recognition system responds by announcing the location and subsequently loading the active voice recognition vocabulary database including the names of all the employees at the announced location of the called party. The voice recognition system then selects the name in the loaded database that most closely matches the name of the called party. The selected name is announced to the calling party and the call is automatically connected to the line associated with the telephone number assigned to the called party unless the calling party interrupts the process by saying, "No." Thereafter the voice recognition system releases from the call.

Because the active voice recognition vocabulary set associated with each ESN or location contains only a portion of the total number of employees of the corporation or company, the WER is much lower than it would be if the complete employee directory was loaded in the database.

The actual decrease in the corporate wide WER (C_WER) is contingent upon how evenly the employees are spread over the different sites. In the best case where the employees are evenly distributed in each site, C_WER will decrease according to the following relation.

$$C\_WER = \frac{WER}{\sqrt[4]{n}}$$

where: n is the number of sites.

In the worst case, where there is only one employee in each site, except for one site which holds all of the remaining employees, there will be a negligible decrease in the C_WER.

$$C\_WER \propto \sqrt[4]{(m-n)}$$

where: m is the number of employees in the company.

$$C\_WER \approx WER$$

for: m >> n

In a similar way that ESN information is used by the speech recognition system to dynamically load the active vocabulary set, the ESN information can also be used by the speech recognition system to select the appropriate acoustic model set. Speech recognition systems use previously collected speech samples to serve as reference templates against which new spoken speech samples are matched for classification. Statistical pattern recognition techniques are used to match new speech samples against reference templates to determine the closest match. These reference templates are refereed to as acoustic models in the speech recognition system. Acoustic models may vary according to the regional accent and subsequently according to ESN locations. The speech recognition system can use site-specific acoustic models that are dynamically loaded based on the ESN information presented at the time of the call. Having site-specific acoustic models will also decrease the WER of the system.

The following specification illustrates an example of the NORTEL Speech Directory Calling Service. The state diagram shown in FIGURES 3 and 4 describes the user interface that users of the service experience and is not meant as an implementation specification. Some parts of the system, such as error recovery and instructions have been omitted.

In the description that follows, the use of *italics* denotes system state and the use of a dollar sign symbol denotes a parameter.

Description of the States in Alphabetical Order:

Cancel:

```
Play Who
go to Listening Timeout
```

Idle:

```
/* Go to Idle  anytime a user hangs up */
On an incoming call
Get ESN information
Set $Location based on ESN information
go to Listening Timeout
```

KeyWord Handler:

```
Case
     Service Locations:   go to Service Location
     Receptionist:        go to Transfer Receptionist
     Cancel:              go to Cancel
End Case
```

Known Loc:

```
Set $Location to $RecognizedWord
Play $Location
Play EmployeeName
go to Listening Timeout
```

Listening Timeout:

```
Listen for $Timeout
If the user speaks
      go to Speech
Else
      go to Prompt
```

```
Loc Handler:
If $Location is known location
     go to Known Loc
Else
     go to Unknown Loc

Prompt:

Case (state before Listening Timeout )
     Idle:
          Play Who
          go to Listening Timeout

     The rest of the states:
          When $Timeout expires on the first two times
               Play TimeoutHelp.$Location
               $Timeout = 4 sec
               go to Listening Timeout
          When $Timeout expires on the third time
               Play Difficulties
               go to Transfer Receptionist
End Case

Service Location:
Play ServiceAvailable
Play $Location list
Play Who
go to Listening Timeout

Speech:
Load the active vocabulary set from $Location
Get $RecognizedWord from Speech Recognizer
```

```
Case ($RecognizedWord)
     Rejection:      go to Rejection Handler
     $Name:          go to Transfer Call
     $Location:      go to Loc Handler
     Key Word:       go to KeyWord Handler
End Case
```

Transfer Call:

Database Lookup for Employee Phone Number
Transfer the call
go to *Idle*

Transfer Receptionist :

Play TransferReceptionist
Transfer the call to the receptionist
go to *Idle*

Unknown Loc:

Play NotServiced.$Location
go to *Listening Timeout*

Index of the Prerecorded Prompts in Alphabetical Order :

Calling:

Calling $Name?

Difficulties:

The system is having difficulties with your request. Transferring to a receptionist.

EmployeeName:

Employee name?

NotServiced:

This service is not available in $Location. Choose another location or for a list of serviced ESN locations, say "Service Locations".

ServiceAvailable:

This service is available for the following Nortel/BNR locations: $Location list.

TransferReceptionist:

Transferring to a receptionist.

Who:

Who would you like to call?

A specific embodiment of the invention has been disclosed and illustrated. It will be apparent to one skilled in the art that various changes in methodology and/or approach can be made without departing from the spirit and scope of this invention as defined in the appended claims.

I CLAIM:

1. A telephone network including:

a plurality of telephone exchanges each for serving a plurality of telephone instruments and each being interconnected with at least one other of the telephone exchanges, for providing telephone communications between telephone users associated with the telephone instruments; and

a simulated telephone operator apparatus for receiving a voiced speech request from a user for connection to another of the telephone users and translating said request into a directory number for use by one of the telephone exchanges in accordance with a speech recognition algorithm and an active speech recognition vocabulary selected in accordance with the origin of the request.

2. A simulated telephone operator server for a telephone network comprising:

means for storing voice utterances of a calling party telephone user;

means responsive to location information in association with the telephone user for selecting an active speech recognition vocabulary;

speech detection means for processing the stored voice utterances in accordance with a speech recognition algorithm and said active speech recognition vocabulary;

directory lookup means for identifying a directory listing of a called party corresponding to a result of said processing by the speech detection means; and

means for transmitting the directory listing to a telephone exchange serving said called party.

3. A simulated telephone operator server as defined in claim 2, wherein the directory lookup means defaults to identification by a telephone attendant directory listing in the event of there being no called party directory listing

corresponding to the result of said processing by the speech detection means.

4. A telephone exchange comprising:

a plurality of ports for serving a plurality of telephone users' telephone instruments via telephone lines;

a trunk facility for connection to another telephone exchange;

a switching network for connecting and disconnecting the telephone instruments;

a controller means for causing a newly OFF HOOK telephone instrument to be coupled via the switching network with a solicitation signal, and subsequently for being responsive to a telephone number received in association with the newly OFF HOOK telephone instrument for completing a telephone call via the switching network;

an originating register means for storing voice band signals received from the newly OFF HOOK telephone instrument via the switching network;

means for detecting digits represented by frequency signals, within the stored voice band signals, in accordance with a standard for key pad dialed telephone numbers, and for transmitting detecting digits to the call controller;

a simulated telephone operator apparatus for receiving and translating voice band signals in accordance with a speech recognition algorithm and an active speech recognition vocabulary selected in accordance with the origin of the voice band signals into a directory number for use by the controller means; and

an interface facility for transmitting the stored voice band signals via the switching network to the simulated telephone operator server apparatus in an event wherein the voice band signals did not include a key pad dialed digit.

5. A telephone exchange as defined in claim 4, wherein the call controller means is operative to cause the interface means to transmit said stored voice band signals via the switching network to the simulated telephone operator server apparatus in an event wherein the voice band signals included a key pad dialed digit designating the simulated telephone operator apparatus.

6. A simulated telephone operator apparatus for receiving a user voiced speech request for connection to another user of a telephone network and translating said request into a directory number for use by an automatic telephone exchange, in accordance with a speech recognition algorithm and an active speech recognition vocabulary selected in accordance with the origin of the request.

7. A method for detecting a calling telephone user voiced speech request for connection to another telephone user via an automatic telephone exchange comprising:

storing a plurality of speech recognition vocabularies in association with geographic locations of users;

receiving the voiced speech request and information as to the geographic location of the user having voiced the speech request from the automatic telephone exchange;

selecting an active speech recognition vocabulary in accordance with the information as to the geographic location of the user; and

in accordance with a speech recognition algorithm and the selected active speech recognition vocabulary, translating the received request into a directory number for use by the automatic telephone exchange in setting up a telephone connection between the calling telephone user and said another telephone user.

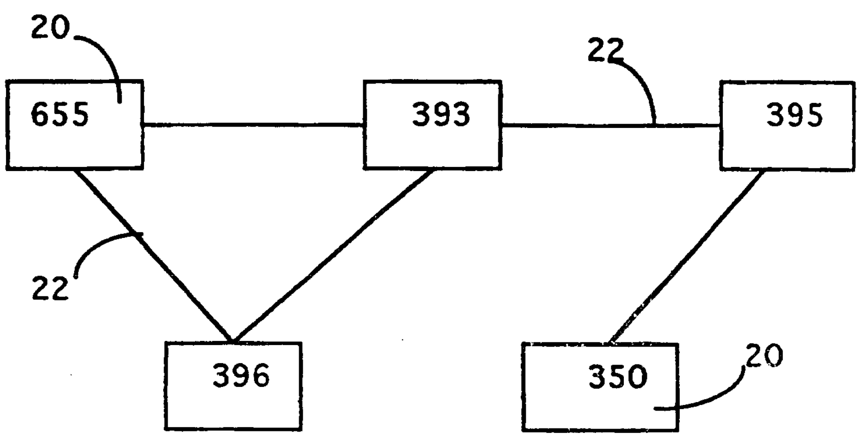

FIGURE 1

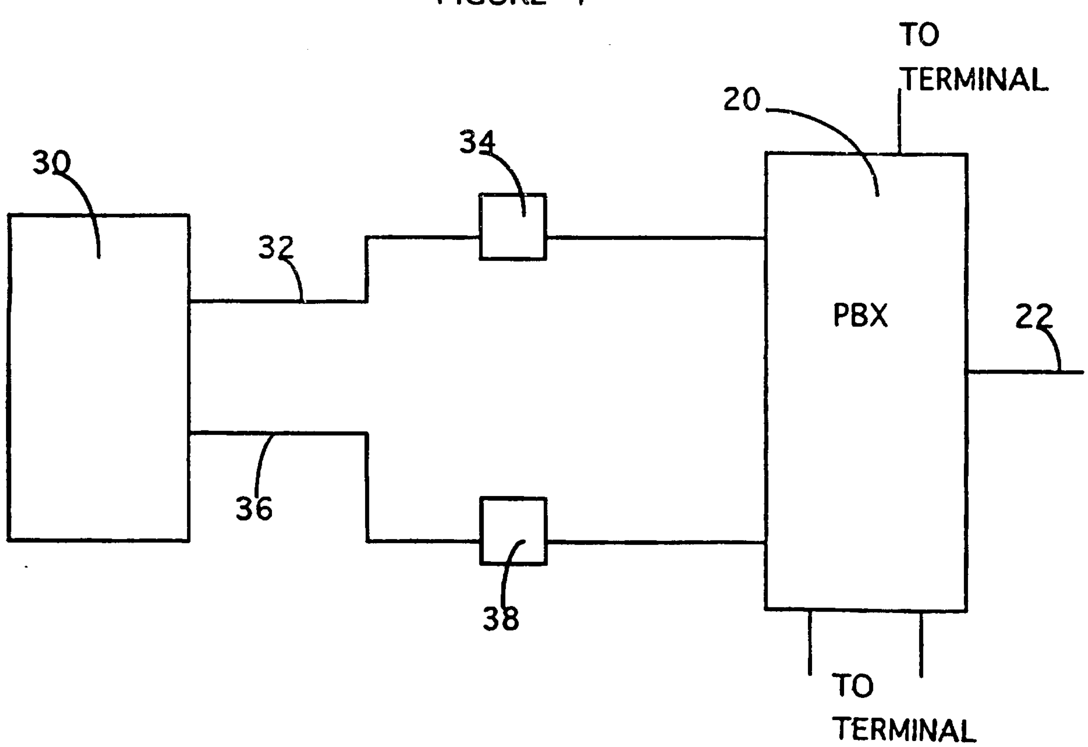

FIGURE 2

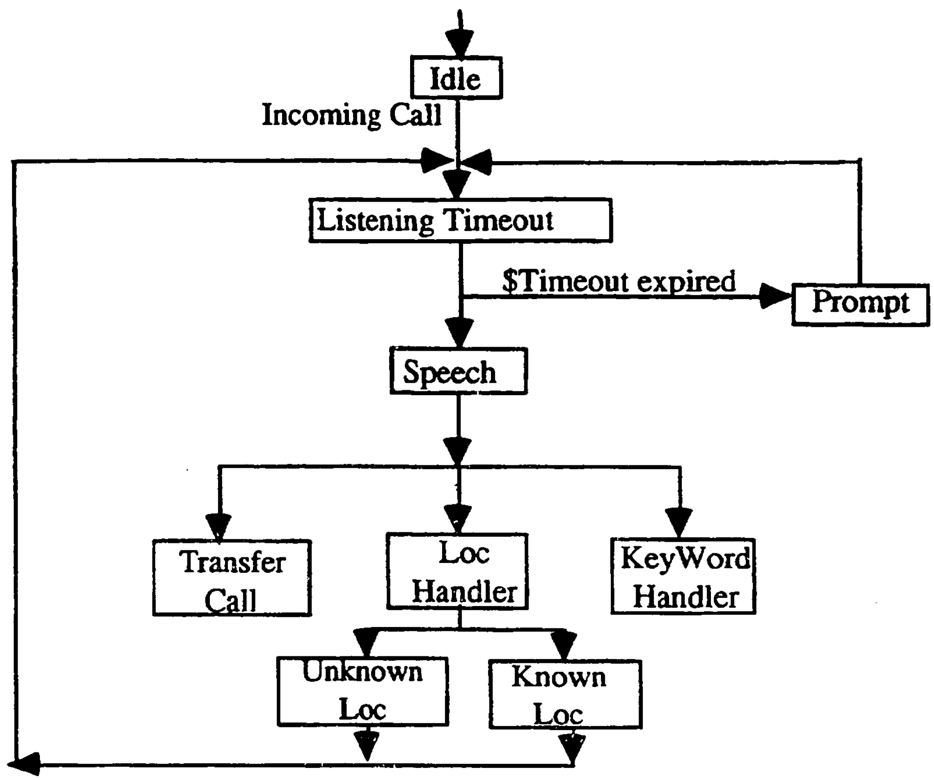

FIGURE 3

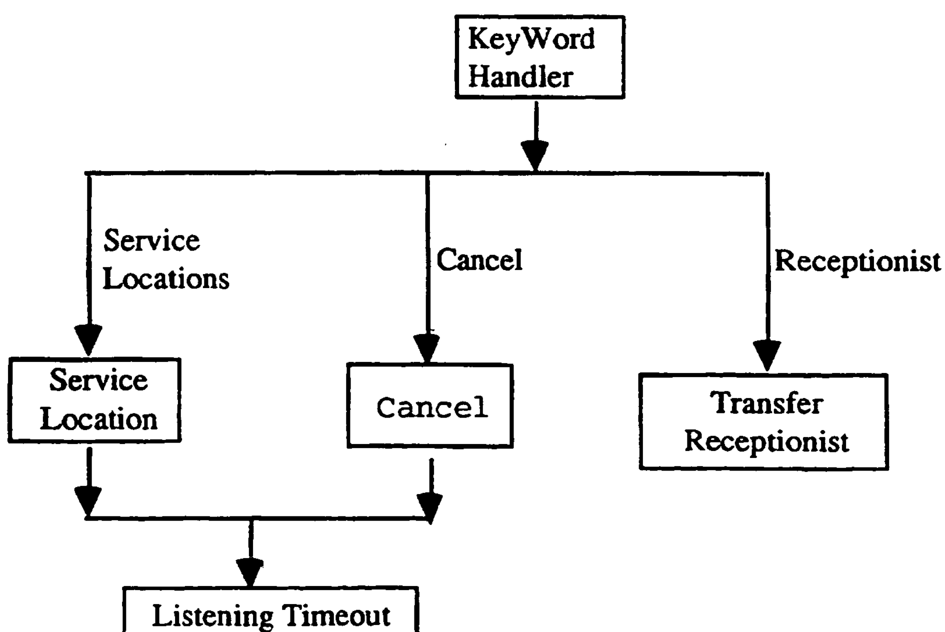

FIGURE 4

INTERNATIONAL SEARCH REPORT

International Application No
PCT/CA 97/00008

A. CLASSIFICATION OF SUBJECT MATTER

H 04 M 1/27,G 10 L 5/06

According to International Patent Classification (IPC) or to both national classification and IPC 6

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

H 04 M,G 10 L,H 04 Q

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

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

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	EP, A, 0 105 441 (SIEMENS) 18 April 1984 (18.04.84), page 1, line 1 - page 3, line 32; claim 1. --	1,2,4, 6,7
A	US, A, 5 165 095 (BORCHERDING) 17 November 1992 (11.11.92), abstract; column 1, line 6 - column 2, line 37; column 3, line 22 - column 6, line 27; fig. 1,2 (cited in the application). --	1,2,4, 6,7
A	EP, A, 0 568 979 (SONY CORPORATION) 10 November 1993 (10.11.93), abstract; column 1, lines	1,2,4, 6,7

[X] Further documents are listed in the continuation of box C. [ ] Patent family members are listed in annex.

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Date of the actual completion of the international search	Date of mailing of the international search report
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-2-

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
	43-50. --	
A	US, A, 5 390 278 (GUPTA et al.) 14 February 1995 (14.02.95), abstract (cited in the application). --	1,2,4, 6,7
A	EP, A, 0 045 941 (SIEMENS) 17 February 1982 (17.02.82), page 1, line 7 - page 4, line 9; fig.. --	1,2,4, 6,7
A	US, A, 5 091 947 (ARIYOSHI et al.) 25 February 1992 (25.02.92), abstract; column 1, line 10 - column 3, line 17 (cited in the application). ----	1,2,4, 6,7

ANHANG

zum internationalen Recherchenbericht über die internationale Patentanmeldung Nr.

ANNEX

to the International Search Report to the International Patent Application No.

ANNEXE

au rapport de recherche international relatif à la demande de brevet international n°

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Im Recherchenbericht angeführtes Patentdokument Patent document cited in search report Document de brevet cité dans le rapport de recherche	Datum der Veröffentlichung Publication date Date de publication	Mitglied(er) der Patentfamilie Patent family member(s) Membre(s) de la famille de brevets	Datum der Veröffentlichung Publication date Date de publication
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		DE A1 [illegible]	05-04-84
		DE C0 3377193	28-07-88
		EP B1 105441	22-06-88
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		EP A3 427680	[illegible]
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		DE C2 3819178	20-06-91

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(81) De ites: AL, AU, BA, BB, BG, BR, CA, CN, CU, E, GH, HU, ID, IL, IS, JP, KP, KR, LC, LK, LR, L. IG, MK, MN, MX, NO, NZ, PL, RO, SG, SI, SK, SL, IT, UA, US, UZ, VN, YU, ARIPO patent (GH, KE, L, MW, SD, SZ, UG, ZW), Eurasian patent (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European patent (AT, BE, CH, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE), OAPI patent (BF, BJ, CF, CG, CI, CM, GA, GN, ML, MR, NE, SN, TD, TG).

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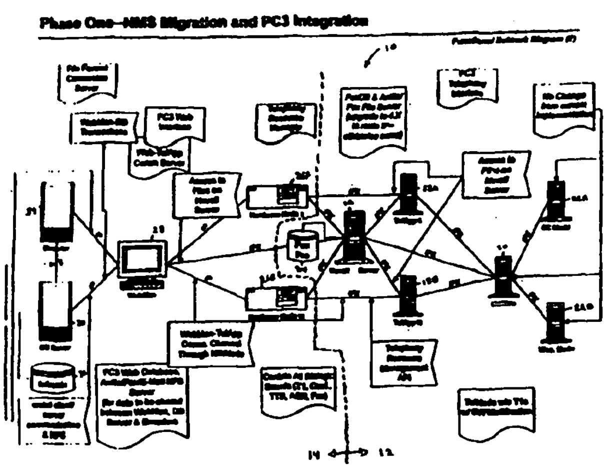
(54) Title: SYSTEM FOR INTEGRATED MANAGEMENT OF MESSAGING AND COMMUNICATIONS



(57) Abstract

Systems for providing to a user an integrated interface for accessing telephony- and computer network-based communications resources. A user may access the systems via telephone or computer (via modem or Internet). The user may access messages addressed to him or her in various media, such as e-mail, voice mail, fax, etc. The user may respond to these messages in the same media, or different media, including real-time communications such as direct telephony or conference calling. The interface may also allow the user to present others with a web page which displays information selected by the user. The system may consist of a stand alone system which interfaces with the Internet and the telephony network. The system may also be integrated with existing telephony-based communication management infrastructures.

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CA	Canada	IT	Italy	MX	Mexico	UZ	Uzbekistan
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SYSTEM FOR INTEGRATED MANAGEMENT OF MESSAGING AND COMMUNICATIONS

This application claims priority to U.S. Provisional Application Serial No. 60/031,301, filed November 18, 1996.

Background of the Invention

The present invention relates generally to communication technologies such as electronic mail (e-mail), voice mail, paging, conference calling and facsimile (fax) mail messaging. Particularly, the present invention relates to systems for integrating conventional telephony-based communication management services and capabilities with Internet-based communication management applications and capabilities.

Current communication technology utilizes several media, such as e-mail, voice, voice messaging, paging, video, video messaging, conference calling and facsimile. Such messages and media may be accessed through a wide variety of interfaces such as pagers, wireline telephones, cellular and PCS telephones and computers (through direct dial modem, network or Internet access). With such a wide array of options in both medium and interface, systems have been provided for centralizing a user's communications in order to simplify access to these communications while at home, in the office or traveling. For example, U.S. Patent Application No. 08/665,564, filed June 18, 1996 entitled "System For Integrated Electronic Communications" (James Arthur Kitchen, David Gregory Smith, Leonard A. DeNittis, Kyle S. Brown, Michael S. Finney, Thomas Francis Johnson, III, Steve Feinstein, and Stephen J. H. Owens, inventors) which is incorporated herein in its entirety by this reference, describes a "universal mailbox" which comprises a fax server, an e-mail server and a voice mail server. Each server receives and stores messages for a given user. The messages can thereafter be retrieved by the user at his or her convenience, using only a telephone interface. The servers are networked together and may be accessed from a single telephone interface; cross media capability allows e-mail messages, for instance, to be converted to voice. Moreover, the user is able to route messages to desired destinations; for instance, the user can access the system, designate a nearby telephone number, and have his or her communications sent to devices at that number. Thus, communications directed to a user are brought together in one place. Nonetheless, the means for accessing the communications are restricted to telephones and, in some cases, direct dial modem hookups.

Universal mailboxes are limited in that they typically do not support real-time communications. For instance, upon receiving a message from a universal mailbox, a user may wish to contact one or more persons, either sequentially or simultaneously. In the current environment, the user must sever his or her connection with the universal mailbox and then individually dial each person he or she wishes to contact. This can be time consuming and inefficient. Performing a conference call can be, paradoxically, even more difficult, requiring the assistance of an operator for dealing with complex command sets through a PBX. Thus, it would be desirable to provide a user-friendly interface which allows a user to access the features of a universal mailbox and engage in real-time communications without having to leave the mailbox environment.

Accessing universal mailboxes, even through computers, also provides limited flexibility because specialized software is required. Thus, users must learn how to operate the software, which may be time consuming. Also, as new versions of the software are released, compatibility problems between the users' platforms and the universal mailbox arise. These problems may be related to inconsistent software compatibility across the multitude of platforms employed by users or compatibility problems between the user and the mailbox as different users use different versions of the access software. Thus, it would be desirable to provide a network-based interface that relies on global standards, such as World Wide Web (WWW) protocols, so that each user need not have specialized hardware or software in order to access an use the system. Thus, a user could employ any browser or Internet protocol to establish communications with and use the system to manage his or her communications and messaging.

Communications delivered to a user may be multiple messages delivered over various media from one sender or many. Thus, a user may need to respond to various senders' e-mails, voice messages and faxes. Keeping track of the e-mail address, fax number and voice number of even a single person can be very difficult, particularly when one is away from the office. This problem is compounded when a user must maintain communications over multiple media with multiple persons. Thus, it would be desirable to provide an interface which allows a user to direct messages and communications to any number of individuals across any number of media while relying on information stored in the system to direct the messages and communications appropriately.

Furthermore, while a message may be received in one medium, such as e-mail, it may be desirable to receive it and/or direct a response in a different medium, such as voice mail. This may be because the individual to whom the message is addressed may prefer one medium over another; it may simply be because at that point in time, the user has no computer handy. Also, if the same message is to be delivered to several individuals, the user may wish to deliver the message via e-mail to some, fax to others and voice mail to the remainder. Thus, it would be desirable to provide functionality for directing a single message to one or more individuals in multiple formats.

Mass communications, such as fax broadcasting and conference calling, while particularly useful, are often complicated and cumbersome to implement. Thus it would be desirable to provide a communications management interface that allows mass communications to be easily and efficiently implemented.

While the Internet has emerged as a source of information and communications for many, the current universal mailboxes do not provide accessibility to the Internet. It would be desirable to provide a system which allows users to access their communications via the Internet, and, when possible, take advantage of the lower communications costs associated with the Internet to communicate with others. For instance, in sending out multiple responses to a message, a user may wish to attach a voice file to an e-mail message for delivery over the Internet, rather than pay the cost of multiple traditional telephone hookups. Furthermore, it would be desirable to allow others to contact the user via the Internet. This would allow the user to present to others considerably more information than would be traditionally available, for instance, on a voice mail system.

Existing systems provide the user only with information specifically addressed to the user. Thus, in order to access generally available information, such as news, sports, weather, financial, or other information, a user must subscribe to services which will identify and forward that information to the user's address. Much of this information is available, however, from publicly accessible areas, such as Internet World Wide Web (WWW or Web) pages. Thus, it would be desirable to allow the user to customize the interface such that he or she may easily access such sources of information, either using a computer or a telephone interface.

Although a computer is often the simplest and most powerful means of accessing information, situations may arise in which access to a computer may be limited or non-existent. Thus, it would be desirable to provide an interface which, while accessible via computer, is still accessible from other interface units such as telephones.

Current telephony-based universal mailbox systems are relatively mature technologies and therefore have many desirable features that are established and fully operational. Thus, while it may be desirable to provide standalone systems for integrated computer/telephony communications, it may also be desirable to provide computer communication systems which can be added to existing computerized telephony communications systems in order to take advantage of the mature technology embedded in the exiting telephony systems.

Summary of the Invention

The present invention provides network-based communication and information systems that leverage existing computer telephony information processing, storage and retrieval platforms with the power and economies of the Internet and private data networks. The systems make communication and access to information more efficient and save users time and money. In a first embodiment, certain communications hardware and features of a computer telephony-based system are off-loaded to a separate hardware node. For instance, fax boards, voice mail boards and text to speech boards are removed from the telephony management node and placed in one or more stand alone computers, such as "hardware nodes." An Internet node is then added to the system. Computer telephony communication management nodes, which can be conventional telnodes that are thin on hardware (the hardware having been placed in the hardware nodes) but on which the applications reside (the "telapps"), and the Internet node then access the hardware nodes as various functionalities are required, using Internet protocols. The applications on the telnodes are therefore accessible to fax boards, T1 boards, and other components of the hardware nodes in a manner that is far more flexible than in conventional computer telephony systems which rely on conventional networking techniques and protocols. For instance, if a user, accessing the system via the Internet wishes to send a fax, a command is sent through the Internet node to the appropriate hardware node, which handles the actual fax process; the hardware node runs the application from the telapp, and the telapp interfaces with the network server for billing and administration purposes. Such billing systems have the capacity, preferably, to interface with remote credit or payment sources, such as those of banks or credit card companies, in real time, in order to transfer funds as required or otherwise maintain proper account status.

Likewise, a user accessing the system through telephony generates a command to the telapp, which commands the hardware node to process and deliver the fax; billing and administration can be handled in conventional ways or as otherwise desired. Both the telapps and the Internet node have access to data in the billing information systems concerning customer accounts, preferences, contact information, etc. in order to process and bill the transaction. Common or distributed databases may be provided to serve this need. If distributed databases are employed, i.e., one database for the Internet node and one for the telapp, it may be desirable to have the information mirrored from one database to the other to ensure synchronization of customer data. Either the telephony network manager or the Internet network manager may be responsible for synchronizing the distributed databases.

An integrated computer/telephony communications system may also be provided which does not utilize existing telephony-based communications management systems infrastructure. In such a system, a single database may be used. One or more hardware nodes can be provided which use Internet protocols to communicate with applications functionality for flexibility, and a single network management system can be utilized. The network management system draws on a plurality of applications which in turn access hardware elements on the hardware node to provide system functionality. In other words, a user can access the network manager via either telephony or the Internet. User commands to the network manager access applications stored therein or on a separate applications node. The applications then access hardware (such as fax boards TTS boards, voicemail boards, etc.) on the hardware node and the user's commands are implemented.

Regardless of which embodiment is used, the combination of computer and telephony communications provides a user with powerful tools and features for managing his or her communications needs. Because the computer and telephony networks are integrated, a user no longer need sever his or her connection with the computer network to conduct real-time communications via the telephony network - live phone calls and even conference calls can be placed via the computer interface. Likewise, with the advent of text to speech conversion capabilities, e-mail- and word processing document-to-fax conversion capabilities and other media conversion capabilities, access to communications from a telephone are not limited to voice mail messages. A user can direct e-mails, faxes and other communications to a fax machine of his or her choice. A user can even "listen" to his or her text based communications.

Traditionally, users of computer telephony based servers could instantly retrieve all voice, fax, e-mail, or other information from any telephone. The present invention allows users to retrieve all messages and other information from their desktop PC and also respond or forward messages to other people with a click of their mouse or a simple keystroke from any telephone. Conference calling is so simple that two clicks of the mouse creates a normal telephone conference call in seconds and for a fraction of the cost of using an operator.

Because the system is network-based, users may access advanced computer telephony functionality without purchasing or installing any new hardware or software. The system can be completely platform independent and can be effectively used by customers with literally no computer or Internet skills. This approach gives users the ability to communicate with anyone and access information anytime, anywhere. Whether on the Internet or by any telephone, the system allows users to access their personal on-line phone directory, custom information such as stock reports and news, fully automated conference call and fax capabilities, and the power of "people centered communication".

People centered communication is a data base structure and program which simplifies the process of communicating and accessing information. The system eliminates the need to remember and organize the myriad of fax machine numbers, pagers, cell phones, multiple office and home numbers, and voice mail box addresses that now clutter business communications. Its on-line personal directory allows the user to send voice, fax or page messages to anyone without looking up or dialing any numbers. The on-line directory organizes personal and business contacts into logical groups to which the user can instantly broadcast voice, fax, or pager notifications.

Communications preferences and technical skills vary widely. While some prefer pagers to cell phones, others use e-mail instead of faxes. Every company seems to have a unique system for receiving and responding to business communications. For this reason, systems according to the present invention employ a "cross media messaging" strategy. Cross media messaging means that each person receives communications according to their own preferences - independent of how the message was originally sent or the technology employed. If a message were forwarded to a group of nine people; four might receive a fax at their office, three might receive it as an attachment to an e-mail message (cost savings and convenience), and two would be paged. Likewise, voice messages and other communications or information can be instantly sent to individuals or groups of people using the system's innovative cross media messaging component.

The distributed computer telephony systems according to the present invention utilize Internet or conventional network-based platforms that allow users direct access to existing telecommunication infrastructures without any investment in premise-based hardware or any specialized software. By leveraging the open standards of the Internet, the systems allow users to access and/or configure their personal assistant from any standard Web browser without purchasing or installing any software on their computer, private network, or existing phone system. Thus, systems according to the present invention can be completely platform independent.

Since all data and equipment are resident in a single facility, users can access their information and communication resources from anywhere in the world – from either a PC connected to the Internet or any touch tone telephone. When at the office or connected to the Internet, users can take advantage of the economies of packet-switched communications such as no cost e-mail transport, low cost retrieval of voice and other data, and direct access to their personal on-line directory. Significantly, if a user logs off the Internet or does not even own a computer, he or she is still in touch with all of the system's communication and information resources from any touch tone phone -- worldwide.

For example, a business person on the road or in an airplane can send multiple messages with a single phone call, set up conference calls from a cell phone, or send a 30-person fax broadcast from any telephone or PC within seconds. All of this is made possible through computer telephony platforms according to the present invention which allow the customer to initiate complex telecommunication transactions with great ease.

Systems according to the present invention can include a universal in-box which provides a single mailbox where users can retrieve, listen to, view, download, forward and/or save all of their inbound communications including voice mail, faxes, pages, and e-mail. All messages can be viewed or downloaded from any PC with an Internet connection and can be forwarded to others by e-mail, fax, or voice regardless of whether or not they have Internet access or are customers of the system provider. Since the system is wholly Internet, intranet or network-based, all features of the universal in-box can be accessed from any touch tone telephone.

Systems according to the present invention can also provide simplified conference calling. Whether at a PC or at any touch tone telephone, the user finds setting up conference calls a procedure that is simplified to a few mouse clicks or telephone keystrokes. The user eliminates the need for a conference call operator and takes control of the call. This feature replaces complex, confusing PBX systems. Seconds after requesting a conference call with a group of people, the user's phone rings and the system's automated attendant does the rest. Each party is called by the system which retrieves all required numbers and automatically places all phone calls. Since the customer is actually in control of the system's resources, muting and dropping of callers can be accomplished with a single click of the mouse.

