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EXAMINER	
CHAWAN, VIJAY B	

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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

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PTOL-90A (Rev. 04/07)

Office Action Summary	**Application No.** 16/155,523	**Applicant(s)** Kurganov, Alexander	
	Examiner VIJAY B CHAWAN	**Art Unit** 2658	**AIA (FITF) Status** No

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTHS FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133).
  Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☐ Responsive to communication(s) filed on _____.
☐ A declaration(s)/affidavit(s) under **37 CFR 1.130(b)** was/were filed on _____.
2a)☐ This action is **FINAL.** 2b) ☑ This action is non-final.
3)☐ An election was made by the applicant in response to a restriction requirement set forth during the interview on _____; the restriction requirement and election have been incorporated into this action.
4)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims*

5) ☑ Claim(s) 1-20 is/are pending in the application.
5a) Of the above claim(s) _____ is/are withdrawn from consideration.
6) ☐ Claim(s) _____ is/are allowed.
7) ☑ Claim(s) 1-20 is/are rejected.
8) ☐ Claim(s) _____ is/are objected to.
9) ☐ Claim(s) _____ are subject to restriction and/or election requirement

* If any claims have been determined allowable, you may be eligible to benefit from the **Patent Prosecution Highway** program at a participating intellectual property office for the corresponding application. For more information, please see http://www.uspto.gov/patents/init_events/pph/index.jsp or send an inquiry to **PPHfeedback@uspto.gov.**

Application Papers

10)☐ The specification is objected to by the Examiner.
11)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.
Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).
Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).
Certified copies:
a)☐ All b)☐ Some** c)☐ None of the:
1.☐ Certified copies of the priority documents have been received.
2.☐ Certified copies of the priority documents have been received in Application No. _____.
3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

** See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☑ Notice of References Cited (PTO-892)
2) ☑ Information Disclosure Statement(s) (PTO/SB/08a and/or PTO/SB/08b)
Paper No(s)/Mail Date ______.
3) ☐ Interview Summary (PTO-413)
Paper No(s)/Mail Date ______.
4) ☐ Other: ______.

DETAILED ACTION

Notice of Pre-AIA or AIA Status

1. The present application is being examined under the pre-AIA first to invent provisions.

Double Patenting

2. The nonstatutory double patenting rejection is based on a judicially created doctrine grounded in public policy (a policy reflected in the statute) so as to prevent the unjustified or improper timewise extension of the “right to exclude” granted by a patent and to prevent possible harassment by multiple assignees. A nonstatutory double patenting rejection is appropriate where the conflicting claims are not identical, but at least one examined application claim is not patentably distinct from the reference claim(s) because the examined application claim is either anticipated by, or would have been obvious over, the reference claim(s). See, e.g., *In re Berg*, 140 F.3d 1428, 46 USPQ2d 1226 (Fed. Cir. 1998); *In re Goodman*, 11 F.3d 1046, 29 USPQ2d 2010 (Fed. Cir. 1993); *In re Longi*, 759 F.2d 887, 225 USPQ 645 (Fed. Cir. 1985); *In re Van Ornum*, 686 F.2d 937, 214 USPQ 761 (CCPA 1982); *In re Vogel*, 422 F.2d 438, 164 USPQ 619 (CCPA 1970); *In re Thorington*, 418 F.2d 528, 163 USPQ 644 (CCPA 1969).

A timely filed terminal disclaimer in compliance with 37 CFR 1.321(c) or 1.321(d) may be used to overcome an actual or provisional rejection based on nonstatutory double patenting provided the reference application or patent either is shown to be commonly owned with the examined application, or claims an invention made as a result of activities undertaken within the scope of a joint research agreement. See MPEP § 717.02 for applications subject to examination under the first inventor to file provisions of the AIA as explained in MPEP § 2159. See MPEP §§ 706.02(l)(1) - 706.02(l)(3) for applications not subject to examination under the first inventor to file provisions of the AIA. A terminal disclaimer must be signed in compliance with 37 CFR 1.321(b).

The USPTO Internet website contains terminal disclaimer forms which may be used. Please visit www.uspto.gov/patent/patents-forms. The filing date of the application in which the form is filed determines what form (e.g., PTO/SB/25, PTO/SB/26, PTO/AIA/25, or PTO/AIA/26) should be used. A web-based eTerminal Disclaimer may be filled out completely online using web-screens. An eTerminal Disclaimer that meets all requirements is auto-processed and approved immediately upon submission. For more information about eTerminal Disclaimers, refer to www.uspto.gov/patents/process/file/efs/guidance/eTD-info-I.jsp.

3. Claims 1-20 are rejected on the ground of nonstatutory double patenting as being unpatentable over claims 1-28 of U.S. Patent No. 10,096,320. Although the claims at issue are not identical, they are not patentably distinct from each other because, the claims of the instant application are similar in scope and content of the patented claims issued to the same applicant.

Application No: 16/155,523	Patent No: 10,096,320
1. A system comprising: (a) at least one data processor, the at least one data processor operatively coupled to a plurality of communication data networks; (b) at least one speaker-independent speech-recognition engine operatively coupled to the data processor; (c) memory accessible to the at least one data processor and storing at least: (i) an instruction set for querying of information to be retrieved from a plurality of sources coupled to the plurality of communication data networks, the instruction set comprising: an indication of the plurality of sources, each identified by a source identifier, and each identifying certain information to be retrieved from the source identifier, and (ii) at least one recognition grammar executable code corresponding to each instruction set and corresponding to data characterizing audio containing a naturally-spoken-speech command including an information request, (d) wherein the at least one speaker-independent-speech-recognition engine is adapted to: (1) receive the data characterizing audio containing the naturally-spoken-speech command from a voice-enabled device via a first of the plurality of communication data networks; (2) to recognize phenomes in the data characterizing audio containing naturally-spoken-speech commands to understand spoken words; and (3) to generate recognition results data, (e) wherein the at least one data processor is adapted to: (1) select the corresponding at least one recognition grammar executable code upon receiving the data characterizing the audio containing the naturally-spoken-speech command and to convert the data characterizing the audio containing the naturally-spoken-speech command into a data message for transmission to a network	1. A system comprising: (a) at least one data processor, the at least one data processor operatively coupled to a plurality of communication data networks; (b) at least one speaker-independent speech-recognition engine operatively coupled to the data processor; (c) memory accessible to the at least one data processor and storing at least: (i) an instruction set querying of information to be retrieved from a plurality of sources, the instruction set comprising: an indication of the plurality of sources, each identified by a source identifier, and each identifying certain information to be retrieved from the source identifier, and (ii) at least one recognition grammar executable code corresponding to each instruction set and corresponding to data characterizing audio containing a naturally-spoken-speech command including an information request, (d) wherein the at least one speaker-independent-speech-recognition engine is adapted (i) to receive the data characterizing audio containing the naturally-spoken-speech command from a voice-enabled device via a first of the plurality of communication data networks, (ii) to recognize phenomes in the data characterizing audio containing naturally-spoken-speech commands to understand spoken words, and (iii) to generate recognition results data, (e) wherein the at least one data processor is adapted to (i) select the corresponding at least one recognition grammar executable code upon receiving the data characterizing the audio containing the naturally-spoken-speech command and to convert the data characterizing the audio containing the naturally-spoken-speech command into a data message for transmission to a network interface adapted to access a second of the