Each system user may have a personal on-line directory which is fully accessible from any touch tone telephone or any PC with an Internet connection. The on-line directory can store and organize contact names, addresses, phone numbers, pager information, notes, and other data. To place a call, send a fax, or page anyone in the on-line directory, the user simply clicks on the person's name from a PC or spells the name using a telephone key pad. Users can easily communicate with anyone in their on-line directory without having to keep track of multiple phone numbers and other contact information.

Each user may receive a personalized public home page on the Internet which is simple to configure and which serves as virtual receptionist to the world. The public page may be turned on or off by the user and may contain a wide array of information which the user chooses to place on the page. The user can select the most convenient way for people to contact them at any time. Real time messages, for example, can be sent to the user by simply clicking on the "contact-me-now" button. Users may publish their name, address, contact information, automated street level maps and/or driving instructions, and direct access telecommunication and e-mail functions. If selected, direct e-mail, pager, and fax links can be placed on the public page giving Internet users a simple way to find the user and contact him or her directly from the Internet. In addition memos and a voice greeting can be placed on the public page making it a personal message board to the world.

Systems according to the present invention can provide all users direct access to news, sports, financial, travel and other custom content directly from their PC or from any telephone. After using a PC to select and customize the content the user desires, the content can be linked directly to the public page and is made available from any touch tone telephone.

Accordingly, an object of the present invention is to provide integrated interfaces, functionality and processes for managing, generating, accessing and manipulating communications and messages across one medium or across various media to and from various persons.

Another object of the present invention is to provide integrated interfaces, functionality and processes for managing, generating, accessing and manipulating communications and messages which can be accessed via telephony or computer.

Another object of the present invention is to provide integrated interfaces, functionality and processes for managing, generating, accessing and manipulating communications and messages which are platform independent.

Another object of the present invention is to provide integrated interfaces, functionality and processes for managing, generating, accessing and manipulating communications and messages which manage communications over multiple media including, but not limited to, telephony networks and the Internet.

Another object of the present invention is to provide integrated interfaces, functionality and processes for managing, generating, accessing and manipulating communications and messages which manage communications over multiple media including, but not limited to, telephony networks and the Internet and may intelligently select one available medium over another to optimize communication efficiency.

Another object of the present invention is to provide integrated interfaces, functionality and processes for managing, generating, accessing and manipulating communications and messages which provide cross media messaging capability for message broadcasts.

Another object of the present invention is to provide integrated interfaces, functionality and processes for managing, generating, accessing and manipulating communications and messages in which the interface and functionality is network-based and platform independent.

Another object of the present invention is to provide integrated interfaces, functionality and processes for managing, generating, accessing and manipulating communications and messages in which the interface and functionality distribute applications and system functionality so that resources may be allocated in the system flexibly and reliably using Internet or other distributed protocol and techniques.

Another object of the present invention is to provide integrated interfaces, functionality and processes for managing, generating, accessing and manipulating communications and messages in which the generation of message broadcasts and conference calls is greatly simplified.

Another object of the present invention is to provide integrated interfaces, functionality and processes for managing, generating, accessing and manipulating communications and messages in which one portion of the interface may be dedicated to an individual user and a second portion of the interface may be accessible to the public, thereby allowing members of the public to contact the user or access information provided by the user.

Another object of the present invention is to provide integrated interfaces, functionality and processes for managing, generating, accessing and manipulating communications and messages in which one portion of the interface may be dedicated to an individual user and the user may customize that portion so that information resources may be accessed directly from that portion.

Another object of the present invention is to provide integrated interfaces, functionality and processes for managing, generating, accessing and manipulating communications and messages which may be implemented in conjunction with existing computer telephony based message server systems.

Another object of the present invention is to provide an Internet or other network based graphical or computer interface to existing or new computer telephony systems.

Another object of the present invention is to provide computer network (Internet or otherwise) access to individuals' telephone numbers, pagers, fax machines and other devices which allow the Internet user to contact individuals in real time using the computer network.

Other objects, features, and advantages of the present invention will become apparent with reference to the remainder of the written portion and the drawings of this document.

Brief Description of the Drawings

FIG. 1 illustrates one embodiment of a system for managing, generating, accessing and manipulating communications and messages in accordance with the present invention.

FIG. 2 illustrates the universal mailbox portion of the system of FIG. 1.

FIG. 3 illustrates the telapp of the system of FIG. 1

FIG. 4 illustrates the interface portion of the system of FIG. 1.

FIG. 5 illustrates the network management system file system organization of the system of FIG. 1.

FIG. 6 illustrates the object, structure, and information flow of the system of FIG. 1.

FIG. 7. illustrates the structure of a second embodiment of a system for managing, generating, accessing and manipulating communications and messages in accordance with the present invention.

FIG. 8 illustrates a process by which the system of FIG. 7 can initiate a telephone action.

FIG. 9 illustrates a process by which the system of FIG. 7 can initiate an Internet action.

FIG. 10 illustrates one form of communication structure and applications development structure of the system of FIG. 7.

FIG. 11 illustrates a user screen interface which can be generated by the systems of FIGS. 1 and 7.

FIG. 12A illustrates an alternative embodiment of the screen of FIGS. 11.

FIG. 12 illustrates the configuration screen of the screen interface of FIG. 11.

FIG. 13 illustrates the personal page configuration screen of the screen interface of FIG. 11.

FIG. 13A illustrates the default configuration screen accessed via the screen of FIG. 13.

FIG. 13B illustrates a billing screen of the interface of the system of FIGS. 1 and 7.

FIG. 13C illustrates the configuration selection screen of the interface of the system of FIGS. 1 and 7.

FIG. 14 illustrates a portion of the screen of FIG. 13 in which WWW hyperlinks may be entered.

FIG. 15 illustrates a portion of the screen of screen 13 in which a personal memo may be entered.

FIG. 16 illustrates the personal page generated by the interface screens illustrated in FIGS. 13-15.

FIG. 17 illustrates a configuration screen of a screen interface which may be generated by the systems of FIGS. 1 and 7 and in which addressing groups may be selected.

FIG. 17A illustrates the group editing option screen of the interface of FIG. 17.

FIG. 17B illustrates the group creation option screen of the interface of FIG. 17.

FIG. 17C illustrates the message creation screen of the interface of FIG. 17.

FIG. 18 shows a screen portion of the screen of FIG. 17 in which the information relating to addressing groups may be edited.

FIG. 19 shows another portion of the screen of FIG. 18.

FIG. 20 illustrates a configuration screen of a screen interface which may be generated by the systems of FIGS. 1 and 7 and in which individual persons' contact information may be selected and edited.

FIG. 20A shows an "Add Name" accessed via the screen of FIG 20.

FIG. 21 illustrates a personal inbox screen of a screen interface which may be generated by the systems of FIG. 1 and 7.

FIG. 22A illustrates a screen displaying an e-mail message accessed via the personal inbox screen of FIG. 21.

FIG. 22B illustrates an alternative embodiment of a screen displaying an e-mail message accessed via the personal inbox screen of FIG. 21.

FIG. 23A illustrates a screen showing a fax message accessed via the personal inbox of FIG. 21.

FIG. 23B illustrates an alternative embodiment of a screen showing a fax message accessed via the personal inbox of FIG. 21.

FIG. 24A illustrates a screen showing a voicemail message accessed via the screen FIG. 21

FIG. 24B illustrates an alternative embodiment of a screen showing a voicemail message accessed via the screen FIG. 21

FIG. 25A illustrates a contact screen of a screen interface which may be generated by the systems of FIGS. 1 and 7 and by which messages may be distributed to individual or group contacts.

FIG. 25B illustrates a contact screen of a screen interface which may be generated by the systems of FIGS. 1 and 7 and by which messages may be distributed to individual or group contacts with an area in which to enter the recipients address.

FIG. 26 illustrates a contact screen of a screen interface which may be generated by the systems of FIGS. 1 and 7 and by which conference calls may be established.

FIG. 27 illustrates the screen of FIG. 26 once a conference call has been established.

Detailed Description of the Drawings

FIG. 1 shows one embodiment of a system 10 for managing, generating, accessing and manipulating communications and messages in accordance with the present invention. It should not be viewed as limiting, but rather to disclose in detail one particular way for carrying out the invention. The illustrative embodiment illustrated in FIGS. 1-6 reflects a system configuration driven by the desire to take advantage of the hardware and capabilities of existing conventional telephony based messaging systems. As illustrated in FIGS. 7-10, another embodiment of a system according to the present invention may be provided which does not rely on conventional or existing infrastructure.

Referring to FIG. 1, telephony subsystem 12 of system 10 comprises many of the components of a conventional telephony-based universal mailbox system. Internet subsystem 14 of system 10 comprises the components for providing the integrated interface through which telephony subsystem 12 may be accessed and information stored therein manipulated.

FIG. 2 illustrates telephony subsystem 12 of the system of FIG. 1. Telephony subsystem 12 comprises server 16, telephony application nodes (telapps) 18A-N, network manager 20, special purpose telephony nodes (telnodes) 22A-N and database 24. In the illustrative embodiment, server 16 is a Novell Server, consequently, communication among the elements of telephony subsystem 12 is via SPX, IPX or other desirable network protocol. Of course other networking standards and components could be employed, such as Microsoft NT or Unix based systems, in which case other protocols, including IP, could be used. Server 16, in combination with database 24, stores all of the messages delivered to any users as well as all information relating to user accounts for configuration control and billing purposes.

In the present invention, in order to provide Internet functionality to existing computer telephony systems, the inventors have replaced certain conventional telnodes with hardware nodes and telephone application nodes, or "Telapps." A primary reason for this approach is to provide a first platform which is primarily hardware-centric and a second in which the applications are made available to the hardware node across an Internet protocol or other suitable transfer protocol network. Telapps 18A-N manage Internet based system functionality according to the present invention such as voicemail, e-mail, and other media messaging and communications functions. Conventional telnodes 22A-N also continue to manage conventional computer telephony functionality such as provision of voice, e-mail, T1, and fax communications to system 10 via a conventional or other telephone switch.

As illustrated in FIGS. 1 and 3, hardware nodes 26A-N provide standalone hardware units. For instance, all communications boards, such as digital network interface (T1), fax, text-to-speech (TTS), conference, etc., may be transferred to hardware node 26A. Thus, telapps 18A-N are platforms for applications which access the hardware elements as necessary to implement functionality, such as control of call flow, TTS processing, etc. Telapps 18A - N communicates with hardware nodes 26A - N, for instance, through a suitable high level API for controlling call flow and telephony resources. A user communicates with telapps 18A-N to access and manipulate communications. Telapps 18A-N, upon receiving user commands, access applications which in turn access hardware on hardware nodes 26A-N to perform the desired functionality. All transactions are tracked from, and relevant information for each transaction is provided by server 16. Network manager 20 coordinates the interaction between telapps 18A-N, server 16 and, when used, specialized nodes 22A-N.

Hardware nodes 26A-N of the illustrative embodiment are Pentium® based SCO/UNIXWARE units with rack-mount passive backplane systems (such as those with an SCSA bus) populated with T1, voice, conference, fax, TTS, voice recording and voice recognition boards. The use of SCO/UNIXWARE is desirable in the illustrative embodiment because it allows compatibility between the UNIX network file system (NFS) used in Internet subsystem 14 and the Novell proprietary file system (which supports NFS) employed in telephony subsystem 12. Hardware node 26A will be completely configurable based on the boards that are populated therein.

A benefit of dividing the resources among telapps 18A - N and hardware nodes 26A - N is to allow the platforms to become independent of the DOS environment, be supported by industry standards such as UNIX, and increase scalability. The functionality split also serves to eliminate the need to transfer calls through the switch and coordinate function of the net manager 20 eliminating cost, time consumption and inefficiency associated with providing conference, fax, TTS, and other features during a call. The functionality split also provides a method for allowing a conventional telephony subsystem to communicate with an Internet-based subsystem thereby enabling the interface according to the present invention. In other words, an existing system, with established telephony capability, a customer base and an established database can easily be integrated with a computer communications network, such as the Internet, without losing the benefit of those valuable assets.

FIGS. 4 and 5 illustrate Internet subsystem 14 of system 10, which comprises web manager 28, DB server 30, database 32 and encoder 34. All communications within Internet subsystem 14 is via IP protocol. Web manager 28 communicates with hardware nodes 26A-N via IP protocol. Web manager 28 communicates with server 16, however, via IPX protocol since server 16 is on the existing computer telephony Novell net. Web manager 28 acts as the conduit for all communications between telephony subsystem 12 and any processes which utilize the Internet or communicate via TCP/IP. Web manager 28 also acts to coordinate and synchronize all databases in system 10.

DB server 30 stores all of the information required for operation of the Internet related functions of system 10. Server 30 holds all information necessary for the Internet interface of system 10, including messages, contact data, customer records, etc. Server 30 is kept synchronized by web manager 28 which coordinates with telapps 18A-N. Web manager 28 also keeps the files stored on server 30 in synch with those stored on server 16. Also, server 30 stores all of the messaging information for the messages stored on the system for a user. In other words, although the message itself is stored on server 16, descriptive information regarding the message, such as time received, return address, etc. may be generated and stored on server 30 for access via a graphical interface as described more fully below. Database 32 is a storage system for data stored by server 30 and in the illustrative embodiment comprises an Informix database. Encoder 34 is responsible for converting file formats of information sent out or received so that it is compatible for its ultimate use.

Server 30 serves as centralized storage for subsystem 14. Web manager 28, e-mail server 36 and encoder 34 may refer to files thereon regardless of on which platform they reside. Voice mail, fax and other customer data stored on server 16 may be mirrored on server 30.

Web manager 28 handles the transfer of files from server 16 to server 30 upon command from telapps 18. E-mail server 36 stores new e-mails on server 30 and inserts a message record in database 32, which makes it accessible to the Internet interface process which require the information so that it may be displayed to the user. In order for telapps 18 to access the message, the e-mail directory may be mapped to as a shared directory with server 16. Telapps 18 query web manager 28 in order to query server 30 for a given user's list of e-mail messages.

As illustrated in FIGS. 1 and 6 (which illustrates the object structure of the system of FIG. 1), it can be seen that system 10 comprises two primary elements - an Internet communications system (Internet subsystem 14) and a telephony system (telephony subsystem 12). Both of these elements draw on hardware nodes 26A-N for hardware resources used in performing services required by the user. Web manager 28, in addition to managing the Internet functions, manages the communication and cooperation of the telephony and Internet portions to create the desired functionality of the present invention. Thus, when a message, such as an e-mail message, arrive via the Internet addressed to a user, the message is delivered from web manager 28 to servers 16 and 30. The message itself is stored on server 16, while information identifying the message and its location on server 16 are generated by web manager 28 and stored on server 30. Server 30 has the drive of server 16 mounted via NFS. Thus, database 46 is a single shared disk which is accessed from the IPX/DOS nodes (such as telapp 18) as a Netware Directory and from the IP/UNIX nodes (such as web manager 28) as an NFS mounted directory. The one directory is shared by two different means for different types of clients or nodes. If a user accesses the message via the Internet connection, appropriate information may be forwarded to server 16 and on to net manager 20 which operates node 22A to generate a bill for the user.

The information stored on server 30 may then be displayed to the user via a visual display on a computer connected to web manager 28 over the Internet. In other words, a user may install a web browser on his or her computer. By accessing the Internet and his or her account with system 10 therethrough the user can manipulate his or her messages and generate and deliver new messages. No interface software need be installed on the user's computer as the interface is a network application made available by the web manager and is compatible with web browser protocols. Thus, regardless of what type of computer is used by the user he or she will see a consistent interface (provided, of course, that the computer has a web browser installed thereon). An illustrative example of an interface of the present invention is discussed below with reference to FIGS. 11-27. Alternatively, a user may request, via telephone commands to network 12 such as through telapps 18, that the message be converted to voice mail, in which case the telapps will retrieve the message from server 16 and direct it to hardware node 26A where it is converted per the user's command. The message, in its new form is then delivered to the user over the telephone line.

In another example transaction, a fax message may be delivered to the user via a telephone line. As before, the message itself is stored on server 16, while identifying information is stored on server 30. If the user commands the fax to be delivered via telephony, it is retrieved by the telapps 18 which forward it to hardware node 26, whereupon it is sent out over the telephony network to the desired address (phone number). If the user wishes to look at the fax via the Internet interface, telapps 18 retrieve the message and forward it to hardware node 26A which converts the fax to a bitmap image which can be displayed as desired.

A user may also perform mass communications through one or more media. For example, a user, accessing the system via the Internet, may wish to deliver a text message to one or more recipients. The user creates the message in the interface program and addresses the message to various recipients. The user can indicate which medium is to be used for each recipient. Assuming that three recipients are selected, one to receive a voicemail version, one to receive an e-mail version and one to receive a fax version, the process is implemented as follows. The e-mail version is sent directly through the web to the intended recipient and a marker is forwarded to NFS share database 46. When database 46 and database 24 are synchronized by web manager 28, the transaction is logged to the billing system for billing to the user. Both the voice mail and fax messages are sent, in text form, to telapps 18 which accesses hardware nodes 26 to convert the text to a bitmap (for faxing) and to speech (for voice mailing). Once the files are converted, telapps 18 direct hardware nodes 26 to deliver the messages via the telephony network. The transactions are logged in database 24 for billing purposes.

If the user, accessing system 10 through the Internet, wishes to conduct a conference call, he or she accesses a portion of the interface which allows the selection of intended conferees. Once the conferees are selected, web manager 28 notifies telapps 18 of the need for conference telephony. Telapps 18 access the conference call hardware, which may be resident on hardware node 26 or on a specialized node 22. The hardware establishes the connection with each of the conferees and the connection is routed to the user's computer (which is configured to allow audio communication) and the conference call takes place. Again, the transaction is logged to database 24 for billing purposes.

These features may be accessed via telephone as well. For most functions, none of the commands or messages need be routed through any portion of Internet subsystem 14. Of course, any information routed to an Internet address or recipient will be routed appropriately. For instance, a user may wish to deliver a fax to a recipient's e-mail address. The fax is sent through the telephony system, converted to an e-mail attachment and forwarded by web manager 28 to the recipient's e-mail address. Conversion may be performed by encoder 34 which encodes the fax to its final form, such as a GIF image, text (using optical character rccognition), etc., and e-mail server 36 or web manager 28 may attach the file (in its final format) and send it as an e-mail. The same process is also used for encoding and e-mail attaching audio messages.

In this illustrative embodiment, all transactions are routed through telapps 18A-N, even if they are wholly Internet transactions, primarily because billing functions are performed in telephony subsystem 12.

As illustrated in FIGS 7-10, in an alternative illustrative embodiment, system 10 may be implemented with systems independent of conventional telephony systems.

System client 38, or service, uses library 40 which resides on the local host on which the application runs. A service may be transient or permanent. Transient services are applications which are invoked to perform some task, and which terminate on completion of that task. Examples of such services would be a CGI script that faxes the input from a HTML form to a specific number. Permanent services are services that are always running, and which respond to events. Those events invoke some sort of action. Examples of such services include a service which answers calls on a given range of DNIS values. This application can handle from 0 to the maximum number of channels worth of calls. The service starts up, initializes and registers itself, and waits for an event - an incoming call. Another example would be a process that handles all audio file encoding/conversion for a machine. This process waits for an event, an audio file that requires encoding, and responds by encoding the file.

Services may or may not require the telephony resources managed by server 42; they may simply perform a task for other services. And it is for this purpose that services may communicate with one another through the computer telephony interface ("CT/I") API with *peer messages*.

Any desired Internet or other distributed transfer protocol (here, simply for shorthand reference called, "Computer Telephony Transfer Protocol" or "CTTP") allows access of the library 40 functions by the Server 42 and vice versa. CTTP is an application layer protocol which can use TCP/IP as transport. The protocol itself need not be defined since the API allows for the abstraction of all communication between the CT/I Service and the CT/I Server.

Each service 38 can have a distinct connection to server 42. Server 42 may have connections to multiple client applications, and a service may even connect to multiple servers 42. Anything having to do with the actual connection to server 42 is hidden from service 38 through API 40.

The client/server relationship allows for very complex and diverse computer telephony related applications for intranets as well as for the Internet. The model allows for resource sharing between servers on a network. It gives the application developer an abstract view of their servers 42 (as well as other servers) with disparate hardware and operating systems, as resources providing specific function and generating specific events

- all handled through a single API. It allows applications 38 (the services) to run locally or remotely.

Under this architecture server 42 is highly scaleable in that it can be a complete system unto itself for small operations - it could contain a database, web services, and call flow applications, or it may be several servers 42, all performing no other function other than managing telephony resources and outsourcing the actual applications to other machines on the network.

Example of Services 38 include:

- An application that waits for an incoming fax on a predetermined 800 number, and which encodes the fax from the .TIF image format to .GIF and posts it to the web.
- A CGI script that places an outbound call, page and fax notifying the recipient of the fields filled out in the HTML form.
- An E-Mail reader application with the capability to respond with an audio message to the sender.
- A telephone web browser which allows the user to select a URL, and which reads the corresponding document using text to speech functionality.
- An audio encoding process which waits for a peer message that transfers the audio file, and then encodes the audio file into the appropriate format.
- A web based application generator written in Java or other applet inclusive language that allows a user to map out a call flow and then assign that call flow to run on a specific 800 number.

API 40 allows services 38 to communicate with one another using peer messages. This allows distributed processing which offers several advantages:

Applications which control call flow can send messages to another service (a helper service) to perform a function that may block (a database call for example). This keeps the call flow application in control of the call at all times, and greatly reduces latency.

Operations which take a lot of processing power or which need to run on specific operating systems can be accessed through peer messaging from a call flow application running on separate machines.

Users may also extend API 40 to include their own functions. These functions are wrappers for peer messaging, which allow the user to provide their own API functions and events to their services. An example of such an extension would be the addition of a function cti_amail(email_address, audio_file) which mime encodes the audio file and attaches it to an e-mail message which is sent to the given e-mail address. This function would receive an event indicating, at a minimum, success or failure. This function may actually do nothing more than send a peer message to the delivery service and deliver the responding peer message as an event to the calling service. In fact, standard services may also be added to the API to provide functions such as audio and graphic file encoding/conversion.

The CT/I Server program forks into a Communications Server and a computer telephony "CT" Resource Manager. The Communications Server binds to a port and listens for incoming connections. When a client program (a CT/I service) connects to the port, the Communications Server forks off a relay process that attaches the client to the CT/I Resource Manager through a pipe. When the CT/I service quits, the forked relay process also quits and returns the pipe to the Communications Server for use by another service.

The CT Resource Manager process contains the main event loop. Events from the telephony hardware spur signals to the client programs, and commands from the client programs trigger function calls to the CT hardware through the vendor specific API.

The CT/I Server serves as the glue binding these two interfaces (Communications Server and the CT Resource Manager). The CT/I API (ti_* functions and events) is a hardware independent API through which the CT/I service can control call flow, fax, page, etc. Each distinct CT platform (Dialogic, NMS, conversant, etc.) has its own set of corresponding hw_* events and related events. These functions and events correspond directly to the similarly named CT/I API functions.

The CT/I API is a library which may be written in C or other suitable language that can be compiled into a client program on any machine on the Internet, and from the remote client program can attach to the CT/I server and handle a multitude of calls. The CT/I API functions are designed to be easy to use, so that client application programs can be written by programmers without specific knowledge of the intrinsics of the specific CT platform and API. The ti_* functions are asynchronous and will not block, so a client program can handle multiple connections at once. Some of the CT/I API functions are listed below:

int*ti_initialize(name,host,port,digit_event_mode);

```
char *name;
char *host;
int port;
int digit_event_mode;
```

Description:

ti_initialize initiates a connection to a dispatch server running on a Dialogic™ host. The client must also provide a string identifier name by which the dispatcher will know the client. host is a string which identifies the host machine, for example "dialog.tc.net". port is an integer that specifies the port number on the host machine. digit_event_mode is an integer which specifies how dtmf digit events are reported to the client; a value of 0 puts the connection into immediate single-digit reporting mode. If digit_event_mode is non-zero, the connection is put into digit-block request mode. ti_initialize returns a pointer which is the "handle" used for further communication with the dispatch server.

Events:

event connect, subevent_incoming

event_disconnect, subevent_incoming

event_digit (if digit_event_mode is ON)

struct ti_event*ti_wait_event(conn,timeout);

int *conn;

struct timeval *timeout;

Description:

ti_wait-event waits for a duration of timeout for an event. conn specifies the handle of a dialogic dispatch server (returned by ti_initialize). If an event is obtained, the event structure referenced in the return value will specify thc dctails of the event. If no event was obtained before timeout microseconds elapsed, a NULL pointer is returned. The event structure is composed of a major number, a minor number, and an auxiliary pointer.

Events:

event_ring, event_connect, event_disconnect

subevent_incoming

subevent_outgoing

event_digit

subevent_dtmf,subevent_mf,subevent_pulse

aux = digit string

event_play_done

subevent_EOF

subevent_digit

event_record_done

```
            subevent_maxtime
            subevent_maxsilence
            subevent_digit
        event_prompt_done
            subevent_maxdigits
            subevent_finaldigit
            subevent_maxtime
            subevent_maxsilence
                aux = digit string
        event_peer_message
            aux = message

void ti_event_timeout(conn,channel,major,timeout);
    int *conn;
    int major;
    struct timeval *timeout;
```

Description:

ti_event_timeout requests notification if, after timeout, a specific event is not received. If the event is received before timeout, the event_timeout request is canceled. If the event is not received before timeout, then the specific event is generated with a minor event identifier of subevent_timeout. conn specifies the handle of a dialogic dispatch server (returned by ti_initialize).

Events:

event_*,subevent_timeout

void ti_play(conn,channel,filename,interruptable_by_digit);

int *conn;

int channel;

char *filename;

int interruptable_by_digit;

Description:

ti_play plays the audio file filename over a desired channel. The format of the audio file is automatically determined from either the filename extension, or the first few bytes of the file. conn is the connection handle of a dialogic dispatch server (returned by ti_initialize). If interruptable_by_digit is non-zero, then the play operation can be interrupted by a digit event. ti_stop_io() may be called to interrupt the playing process.

Events:

event_play_done,subevent_EOF

event_play_done,subevent_digit

```
void ti_record(conn,channel,filename,maxtime,maxsilence,int_by_digit);
        int *conn;
        int channel;
        char *filename;
        long maxtime;
        long maxsilence;
        int int_by_digit;
```

Description:

ti_record creates an audio file filename with data from a desired channel. The recording process continues until ti_stop_io() is called, or until maxtime milliseconds have elapsed, or until maxsilence milliseconds have elapsed, or until a digit event occurs (if int_by_digit is nonzero). conn is the connection handle of a dialogic dispatch server (returned by ti_initialize). If maxtime is zero, then the recording time is unlimited. If maxsilence is zero, then silence will not stop the recording. If int_by_digit is zero, then a digit event will not stop the recording.

Events:

event_record_done,subevent_maxtime

event_record_done,subevent_maxsilence

event_record_done,subevent_digit

```
void ti_prompt(conn,channel,filename,interruptable_by_digit),
max_digits,final_digit,initial_timeout,idle_timeout);
        int *conn;
        int channel;
        char *filename;
        int interruptable_by_digit;
        int max_digits;
        int final_digit;
        long initial_timeout;/*milliseconds*/
        long idle_timeout;/*milliseconds*/
```

Description:

ti_prompt plays the audio file filename over a desired channel. The format of the audio file is automatically determined from either the filename extension, or the first few bytes of the file. conn is the connection handle of a dialogic dispatch server (returned by ti_initialize). If interuptable_by_digit is nonzero, then the play operation can be interrupted by a digit event. After the play is completed or interrupted, a digit block of max_digits is expected. If no digits are received after initial_timeout milliseconds, or after a special digit final_digit is received, or after an idle (silence) time of idle_timeout milliseconds after one or more digits were received, a digit block event is returned. ti_stop_io() may be called to interrupt the prompting process.

Events:

event_prompt_done,subevent_maxdigits

event_prompt_done,subevent_finaldigit

event_prompt_done,subevent_maxtime

event_prompt_done,subevent_maxsilence

```
void ti_speak_string(conn,channel,text,interruptable_by_digit);
        int *conn;
        int channel;
        char *text;
        int interruptable_by_digit;
```

Description:

ti_text_to_speech_string converts the string text into a voice signal which is output over channel. conn is the connection handle of a Dialogic dispatch server returned by ti_initialize). If interruptable_by_digit is nonzero, then the speech can be interrupted by a digit event.

```
void ti_speak_file(conn,channel,filename,interruptable_by_digit);
        int *conn;
        int channel;
```

char *filename;

int interruptable_by_digit;

Description:

ti_text_to_speech_file converts the text file filename into a voice signal which is output over channel. conn is the connection handle of a dialogic dispatch server returned by ti_initialize). If interruptable_by_digit is nonzero, then the speech can be interrupted by a digit event.

Events:

event_speak_done

void ti_stop_io(conn,channel);

int *conn;

int.channel;

Description:

ti_stop_io stops all play, record, and get_digit_block operations on channel. conn is the connection handle of a dialogic dispatch server (returned by ti_initialize).

Events:

(none)

```
void ti_hangup(conn,channel);
        int *conn;
        int channel;
```

Description:

ti_hangup disconnects the call on channel. conn is the connection handle of a dialogic dispatch server (returned by ti_initialize).

Events:

event_disconnect,subevent_incoming

event_disconnect,subevent_outgoing

```
void ti_exit(conn);
        int *conn;
```

Description:

ti_exit disconnects the client from the dialogic dispatch server. conn is the connection handle of a dialogic dispatch server (returned by ti_initialize).

Events:

(none)

```
void ti_die(s);
        char *s;
```

Description:

ti_die prints the error message s, prints a perror message, and exits the client program abnormally with an exit value of -1.

Events:

(none)

FIGS. 11-27 illustrate various aspects and functionalities of an illustrative embodiment of an Internet interface presented to a user of system 10. Because of the myriad of combinations of communications media and message distribution options, it is impossible to catalog each and every possible function that may be performed via the present invention. The following description identifies many options for generating, accessing, manipulating and delivering messages and information via system 10; however, many other capabilities and functionalities may be added or included in such a system which take advantage of the integration of telephony and computer network capabilities and function.

The interface may allow a user to provide information about himself or herself (including how s/he may be contacted), information about his or her communications preferences, and information about individuals and groups with whom s/he plans to communicate. These communication preferences may be configured by selecting certain options that define rules to be applied to inbound and outbound communications. For example, the user may request that inbound voice messages be forwarded to an e-mail mailbox for later viewing. Alternatively, the user may request that e-mail messages be forwarded to a voice mailbox for later retrieval. The user may also request pager notification any time s/he receives a new e-mail message. This cross-media messaging and notification feature of the present invention (which are provided by the message servers) automatically perform as part of the forwarding operation any reformatting or conversion of messages that may be required.

Computer access to inbound voice, fax, and e-mail communications is provided through a series of personal, secure Web pages. FIG. 11 shows a screen shot of a login page which can be generated by a system 10 according to the present invention. In this example, the Netscape Navigator Web browser is used to access the Web site through which the features and functionality of the present invention are provided. The Web site may be accessed through a Uniform Resource Locator (URL) such as "http://www.tc.net/pc3" or "http://www.premierecomm.com" 70. At this page, the user may be asked to provide a login identifier (such as a name) 72 and a password 74. Alternatively, as shown in FIG. 11A, the screen of FIG. 11 may include button by which destinations within the interface may be selected.

FIG. 12 shows a screen shot of a configuration management page. Information line 80, identifies the user, his or her location, the date and time, and an indicator of the current task (e.g., Configuration). Tab 82 also indicates the current task. From the main configuration management page, the user may select one of several options as indicated by the buttons on page 84. The user may also select one of the buttons at the bottom of page 86.

FIG. 13 shows a screen shot of a personal contact information page. The user provides contact information such as a telephone number, a pager number, fax number, and e-mail address by completing the fields in form 90. The user may also provide a preferred contact method such as e-mail 92. As shown if FIG. 13A, the user may also configure his or her default contact information. Referring to FIG. 14, the user also may provide in additional fields URLs in field 100. These URLs on "hyperlinks" will provide links to other Web sites through which s/he may be contacted directly. By completing field 110, a user may provide street maps and driving instructions that may be useful to other users (not shown). Referring to FIG. 15, the user may enter a short memo 110 - for example, regarding his or her plans for the day.

The user may also access account information as illustrated in FIG. 13B. As shown in FIG. 13C, a user may have several different configuration for varied personal scenarios, i.e., vacation, travel, etc., which will result in different setups specific to those needs.

In addition to providing the personal contact information (so inbound messages may be routed properly), the URL information, and the memo, the user may indicate whether s/he would like any or all of the information he provided (e.g., name, address, contact information, street maps, driving instructions, etc.) to appear on a "Virtual

Receptionist" Web page. The "Virtual Receptionist" Web page provides personal contact information for an individual user and may be accessed by other users of the present invention or other Internet users so they may contact the individual preferably, by simply selecting a "contact-me-now" button. FIG. 16 shows an example of a "Virtual Receptionist" Web page. The page contains the same personal contact information provided by the user, but reformats it so other users may easily find the information and interact with the page through the buttons and links appearing on the page.