interface adapted to access a second of the plurality of communication data networks; and (2) retrieve the instruction set corresponding to the recognition grammar executable codes provided by the at least one speaker-independent-speech-recognition engine and to access the information source queried by the instruction set to obtain at least a part of the information to be retrieved, and (f) at least one speech-synthesis device operatively coupled to the at least one data processor, the at least one speech-synthesis device configured to produce an audio message relating to any resulting information retrieved from the plurality of information sources including a text-to-speech conversion of at least certain data in said any resulting information retrieved from the plurality of information sources, and to convey the audio message via the voice-enabled device.	plurality of communication data networks; and (ii) to retrieve the instruction set corresponding to the recognition grammar executable codes provided by the at least one speaker-independent-speech-recognition engine and to access the information source queried by the instruction set to obtain at least a part of the information to be retrieved, and (f) at least one speech-synthesis device operatively coupled to the at least one data processor, the at least one speech-synthesis device configured to produce an audio message relating to any resulting information retrieved from the plurality of information sources, and to transmit the audio message to the voice-enabled device.
2. The system of claim 1, wherein the plurality of communication data networks includes the Internet.	2. The system of claim 1, wherein the plurality of communication data networks includes the Internet.
3. The system of claim 1, wherein the plurality of communication data networks include a local-area network.	3. The system of claim 1, wherein the plurality of communication data networks include a local-area network.
4. The system of claim 1, wherein the voice-enabled device is a home device.	4. The system of claim 1, wherein the voice-enabled device is a telephone.
5. The method of claim 1, wherein the voice-enabled device is at least one of a group of an IP telephone, a cellular phone, a personal computer, a media player appliance, and a television or other video display device.	4. The system of claim 1, wherein the voice-enabled device is a telephone. 10. The system of claim 1, wherein the voice-enabled device is an IP telephone. 11. The system of claim 1, wherein the voice-enabled device is a cellular phone. 12. The system of claim 1, wherein the voice-enabled device is a personal computer. 13. The system of claim 1, wherein the voice-enabled device is a media player appliance.

	14. The system of claim 1, wherein the voice-enabled device is a television or other video display device.
6. The system of claim 1, wherein the speaker-independent-speech-recognition engine is adapted to analyze the phonemes to recognize conversational naturally-spoken-speech commands.	5. The system of claim 1, wherein the speaker-independent-speech-recognition engine is adapted to analyze the phonemes to recognize conversational naturally-spoken-speech commands.
7. The system of claim 1, wherein the speaker-independent-speech-recognition engine is adapted to recognize the naturally-spoken-speech commands.	6. The system of claim 1, wherein the speaker-independent-speech-recognition engine is adapted to recognize the naturally-spoken-speech commands.
8. The system of claim 1, wherein the instruction set executable code further comprises: a content descriptor associated with each information-source identifier, the content descriptor pre-defining a portion of the information source containing the information to be retrieved.	7. The system of claim 1, wherein the instruction set executable code further comprises: a content descriptor associated with each information-source identifier, the content descriptor pre-defining a portion of the information source containing the information to be retrieved.
9. The system of claim 1, further comprising: a database operatively connected to the data processor, the database adapted to store the information gathered from the information sources in response to the information requests.	8. The system of claim 1, further comprising: a database operatively connected to the data processor, the database adapted to store the information gathered from the information sources in response to the information requests.
10. The system of claim 8, wherein each recognition grammar executable code and each instruction set for querying of information to be retrieved are stored in the database.	9. The system of claim 8, wherein each recognition grammar executable code and each instruction set for querying of information to be retrieved are stored in the database.
11. A method comprising: (a) providing at least one data processor, the data processor operatively coupled to a plurality of communication data networks; (b) providing at least one speaker-independent-speech-recognition engine operatively coupled to the at least one data processor (c) providing memory accessible to the data processor storing at least: (i) an instruction set for querying of the information to be retrieved from a plurality of sources coupled to the plurality of communication data networks, the instruction set comprising: an indication	15. A method comprising: (a) providing at least one data processor, the data processor operatively coupled to a plurality of communication data networks; (b) providing at least one speaker-independent-speech-recognition engine operatively coupled to the at least one data processor (c) providing memory accessible to the data processor storing at least: (i) an instruction set for querying of the information to be retrieved from a plurality of sources, the instruction set comprising: an indication of the plurality of sources, each identified by a information-

of the plurality of sources, each identified by an information-source identifier, and each identifying certain information to be retrieved from the information-source identifier, and (ii) at least one recognition grammar executable code corresponding to each instruction set and corresponding to data characterizing audio containing a naturally-spoken-speech command including an information request, (d) the at least one speaker-independent-speech-recognition engine: (i) receiving the data characterizing audio containing the naturally-spoken-speech command from the voice-enabled device via a first of the communication data networks, (ii) recognizing phenomes in the data characterizing audio containing the naturally-spoken-speech commands to understand spoken words, and (iii) generating recognition-results data, (e) the least one data processor programmed to: (i) select the corresponding at least one recognition grammar executable code upon receiving the data characterizing audio containing the naturally-spoken-speech command and convert the data characterizing audio containing naturally-spoken-speech command into a data message for transmission to a network interface adapted to access a second of the one communication networks; and (ii) retrieve the instruction set corresponding to the recognition grammar executable provided by the at least one speaker-independent-speech-recognition device and access the information source identified by the instruction set to obtain at least a part of the information to be retrieved; and (f) providing at least one speech-synthesis device operatively connected to the at least one data processor, and by the at least one speech-synthesis device adapted to: (i) produce an audio message relating to any resulting information retrieved from the	source identifier, and each identifying certain information to be retrieved from the information-source identifier, and (ii) at least one recognition grammar executable code corresponding to each instruction set and corresponding to data characterizing audio containing a naturally-spoken-speech command including an information request, (d) the at least one speaker-independent-speech-recognition engine: (i) receiving the data characterizing audio containing the naturally-spoken-speech command from the voice-enabled device via a first of the communication data networks, (ii) recognizing phenomes in the data characterizing audio containing the naturally-spoken-speech commands to understand spoken words, and (iii) generating recognition-results data, (e) the least one data processor programmed to: (i) select the corresponding at least one recognition grammar executable code upon receiving the data characterizing audio containing the naturally-spoken-speech command and convert the data characterizing audio containing naturally-spoken-speech command into a data message for transmission to a network interface adapted to access a second of the one communication networks; and (iii) retrieve the instruction set corresponding to the recognition grammar executable provided by the at least one speaker-independent-speech-recognition device and access the information source identified by the instruction set to obtain at least a part of the information to be retrieve; and (f) providing at least one speech-synthesis device operatively connected to the at least one data processor, and by the at least one speech-synthesis device: (i) produce an audio message relating to any resulting information retrieved from the plurality of information sources, and (ii)

plurality of information sources including text-to-speech conversion of said any resulting information retrieved from the plurality of information sources, and (ii) transmit the audio message via the voice-enabled device.	transmit the audio message to the voice-enabled device.
12. The method of claim 11, wherein the plurality of communication data networks includes the Internet.	16. The method of claim 15, wherein the plurality of communication data networks includes the Internet.
13. The method of claim 11, wherein the plurality of communication data networks include a local-area network.	17. The method of claim 15, wherein the plurality of communication data networks include a local-area network.
14. The method of claim 11, wherein the voice-enabled device is a telephone.	18. The method of claim 15, wherein the voice-enabled device is a telephone.
15. The method of claim 11, wherein the speaker-independent-speech-recognition engine is adapted to analyze the phonemes to recognize conversational naturally-spoken-speech commands.	19. The method of claim 15, wherein the speaker-independent-speech-recognition engine is adapted to analyze the phonemes to recognize conversational naturally-spoken-speech commands.
16. The method of claim 11, wherein the speaker-independent-speech-recognition engine is adapted to recognize the naturally-spoken-speech commands.	20. The method of claim 15, wherein the speaker-independent-speech-recognition engine is adapted to recognize the naturally-spoken-speech commands.
17. The method of claim 11, wherein the instruction set executable code further comprises: a content descriptor associated with each information-source identifier, the content descriptor pre-defining a portion of the information source containing the information to be retrieved.	21. The method of claim 15, wherein the instruction set executable code further comprises: a content descriptor associated with each information-source identifier, the content descriptor pre-defining a portion of the information source containing the information to be retrieved.
18. The method of claim 11, further comprising: providing a database and operatively connecting the database to the data processor and storing the information gathered from the information sources in response to the information requests in the database.	22. The method of claim 15, further comprising: providing a database and operatively connecting the database to the data processor and storing the information gathered from the information sources in response to the information requests in the database.
19. The method of claim 18, wherein each recognition grammar executable code and each instruction set are stored in the database.	23. The method of claim 22, wherein each recognition grammar executable code and each instruction set are stored in the database.