In addition to providing personal contact information, a user of the present invention has a personal on-line directory in which to store and organizc the names, addresses, telephone numbers, pager information, notes, and other data regarding the individuals with whom the user communicates regularly. These individuals may provide their own personal contact information so that they may be contacted according to their own personal preferences. At the on-line directory, an individual may be contacted by simply selecting the individual's name from a list. A user may also define groups of people with whom he communicates frequently. The user may send the same communication to each person in the group simply by selecting the group name from a list of groups. Members of the group who have provided their own personal contact information may then receive the communication according their preferred communication method.

FIG. 17 shows a screen shot of an on-line directory page. Initially, the user is presented with the list of groups 120 s/he has defined previously and the list of individuals (Contacts) 122 with whom s/he communicates. In addition, the user is presented with options 124 for managing the Groups and Contacts. For example, the user may add or delete names for either list (see FIG. 17A for group editing screen), add or remove entire groups (see FIG. for screen configuration for group creation option) or create a message for a selected group (see FIG. 17C for message screen). FIG. 18 shows a screen shot of an "Add Namc" page. After selecting the group to which the user would like to add a name (e.g., Sales Group) 130, a form for adding a new name is presented 132. Referring to FIG. 19, the form 140 continues so contact information for reaching the individual through a variety of methods may be provided (e.g., more than one office number, fax number, home phone number, pager number, e-mail address, cellular phone number, etc.). FIG. 20 shows a screen shot of a Contacts page for individuals. Information regarding individuals with whom the user communicates 150 may be provided by completing an "Add Name" form as shown in FIG. 20A. A user may communicate easily with anyone in his on-line directory without having to keep track of multiple phone numbers and other contact information.

FIG. 21 shows a screen shot of a message center screen or page or "personal inbox." All of a user's communications - regardless of the type - may be accessed through the personal inbox. Preferably, the page lists in field 160 the type, time, date, sender, and subject of each communication that has been addressed to the user. Additional information may be accessed from this page by selecting one of the buttons 162 at the bottom of the page. As the user selects each communication, a communication display page for the individual communication is presented. The communication display page changes based on the type of message.

FIG. 21 show the message center screen of the interface. Message Center is intended to be the default location for the interface. The principle sections of this layout (which are analogous to the layout of other screens of the interface, are as follows:

Sidebar 161 - on the left of the screen is a side bar consisting of three tabs labeled Inbox, Contact, and Configure. By pressing on any one of these three tabs, the program opens up to the corresponding interface for that action.

Work Area 160 - The large center section is work area 160. There is a default work area for each of the primary sections. Within the work area appears content appropriate for the section of the program that is active (i.e., messages in the Inbox, faxes in the fax viewer, names in the Contact Manager, etc.). The Work Area can contain hyperlinks that call other sections of the program depending on current status. The current work area determines the current function bar.

Function Bar 163 - On the top of the browser's display area is the function bar. It contains the Exit button which will shut down the program, and leave the user at an Exist screen (which may show advertising, account status information, or may be a return to Licensee's content). In addition it will contain several buttons whose functions are determined by the current Work Area (i.e., the Inbox Area will have related function buttons such as Forward Reply, etc.). The Function buttons call actions which effect their specific work area's content. The Exit button will be common to all Function bars in all screens of the interface.

Status Bar 164 - Just below the Function bar is the Status Bar - by default it indicates the User's name, the current date, and the time as of the most recent refresh.

The Status bar could also potentially include any combination of the following: Banner Advertising, Stock Ticker, News or Sports wire, License specific content, etc.

Menu Bar 165 - At the bottom of the browser's display area is the Menu bar. The Menu Bar contains link buttons 162 for various Work Areas to be displayed. The Menu bar is determined by the sidebar tax - each of the tree primary sections has only one related Menu bar (i.e., the Menu bar for the Inbox contains buttons for each of the content sections of the Inbox). In addition the Menu Bar contains links to an Audio Tip (a streaming audio file giving a tip or hint for ease of use in the current section) and Help (a separate html section giving detailed help on how to use the interface. Help are available from every Menu bar of all screens of the interface. The buttons corresponding with each work area shall appear highlighted when that button is selected.

The message center is the first primary section. It delivers inbound messages to the user, and allows the user to respond to and re-direct those messages. Messages may be in the following forms.

Fax - A Fax, .tif file converted to one or more .gif files for viewing on the web.

E-mail - Standard E-mail - attachments will appear as download links.

Voice Mail - A voice message that has been digitally encoded and electronically delivered.

The message center is also the section which delivers customized content to the user. For example, content may consist of any one or more of the following:

Sports - linking to a Sports provider such as ESPN.

News - linking to a news provider such as CNN.

Weather - linking to a Weather provider such as The Weather Channel.

Travel - linking to an airline or other Travel service such as Delta

Stocks - linking to NASDAQ or Quote.com

Of course, other information providers may also be included:

Menu Bar 165 for the message center contain links to any of the above content as determined by the users configuration. There is also a link to the incoming Message (Inbox). Each of the content links brings it's corresponding content into the interface frameset, in the frame set aside for the Work Area.

The Messages will be displayed in a Work Area in a table sorted by arrival time of the messages wit the most recent at the bottom. Each Message will occupy a row in the table indicting the following:

1. Type of message 166A - Fax, Voice Mail, or E-mail. This will be indicated by a graphic, which will also be a link to a message viewing Work Area for that particular type.

2. Time of Arrival 166B

3. Date of Arrival 166C

4. Sender 166D (for E-mail) or phone number (for fax)

5. Subject 166E (for E-Mail, # of pages for fax)

6. Select Box 166F- that is used to indicate if a particular message or group of messages is tagged to be acted on by a function.

Function bar 163 is inactive when custom content is displayed. When the Messages are displayed, the following functions are available:

1. Refresh 167A- Checks the server for any new messages and redraws the screen.

2. Select All 167B - Selects all messages to be acted on.

3. Deselect all 167C - De-selects all messages.

4. Download 167D - forwards all selected files as E-mail attachments to an E-mail address pre-configured by the user. A user may, as an option, choose to have all transferred messages, removed from the server.

5. Delete 167E - Deletes all selected files from the server (the files may be expunged immediately, or as part of the Exit process).

6. Exit 167F - exits system.

As shown in FIGS. 22A-B, an e-mail message may be displayed in a scrolling text window 170. The user may then reply to the message, store it, or forward it by selecting one of the buttons at the bottom of the page 172. FIGS 22A-B show screen shots of the E-mail viewer portion of the interface. By clicking on the image used to indicate an E-mail message in the Inbox Message Work Area 160, a new Work Area 170 and Function bar 171 are called. Sidebar 161, Menu 165 and Status bar 164 may remain the same. Alternatively, Menu bar 165 may call additional buttons for accessing and manipulating messages. The new Work Area 170 is the E-mail viewer which contains the header and body of the corresponding E-mail message.

Available Functions for the E-mail Viewer are:

1. **Forward** - this commences a forward mail sequence allowing the user to re-direct the E-mail with an added memo.

2. **Reply** - similar to Forward, but directed specifically to the sender.

3. **Download** - send the current E-mail to a pre-configured E-mail address. The message may then be removed from the server or returned in fax or message.

4. **Delete** - delete selected E-mail from server with Confirmation Dialog (e.g. Are you sure?) return to Messages Work Area.

5. **Next Message** - Jump to next Message in chronological queue (may require shifting to another type of message viewer)

6. **Previous Message** - jump to previous Message in chronological queue (may require shifting to another type of message viewer).

These functions may be provided as buttons on function bar 171 or menu bar 165 or other desirable locations on the screen.

As shown in FIGS. 23A-B, a fax message may be displayed as a graphical image (e.g., TIFF image) in a scrolling window 180. In addition, the fax message may be printed or forwarded by selecting a button and entering a fax number in a form field.

FIGS. 23A-B show screen shots of the fax viewer portion of the interface. By clicking on the image used to indicate a Fax message in the Inbox Message Work Area 160 (FIGS. 21A-B), a new Work Area and Function bar are called (Sidebar 161, Menu 165 and Status 164 bars may remain the same). The new Work Area 180 is the Fax viewer which indicates all available information about the sender: Name (if caller ID is available). Time and Date, and number of pages in transmission.

Available Functions for the Fax Viewer are:

1. **Forward** - this will commence a forward mail sequence allowing the user to re-direct the fax with an added memo.

2. **Reply** - similar to Forward, but directed specifically to the sender.

3. **Download** - send the current Fax to a pre-configured E-mail address - the message may then be removed from the server or returned in fax or message.

4. **Delete** - delete selected Fax from server with Confirmation Dialog (e.g. Are you sure?) return to Messages Work Area.

5. **Next Message** - Jump to next Message in chronological queue (may require shifting to another type of message viewer)

6. **Previous Message** - jump to previous Message in chronological queue (may require shifting to another type of message viewer).

7. **Next Page** - display next page of Fax (if any).

8. **Previous Page** - display previous page of Fax (if any)

These functions may provided in the menu bar 165 (FIG 23A) or in the function bar 163 (FIG. 23B) or other desirable locations on the screen.

Information regarding an inbound voice mail message (e.g., date, time, sender's number) may be displayed in a page as shown in FIGS. 24A-B. For computers equipped with a sound card, following selection of a button 190, a digitized voice message may be played back to the user. For computers without a sound card, the user may be given an 800 number to cal' 'o listen to the voice message. The voice mail message may then be saved, stored, or forwarded to individuals or groups.

FIGS. 24A-B show screen shots of the voice mail viewer portion of the interface. By clicking on the image used to indicate a Voice mail message in the Inbox Message Work Area 160 (FIGS. 21A-B), a new Work Area 191 and Function bar 163 are called

(Sidebar 161, Menu 165 and Status 164 bars may remain the same). The new Work Area 191 is the Voice Mail viewer which indicates all available information about the sender: Name (if caller ID is available). Time and Date of arrival, and length of message. The work Area will feature a DialWeb link and will have a RealAudio player embedded therein which will both be used to call a ram file which will play the appropriate audio stream.

Available Functions for the Voice mail Viewer area:

1. **Forward** - this will commence a forward mail sequence allowing the user to re-direct the Voice mail with added memo.

2. **Reply** - similar to Forward, but directed specifically to the sender.

3. **Download** - send the current Voice mail to a pre-configured E-mail address. The message may then be removed from the server or returned in fax or message.

4. **Delete** - delete selected Voice mail from server with Confirmation Dialog (e.g. Are you sure?) return to Messages Work Area.

5. **Next Message** - Jump to next Message in chronological queue (may require shifting to another type of message viewer)

6. **Previous Message** - jump to previous Message in chronological queue (may require shifting to another type of message viewer).

These functions may provided in the menu bar (not shown) or in the function bar 163 (FIG. 23B) or other desirable locations on the screen.

In addition to retrieving inbound messages, the user may compose and send outbound messages. FIG. 25A shows an outbound message page where the message is a reply (thus the user need not enter an address). FIG. 25B shows an outbound message page where the message is not a reply and must therefore be addressed. The user first selects the intended recipients by selecting one or more groups or individuals to whom the message may be sent 200. The user then enters the text of the message in a scrolling window 202. The message may then be sent to each of the selected recipients by selecting one of the delivery modes appearing in the buttons at the bottom of the page 204. Using the "Preferred Delivery" option, the message may converted or reformatted according to the preferred communication method of the recipient, preferably upon retrieval of the message by the recipient.

In another aspect of the present invention, the computer interface may be used to place outbound calls - for example, conference calls. Referring to FIG. 26 to place a conference call, a user first selects a list of participants by selecting one or more of the groups and/or individuals from the Contacts list 210. The user then selects the conference call button 212 to establish the call. The user waits for his telephone to ring. The user is informed of the status of the call through a page as shown in FIG. 27.

Another outbound messaging feature of the present invention that is available via the computer interface is fax broadcasting. To take advantage of this feature, the user simply enters the text of his fax message and provides a list of recipient fax numbers, preferably by selecting a group or individuals from his on-line directory. The message and fax numbers are then forwarded to the telecommunications switch so the necessary calls may be made. For those recipients who prefer e-mail over fax messaging, the cross-media messaging feature of the present invention processes the fax message as an attachment to an e-mail message so that the recipient may retrieve the message according to his or her preferred communication medium. Upon retrieval by the recipient, the fax message may be displayed as an image in a scrolling window. In another aspect of the present invention, the fax message may be sent as e-mail to any recipient who happens to be logged in to the Web site of the present invention at the time the fax is sent. This approach may result in a significant cost savings as fewer phone calls are needed to deliver the message to the intended recipients.

In addition to using the computer interface to present invention, a user may access his personal on-line directory, his personal inbox, and his personal contact information using a telephone. The same information that is available to the user via the computer interface is accessible to the user via the telephone interface. Therefore, the user is assured of access to his communications regardless of the interface he chooses at any given time. The user may also arrange for conference calling and fax broadcasting through the telephone interface. The telephone thus gives the user an alternative device through which the features and functionality of the present invention may be accessed.

Although the invention is described in detail with specific reference to illustrative embodiments, it is not limited to those particular embodiments. For example, the configuration of the components that provide the features and functionality of the present invention may change and nevertheless fall within the scope and spirit of the present invention. The scope of the present invention is defined by the following claims.

Claims:

1. A system for accessing, creating and managing communications and messages comprising:

   a) a server;

   b) a first database in communication with the server;

   c) at least one telephone application node in communication with the server;

   d) at least one telephony node in communication with the server;

   e) at least one Internet node in communication with the server; and

   f) at least one hardware node in communication with the at least one telephone application node, at least one Internet node and server.

2. The system of claim 1 further comprising at least one Internet based messaging and communications management application resident on the at least one telephone application node.

3. The system of claim 2 in which the at least one Internet based messaging and communications management application provides voicemail functionality.

4. The system of claim 2 in which the at least one Internet based messaging and communications management application provides electronic mail functionality.

5. The system of claim 2 in which the at least one Internet based messaging and communications management application provides facsimile functionality.

6. The system of claim 1 further comprising at least one telephony based messaging and communications management application resident on the at least one telephone application node.

7. The system of claim 6 in which the at least one telephony based messaging and communications management application provides voicemail functionality.

8. The system of claim 6 in which the at least one telephony based messaging and communications management application provides electronic mail functionality.

9. The system of claim 6 in which the at least one telephony based messaging and communications management application provides facsimile functionality.

10. The system of claim 1 in which the hardware node further comprises at least one communications processing board.

11. The system of claim 10 in which the at least one communications processing board is a digital network interface.

12. The system of claim 10 in which the at least one communications processing board is a facsimile board.

13. The system of claim 10 in which the at least one communications processing board is a text to speech processor.

14. The system of claim 10 in which the at least one communications processing board is a conference call board.

15. The system of claim 10 in which the at least one communications processing board is a voice recording board.

16. The system of claim 1 further comprising at least one user interface.

17. The system of claim 16 in which the interface is telephony based.

18. The system of claim 16 in which the interface is keypad based.

19. The system of claim 16 in which the interface is an Internet browser.

20. The system of claim 16 in which the interface provides access to stored messages and real-time communications.

21. The system of claim 16 in which the interface provides people centered communications.

22. The system of claim 16 in which the interface provides cross media messaging.

23. The system of claim 16 in which the interface allows a user to initiate and control conference call.

24. The system of claim 16 in which the interface allows a user to direct a single message to multiple recipients.

25. The system of claim 16 in which the interface allows a user to create a publicly accessible web page.

26. The system of claim 16 in which the interface allows a user to identify at least one information source and periodically retrieve selected data from the information source.

27. The system of claim 16 further comprising an application which selects the optimal communications medium for sending or receiving a message.

28. The system of claim 1 in which the Internet node further comprises:

a) a web manager;

b) an encoder in communication with the web manager; and

c) a second database in communication with the web manager.

29. The system of claim 28 in which the web manager periodically synchronizes the first and second databases.

30. A system for managing personal communications comprising:

a) telephony subsystem;

b) an Internet subsystem in communication with the telephony subsystem; and

c) a hardware node in communication with the telephony subsystem and thc Internet subsystem.

31. The system of claim 30 further comprising at least one user interface.

32. The system of claim 31in which the interface is telephony based.

33. The system of claim 31in which the interface is keypad based.

34. The system of claim 31in which the interface is an Internet browser.

35. The system of claim 31 in which the interface provides access to stored messages and real-time communications.

36. The system of claim 31 in which the interface provides people centered communications.

37. The system of claim 31 in which the interface provides cross media messaging.

38. The system of claim 31 in which the interface allows a user to initiate and control conference call.

39. The system of claim 31 in which the interface allows a user to direct a single message to multiple recipients.

40. The system of claim 31 in which the interface allows a user to create a publicly accessible web page.

41. The system of claim 31 in which the interface allows a user to identify at least one information source and periodically retrieve selected data from the information source.

42. The system of claim 31 further comprising an application which selects the optimal communications medium for sending or receiving a message.

43. A system for accessing, creating and managing communications and messages comprising:

a) a system client;

b) a computer telephony interface library in communication with the client; and

c) a computer telephony interface server in communication with the library.

44. The system of claim 43 in which the system client provides at least one service.

45. The system of claim 44 in which the service is facsimile message management.

46. The system of claim 44 in which the service is automated message delivery to a predetermined recipient.

47. The system of claim 44 in which the service is electronic mail management

48. The system of claim 44 in which the service is electronic mail text to speech conversion.

49. The system of claim 44 in which the service is an automated response to electronic mail.

50. The system of claim 44 in which the service is call flow mapping.

51. The system of claim 43 further comprising at least one user interface.

52. The system of claim 51 in which the interface is telephony based.

53. The system of claim 51 in which the interface is keypad based.

54. The system of claim 51 in which the interface is an Internet browser.

55. The system of claim 51 in which the interface provides access to stored messages and real-time communications.

56. The system of claim 51 in which the interface provides people centered communications.

57. The system of claim 51 in which the interface provides cross media messaging.

58. The system of claim 51 in which the interface allows a user to initiate and control conference call.

59. The system of claim 51 in which the interface allows a user to direct a single message to multiple recipients.

60. The system of claim 51 in which the interface allows a user to create a publicly accessible web page.

61. The system of claim 51 in which the interface allows a user to identify at least one information source and periodically retrieve selected data from the information source.

62. The system of claim 51 further comprising an application which selects the optimal communications medium for sending or receiving a message.

63. A method for permitting a user to manage personal communications comprising the steps of:

a) providing a telephony subsystem;

b) providing an Internet subsystem in communication with the telephony subsystem;

c) providing a hardware node in communication with the telephony subsystem and the Internet subsystem; and

d) providing a user interface.

64. The system of claim 63 further comprising the step of providing access to stored messages and real-time communications.

65. The system of claim 63 further comprising the step of providing people centered communications.

66. The system of claim 63 further comprising the step of providing cross media messaging.

67. The system of claim 63 further comprising the step of allowing a user to initiate and control conference call.

68. The system of claim 63 further comprising the step of allowing a user to direct a single message to multiple recipients.

69. The system of claim 63 further comprising the step of allowing a user to create a publicly accessible web page.

70. The system of claim 63 further comprising the step of allowing a user to identify at least one information source and periodically retrieve selected data from the information source.

71. The system of claim 63 further comprising the step of automatically selecting the optimal communications medium for sending or receiving a message.

72. A system for managing personal communications comprising:

a) telephony subsystem;

b) an Internet subsystem in communication with the telephony subsystem;

c) a hardware node in communication with the telephony subsystem and the Internet subsystem;

d) at least one keypad based interface in communication with the telephony subsystem;

e) at least one web browser interface in communication with the Internet subsystem; and

f) a billing information system in communication with the telephony and Internet subystems and with a payment source.

73. The system of claim 72 in which the payment source is a credit card account.

74. The system of claim 72 in which the payment source is a bank account.

75. The system of claim 72 in which the keypad interface is a telephone.

76. The system of claim 72 in which the interfaces provides access to stored messages and real-time communications.

77. The system of claim 72 in which the interfaces provide people centered communications.

78. The system of claim 72 in which the interfaces provide cross media messaging.

79. The system of claim 72 in which the interfaces allows a user to initiate and control conference call.

80. The system of claim 72 in which the interfaces allows a user to direct a single message to multiple recipients.

81. The system of claim 72 in which the interfaces allow a user to create a publicly accessible web page.

82. The system of claim 72 in which the interfaces allow a user to identify at least one information source and periodically retrieve selected data from the information source.

83. The system of claim 72 further comprising an application which selects the optimal communications medium for sending or receiving a message.

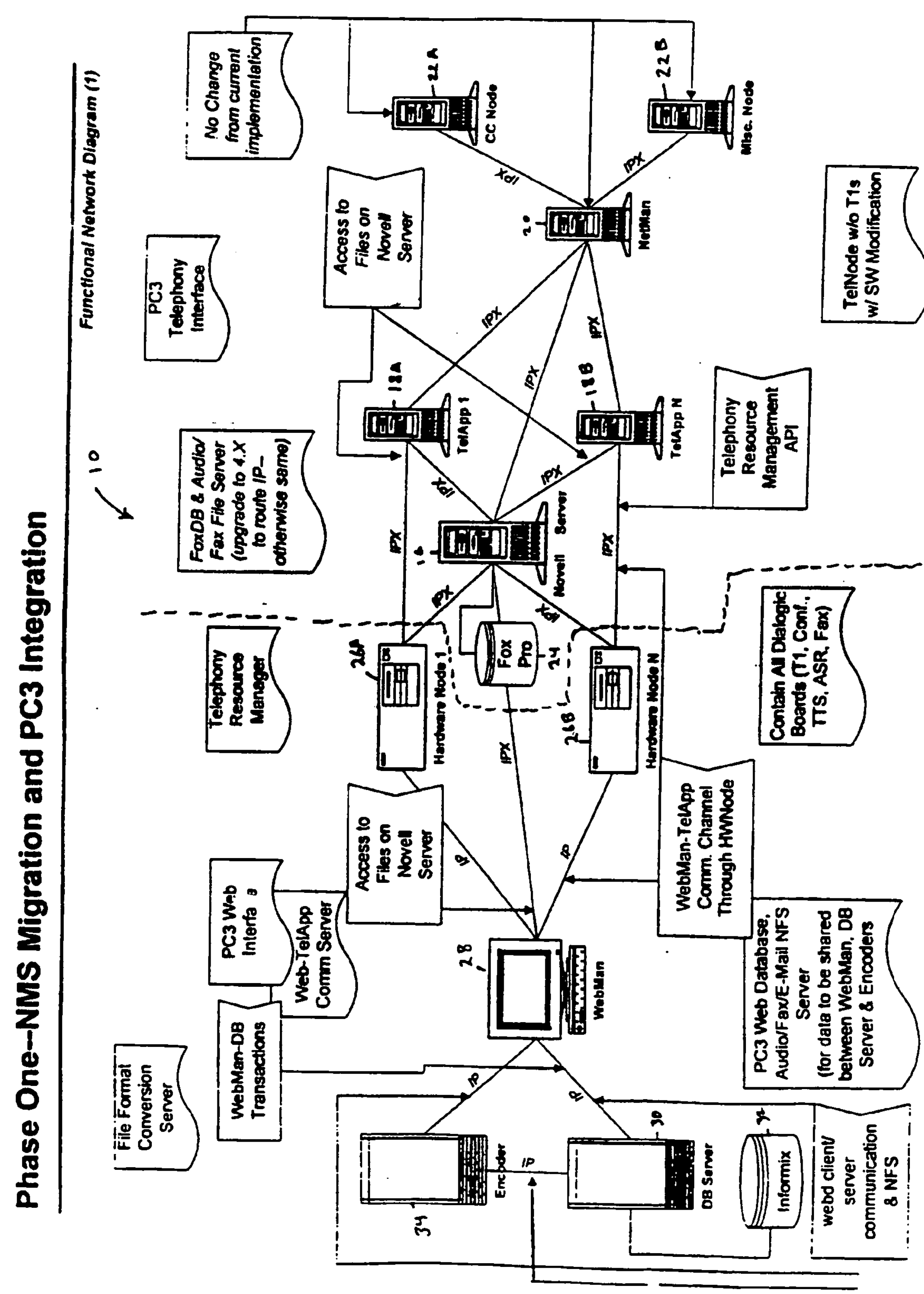

Phase One--NMS Migration and PC3 Integration
Functional Network Diagram (1)
10
File Format Conversion Server
WebMan-DB Transactions
PC3 Web Interfa a
Web-TelApp Comm Server
Access to Files on Novell Server
Telephony Resource Manager
FoxDB & Audio/ Fax File Server (upgrade to 4.X to route IP-- otherwise same)
PC3 Telephony Interface
Access to Files on Novell Server
No Change from current implementation
Telephony Resource Management API
TelNode w/o T1s w/ SW Modification
Contain All Dialogic Boards (T1, Conf., TTS, ASR, Fax)
WebMan-TelApp Comm. Channel Through HWNode
PC3 Web Database, Audio/Fax/E-Mail NFS Server (for data to be shared between WebMan, DB Server & Encoders
webd client/ server communication & NFS
Encoder
DB Server
Informix
WebMan
Hardware Node 1
Hardware Node N
Fox Pro
Novell Server
TelApp 1
TelApp N
NetMan
CC Node
Misc. Node
IPX
IP
34
30
28
26A
26B
24
16
18A
18B
20
22A
22B
14
12

Fig. 1

Phase One--Uneffected NMS Segment

Functional Network Diagram (2)

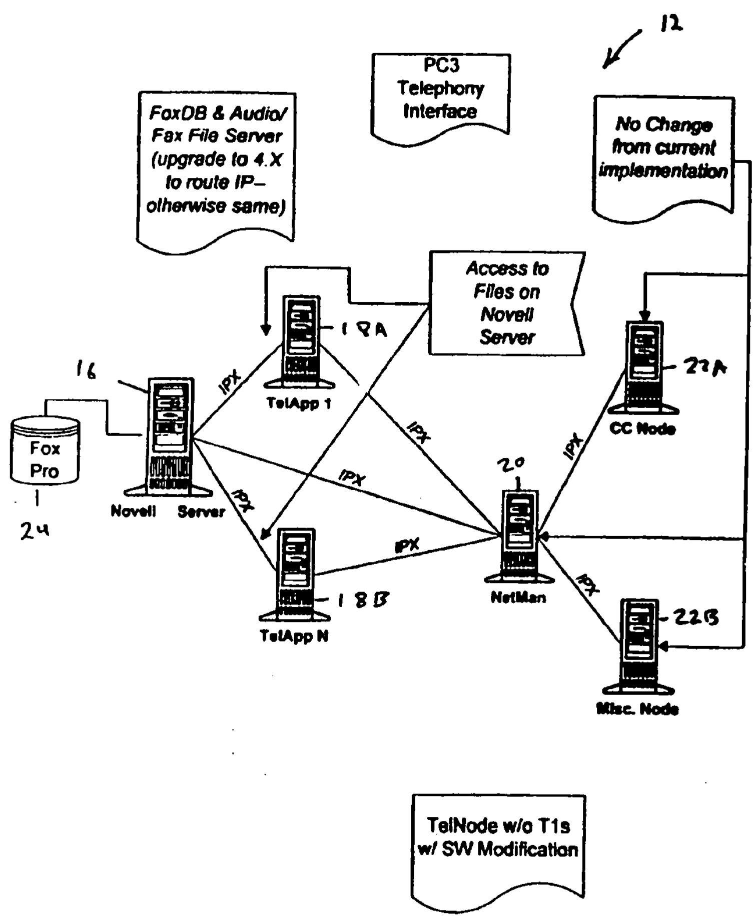


Fig. 2

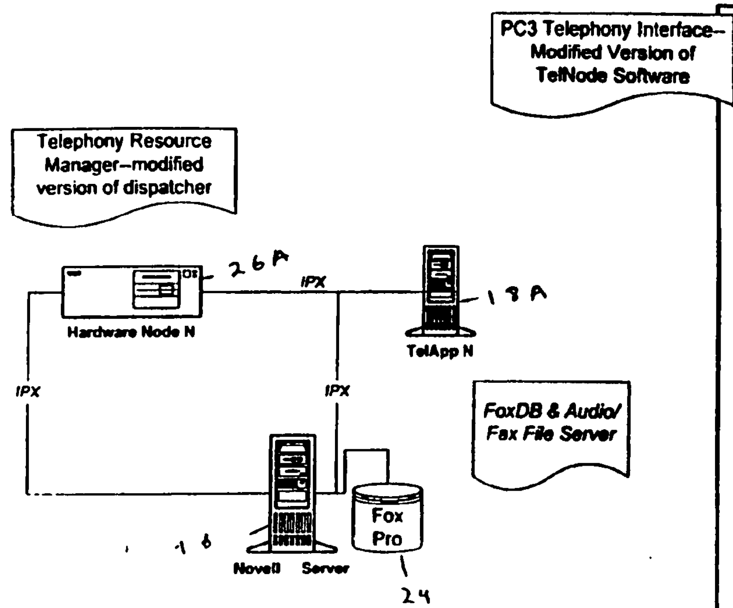
This Section of the NMS is relatively untouched by Phase I of the migration. The NetMan, Novell File Server/Fox Pro Database, Credit Card/Billing Nodes, and TelNodes (now TelApps) interact with each other just as they always have.

Special Purpose TelNodes (FaxNode, TTS Node, Conference Node) are removed from the system, as this role will be distributed between the Hardware Node and the TelApp.

The TelApp is a TelNode stripped of all Digital Network Interface (T1) boards. A TelApp is dependent upon it's Hardware Node which contains all necessary boards (T1, fax, TTS, conference, etc.) and which manages those resources.

Phase One--TelApp & Hardware Node (TelNode Split)

Functional Network Diagram (3)



The intention of reworking the telnode (to include all special case telnodes such as fax or conference nodes) is.

1) to start a natural migration away from the DOS platform towards a platform which is more scaleable and better supported by Dialogic (unix at this point in time)

2) to eliminate the need to transfer calls through the switch & coordinate through the NetMan in order to provide conference, fax, tts and other features during a call (current strategy is costly, time consuming, and limiting)

3) to provide an initial method for allowing the telephony subsystem to communicate with the web subsystem thereby enabling the development of the PC3

The Plan for reworking the TelNode is as follows

1) All special purpose TelNodes (Fax, Conference, TTS, ...) will not be included in the new telephony subsystem.

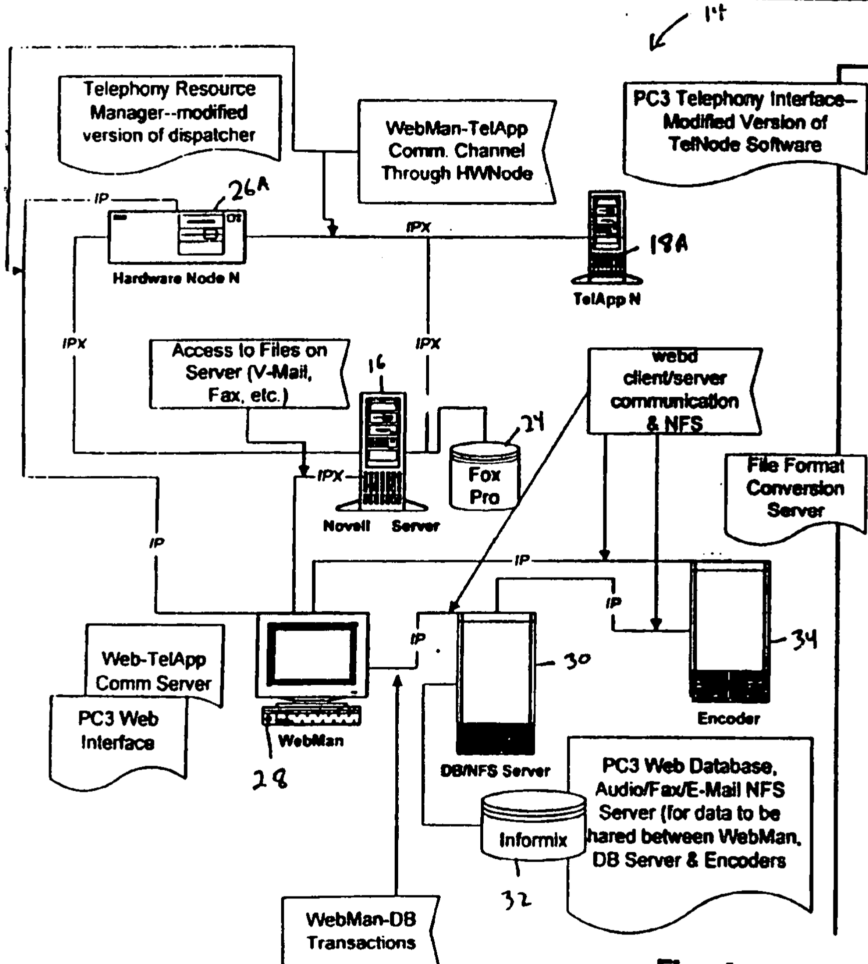
2) The TelNode itself will be stripped of all telephony resource boards (T1). It will be a simple computer which controls call flow much as it does today by communicating with a Hardware Node.

3) The Hardware Node will be a UnixWare rack mount passive backplane system stocked with T1, voice, conference, fax, tts, & voice rec boards. The Hardware Node will eventually be completly configurable as to how much of each it has. For Phase I there will be two or three styles of Hardware Nodes...