20. The method of claim 18, wherein the voice-enabled device is at least one of a group of an IP telephone, a cellular phone, a personal computer, a media player appliance, and a television or other video display device.	24. The method of claim 15, wherein the voice-enabled device is an IP telephone. 25. The method of claim 15, wherein the voice-enabled device is a cellular phone. 26. The method of claim 15, wherein the voice-enabled device is a personal computer. 27. The method of claim 15, wherein the voice-enabled device is a media player appliance. 28. The method of claim 15, wherein the voice-enabled device is a television or other video display device.

4. Claims 1-8 and 11-20 are rejected on the ground of nonstatutory double patenting as being unpatentable over claims1-4, 11-18 of U.S. Patent No. 10,380,981. Although the claims at issue are not identical, they are not patentably distinct from each other because, the claims of the instant application are similar in scope and content of the patented claims issued to the same applicant.

Application No: 16/155,523	Patent No: 10,320,981
1. A system comprising: (a) at least one data processor, the at least one data processor operatively coupled to a plurality of communication data networks; (b) at least one speaker-independent speech-recognition engine operatively coupled to the data processor; (c) memory accessible to the at least one data processor and storing at least: (i) an instruction set for querying of information to be retrieved from a plurality	5. A voice-browsing system for retrieving information from an information source that is periodically updated with current information, by speech commands received from a particular user provided via a voice-enabled device after establishing a connection between the voice-enabled device and a media server of the voice-browsing system, said voice-browsing system comprising: (a) a speech-recognition engine

of sources coupled to the plurality of communication data networks, the instruction set comprising: an indication of the plurality of sources, each identified by a source identifier, and each identifying certain information to be retrieved from the source identifier, and (ii) at least one recognition grammar executable code corresponding to each instruction set and corresponding to data characterizing audio containing a naturally-spoken-speech command including an information request, (d) wherein the at least one speaker-independent-speech-recognition engine is adapted to: (1) receive the data characterizing audio containing the naturally-spoken-speech command from a voice-enabled device via a first of the plurality of communication data networks; (2) to recognize phenomes in the data characterizing audio containing naturally-spoken-speech commands to understand spoken words; and (3) to generate recognition results data, (e) wherein the at least one data processor is adapted to: (1) select the corresponding at least one recognition grammar executable code upon receiving the data characterizing the audio containing the naturally-spoken-speech command and to convert the data characterizing the audio containing the naturally-spoken-speech command into a data message for transmission to a network interface adapted to access a second of the plurality of communication data networks; and (2) retrieve the instruction set corresponding to the recognition grammar executable codes provided by the at least one speaker-independent-speech-recognition engine and to access the information source queried by the instruction set to obtain at least a part of the information to be retrieved, and (f) at least one speech-synthesis device operatively coupled to the at least one data processor,	including a processor and coupled to the media server, the media server initiating a voice-response application once the connection between the voice-enabled device and the voice-browsing system is established, the speech-recognition engine adapted to receive a speech command from a particular user via the voice-enabled device, the media server configured to identify and access the information source via a network, the speech-recognition engine adapted to convert the speech command into a data message by selecting speech-recognition grammar established to correspond to the speech command received from the particular user and assigned to perform searches; (b) the media server further configured to select at least one information-source-retrieval instruction corresponding to the speech-recognition grammar established for the speech command, the at least one information-source-retrieval instruction stored in a database associated with the media server and adapted to retrieve information; (c) a web-browsing server coupled to the media server and adapted to access at least a portion of the information source to retrieve information indicated by the speech command, by using a processor of the web-browsing server, which processor (i) performs an instruction that requests information from an identified web page within the information source, and (ii) utilizes a command to execute a content extractor within the web-browsing server to separate a portion of the information from other information, the information derived from only a portion of a web page containing information relevant to the speech command, wherein the content extractor uses a content-descriptor file containing a description of the portion of information and wherein the content-descriptor file indicates

the at least one speech-synthesis device configured to produce an audio message relating to any resulting information retrieved from the plurality of information sources including a text-to-speech conversion of at least certain data in said any resulting information retrieved from the plurality of information sources, and to convey the audio message via the voice-enabled device.	a location of a portion of the information within the information source, and selecting, by the web-browsing server, an information type relevant from the information source and retrieving only a portion of the information that is relevant according to the at least one information-source-retrieval instruction; and (d) a speech-synthesis engine including a processor and coupled to the media server, the speech-synthesis engine adapted to convert the information retrieved from the information source into audio and convey the audio by the voice-enabled device.
2. The system of claim 1, wherein the plurality of communication data networks includes the Internet.	6. The voice-browsing system claim 5, further comprising: an interface to an associated website by the network to locate requested information. 7. The voice-browsing system of claim 5, wherein the voice-enabled device accesses the voice-browsing system by at least one of a landline telephone, a wireless telephone, and an Internet Protocol telephonic connection and wherein the media server operatively connects to the network, by at least one of a local-area network, a wide-area network, and the Internet.
3. The system of claim 1, wherein the plurality of communication data networks include a local-area network.	6. The voice-browsing system claim 5, further comprising: an interface to an associated website by the network to locate requested information. 7. The voice-browsing system of claim 5, wherein the voice-enabled device accesses the voice-browsing system by at least one of a landline telephone, a wireless telephone, and an Internet Protocol telephonic connection and wherein the media server operatively connects to the network, by at least one of a local-area network, a wide-area network, and the Internet.
4. The system of claim 1, wherein the voice-enabled device is a home device.	6. The voice-browsing system claim 5, further comprising: an interface to an associated

	website by the network to locate requested information. 7. The voice-browsing system of claim 5, wherein the voice-enabled device accesses the voice-browsing system by at least one of a landline telephone, a wireless telephone, and an Internet Protocol telephonic connection and wherein the media server operatively connects to the network, by at least one of a local-area network, a wide-area network, and the Internet.
5. The method of claim 1, wherein the voice-enabled device is at least one of a group of an IP telephone, a cellular phone, a personal computer, a media player appliance, and a television or other video display device.	
6. The system of claim 1, wherein the speaker-independent-speech-recognition engine is adapted to analyze the phonemes to recognize conversational naturally-spoken-speech commands.	9. The voice-browsing system of claim 5, further comprising: a clipping engine adapted to generate the content-descriptor file, by which, an instruction is used by the web-browsing server to request information from the identified web site and the information is displayed on the voice-enabled device, wherein the information is only the portion of the web page containing information relevant to the speech command.
7. The system of claim 1, wherein the speaker-independent-speech-recognition engine is adapted to recognize the naturally-spoken-speech commands.	8. The voice-browsing system of claim 5, wherein the media server functions as a user-interface system adapted to provide access to a voice-browsing system.
8. The system of claim 1, wherein the instruction set executable code further comprises: a content descriptor associated with each information-source identifier, the content descriptor pre-defining a portion of the information source containing the information to be retrieved.	9. The voice-browsing system of claim 5, further comprising: a clipping engine adapted to generate the content-descriptor file, by which, an instruction is used by the web-browsing server to request information from the identified web site and the information is displayed on the voice-enabled device, wherein the information is only the portion of the web page containing information relevant to the speech command.
9. The system of claim 1, further comprising: a database operatively connected to the data processor, the database adapted to store the	