4) The TelNode will be modified to talk to the Hardware Node through a high level API for controlling call flow and telephony resources. Any current code which uses the Dialogic API will be reworked to conform to the API. The TelNode will be paired with a HardWare Node. The HardWare Node will run a modified version of the dispatcher software (Telet's system) to manage telephony resources and to communicate with the TelNode (using SPX/IPX).

Fig. 3

Phase One--Integration of NMS and the Internet



The strategy for "Web Enabling" the NMS is as follows:

TelApp/WebMan Communication

1) Add a new node to each "POD" which will act as the conduit for all communication between the TelApp and any processes which need the internet or communicate through TCP/IP. This Node is the WebMan

2) Initially (until the TelApp's use TCP/IP), all communication between the TelApp and the WebMan will flow through the dispatcher--as it will support both TCP/IP and SPX/IPX. A new module will be added to the dispatcher for conversion between WebMan/TelApp TCP and SPX messages.

3) A scheme for groups of TelApp's to connect to a WebMan much as they do a NetMan.

WebMan/Web processes Communication

The WebMan will communicate with several other processes in the Web Subsystem.

Encoded

The encoder daemon, which runs on the encoders, takes care of file format conversion (audio, graphic, etc.)

E-Mail Agent

36

This process runs on the E-Mail Server(s) and takes care of transfering incoming e-mail messages to the NFS server and inserting message info into the Database. It may also handle attachment detachment and encoding (coordinate with encoded) for e-mail to whatever gateways.

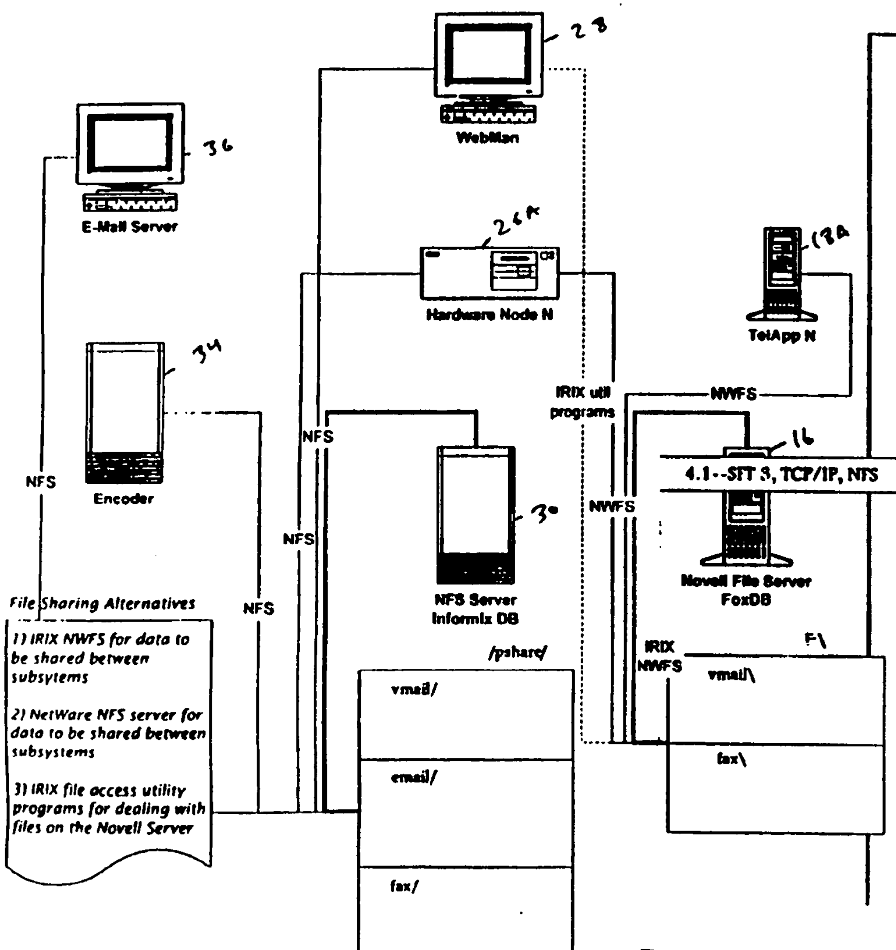
PC3 CGI/Java Apps

The Applications which make up the PC3 Web Interface. A number of CGI programs which may need to coordinate jobs with other processes in the subsystem--conference call control through the WebMan.

Fig. 4

Phase One--NMS File System Organization

Software Diagram (1)



The Telephony Subsystem and the Web Subsystem each have their own specific database and file server. The role of the Telephony Subsystem's File Server (Novell/Fox Pro) remains virtually unchanged in this first phase.

The Web Subsystem Database holds all information necessary in the PC3 Web Interface--messages, contact data, customer records, etc. The Web Subsystem Database is kept synchronized by the WebMan which coordinates with the TelApp. The WebMan also keeps the files stored on the NFS server (voice mail, fax, etc.) in sync with those stored on the Novell Server.

The Web Subsystem File Server serves as centralized storage for the subsystem. The WebMan, Encoder, and E-Mail Server each NFS mount the file system and may refer to files without regard for which machine they are on. Voice Mail, Fax, and other customer data stored on the Novell File Server is duplicated (possibly in different file formats) on the NFS server.

The WebMan handles the transfer of files (Voice Main, Fax) from Novell Server to NFS server on TelApp's command. Incoming E-Mail is handled in the following way. The E-Mail Server stores the new E-Mail on the NFS server and inserts a message record in the database. It is now easily accesable to the web processes (PC3) which need to get at it. In order for the TelApp to access the e-mail, the e-mail directory may be mapped as a Novell Shared Directory (IRIX supports this). The TelApp still needs to communicate with the WebMan in order to query the Web DB for a given customer's list of email messages.

Fig 5

Phase One--Subsystem IPC

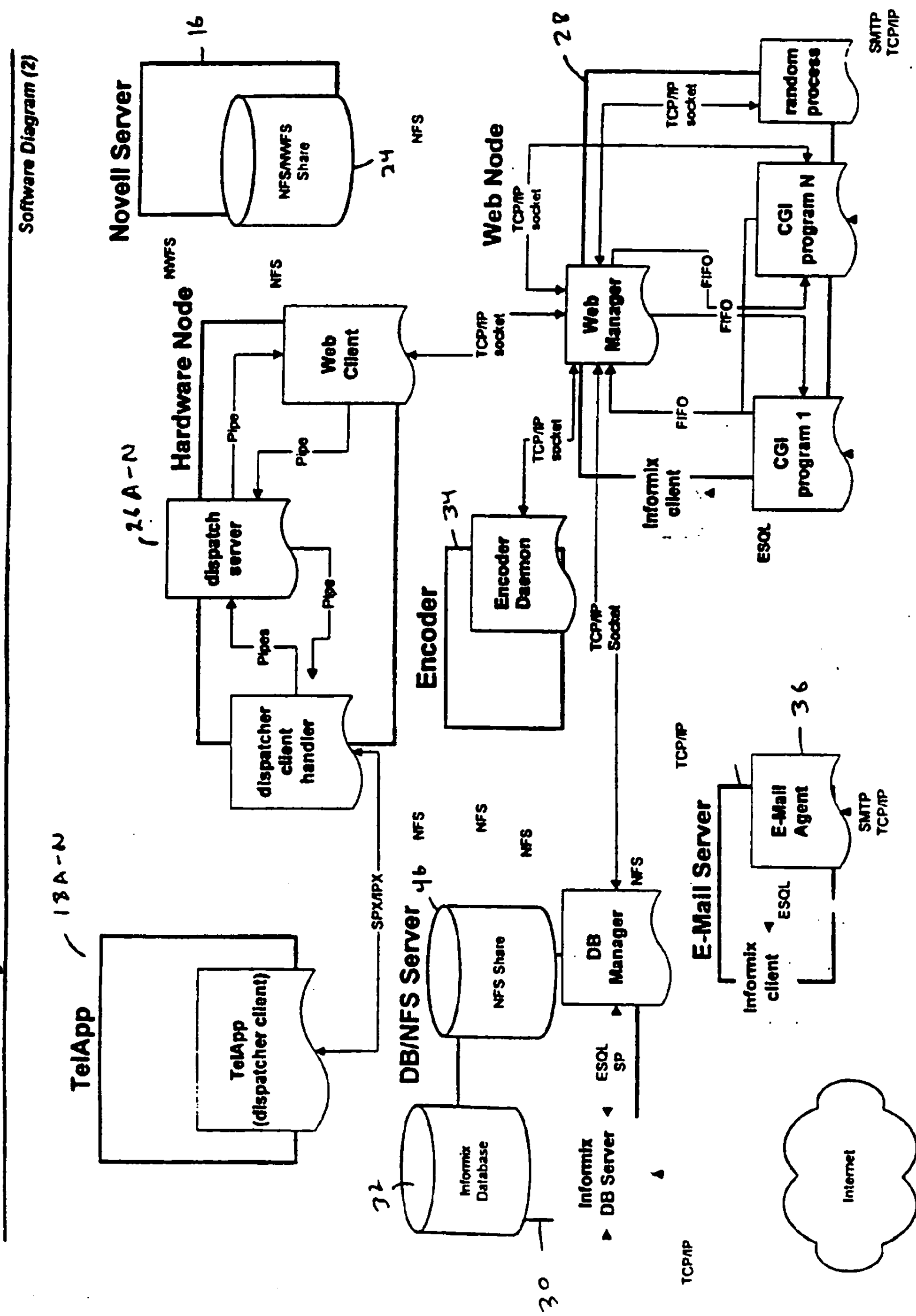
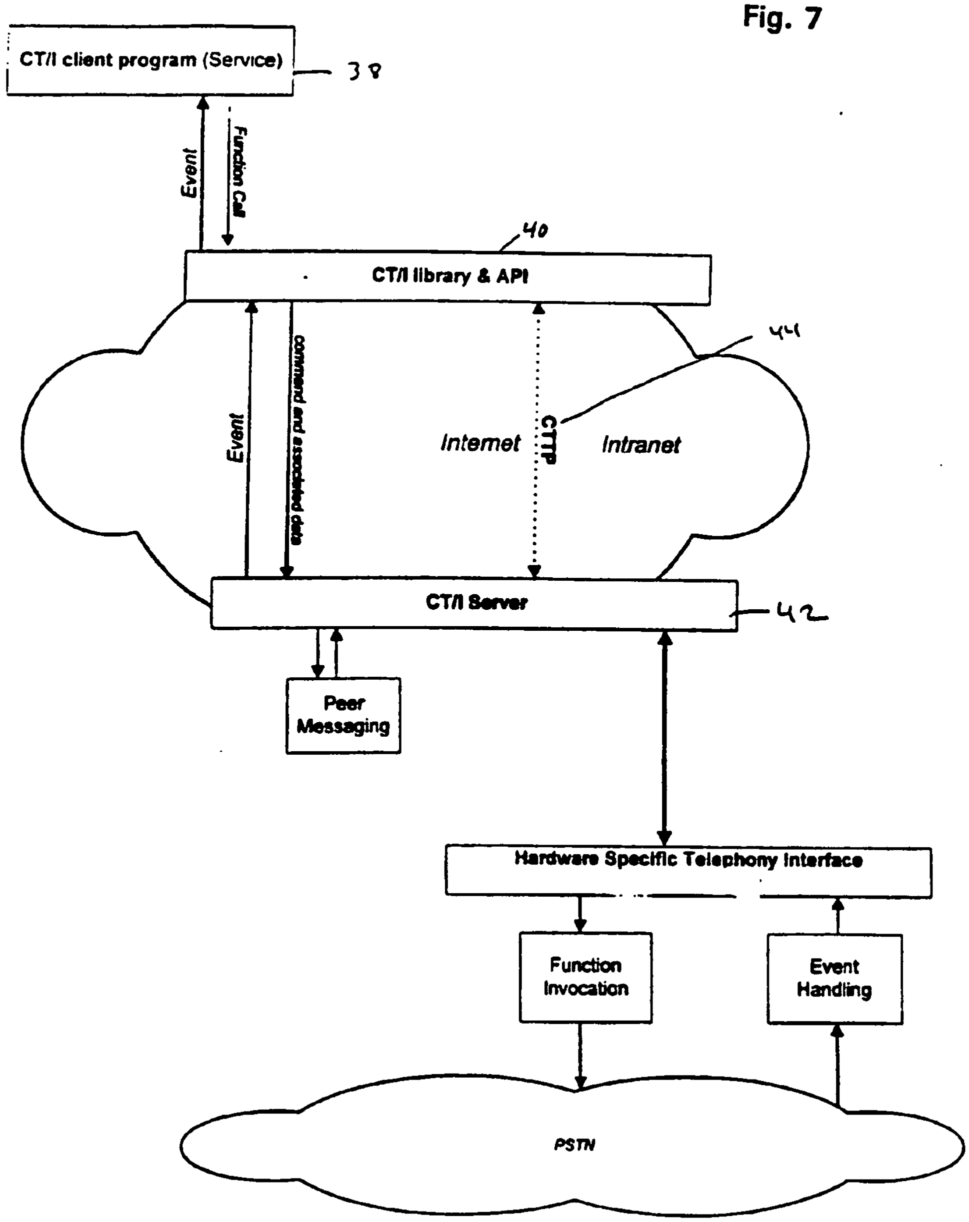


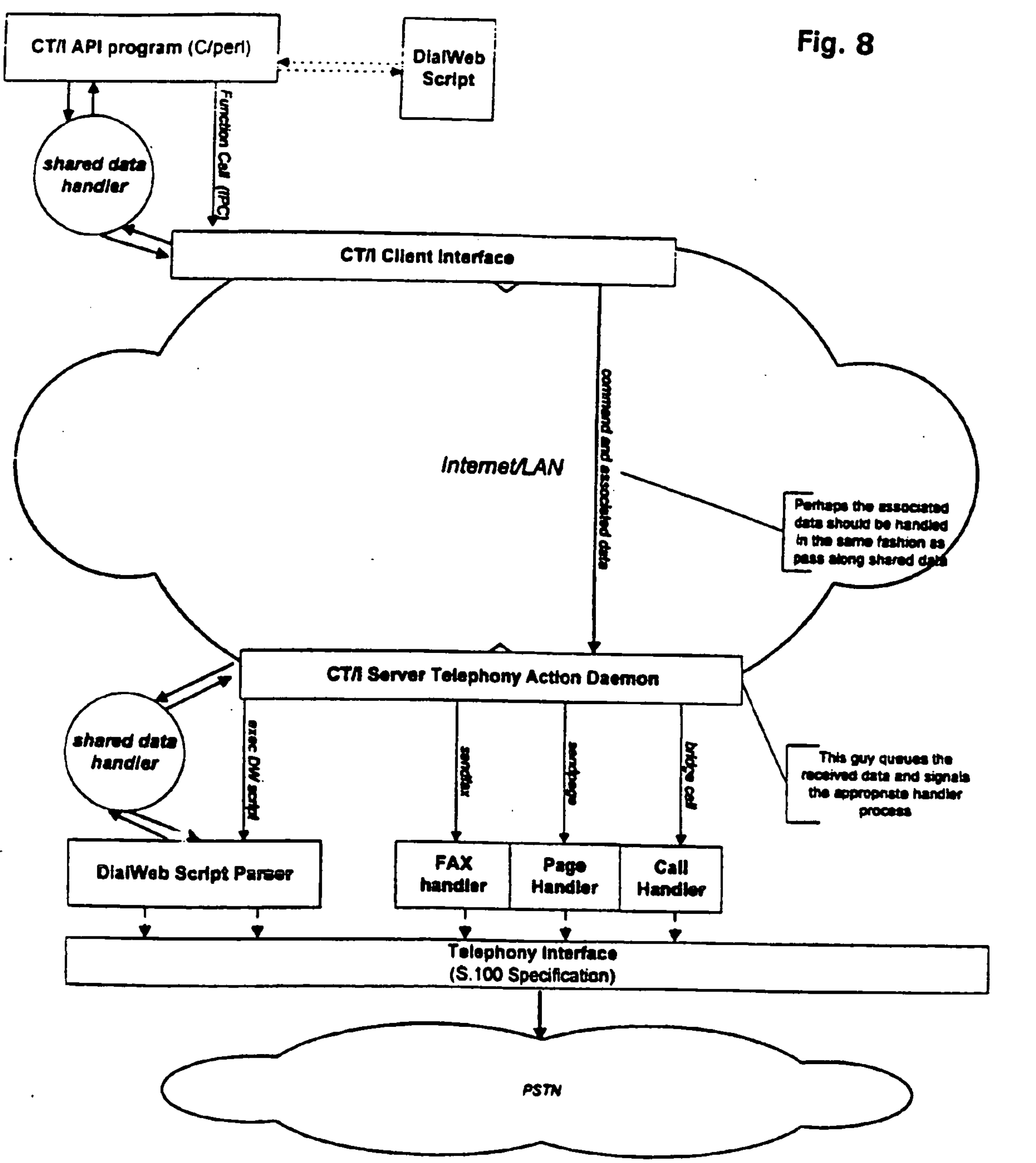
Fig. 6

CT/I Client/Server Diagram

Fig. 7

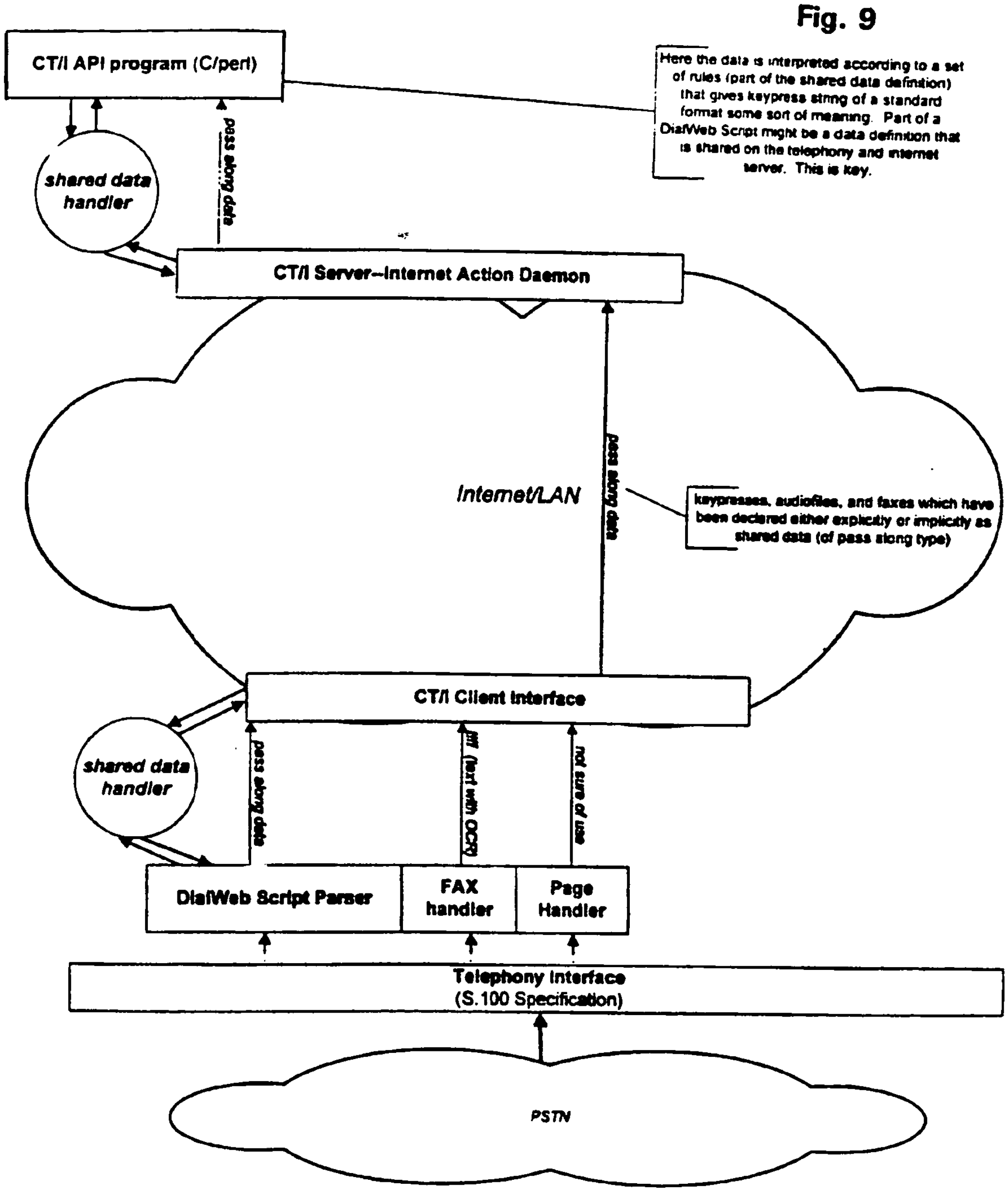


Web process initiates a telephone action



Telephone process initiates an internet action

Fig. 9



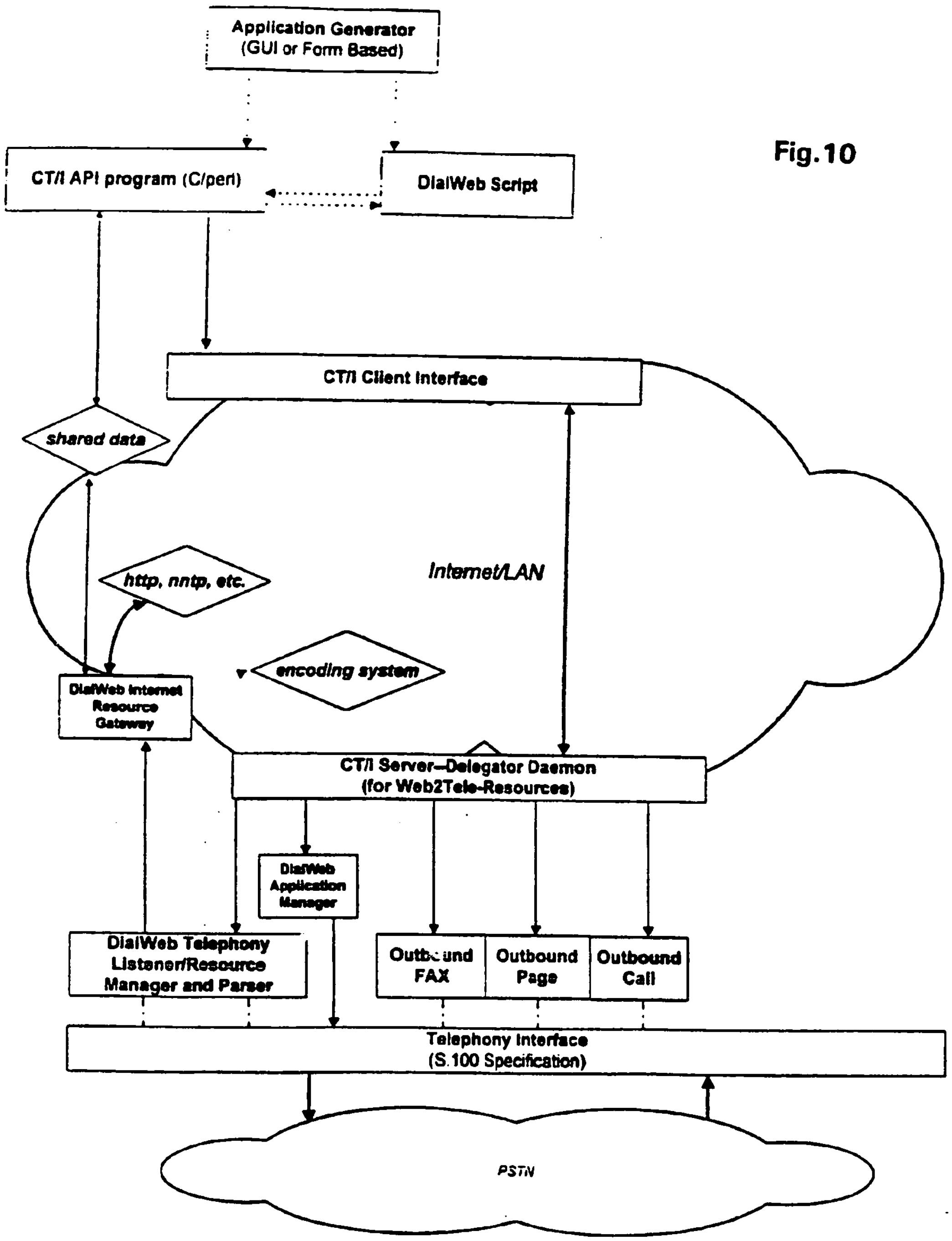

Application Generator
(GUI or Form Based)
CT/I API program (C/perl)
DialWeb Script
CT/I Client Interface
shared data
Internet/LAN
http, nntp, etc.
encoding system
DialWeb Internet Resource Gateway
CT/I Server–Delegator Daemon
(for Web2Tele-Resources)
DialWeb Application Manager
DialWeb Telephony Listener/Resource Manager and Parser
Outbound FAX
Outbound Page
Outbound Call
Telephony Interface
(S.100 Specification)
PSTN

Fig. 10

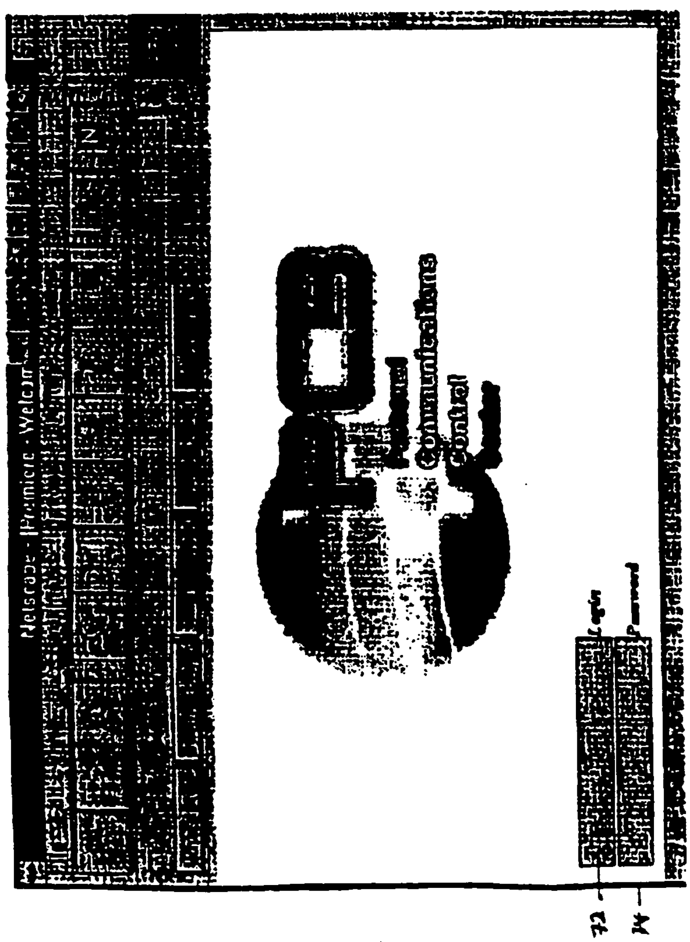


Fig. 11

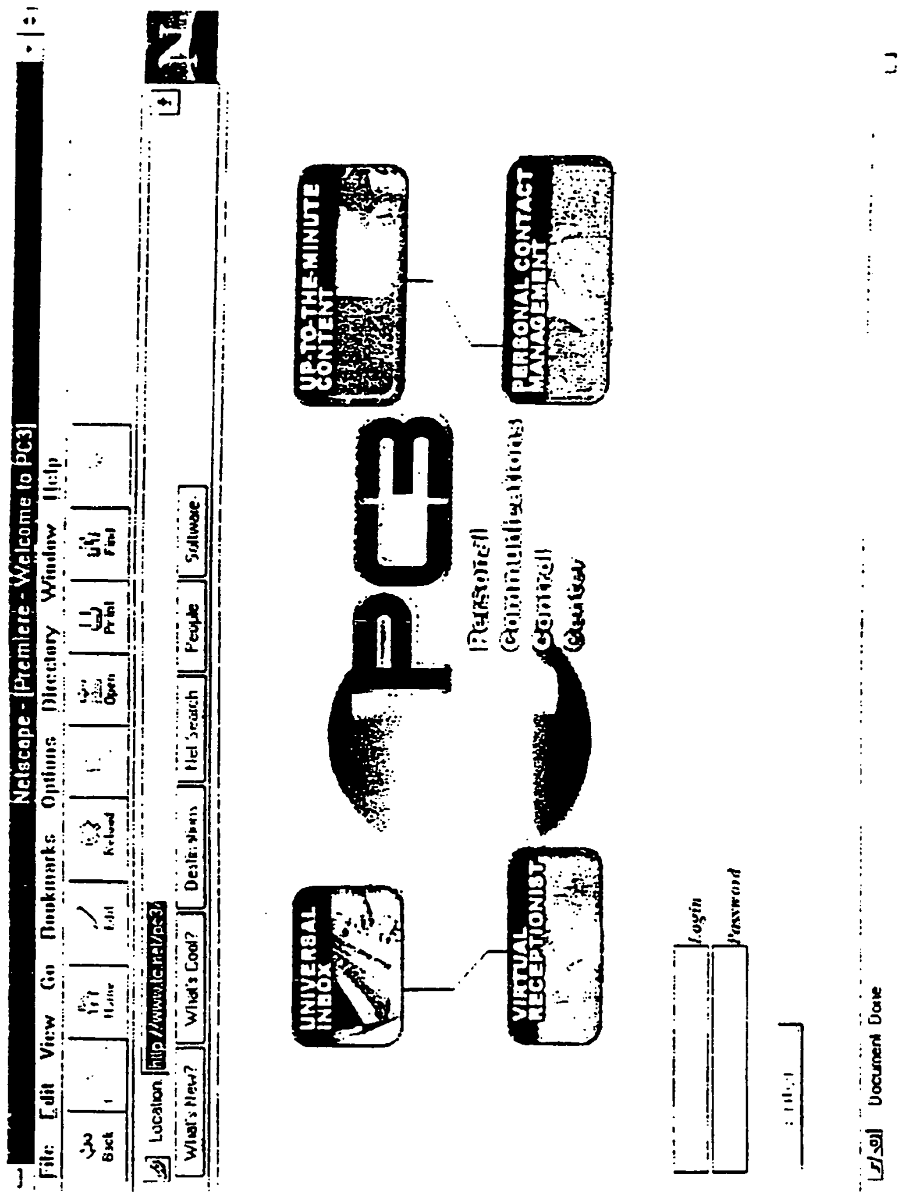

Netscape - [Premiere - Welcome to PC3]
File Edit View Go Bookmarks Options Directory Window Help
Back Home Edit Reload Open Print Find
Location
What's New? What's Cool? Destinations Net Search People Software
UNIVERSAL INBOX
VIRTUAL RECEPTIONIST
UP-TO-THE-MINUTE CONTENT
PERSONAL CONTACT MANAGEMENT
PC3
Login
Password
Document Done

Fig. 11A

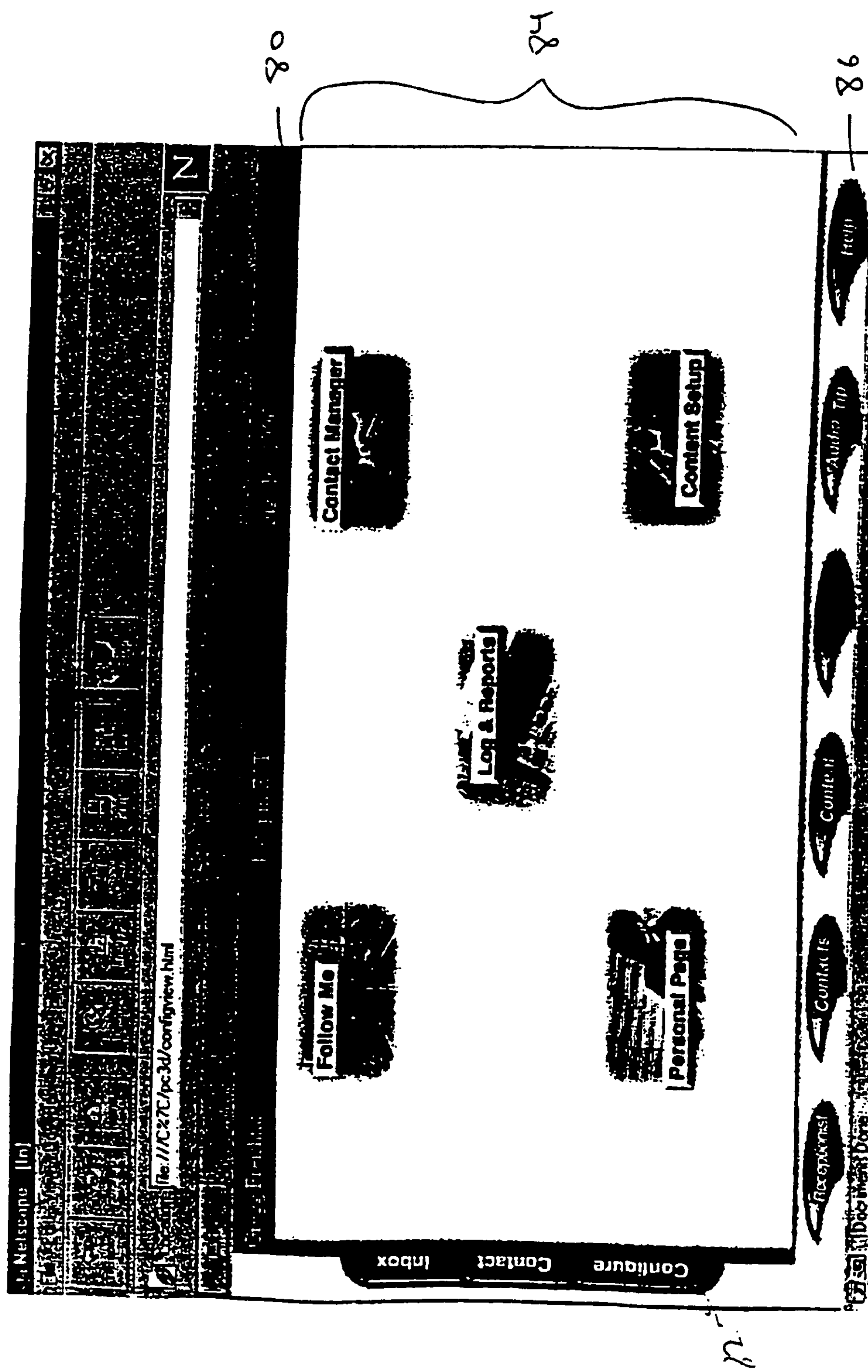

80
84
86
82
Contact Manager
Content Setup
Log & Reports
Follow Me
Personal Page
Inbox
Contact
Configure