information gathered from the information sources in response to the information requests.	
10. The system of claim 8, wherein each recognition grammar executable code and each instruction set for querying of information to be retrieved are stored in the database.	
11. A method comprising: (a) providing at least one data processor, the data processor operatively coupled to a plurality of communication data networks; (b) providing at least one speaker-independent-speech-recognition engine operatively coupled to the at least one data processor (c) providing memory accessible to the data processor storing at least: (i) an instruction set for querying of the information to be retrieved from a plurality of sources coupled to the plurality of communication data networks, the instruction set comprising: an indication of the plurality of sources, each identified by an information-source identifier, and each identifying certain information to be retrieved from the information-source identifier, and (ii) at least one recognition grammar executable code corresponding to each instruction set and corresponding to data characterizing audio containing a naturally-spoken-speech command including an information request, (d) the at least one speaker-independent-speech-recognition engine: (i) receiving the data characterizing audio containing the naturally-spoken-speech command from the voice-enabled device via a first of the communication data networks, (ii) recognizing phenomes in the data characterizing audio containing the naturally-spoken-speech commands to understand spoken words, and (iii) generating recognition-results data, (e) the least one data processor programmed to: (i) select the corresponding at least one recognition grammar executable code upon	1. A method, comprising: (a) receiving a speech command from a voice-enabled device of a particular user, over a network, by a speech-recognition engine coupled to a media server by an interactive voice response application including a user-defined search, the speech-recognition engine adapted to convert the speech command into a data message, the media server adapted to identify and access at least one or more websites containing information of interest to the particular user, the speech-recognition engine adapted to select particular speech-recognition grammar describing the speech command received and assigned to fetching content relating to the data message converted from the speech command and assigned to the user-defined search including a web request, along with a uniform resource locator of an identified web site from the one or more websites containing information of interest to the particular user and responsive to the web request; (b) selecting, by the media server, at least one information-source-retrieval instruction stored for the particular speech-recognition grammar in a database coupled to the media server and adapted to retrieve information from the at least one or more websites; (c) accessing, by a web-browsing server, a portion of an information source to retrieve information relating to the speech command, by using a processor of the web-browsing server, which processor (i) performs an instruction that requests information from an identified web site, (ii)

receiving the data characterizing audio containing the naturally-spoken-speech command and convert the data characterizing audio containing naturally-spoken-speech command into a data message for transmission to a network interface adapted to access a second of the one communication networks; and (ii) retrieve the instruction set corresponding to the recognition grammar executable provided by the at least one speaker-independent-speech-recognition device and access the information source identified by the instruction set to obtain at least a part of the information to be retrieved; and (f) providing at least one speech-synthesis device operatively connected to the at least one data processor, and by the at least one speech-synthesis device adapted to: (i) produce an audio message relating to any resulting information retrieved from the plurality of information sources including text-to-speech conversion of said any resulting information retrieved from the plurality of information sources, and (ii) transmit the audio message via the voice-enabled device.	utilizes a command to execute a content extractor within the web-browsing server to separate a portion of information that is relevant from other information on the web page using a name of a named object including the information, the information derived from only a portion of the web page containing information pertinent to the speech command, the content extractor adapted to use a content-descriptor file containing a description of the portion of information and the content-descriptor file adapted to indicate a location of the portion of the information within the information source; (d) selecting, by the web-browsing server, the information relating to the speech command from the information source and retrieving only the portion of the information requested by the speech command according to the at least one information-source-retrieval instruction; (e) converting the information retrieved from the information source into an audio message by a speech-synthesis engine, the speech-synthesis engine coupled to the media server; and (f) transmitting the audio message by the voice-enabled device to the particular user.
12. The method of claim 11, wherein the plurality of communication data networks includes the Internet.	2. The method of claim 1, wherein the speech command is received by at least one of a landline telephone, a wireless telephone, and an Internet Protocol telephone and the media server is operatively connected to at least one of a local-area network, a wide-area network, and the Internet.
13. The method of claim 11, wherein the plurality of communication data networks include a local-area network.	2. The method of claim 1, wherein the speech command is received by at least one of a landline telephone, a wireless telephone, and an Internet Protocol telephone and the media server is operatively connected to at least one of a local-area network, a wide-area network, and the Internet.
14. The method of claim 11, wherein the voice-enabled device is a telephone.	3. The method of claim 2, wherein the media server functions as a user-interface system