Fig. 12

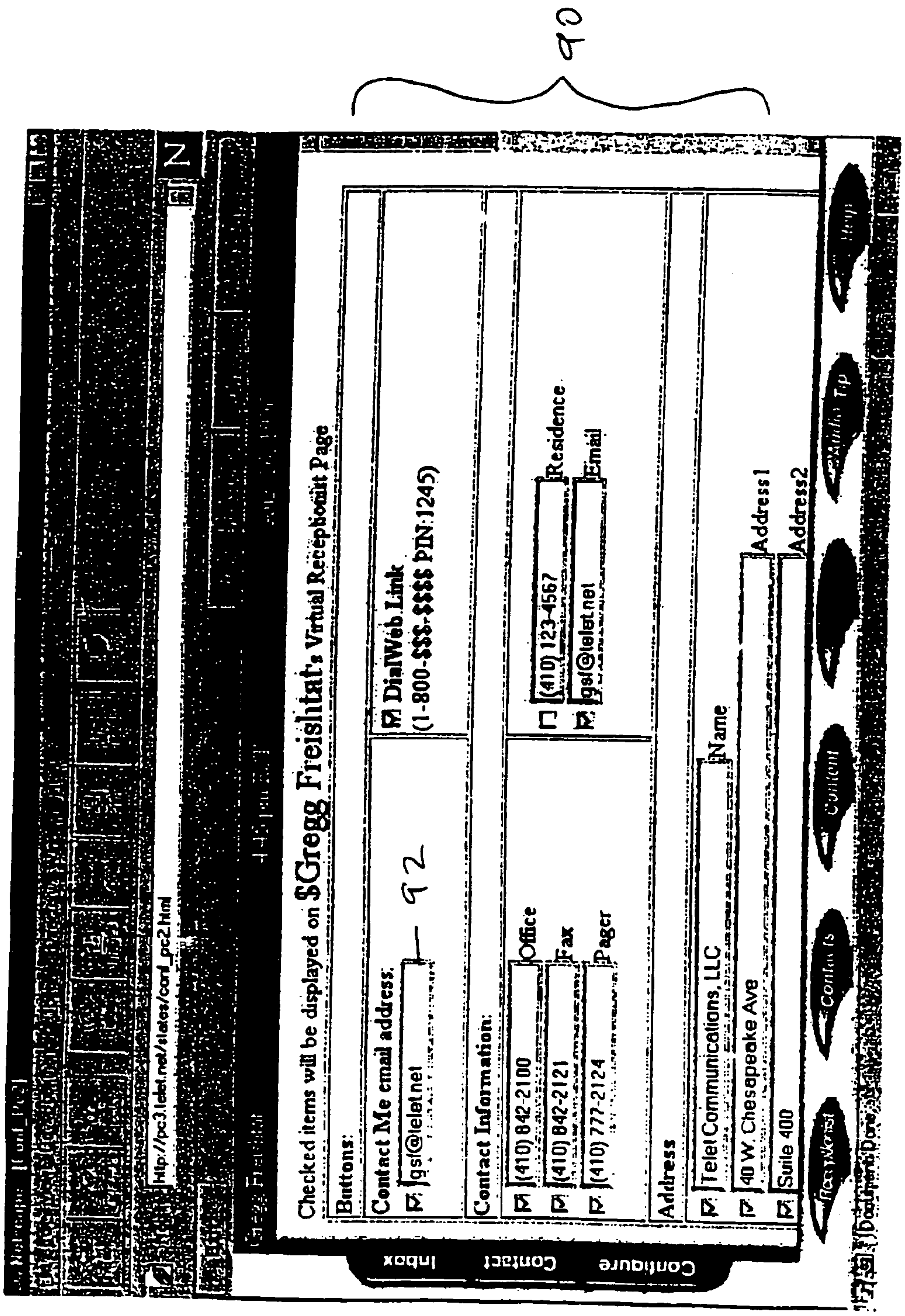

90
Checked items will be displayed on $Gregg Freishtat's Virtual Receptionist Page
Buttons:
Contact Me email address:
gsf@telet.net
92
DialWeb Link
(1-800-$$$-$$$$ PIN:1245)
Contact Information:
(410) 842-2100
Office
(410) 842-2121
Fax
(410) 777-2124
Pager
(410) 123-4567
Residence
gsf@telet.net
Email
Address
Telet Communications, LLC
Name
40 W. Chesapeake Ave
Address1
Suite 400
Address2
Inbox
Contact
Configure
Contacts
Content
Help

Fig. 13

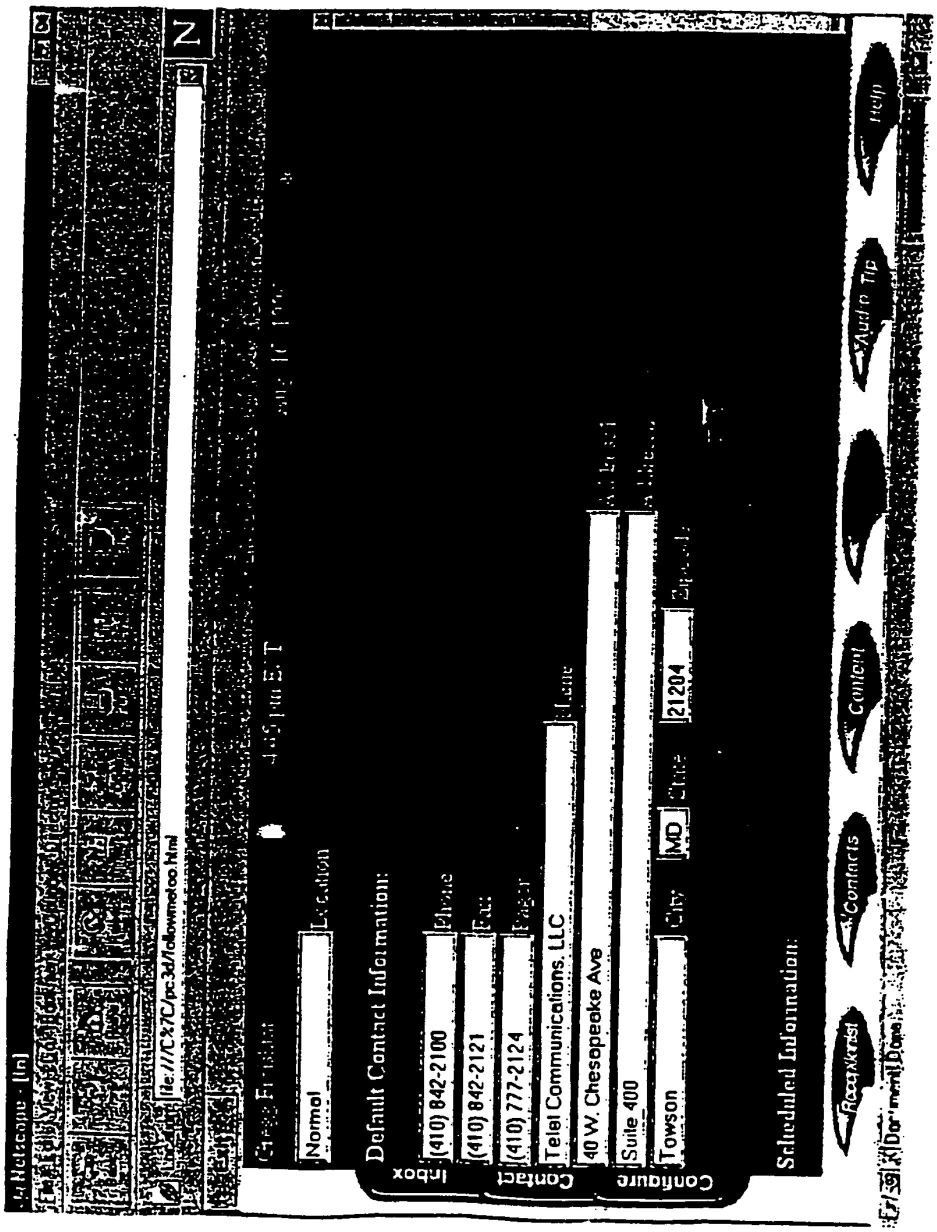

Normal
Location
Default Contact Information:
(410) 842-2100
Phone
(410) 842-2121
Fax
(410) 777-2124
Pager
Telei Communications, LLC
40 W. Chesapeake Ave
Suite 400
Towson
City
MD
State
21204
Scheduled Information:
Inbox
Contact
Configure
Contacts
Content
Audio Tip
Help

Fig. 13A

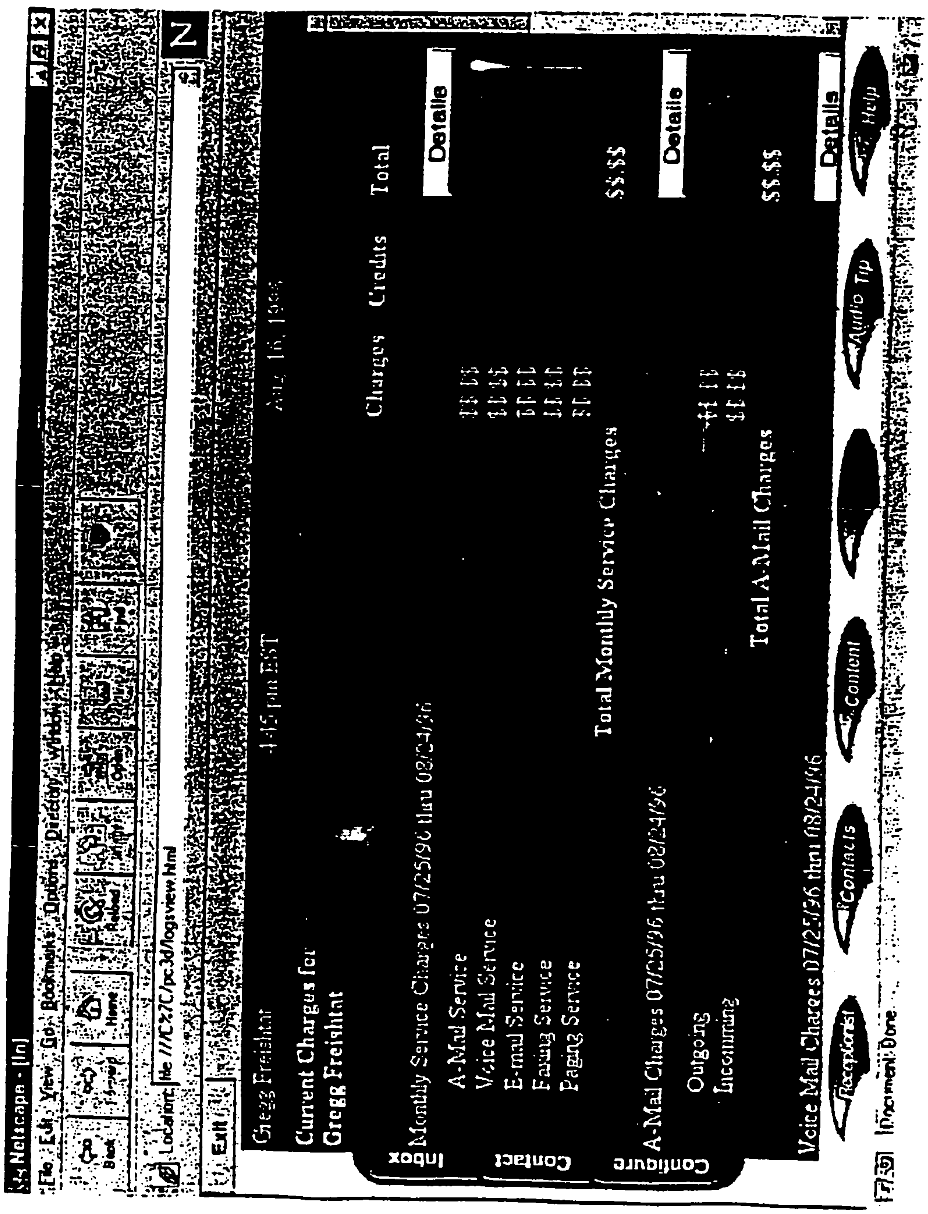

Netscape - [In]
File Edit View Go Bookmarks Options Directory Window Help
Back Forward Home
Exit
Gregg Freishtat
4:45 pm EST
Aug 16, 1996
Current Charges for
Gregg Freishtat
Monthly Service Charges 07/25/96 thru 08/24/96
Charges
Credits
Total
A-Mail Service
Voice Mail Service
E-mail Service
Faxing Service
Paging Service
Total Monthly Service Charges
$$.$$
Details
A-Mail Charges 07/25/96 thru 08/24/96
Outgoing
Incoming
Total A-Mail Charges
$$.$$
Details
Voice Mail Charges 07/25/96 thru 08/24/96
Details
Inbox
Contact
Configure
Receptionist
Contacts
Content
Audio Tip
Help
Document: Done

Fig. 13B

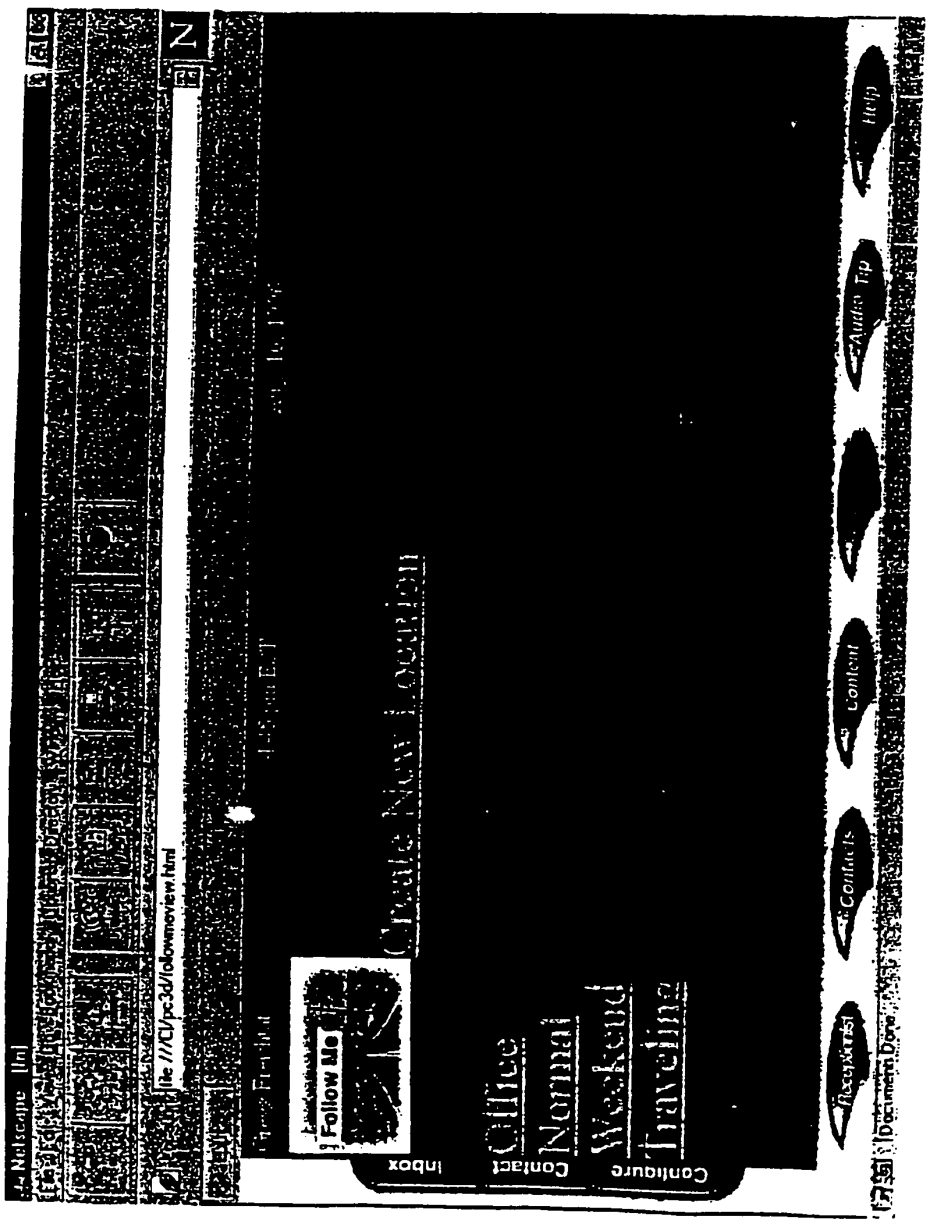

Follow Me
Create New Location
Office
Normal
Weekend
Traveling
Inbox
Contact
Configure
Contacts
Content
Audio Tip
Help

Fig. 13C

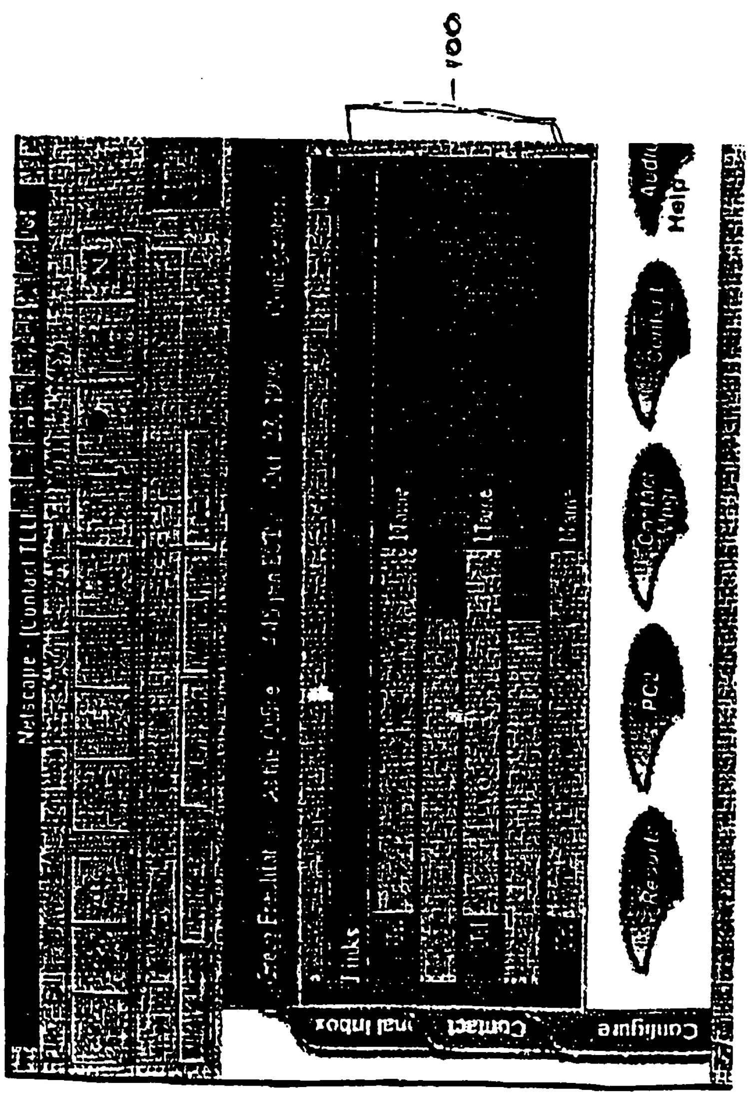


Fig. 14

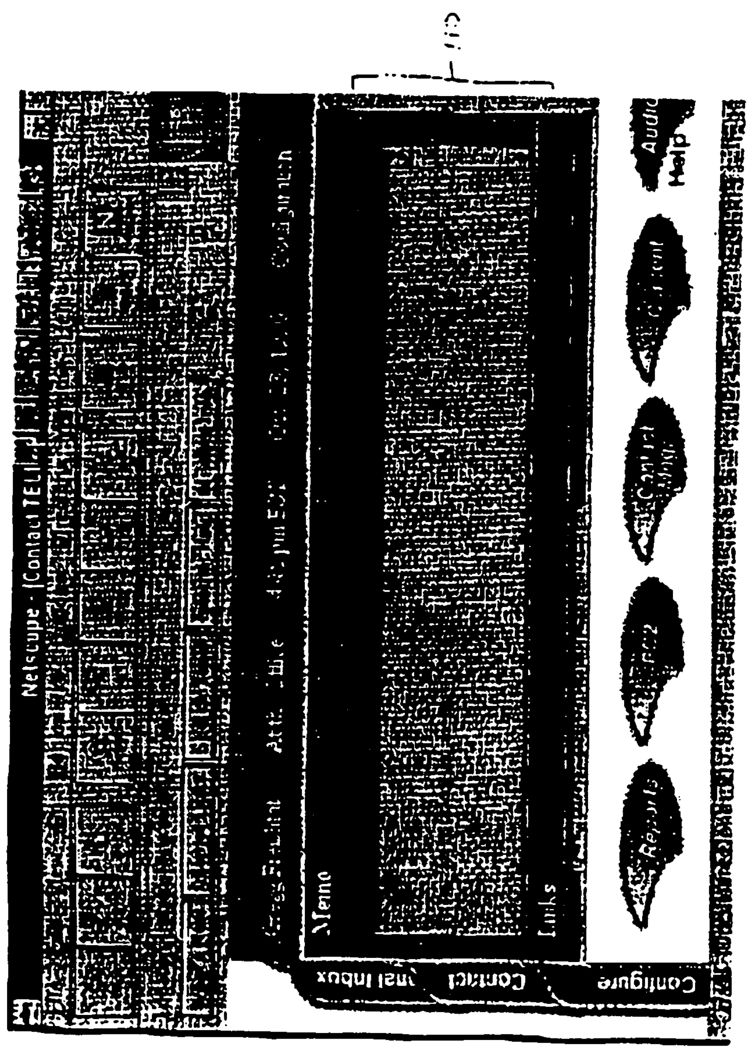

Netscape - [Contact TEL
Memo
Links
Reports
Audio Help
Contact
Configure

Fig. 15

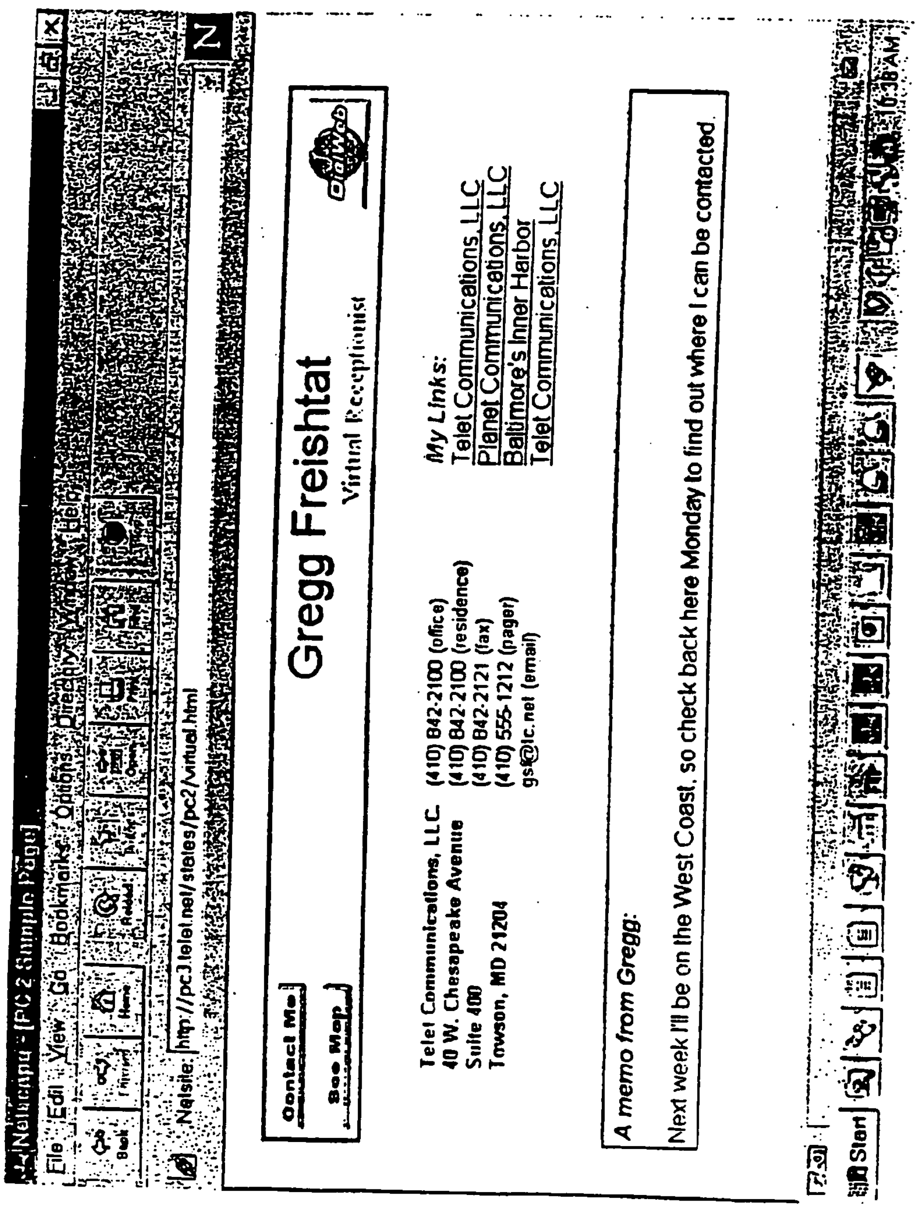

Netscape - [PC 2 Sample Page]
File Edit View Go Bookmarks Options Directory Window Help
Netsite: http://pc3.telet.net/states/pc2/virtual.html
Contact Me
See Map
Gregg Freishtat
Virtual Receptionist
Telet Communications, LLC
40 W. Chesapeake Avenue
Suite 400
Towson, MD 21204
(410) 842-2100 (office)
(410) 842-2100 (residence)
(410) 842-2121 (fax)
(410) 555-1212 (pager)
gs@tc.net (email)
My Links:
Telet Communications, LLC
Planet Communications, LLC
Baltimore's Inner Harbor
Telet Communications, LLC
A memo from Gregg:
Next week I'll be on the West Coast, so check back here Monday to find out where I can be contacted.
Start

Fig. 16

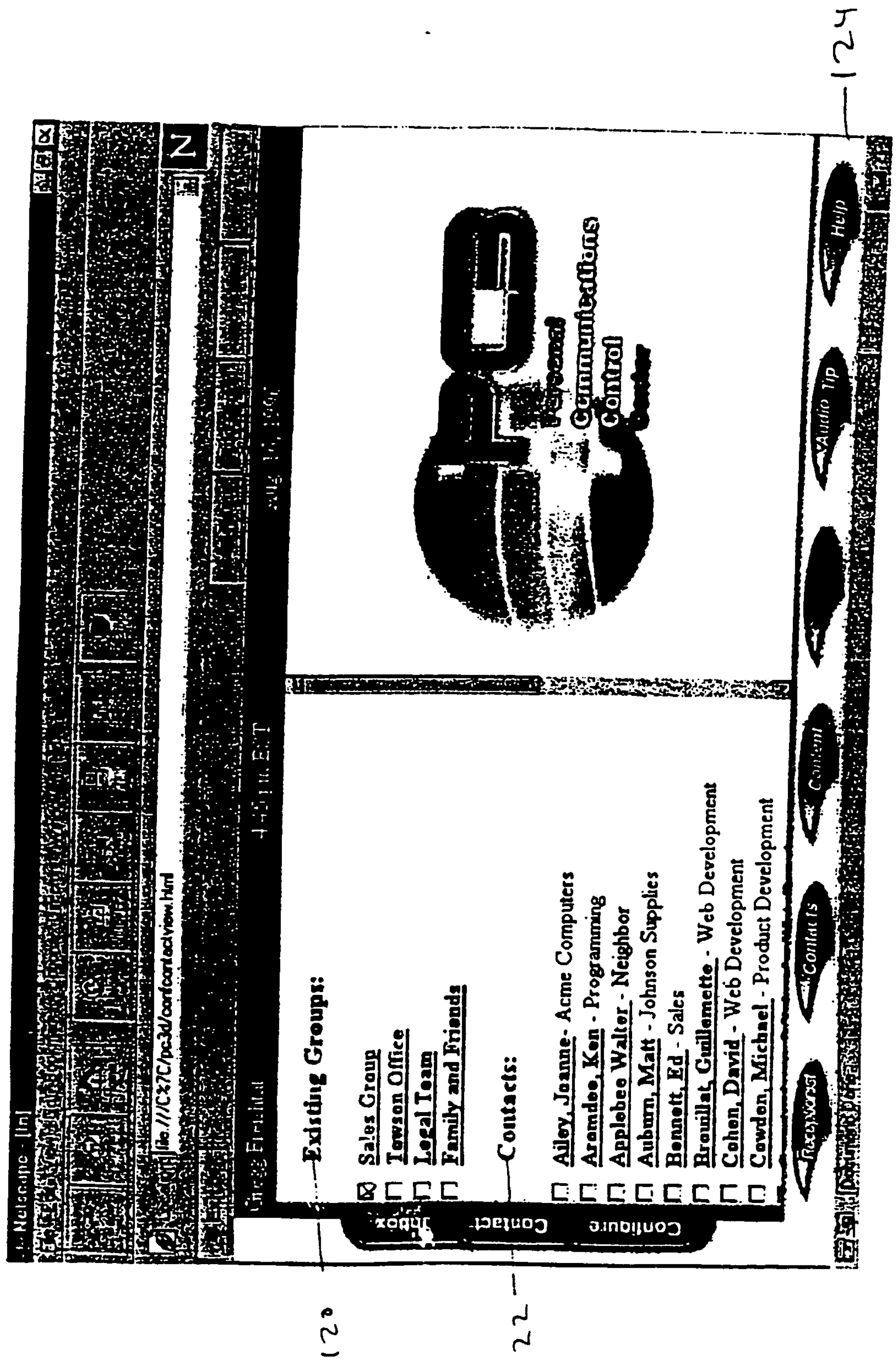

124
file:///C|/C7C/pc3d/confcontactview.html
Existing Groups:
Sales Group
Towson Office
Legal Team
Family and Friends
Contacts:
Ailey, Joanne - Acme Computers
Aramdee, Ken - Programming
Applebee Walter - Neighbor
Auburn, Matt - Johnson Supplies
Bennett, Ed - Sales
Brouillat, Guillemette - Web Development
Cohen, David - Web Development
Cowdon, Michael - Product Development
Personal
Communications
Control
Center
Inbox
Contact
Configure
Contacts
Content
Audio Tip
Help
120
122

Fig. 17

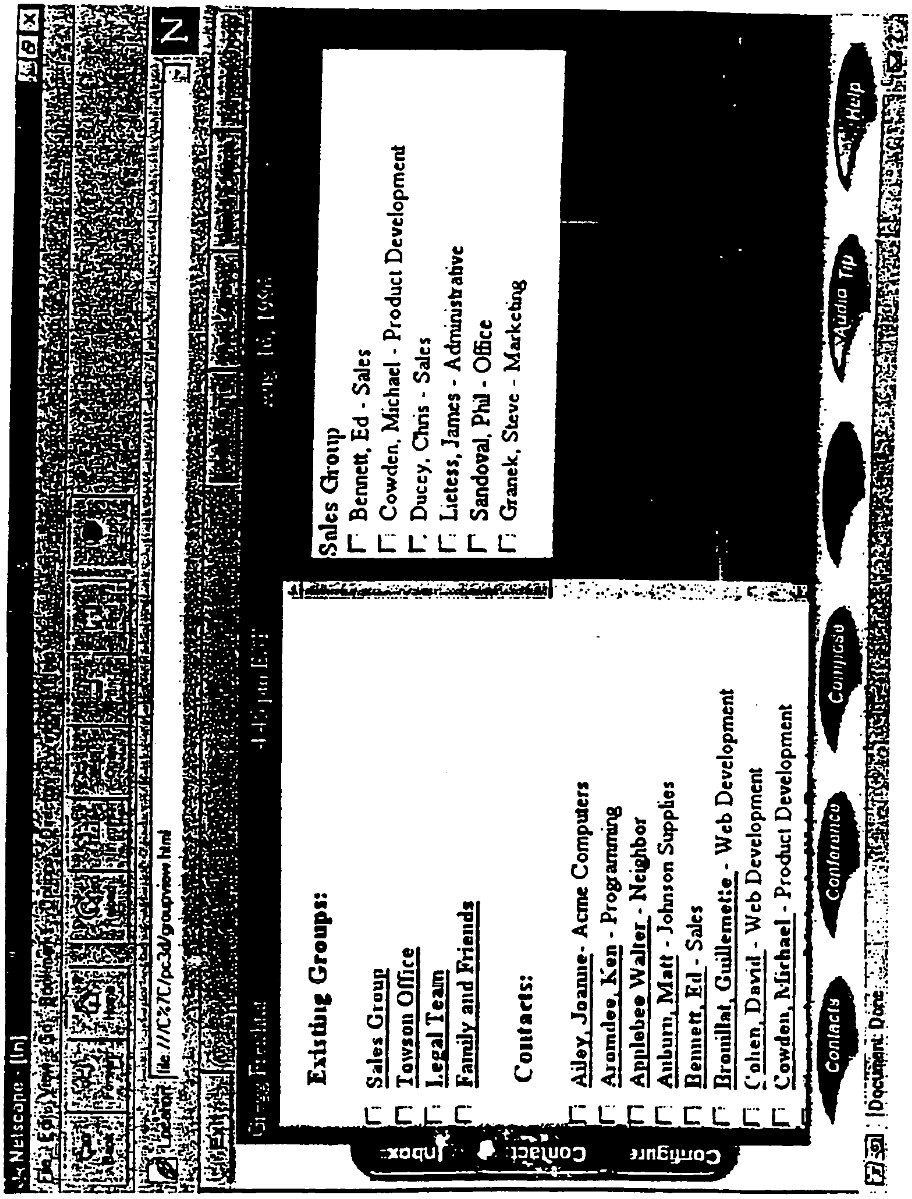

Netscape - [In]
Existing Groups:
Sales Group
Towson Office
Legal Team
Family and Friends
Contacts:
Ailey, Joanne - Acme Computers
Aromdee, Ken - Programming
Applebee Walter - Neighbor
Auburn, Matt - Johnson Supplies
Bennett, Ed - Sales
Bronillat, Guillemette - Web Development
Cohen, David - Web Development
Cowden, Michael - Product Development
Sales Group
Bennett, Ed - Sales
Cowden, Michael - Product Development
Ducey, Chris - Sales
Lietess, James - Administrative
Sandoval, Phil - Office
Granek, Steve - Marketing
Inbox
Contact
Configure
Contacts
Conference
Compose
Audio Tip
Help
Document: Done

Fig. 17A

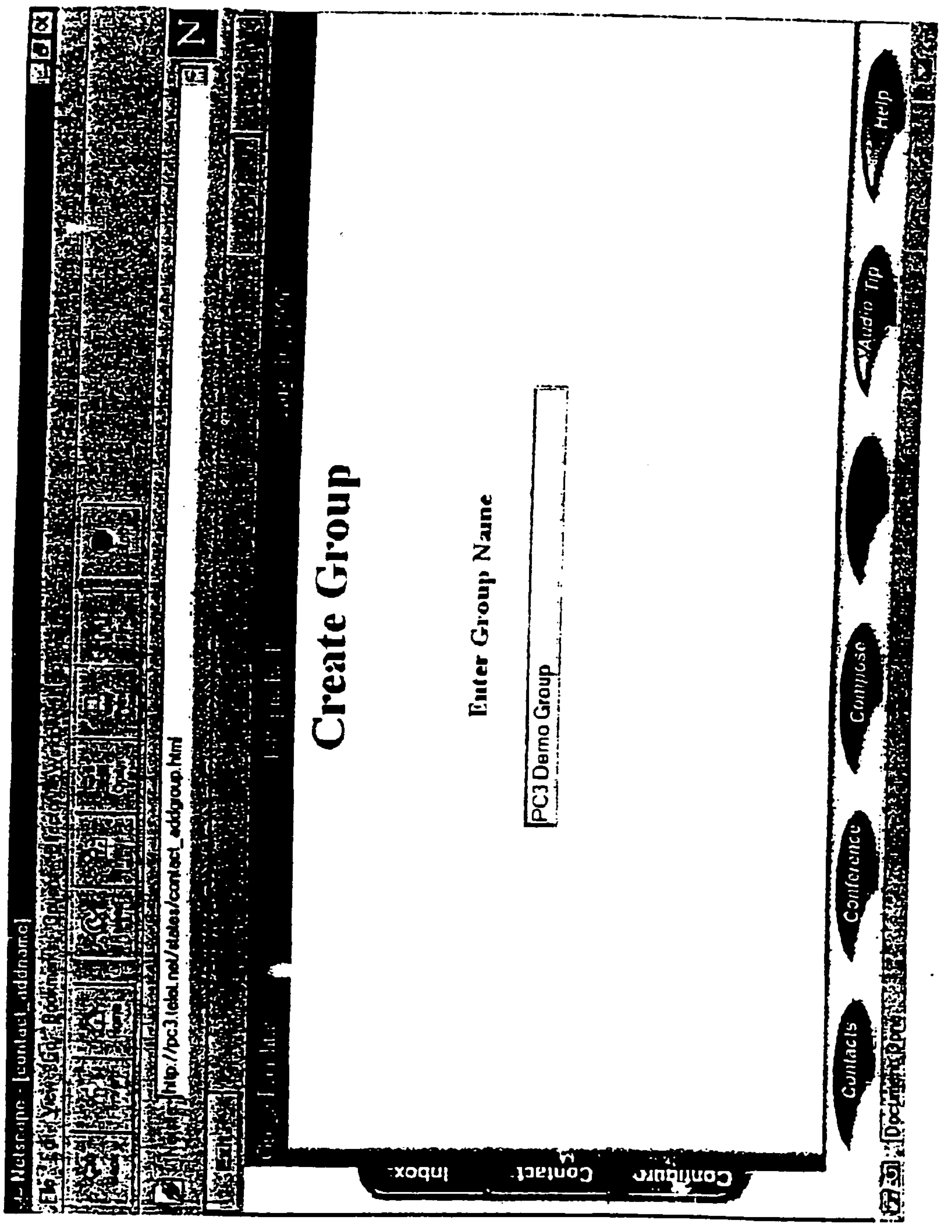

Create Group
Enter Group Name
PC3 Demo Group
Contacts
Conference
Compose
Audio Tip
Help
Inbox
Contact
Configure

Fig. 17B

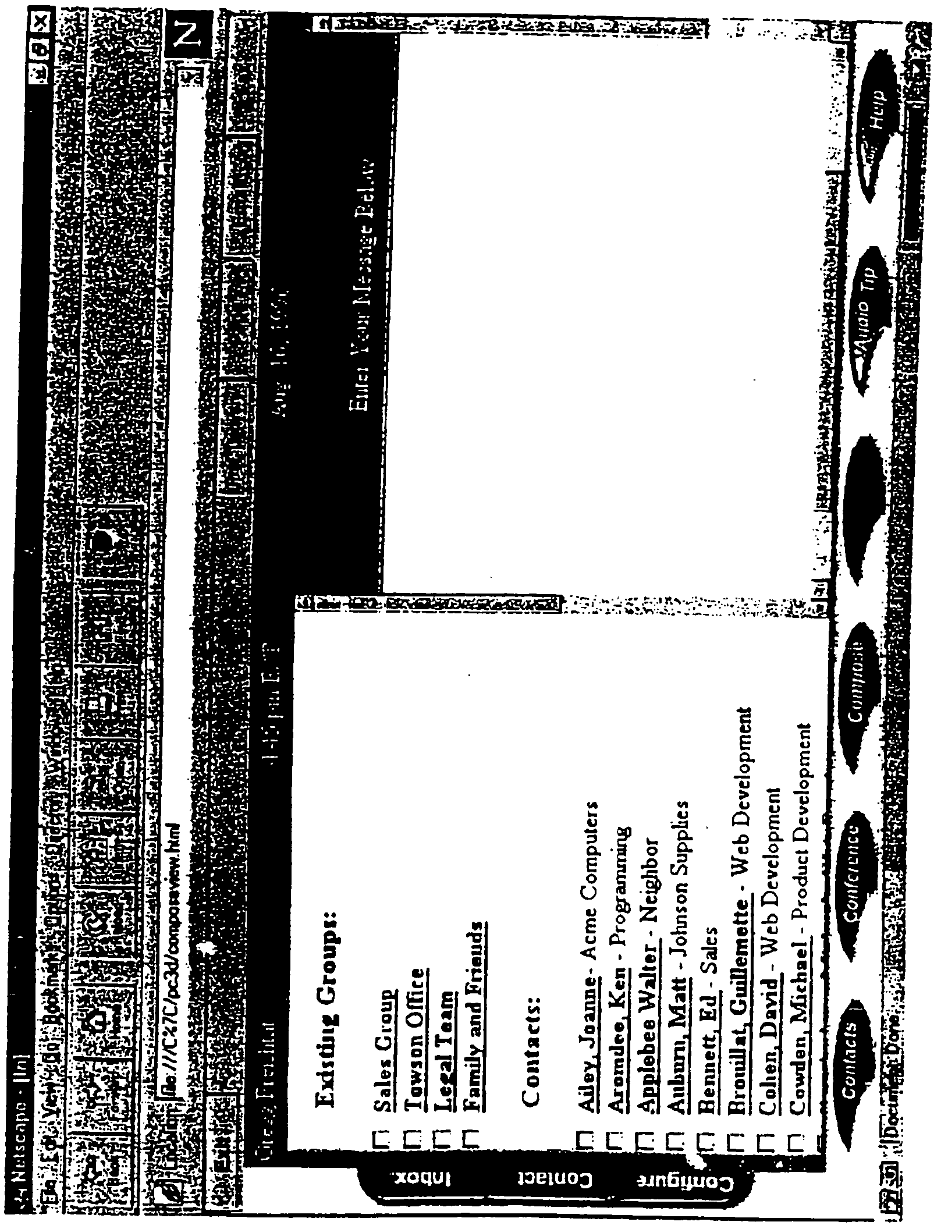
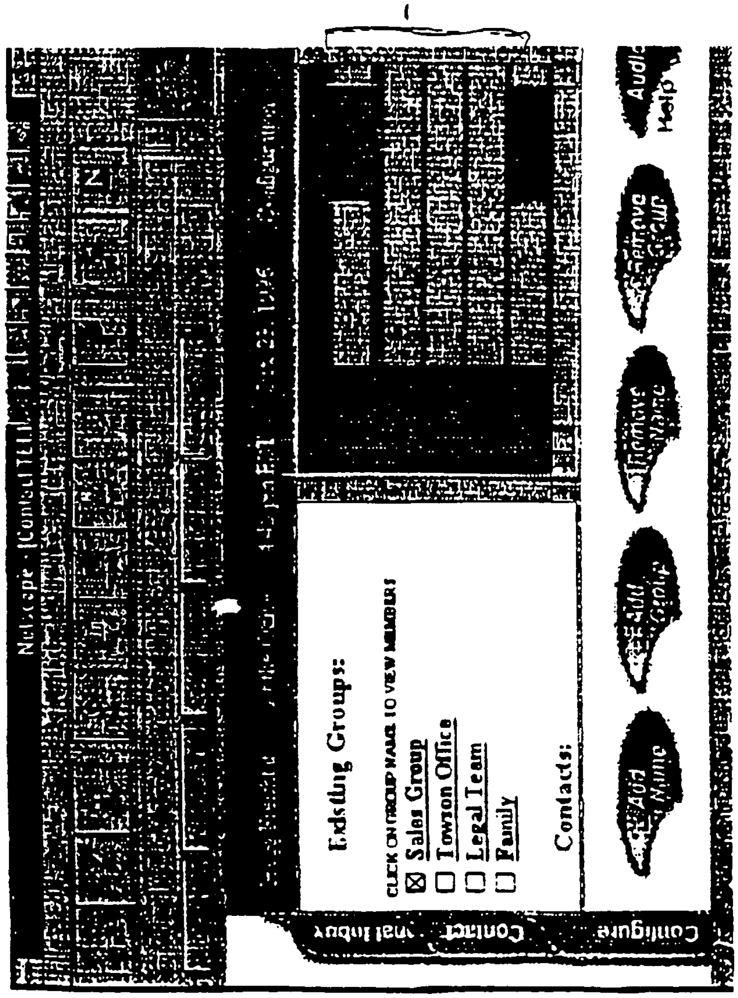

Netscape - [In]
file:///C27C/pc3d/composeview.html
Inbox
Contact
Configure
Existing Groups:
Sales Group
Towson Office
Legal Team
Family and Friends
Contacts:
Ailey, Joanne - Acme Computers
Aromdee, Ken - Programming
Applebee Walter - Neighbor
Auburn, Matt - Johnson Supplies
Bennett, Ed - Sales
Brouillet, Guillemette - Web Development
Cohen, David - Web Development
Cowden, Michael - Product Development
Enter Your Message Below
Contacts
Conference
Compose
Audio Tip
Help
Document: Done

Fig. 17C

132



130

Fig. 18

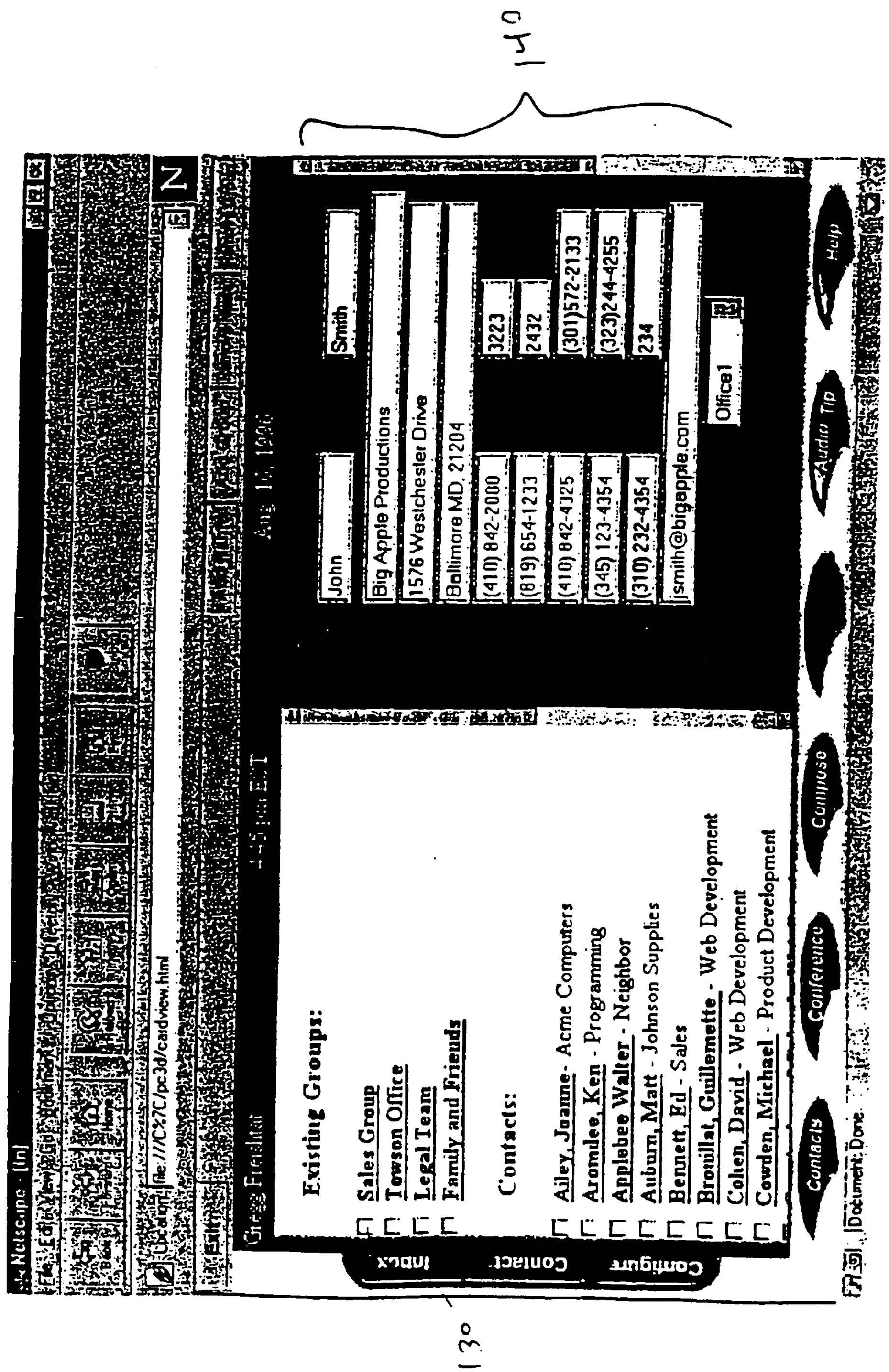
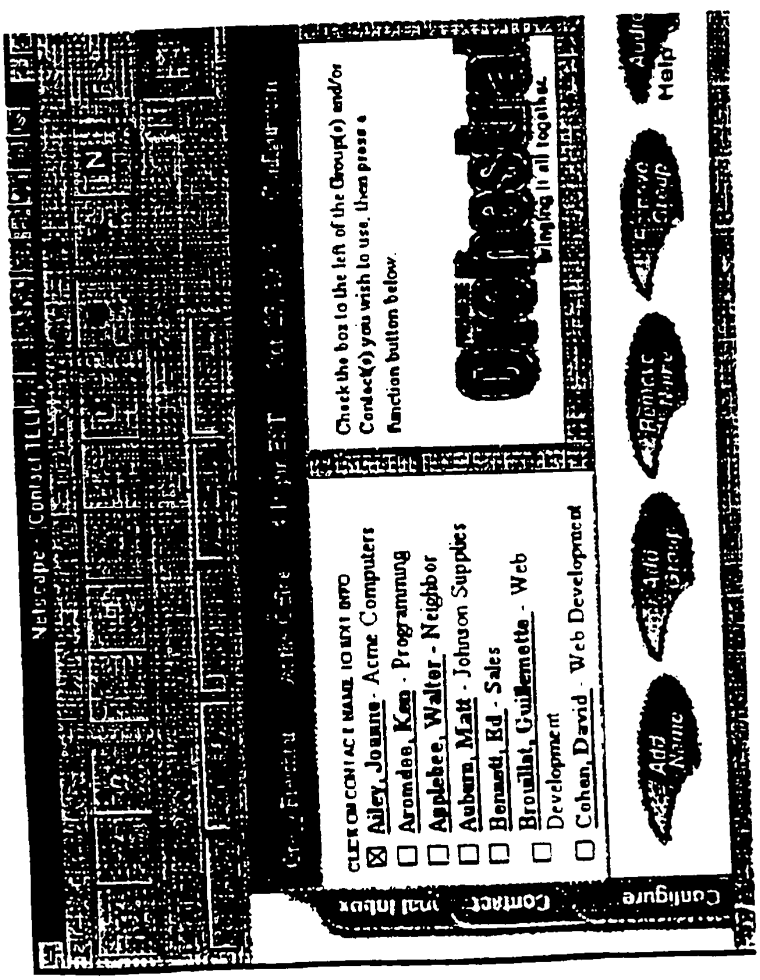
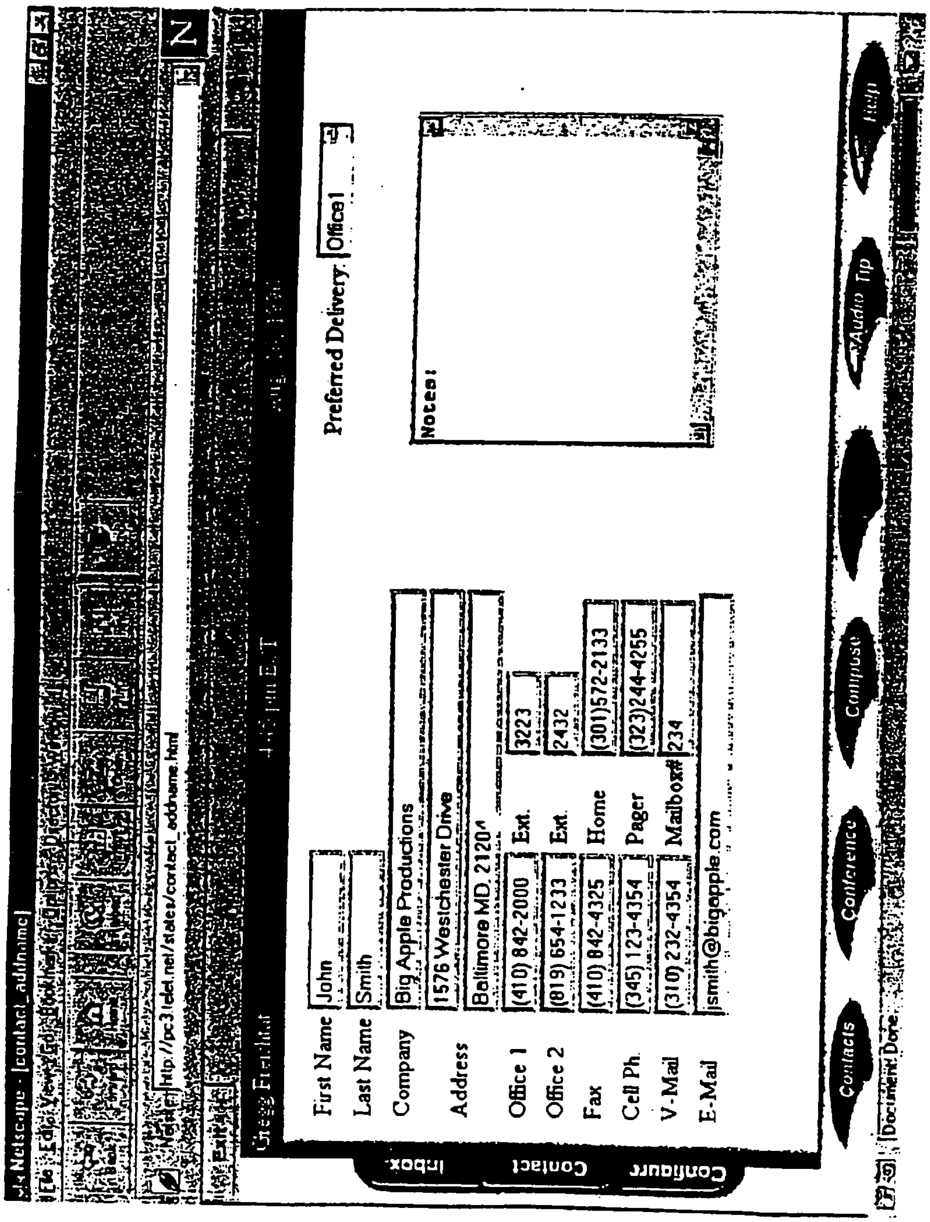

140
130
Netscape
Gregg Freishtat
Aug 16, 1996
Existing Groups:
Sales Group
Towson Office
Legal Team
Family and Friends
Contacts:
Ailey, Joanne - Acme Computers
Aromdee, Ken - Programming
Applebee Walter - Neighbor
Auburn, Matt - Johnson Supplies
Bennett, Ed - Sales
Brouillat, Guillemette - Web Development
Cohen, David - Web Development
Cowden, Michael - Product Development
John
Smith
Big Apple Productions
1576 Westchester Drive
Baltimore MD 21204
(410) 842-2000
3223
(819) 654-1233
2432
(410) 842-4325
(301)572-2133
(345) 123-4354
(323)244-4255
(310) 232-4354
234
jsmith@bigapple.com
Office1
Index
Contact
Configure
Contacts
Conference
Compose
Audio Tip
Help
Document: Done

Fig. 19


Netscape - [Contact
Check the box to the left of the Group(s) and/or Contact(s) you wish to use, then press a function button below.
Bringing it all together.
CLICK ON CONTACT NAME TO EDIT INFO
Ailey, Joanne - Acme Computers
Aromdee, Ken - Programming
Applebee, Walter - Neighbor
Auburn, Matt - Johnson Supplies
Bennett, Ed - Sales
Brouillet, Guillemette - Web Development
Cohen, David - Web Development
Add Name
Add Group
Help
Contact
Configure

150

Fig. 20


Netscape - [contact_addname]
First Name
John
Last Name
Smith
Company
Big Apple Productions
Address
1576 Westchester Drive
Office 1
(410) 842-2000
Ext.
3223
Office 2
(819) 654-1233
Ext.
2432
Fax
(410) 842-4325
Home
(301)572-2133
Cell Ph.
(345) 123-4354
Pager
(323)244-4255
V-Mail
(310) 232-4354
Mailbox#
234
E-Mail
jsmith@bigapple.com
Preferred Delivery:
Office1
Notes:
Inbox
Contact
Configure
Contacts
Conference
Compose
Audio Tip
Help
Document: Done

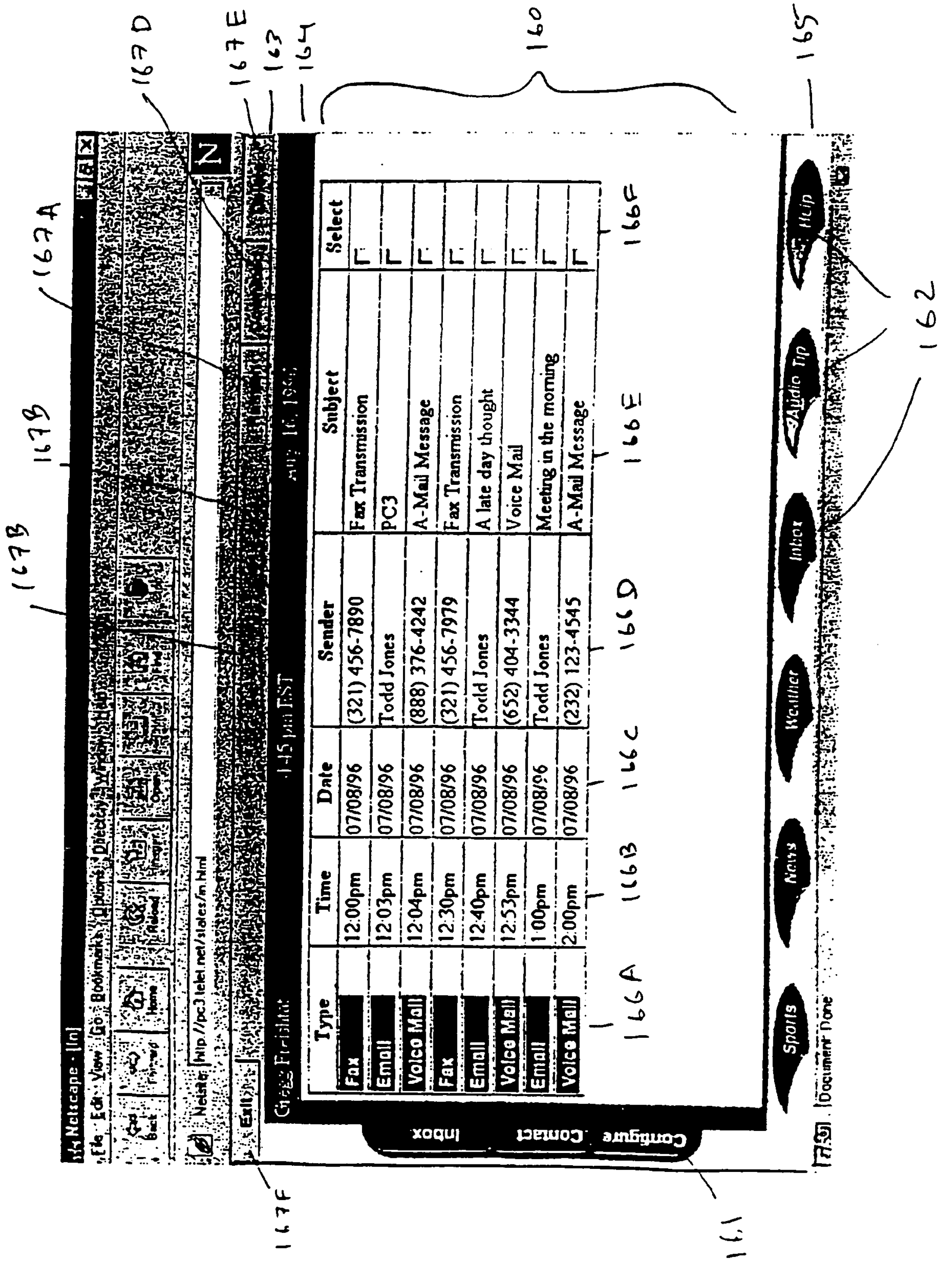

Netscape - [In]
http://pc3.telet.net/states/in.html
1:45 pm EST
Type
Time
Date
Sender
Subject
Select
Fax
Email
Voice Mail
12:00pm
12:03pm
12:04pm
12:30pm
12:40pm
12:53pm
1:00pm
2:00pm
07/08/96
(321) 456-7890
Todd Jones
(888) 376-4242
(321) 456-7979
(652) 404-3344
(232) 123-4545
Fax Transmission
PC3
A-Mail Message
A late day thought
Voice Mail
Meeting in the morning
Inbox
Contact
Configure
Sports
News
Weather
Audio Tip
Help
Document Done
160
161
162
163
164
165
166A
166B
166C
166D
166E
166F
167A
167B
167D
167E
167F

Fig. 21

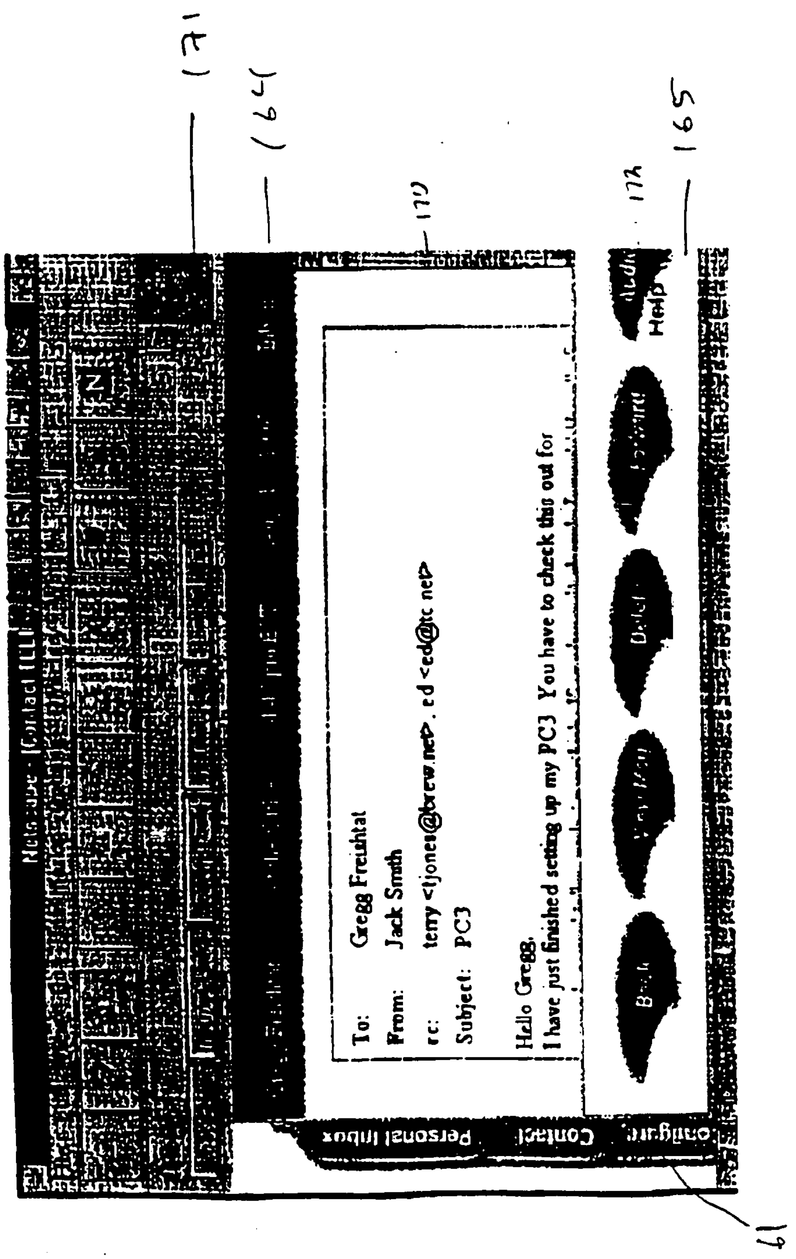

171
164
170
172
165
161
To: Gregg Freishtat
From: Jack Smith
cc: terry <tjones@brew.net>, ed <ed@tc net>
Subject: PC3
Hello Gregg,
I have just finished setting up my PC3 You have to check this out for
Help
Personal Inbox
Contact
Configure

Fig. 22A

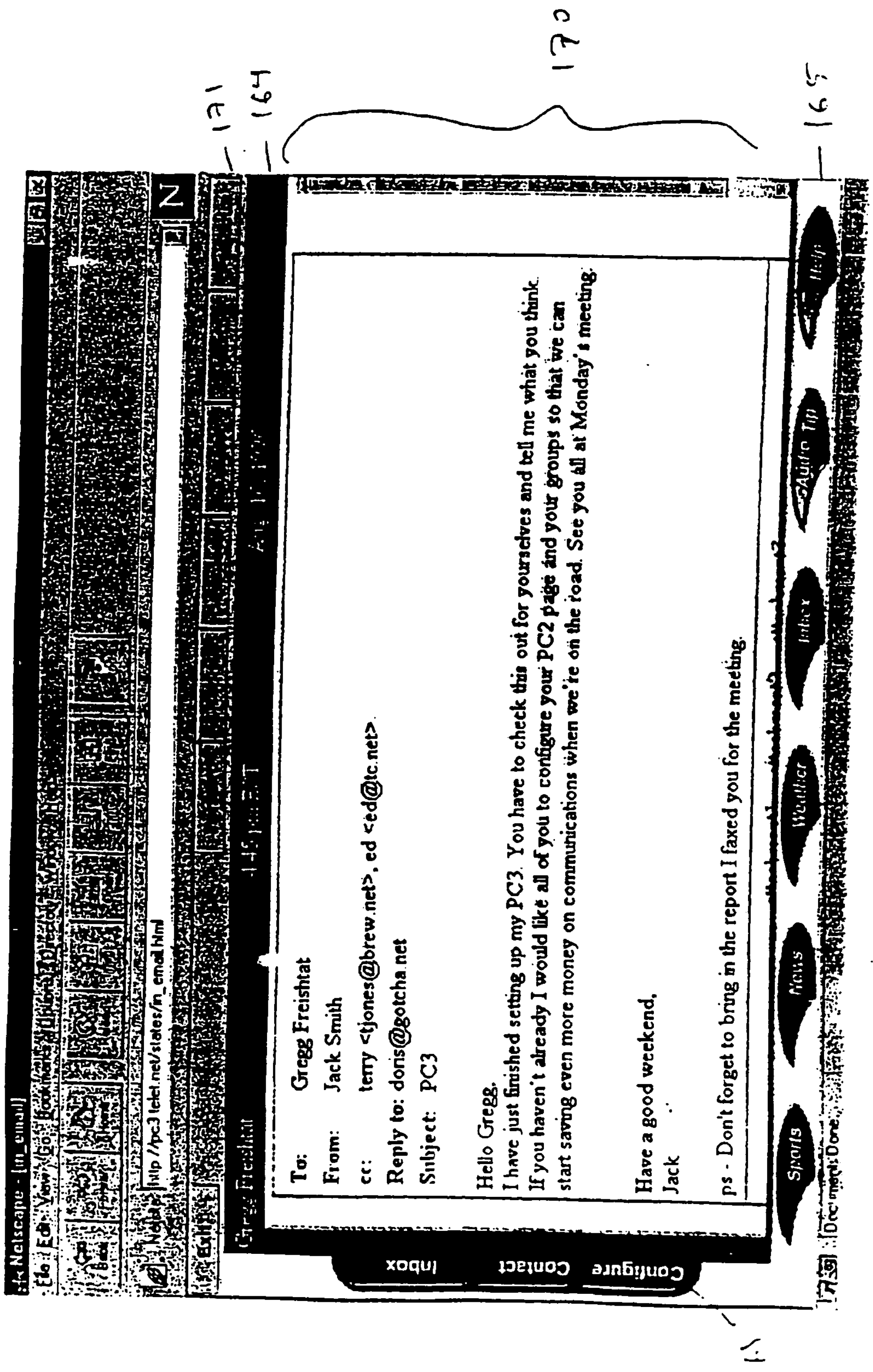

Netscape - [in_email]
http://pc3.telel.net/stoles/in_email.html
Gregg Freishtat
To: Gregg Freishtat
From: Jack Smith
cc: terry <tjones@brew.net>, ed <ed@tc.net>
Reply to: doris@gotcha.net
Subject: PC3
Hello Gregg,
I have just finished setting up my PC3. You have to check this out for yourselves and tell me what you think.
If you haven't already I would like all of you to configure your PC2 page and your groups so that we can
start saving even more money on communications when we're on the road. See you all at Monday's meeting.
Have a good weekend,
Jack
ps - Don't forget to bring in the report I faxed you for the meeting.
Sports
News
Weather
Inbox
Audio Tip
Help
Inbox
Contact
Configure
Document: Done
171
164
170
165

Fig. 22B

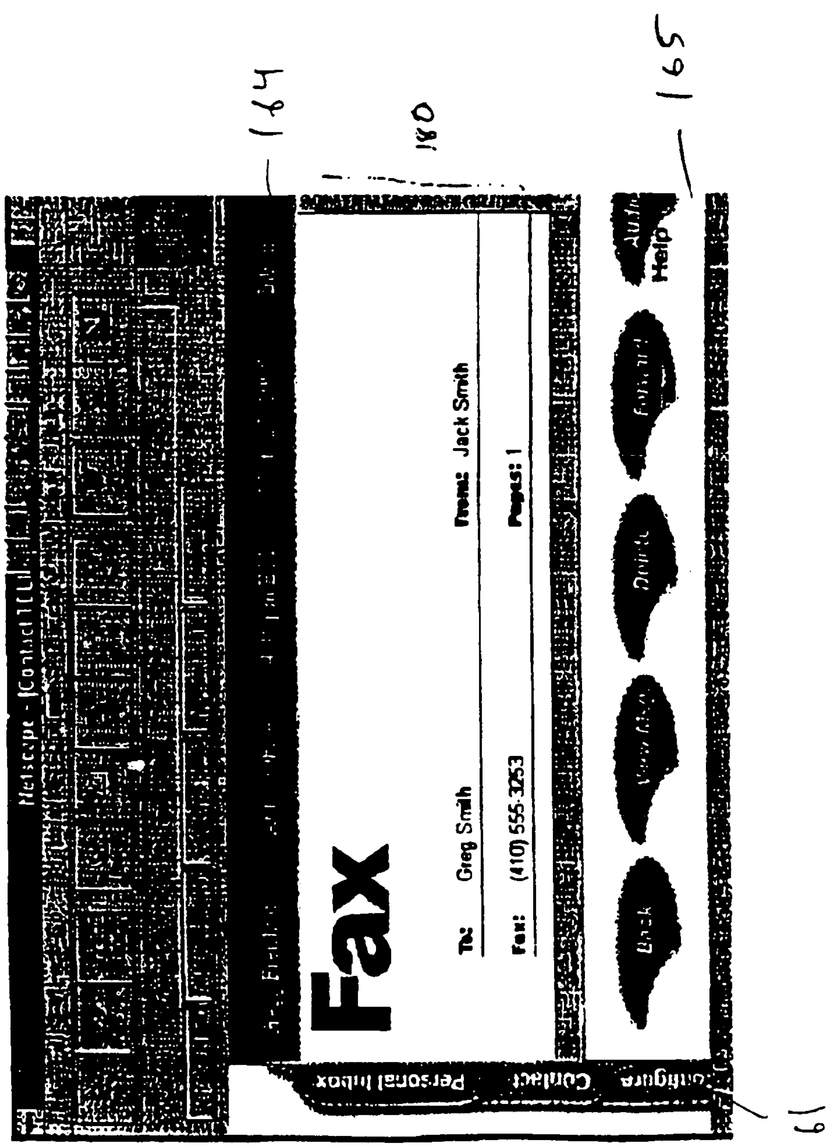

164
180
165
161
Fax
To: Greg Smith
Fax: (410) 555-3253
From: Jack Smith
Pages: 1
Help
Personal Inbox
Contact

Fig. 23A

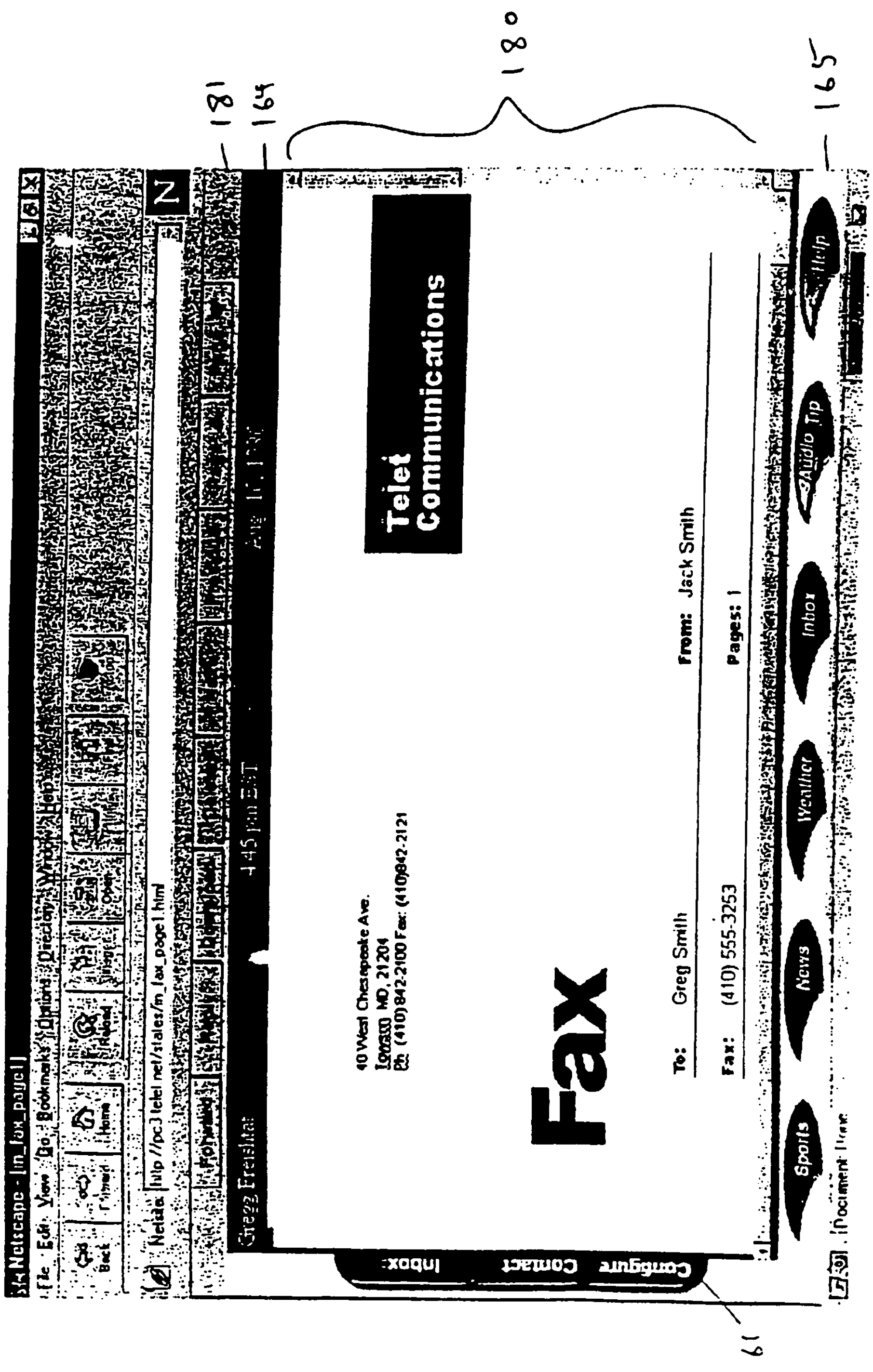
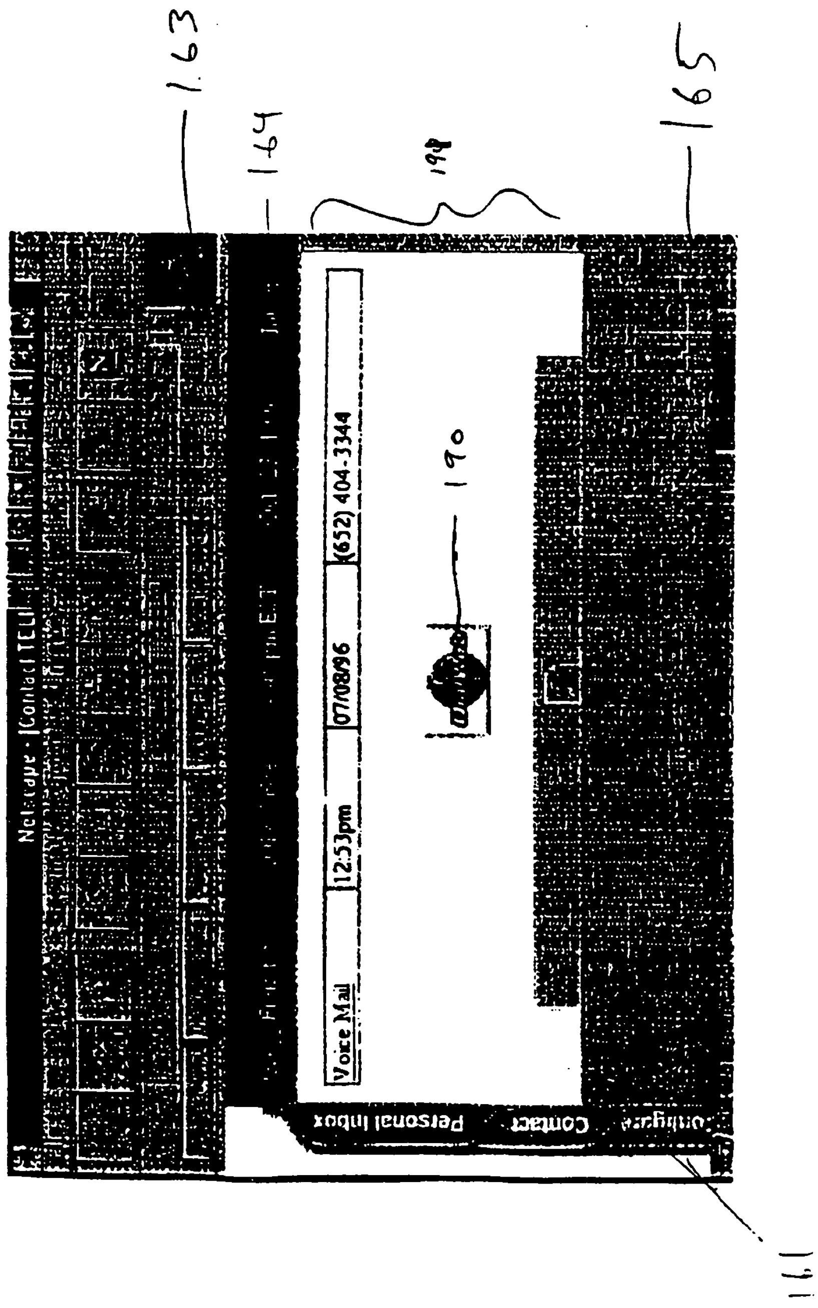
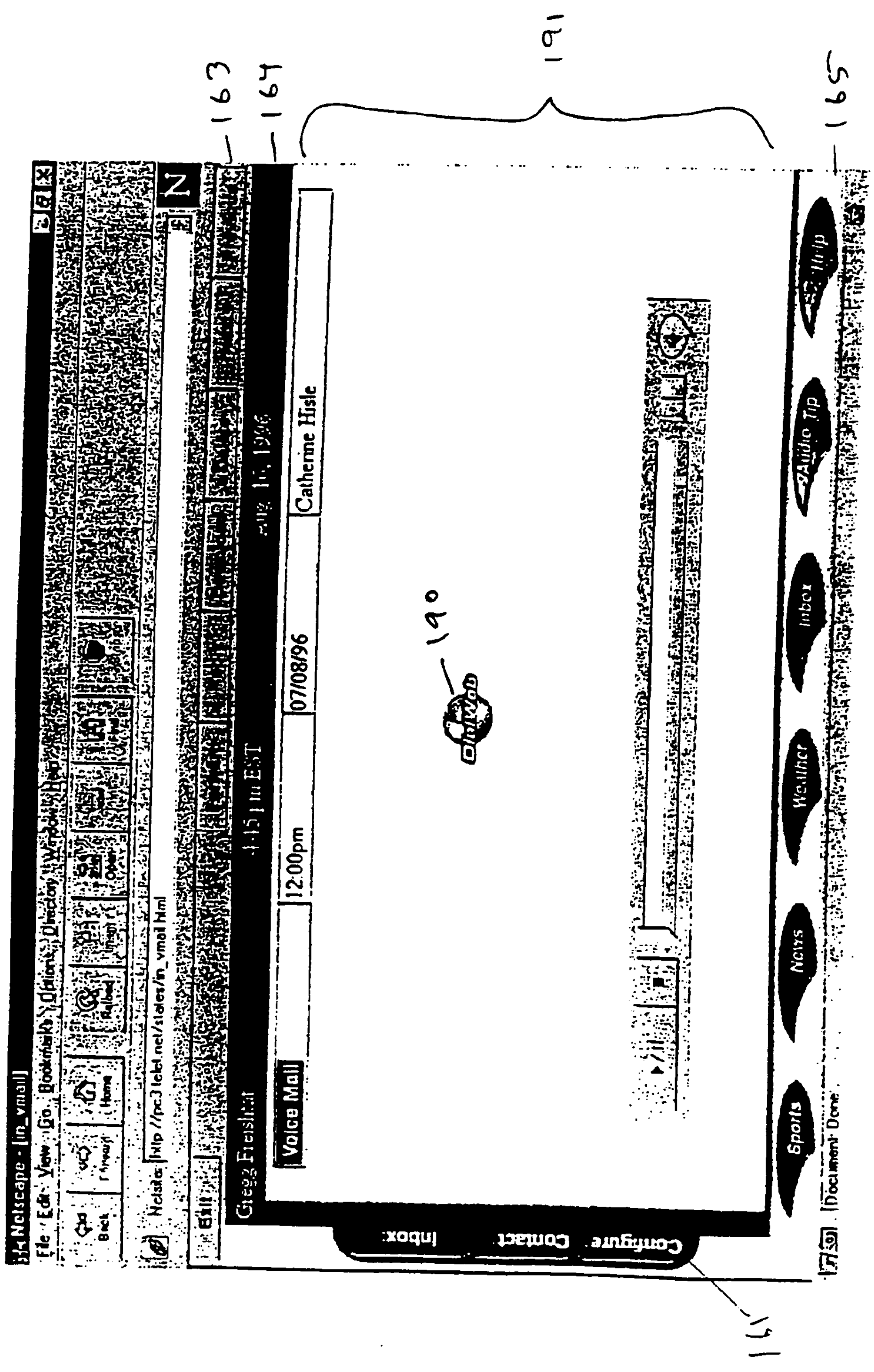

Fax
To: Greg Smith
From: Jack Smith
Fax: (410) 555-3253
Pages: 1
Sports
News
Weather
Inbox
Audio Tip
Help

Fig. 23B



163
164
165
190
161
Voice Mail
12:53pm
07/08/96
(652) 404-3344
Personal Inbox
Contact

Fig. 24A



163
164
191
165
190
171
Netscape - [in_vmail]
Voice Mail
12:00pm
07/08/96
Catherine Hisle
Sports
News
Weather
Inbox
Audio Tip
Help
Configure
Contact
Inbox:
Document: Done

Fig. 24B

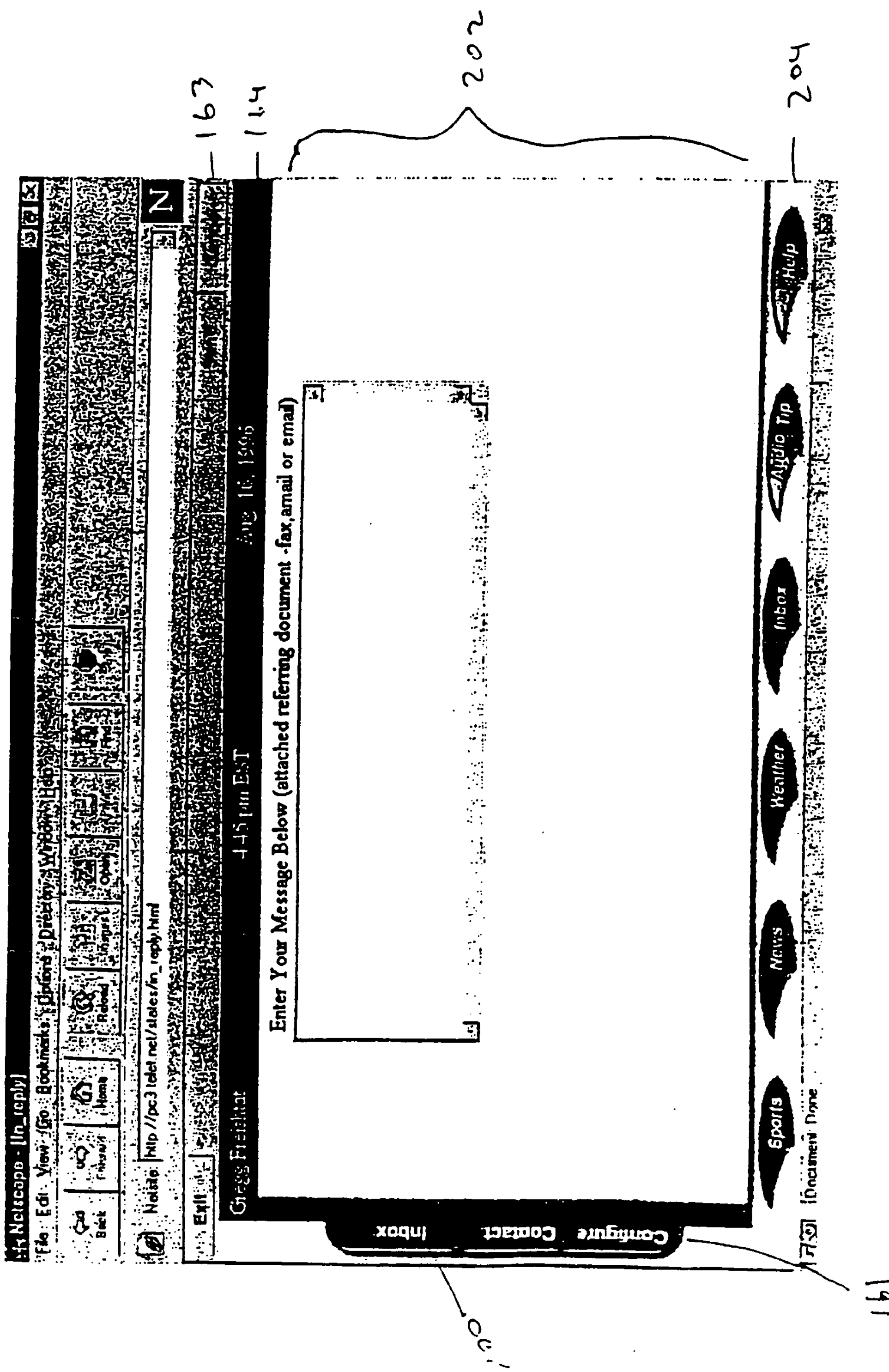

163
114
202
204
161
Netscape - [in_reply]
Gregg Freishtat
4:45 pm EST
Aug 16, 1996
Enter Your Message Below (attached referring document -fax,amail or email)
Sports
News
Weather
Inbox
Audio Tip
Help
Configure
Contact
Inbox
Document: Done

Fig. 25A

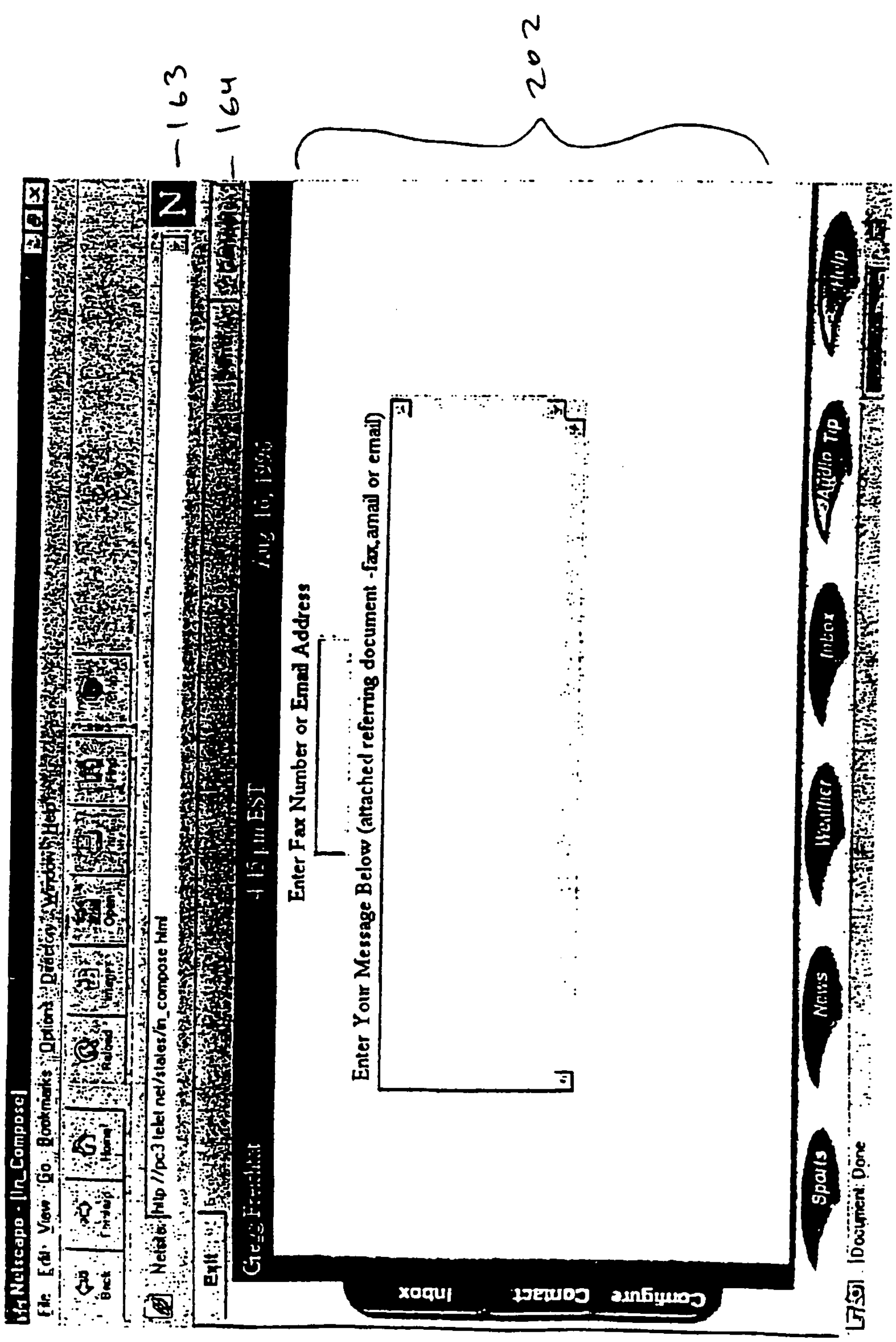

163
164
202
Enter Fax Number or Email Address
Enter Your Message Below (attached referring document -fax,email or email)
Sports
News
Weather
Inbox
Audio Tip
Help
Inbox
Contact
Configure
Document Done

Fig. 25B

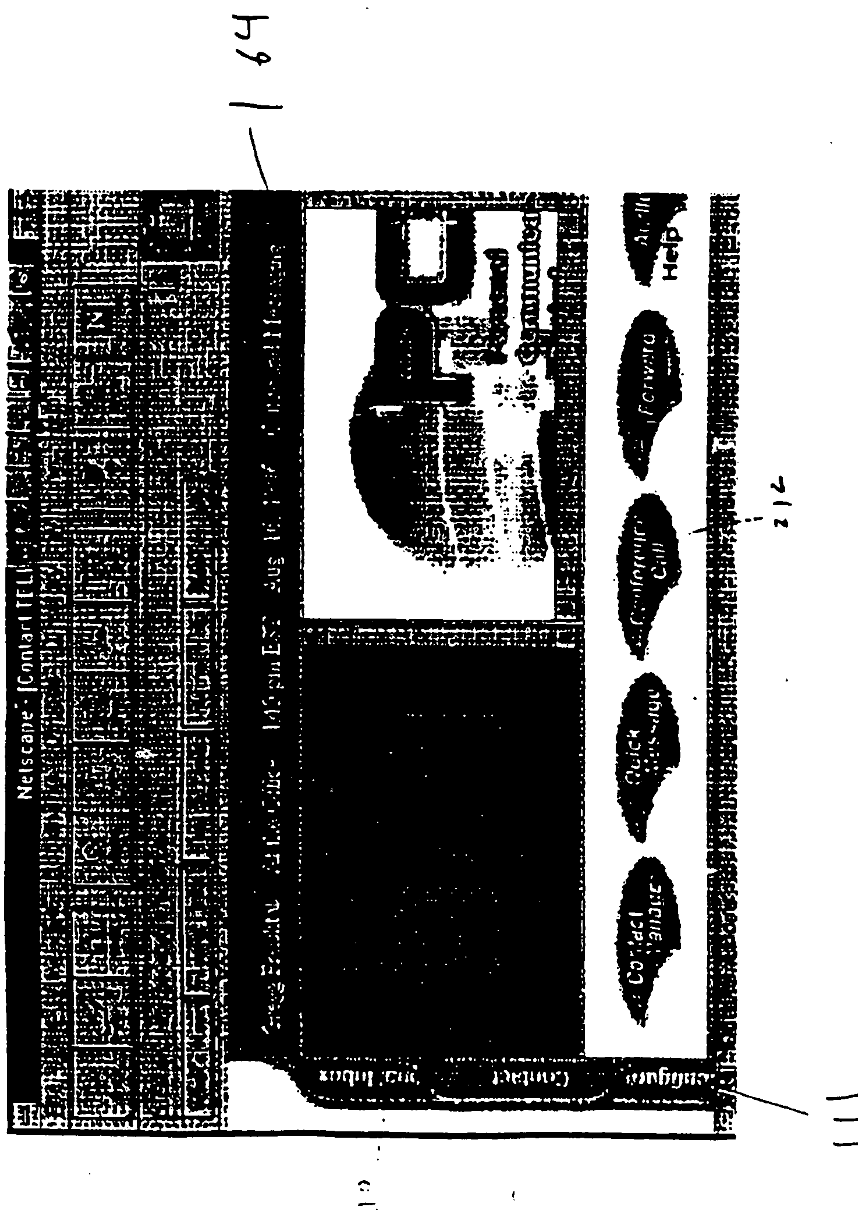

164
Netscape - [Contact TELL
Contact
Quick
Conference Call
Forward
Help
Configure
Contact
Inbox
212
116

Fig. 26

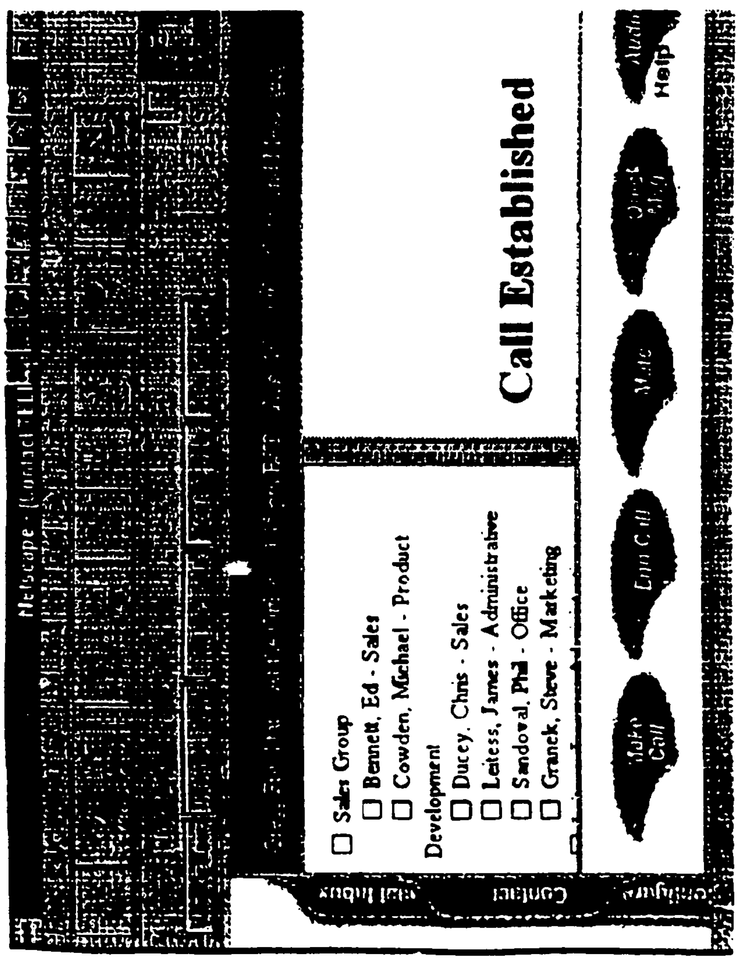

Sales Group
Bennett, Ed - Sales
Cowden, Michael - Product
Development
Ducey, Chris - Sales
Leiless, James - Administrative
Sandoval, Phil - Office
Granek, Steve - Marketing
Call Established
Make Call
End Call
Mute
Help
Contact

Fig. 27

PCT

WORLD INTELLECTUAL PROPERTY ORGANIZATION
International Bureau

INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

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(11) International Publication Number: WO 98/23058

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60/031,301 18 November 1996 (18.11.96) US

(71) Applicant *(for all designated States except US)*: PREMIERE COMMUNICATIONS, INC. [US/US]; The Lenox Building, Suite 400, 3399 Peachtree Road, Atlanta, GA 30326 (US).

(72) Inventors; and
(75) Inventors/Applicants *(for US only)*: FREISHTAT, Gregg, S. [US/US]; 2078 Renfroe Lake Drive, Dunwoody, GA 30338 (US). LEITESS, James, K. [US/US]; 929 Highland Terrace N.E., Atlanta, GA 30306 (US). COWDEN, Michael, J. [US/US]; Unit B-317, 1074 Peachtree Walk, Atlanta, GA 30309 (US). SMITH, David, Gregory [US/US]; 1907 Oakmont, Tampa, FL 33629 (US).

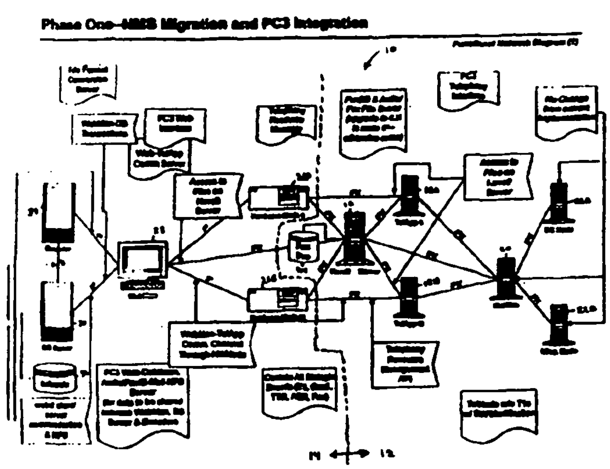
(74) Agents: PRATT, John, S. et al.; Kilpatrick Stockton LLP, Suite 2800, 1100 Peachtree Street, Atlanta, GA 30309-4530 (US).

(81) Designated States: AL, AU, BA, BB, BG, BR, CA, CN, CU, CZ, EE, GE, GH, HU, ID, IL, IS, JP, KP, KR, LC, LK, LR, LT, LV, MG, MK, MN, MX, NO, NZ, PL, RO, SG, SI, SK, SL, TR, TT, UA, US, UZ, VN, YU, ARIPO patent (GH, KE, LS, MW, SD, SZ, UG, ZW), Eurasian patent (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European patent (AT, BE, CH, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE), OAPI patent (BF, BJ, CF, CG, CI, CM, GA, GN, ML, MR, NE, SN, TD, TG).

Published
- *With international search report.*
Before the expiration of the time limit for amending the claims and to be republished in the event of the receipt of amendments.

(88) Date of publication of the international search report: 3 September 1998 (03.09.98)

(54) Title: SYSTEM FOR INTEGRATED MANAGEMENT OF MESSAGING AND COMMUNICATIONS



(57) Abstract

System (10) for providing to a user an integrated interface for accessing telephony- and computer network-based communications resources. A user may access the system (10) via telephone (12), computer, modem or Internet (14). The user may access messages addressed to him or her in various media, such as e-mail (36), voice mail, fax, etc. The user may respond to these messages in the same media, or different media, including real-time communications such as direct telephony or conference calling. The interface may also allow the user to present others with a web page (28) which displays information selected by the user. The system may consist of a stand alone system which interfaces with the Internet (14) and the telephony network (12). The system may also be integrated with existing telephony-based communication management infrastructures.

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Codes used to identify States party to the PCT on the front pages of pamphlets publishing international applications under the PCT.

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BE	Belgium	GN	Guinea	MK	The former Yugoslav Republic of Macedonia	TM	Turkmenistan
BF	Burkina Faso	GR	Greece			TR	Turkey
BG	Bulgaria	HU	Hungary	ML	Mali	TT	Trinidad and Tobago
BJ	Benin	IE	Ireland	MN	Mongolia	UA	Ukraine
BR	Brazil	IL	Israel	MR	Mauritania	UG	Uganda
BY	Belarus	IS	Iceland	MW	Malawi	US	United States of America
CA	Canada	IT	Italy	MX	Mexico	UZ	Uzbekistan
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CG	Congo	KE	Kenya	NL	Netherlands	YU	Yugoslavia
CH	Switzerland	KG	Kyrgyzstan	NO	Norway	ZW	Zimbabwe
CI	Côte d'Ivoire	KP	Democratic People's Republic of Korea	NZ	New Zealand		
CM	Cameroon			PL	Poland		
CN	China	KR	Republic of Korea	PT	Portugal		
CU	Cuba	KZ	Kazakstan	RO	Romania		
CZ	Czech Republic	LC	Saint Lucia	RU	Russian Federation		
DE	Germany	LI	Liechtenstein	SD	Sudan		
DK	Denmark	LK	Sri Lanka	SE	Sweden		
EE	Estonia	LR	Liberia	SG	Singapore		

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US97/20986

A. CLASSIFICATION OF SUBJECT MATTER

IPC(6) :H04L 12/28, 12/56
US CL :370/352
According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

U.S. : 370/352, 383, 389, 390, 392, 401, 408, 410; 379/ 89, 90.01, 03.01, 93.07, 93.08, 93.14, 93.29, 100.11, 100.13, 114

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

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

APS (integrated interface, telephony, computer, communications resources, Internet, e-mail, voice mail, fax, real-time communications)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	US 4,771,425 A (BARAN et al) 13 September 1988.	1-83
Y, P	US 5,610,910 A (FOCSANEANU et al) 11 March 1997, col. 2, line 59 to col. 3, line 22; col. 9, lines 14-39; col. 14, lines 13-20	1, 16, 30, 43, 44, 51, 63, 72
Y, P	US 5,608,786 A (GORDON) 04 March 1997, col. 3, lines 9-29; col. 7, lines 51-67.	1, 16, 30, 43, 44, 51, 63, 72
X	YANG, C. INETPhone: Telephone Services and Servers on Internet, RFC 1789, April 1995.	1-83

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

* Special categories of cited documents:

"A" document defining the general state of the art which is not considered to be of particular relevance

"E" earlier document published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

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"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
09 APRIL 1998	08 JUL 1998

Name and mailing address of the ISA/US	Authorized officer
Commissioner of Patents and Trademarks Box PCT Washington, D.C. 20231	WELLINGTON CHIN
Facsimile No. (703) 305-3230	Telephone No. (703) 305-3900

Form PCT/ISA/210 (second sheet)(July 1992)★

This Page Blank (uspto)

Electronic Patent Application Fee Transmittal

Application Number:	
Filing Date:	
Title of Invention:	Acquiring Information from Sources Responsive to Naturally-Spoken-Speech Commands Provided by a Voice-Enabled Device
First Named Inventor/Applicant Name:	Alexander Kurganov
Filer:	Reena Kuyper/Kristin Takahashi
Attorney Docket Number:	10115-07594 US

Filed as Small Entity

Filing Fees for Utility under 35 USC 111(a)

Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
Basic Filing:				
UTILITY FILING FEE (ELECTRONIC FILING)	4011	1	75	75
UTILITY SEARCH FEE	2111	1	330	330
UTILITY EXAMINATION FEE	2311	1	380	380
Pages:				
Claims:				
Miscellaneous-Filing:				
Petition:				
Patent-Appeals-and-Interference:				

Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
Post-Allowance-and-Post-Issuance:				
Extension-of-Time:				
Miscellaneous:				
		Total in USD ($)		**785**

Electronic Acknowledgement Receipt

EFS ID:	33955145
Application Number:	16155523
International Application Number:	
Confirmation Number:	9544
Title of Invention:	Acquiring Information from Sources Responsive to Naturally-Spoken-Speech Commands Provided by a Voice-Enabled Device
First Named Inventor/Applicant Name:	Alexander Kurganov
Customer Number:	93219
Filer:	Reena Kuyper
Filer Authorized By:	
Attorney Docket Number:	10115-07594 US
Receipt Date:	09-OCT-2018
Filing Date:	
Time Stamp:	17:12:18
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	yes
Payment Type	CARD
Payment was successfully received in RAM	$785
RAM confirmation Number	101018INTEFSW17151100
Deposit Account	
Authorized User	

The Director of the USPTO is hereby authorized to charge indicated fees and credit any overpayment as follows:

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Application Data Sheet	07594US_20181009_ADS.pdf	1256927 a095472a8e9ef86e0a8ce22afae7483929dd0333	no	9
Warnings:					
Information:					
2	Power of Attorney	07594US_20181009_POA.pdf	215909 6e56ec243d7e6d70fc0abb5e34a44de5d9b061d6	no	3
Warnings:					
Information:					
3	Oath or Declaration filed	07594US_20181009_Declaration.pdf	103586 c5f202d707351b035edc65eccb43e22f7dd92f62	no	1
Warnings:					
Information:					
4		07594US_20181009_Specification.pdf	251421 7214e44ccbc89d188154fc3ebfc8e731651c5c8e	yes	44

Multipart Description/PDF files in .zip description

Document Description	Start	End
Specification	1	36
Claims	37	43
Abstract	44	44

Warnings:					
Information:					
5	Drawings-only black and white line drawings	07594US_Drawings.pdf	898122 4df43e97a5bcc69e829b2baed569a54825ab8ce5	no	4

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Information:					
6	Information Disclosure Statement (IDS) Form (SB08)	07594US_20181009_IDS1.pdf	1041436 0e02906c9e822629958ed601878664cfb556ed03	no	30
Warnings:					
Information:					
7	Information Disclosure Statement (IDS) Form (SB08)	07594US_20181009_IDS2.pdf	1040622 e5905b3275ac152e98e67c6157e459ca2d78c54e	no	19
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8	Non Patent Literature	_BusinessPhoneSystemsforAdvancedOffices19901100.pdf	190116 bfbac39251df5f8d46fda71a9fd45cc6793ff4a3	no	3
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9	Non Patent Literature	_DataCommunicationsNetworksMessageHandlingSystems19841008.pdf	1921276 9b9a92a0b2754fe58f7bb98d8f65c27f76b79c6c	no	42
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10	Non Patent Literature	_FaxpakStoreandForwardFacsimileTransmissionService19790000.pdf	392847 1fcd6eb04be88fc8d8c84c232d1115d8397a07df	no	5
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11	Non Patent Literature	_Globecom85IEEEGlobalTelecommunicationsConference19851202.pdf	351326 30b142ad8513b0af15f7e7b19be886dedac3023c	no	7
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12	Non Patent Literature	_IntroducingPICSuperFaxFirstPCFaxSystemtoRunUnderWindows19871103.pdf	52969 b2267b99559d0dbe0e94ac2f692f7f851b48c507	no	1
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13	Non Patent Literature	_JugglerbyPureSpeech20061208.pdf	31413 8a4416c1ce96d5a28145ca0df24c4c1fe8fb8c40	no	1
Warnings:					
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14	Non Patent Literature	_ProceedingsoftheIFIP10thWorldComputerCongress19860901.pdf	1205234 39175425cdb1759ba81881ad9fda6280a14ff4f0	no	16
Warnings:					
Information:					
15	Non Patent Literature	_PureSpeechAnnouncesJugglerPCSystemforFirstQuarterof1997_19960919.pdf	75249 c5a9a5b5651543f2af30d5afedd7208c47a8ce6a	no	1
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Information:					
16	Non Patent Literature	_PurespeechsJugglerAdonpage36ofTeleconnect19961200.pdf	885705 aa8351c0e0867e9dc3b67875cf33cc33a1b03f11	no	2
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18	Non Patent Literature	_WildlfireCommunicationsInc2.pdf	3078377 9a51f402c49a2fc3e3961eaf2363bc6ba130be79	no	23
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Information:					
19	Non Patent Literature	_WordPerfectNewTelephoneyFeaturesBoostOffice.pdf	342511 d5511913dbbf9e40287558ebc51b468a73239caa	no	2
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Information:					

20	Non Patent Literature	APABXthatListensandtalks.pdf	642955 d69febe7ec6a49173bc65432ea2924b447e7219c	no	6
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22	Non Patent Literature	ATTPressReleaseATTCustomersCanTeachSystemstoListenandResponsetoVoice.pdf	91514 22098e0dd9c46d681cf96703e25ec3d984a2adb1	no	2
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24	Non Patent Literature	BrachmanetalFragmentationinStoreandForwardMessageTransfer.pdf	1118487 13ea1cd6b58b66bc76f3cf83b3fdfb371df01e5e	no	10
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25	Non Patent Literature	_TheVMXSystemsProductReferenceManual19940500.pdf	19102843 87990c94f0d42b2d4e68c16bb255c0bd8ecb6fbb	no	577
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27	Foreign Reference	CA1329852AudiofaxInc.pdf	3421682 4ff7de92b825181a50d1214b202f9fd303c07966	no	42
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28	Foreign Reference	EP0572544VMXInc.pdf	2422730 d50bf1e486e18364793819e8682574fd8c1ed9fd	no	54
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29	Foreign Reference	EP0794650A2InternationalBusinessMachinesCorporation.pdf	917208 81727b4a0302a1455280bd6db1481b7f1fd68e0c	no	13
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30	Foreign Reference	GB2211698AmericanTelephoneandTelegraphCompany.pdf	758444 1758b725380d36e47b6e32e6c55b489508fb6b81	no	18
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33	Foreign Reference	JP10821690RicohCoLtd.pdf	260607 8ab4bad4dc547e16b202a751721d3ac2a8a6aba9	no	6
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34	Foreign Reference	WO9107838FullerResearchandDevelopmentCompany.pdf	8663234 89497450703de6e4afdf9946e096247f844496c6	no	177
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35	Foreign Reference	WO9118466MessagerPartners.pdf	767970 8cb23b956beef14eeb8ed2f58670bbcbea44c4ba	no	20
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36	Foreign Reference	WO9609710OctelCommunicationsCorporation.pdf	5203536 2aa58b7cb9912ed5aafaf8160b05c842a8eabc40	no	150
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37	Foreign Reference	WO9734401XantelCorporation.pdf	6493839 c8fe8e540d4d3861fbe4866e3f48e0c0e71928ed	no	177
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38	Foreign Reference	WO9737481NorthernTelecomLimited.pdf	889186 acb8f28b80528b3309a89b105ade2557f5de2270	no	26
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39	Foreign Reference	WO9823058PremiereCommunicationsInc.pdf	3601176 6ff40b96ac92febbf685799522f32e8a24ff6a8b	no	110
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40	Non Patent Literature	BriefofAppellees.pdf	437815 3875f8f4cd918d6a105d4401ee77407d6c8b726d	no	53
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Information:					

41	Non Patent Literature	COLEetalAnArchitectureforaMobileOSIMailAccressSystem.pdf	901851 d4fe260b69f562f384c7ca8d38dd73828de80f27	no	8
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42	Non Patent Literature	DAXSYSTEMSINCPressReleaseSpeechRecognitionSuccessinDaxsGrasp.pdf	91627 f5e6152b230b568dbb90af9a930788ad2c190f4b	no	2
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43	Non Patent Literature	DefendantsAnswertotheAmendedComplaintandDemandforJuryTrial.pdf	53574 dd5a4d54126873599c2ffa13477fbc2d953c24e1	no	1
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44	Non Patent Literature	ExamplesAbstractIdeas.pdf	254755 8c83a5913995027e4701dc384031a085e2be1591	no	20
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48	Non Patent Literature	HUNTetalLongDistanceRemoteControltotheRescue.pdf	98757 59bd811235d80f73adbb62b9aa8c9315fe522fe4	no	1
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49	Non Patent Literature	IBMAIXDirectTalk6000GeneralInfoPt1.pdf	16894452 9fe4bbf03f073045d1fefebdc24481b6dc34a783	no	79
Warnings:					
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50	Non Patent Literature	IBMAIXDirectTalk6000GeneralInfoPt2.pdf	18414055 c1a9ce0deb4bf00a1a553a132588ea3cacd520f6	no	83
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51	Non Patent Literature	IBMAIXDirectTalk6000Release6Version1ImprovesYourVoiceProcessing.pdf	293533 504a4ab1a97873f0a9df997c261f805d26ddb185	no	27
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52	Non Patent Literature	IBMAnnouncementLetter.pdf	251042 33273740c0bf1573d63c40e7c1968d1059aa4b52	no	10
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53	Non Patent Literature	IBMDirectTalkMailAdministrationPt1.pdf	13519332 afee77f3a7ef8330698e3b8aa1a49ff0a2d0f0ed	no	84
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54	Non Patent Literature	IBMDirectTalkMailAdministrationPt2.pdf	24901376 55e6d8bc17afc6d1732ec4156f28a60596c80f1e	no	190
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55	Non Patent Literature	JointAppendix.pdf	9332456 0b4cc75bbf17d4ab7c8e8f941530a799a9031329	no	406
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56	Non Patent Literature	JudgementwithoutOpinion.pdf	39709 ae70db2b2405895fc04ac6311994a25861bb7fa4	no	2
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57	Non Patent Literature	KUBALAetalBYBLOSSpeechRecognitionBenchmarkResults.pdf	566502 e86223087e8fa0f6c1ea77d2f28e44a4f7854cda	no	6
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59	Fee Worksheet (SB06)	fee-info.pdf	35707 e5342f62c67391c3c473db5c58492ae39d1c1410	no	2
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Information:					
		Total Files Size (in bytes):	183800881		

This Acknowledgement Receipt evidences receipt on the noted date by the USPTO of the indicated documents, characterized by the applicant, and including page counts, where applicable. It serves as evidence of receipt similar to a Post Card, as described in MPEP 503.

<u>New Applications Under 35 U.S.C. 111</u>
If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.
<u>National Stage of an International Application under 35 U.S.C. 371</u>
If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.
<u>New International Application Filed with the USPTO as a Receiving Office</u>
If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.

Application Data Sheet 37 CFR 1.76	Attorney Docket Number	10115-07594 US
	Application Number	
Title of Invention	Acquiring Information from Sources Responsive to Naturally-Spoken-Speech Commands Provided by a Voice-Enabled Device	

The application data sheet is part of the provisional or nonprovisional application for which it is being submitted. The following form contains the bibliographic data arranged in a format specified by the United States Patent and Trademark Office as outlined in 37 CFR 1.76.
This document may be completed electronically and submitted to the Office in electronic format using the Electronic Filing System (EFS) or the document may be printed and included in a paper filed application.

Secrecy Order 37 CFR 5.2:

☐ Portions or all of the application associated with this Application Data Sheet may fall under a Secrecy Order pursuant to 37 CFR 5.2 (Paper filers only. Applications that fall under Secrecy Order may not be filed electronically.)

Inventor Information:

Inventor 1 [Remove]

Legal Name

Prefix	Given Name	Middle Name	Family Name	Suffix
	Alexander		Kurganov	

Residence Information (Select One) ● US Residency ○ Non US Residency ○ Active US Military Service

City	Buffalo Grove	State/Province	IL	Country of Residence	US

Mailing Address of Inventor:

Address 1	c/o Parus Holding, Inc.		
Address 2	3000 Lakeside Drive, Suite 110S		
City	Bannockburn	State/Province	IL
Postal Code	60015	Country	US

All Inventors Must Be Listed - Additional Inventor Information blocks may be generated within this form by selecting the **Add** button. [Add]

Correspondence Information:

Enter either Customer Number or complete the Correspondence Information section below. For further information see 37 CFR 1.33(a).

☐ **An Address is being provided for the correspondence Information of this application.**

Customer Number	93219
Email Address	docketing@patentlawworks.net [Add Email] [Remove Email]

Application Information:

Title of the Invention	Acquiring Information from Sources Responsive to Naturally-Spoken-Speech Commands Provided by a Voice-Enabled Device		
Attorney Docket Number	10115-07594 US	Small Entity Status Claimed	☒
Application Type	Nonprovisional		
Subject Matter	Utility		
Total Number of Drawing Sheets (if any)		Suggested Figure for Publication (if any)	

Application Data Sheet 37 CFR 1.76	Attorney Docket Number	10115-07594 US
	Application Number	

Title of Invention	Acquiring Information from Sources Responsive to Naturally-Spoken-Speech Commands Provided by a Voice-Enabled Device

Filing By Reference:

Only complete this section when filing an application by reference under 35 U.S.C. 111(c) and 37 CFR 1.57(a). Do not complete this section if application papers including a specification and any drawings are being filed. Any domestic benefit or foreign priority information must be provided in the appropriate section(s) below (i.e., "Domestic Benefit/National Stage Information" and "Foreign Priority Information").

For the purposes of a filing date under 37 CFR 1.53(b), the description and any drawings of the present application are replaced by this reference to the previously filed application, subject to conditions and requirements of 37 CFR 1.57(a).

Application number of the previously filed application	Filing date (YYYY-MM-DD)	Intellectual Property Authority or Country

Publication Information:

- ☐ Request Early Publication (Fee required at time of Request 37 CFR 1.219)
- ☐ **Request Not to Publish.** I hereby request that the attached application not be published under 35 U.S.C. 122(b) and certify that the invention disclosed in the attached application **has not and will not** be the subject of an application filed in another country, or under a multilateral international agreement, that requires publication at eighteen months after filing.

Representative Information:

Representative information should be provided for all practitioners having a power of attorney in the application. Providing this information in the Application Data Sheet does not constitute a power of attorney in the application (see 37 CFR 1.32).
Either enter Customer Number or complete the Representative Name section below. If both sections are completed the customer Number will be used for the Representative Information during processing.

Please Select One:	● Customer Number	US Patent Practitioner	◯ Limited Recognition (37 CFR 11.9)
Customer Number	93219		

Domestic Benefit/National Stage Information:

This section allows for the applicant to either claim benefit under 35 U.S.C. 119(e), 120, 121, 365(c), or 386(c) or indicate National Stage entry from a PCT application. Providing benefit claim information in the Application Data Sheet constitutes the specific reference required by 35 U.S.C. 119(e) or 120, and 37 CFR 1.78.
When referring to the current application, please leave the "Application Number" field blank.

Prior Application Status	Pending		Remove
Application Number	Continuity Type	Prior Application Number	Filing or 371(c) Date (YYYY-MM-DD)
	Continuation of	15269776	2016-09-19

Application Data Sheet 37 CFR 1.76	Attorney Docket Number	10115-07594 US
	Application Number	

Title of Invention	Acquiring Information from Sources Responsive to Naturally-Spoken-Speech Commands Provided by a Voice-Enabled Device

Prior Application Status	Patented				Remove
Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)	Patent Number	Issue Date (YYYY-MM-DD)
15269776	Continuation of	13462819	2012-05-03	9451084	2016-09-20
Prior Application Status	Patented				Remove
Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)	Patent Number	Issue Date (YYYY-MM-DD)
13462819	Continuation of	12973475	2010-12-20	8185402	2012-05-22
Prior Application Status	Patented				Remove
Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)	Patent Number	Issue Date (YYYY-MM-DD)
12973475	Continuation of	12030556	2008-02-13	7881941	2011-02-01
Prior Application Status	Patented				Remove
Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)	Patent Number	Issue Date (YYYY-MM-DD)
12030556	Continuation of	11409703	2006-04-24	7386455	2008-06-10
Prior Application Status	Patented				Remove
Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)	Patent Number	Issue Date (YYYY-MM-DD)
11409703	Continuation of	10821690	2004-04-09	7076431	2006-07-11
Prior Application Status	Patented				Remove
Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)	Patent Number	Issue Date (YYYY-MM-DD)
10821690	Continuation of	09776996	2001-02-05	6721705	2004-04-13

Prior Application Status	Expired		Remove
Application Number	Continuity Type	Prior Application Number	Filing or 371(c) Date (YYYY-MM-DD)
09776996	Claims benefit of provisional	60233068	2000-09-15
Prior Application Status	Expired		Remove
Application Number	Continuity Type	Prior Application Number	Filing or 371(c) Date (YYYY-MM-DD)
09776996	Claims benefit of provisional	60180344	2000-02-04

Additional Domestic Benefit/National Stage Data may be generated within this form by selecting the **Add** button. Add

Foreign Priority Information:

Application Data Sheet 37 CFR 1.76	Attorney Docket Number	10115-07594 US
	Application Number	

Title of Invention	Acquiring Information from Sources Responsive to Naturally-Spoken-Speech Commands Provided by a Voice-Enabled Device

This section allows for the applicant to claim priority to a foreign application. Providing this information in the application data sheet constitutes the claim for priority as required by 35 U.S.C. 119(b) and 37 CFR 1.55. When priority is claimed to a foreign application that is eligible for retrieval under the priority document exchange program (PDX)[i] the information will be used by the Office to automatically attempt retrieval pursuant to 37 CFR 1.55(i)(1) and (2). Under the PDX program, applicant bears the ultimate responsibility for ensuring that a copy of the foreign application is received by the Office from the participating foreign intellectual property office, or a certified copy of the foreign priority application is filed, within the time period specified in 37 CFR 1.55(g)(1).

Application Number	Country[i]	Filing Date (YYYY-MM-DD)	Remove Access Code[i] (if applicable)

Additional Foreign Priority Data may be generated within this form by selecting the **Add** button. Add

Statement under 37 CFR 1.55 or 1.78 for AIA (First Inventor to File) Transition Applications

☐ This application (1) claims priority to or the benefit of an application filed before March 16, 2013 and (2) also contains, or contained at any time, a claim to a claimed invention that has an effective filing date on or after March 16, 2013.
NOTE: By providing this statement under 37 CFR 1.55 or 1.78, this application, with a filing date on or after March 16, 2013, will be examined under the first inventor to file provisions of the AIA.

Application Data Sheet 37 CFR 1.76	Attorney Docket Number	10115-07594 US
	Application Number	
Title of Invention	Acquiring Information from Sources Responsive to Naturally-Spoken-Speech Commands Provided by a Voice-Enabled Device	

Authorization or Opt-Out of Authorization to Permit Access:

When this Application Data Sheet is properly signed and filed with the application, applicant has provided written authority to permit a participating foreign intellectual property (IP) office access to the instant application-as-filed (see paragraph A in subsection 1 below) and the European Patent Office (EPO) access to any search results from the instant application (see paragraph B in subsection 1 below).

Should applicant choose not to provide an authorization identified in subsection 1 below, applicant **must opt-out** of the authorization by checking the corresponding box A or B or both in subsection 2 below.

NOTE: This section of the Application Data Sheet is **ONLY** reviewed and processed with the **INITIAL** filing of an application. After the initial filing of an application, an Application Data Sheet cannot be used to provide or rescind authorization for access by a foreign IP office(s). Instead, Form PTO/SB/39 or PTO/SB/69 must be used as appropriate.

1. Authorization to Permit Access by a Foreign Intellectual Property Office(s)

A. Priority Document Exchange (PDX) - Unless box A in subsection 2 (opt-out of authorization) is checked, the undersigned hereby **grants the USPTO authority** to provide the European Patent Office (EPO), the Japan Patent Office (JPO), the Korean Intellectual Property Office (KIPO), the State Intellectual Property Office of the People's Republic of China (SIPO), the World Intellectual Property Organization (WIPO), and any other foreign intellectual property office participating with the USPTO in a bilateral or multilateral priority document exchange agreement in which a foreign application claiming priority to the instant patent application is filed, access to: (1) the instant patent application-as-filed and its related bibliographic data, (2) any foreign or domestic application to which priority or benefit is claimed by the instant application and its related bibliographic data, and (3) the date of filing of this Authorization. See 37 CFR 1.14(h)(1).

B. Search Results from U.S. Application to EPO - Unless box B in subsection 2 (opt-out of authorization) is checked, the undersigned hereby **grants the USPTO authority** to provide the EPO access to the bibliographic data and search results from the instant patent application when a European patent application claiming priority to the instant patent application is filed. See 37 CFR 1.14(h)(2).

The applicant is reminded that the EPO's Rule 141(1) EPC (European Patent Convention) requires applicants to submit a copy of search results from the instant application without delay in a European patent application that claims priority to the instant application.

2. Opt-Out of Authorizations to Permit Access by a Foreign Intellectual Property Office(s)

☐ A. Applicant **DOES NOT** authorize the USPTO to permit a participating foreign IP office access to the instant application-as-filed. If this box is checked, the USPTO will not be providing a participating foreign IP office with any documents and information identified in subsection 1A above.

☐ B. Applicant **DOES NOT** authorize the USPTO to transmit to the EPO any search results from the instant patent application. If this box is checked, the USPTO will not be providing the EPO with search results from the instant application.

NOTE: Once the application has published or is otherwise publicly available, the USPTO may provide access to the application in accordance with 37 CFR 1.14.

Application Data Sheet 37 CFR 1.76	Attorney Docket Number	10115-07594 US
	Application Number	

Title of Invention	Acquiring Information from Sources Responsive to Naturally-Spoken-Speech Commands Provided by a Voice-Enabled Device

Applicant Information:

Providing assignment information in this section does not substitute for compliance with any requirement of part 3 of Title 37 of CFR to have an assignment recorded by the Office.

Applicant 1 Remove

If the applicant is the inventor (or the remaining joint inventor or inventors under 37 CFR 1.45), this section should not be completed. The information to be provided in this section is the name and address of the legal representative who is the applicant under 37 CFR 1.43; or the name and address of the assignee, person to whom the inventor is under an obligation to assign the invention, or person who otherwise shows sufficient proprietary interest in the matter who is the applicant under 37 CFR 1.46. If the applicant is an applicant under 37 CFR 1.46 (assignee, person to whom the inventor is obligated to assign, or person who otherwise shows sufficient proprietary interest) together with one or more joint inventors, then the joint inventor or inventors who are also the applicant should be identified in this section.

Clear

● Assignee | Legal Representative under 35 U.S.C. 117 | Joint Inventor

Person to whom the inventor is obligated to assign. | Person who shows sufficient proprietary interest

If applicant is the legal representative, indicate the authority to file the patent application, the inventor is:

Name of the Deceased or Legally Incapacitated Inventor:

If the Applicant is an Organization check here. ☒

Organization Name	Parus Holdings, Inc.		
Mailing Address Information For Applicant:			
Address 1	3000 Lakeside Drive, Suite 110S		
Address 2			
City	Bannockburn	**State/Province**	IL
Country	US	Postal Code	60015
Phone Number		Fax Number	
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Application Data Sheet 37 CFR 1.76	Attorney Docket Number	10115-07594 US
	Application Number	
Title of Invention	Acquiring Information from Sources Responsive to Naturally-Spoken-Speech Commands Provided by a Voice-Enabled Device	

Assignee 1

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This Application Data Sheet **must** be signed by a patent practitioner if one or more of the applicants is a **juristic entity** (e.g., corporation or association). If the applicant is two or more joint inventors, this form must be signed by a patent practitioner, **all** joint inventors who are the applicant, or one or more joint inventor-applicants who have been given power of attorney (e.g., see USPTO Form PTO/AIA/81) on behalf of **all** joint inventor-applicants.

See 37 CFR 1.4(d) for the manner of making signatures and certifications.

Signature	/Reena Kuyper/			Date (YYYY-MM-DD)	2018-10-09
First Name	Reena	Last Name	Kuyper	Registration Number	33,830

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Application Data Sheet 37 CFR 1.76	Attorney Docket Number	10115-07594 US
	Application Number	

Title of Invention	Acquiring Information from Sources Responsive to Naturally-Spoken-Speech Commands Provided by a Voice-Enabled Device

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INFORMATION DISCLOSURE STATEMENT BY APPLICANT (**Not for submission under 37 CFR 1.99)**		
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INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)	Application Number	
	Filing Date	2018-10-09
	First Named Inventor	Alexander Kurganov
	Art Unit	
	Examiner Name	
	Attorney Docket Number	10115-07594 US

	48	YANG, E., "INETPhone - Telephone Services and Servers on the Internet", April 1995, University of North Texas, pp. 1-6.	

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INFORMATION DISCLOSURE STATEMENT BY APPLICANT (**Not for submission under 37 CFR 1.99)**	Application Number		
	Filing Date	2018-10-09	
	First Named Inventor	Alexander Kurganov	
	Art Unit		
	Examiner Name		
	Attorney Docket Number	10115-07594 US	

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Please see 37 CFR 1.97 and 1.98 to make the appropriate selection(s):

That each item of information contained in the information disclosure statement was first cited in any communication from a foreign patent office in a counterpart foreign application not more than three months prior to the filing of the information disclosure statement. See 37 CFR 1.97(e)(1).

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The fee set forth in 37 CFR 1.17 (p) has been submitted herewith.

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Signature	/Reena Kuyper/	Date (YYYY-MM-DD)	2018-10-09
Name/Print	Reena Kuyper	Registration Number	33,830

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EFS ID:	33960560
Application Number:	16155523
International Application Number:	
Confirmation Number:	9544
Title of Invention:	Acquiring Information from Sources Responsive to Naturally-Spoken-Speech Commands Provided by a Voice-Enabled Device
First Named Inventor/Applicant Name:	Alexander Kurganov
Customer Number:	93219
Filer:	Reena Kuyper
Filer Authorized By:	
Attorney Docket Number:	10115-07594 US
Receipt Date:	09-OCT-2018
Filing Date:	
Time Stamp:	18:32:46
Application Type:	Utility under 35 USC 111(a)

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Submitted with Payment	no

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Information Disclosure Statement (IDS) Form (SB08)	07594US_20181009_IDS2.pdf	1040622 e5905b3275ac152e98e67c6157e459ca2d78c54e	no	19

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