	adapted to provide access to a voice-browsing system. 4. The method of claim 2, further comprising: a clipping engine adapted to initially generate the content-descriptor file that indicates the location of the portion of the information within the identified web site.
15. The method of claim 11, wherein the speaker-independent-speech-recognition engine is adapted to analyze the phonemes to recognize conversational naturally-spoken-speech commands.	3. The method of claim 2, wherein the media server functions as a user-interface system adapted to provide access to a voice-browsing system. 4. The method of claim 2, further comprising: a clipping engine adapted to initially generate the content-descriptor file that indicates the location of the portion of the information within the identified web site.
16. The method of claim 11, wherein the speaker-independent-speech-recognition engine is adapted to recognize the naturally-spoken-speech commands.	3. The method of claim 2, wherein the media server functions as a user-interface system adapted to provide access to a voice-browsing system. 4. The method of claim 2, further comprising: a clipping engine adapted to initially generate the content-descriptor file that indicates the location of the portion of the information within the identified web site.
17. The method of claim 11, wherein the instruction set executable code further comprises: a content descriptor associated with each information-source identifier, the content descriptor pre-defining a portion of the information source containing the information to be retrieved.	16. The method of claim 12, wherein the corresponding descriptor file identifies the web-accessible information source and information used to generate proper requests to the information source with a specific URL format including search parameters.
18. The method of claim 11, further comprising: providing a database and operatively connecting the database to the data processor and storing the information gathered from the information sources in response to the information requests in the database.	11. The method of claim 10, wherein the content is located in the response data using the named object regardless of the location of the named object within the response data. 12. The method of claim 11, wherein the fetching occurs on a web browsing server, and wherein the web browsing server

	receives the identified speech command from a different server. 13. The method of claim 12, further comprising: using Internet Protocol to communicate with the electronic-communication device of the user. 14. The method of claim 12, further comprising: using a telecommunication network to communicate with the electronic-communication device of the user. 15. The method of claim 12, wherein the electronic-communication device of the user is a voice-enabled wireless unit that is not a telephone. 17. The method of claim 12, wherein using the request information to fetch comprises fetching the response data from a database stored on a Local Area Network (LAN) or a Wide Area Network (WAN). 18. The method of claim 12, further comprising: using the named object to determine a beginning and an end of the content within the response data.
19. The method of claim 18, wherein each recognition grammar executable code and each instruction set are stored in the database.	11. The method of claim 10, wherein the content is located in the response data using the named object regardless of the location of the named object within the response data. 12. The method of claim 11, wherein the fetching occurs on a web browsing server, and wherein the web browsing server receives the identified speech command from a different server. 13. The method of claim 12, further comprising: using Internet Protocol to

	communicate with the electronic-communication device of the user. 14. The method of claim 12, further comprising: using a telecommunication network to communicate with the electronic-communication device of the user. 15. The method of claim 12, wherein the electronic-communication device of the user is a voice-enabled wireless unit that is not a telephone. 17. The method of claim 12, wherein using the request information to fetch comprises fetching the response data from a database stored on a Local Area Network (LAN) or a Wide Area Network (WAN). 18. The method of claim 12, further comprising: using the named object to determine a beginning and an end of the content within the response data.
20. The method of claim 18, wherein the voice-enabled device is at least one of a group of an IP telephone, a cellular phone, a personal computer, a media player appliance, and a television or other video display device.	11. The method of claim 10, wherein the content is located in the response data using the named object regardless of the location of the named object within the response data. 12. The method of claim 11, wherein the fetching occurs on a web browsing server, and wherein the web browsing server receives the identified speech command from a different server. 13. The method of claim 12, further comprising: using Internet Protocol to communicate with the electronic-communication device of the user. 14. The method of claim 12, further comprising: using a telecommunication

	network to communicate with the electronic-communication device of the user. 15. The method of claim 12, wherein the electronic-communication device of the user is a voice-enabled wireless unit that is not a telephone. 17. The method of claim 12, wherein using the request information to fetch comprises fetching the response data from a database stored on a Local Area Network (LAN) or a Wide Area Network (WAN). 18. The method of claim 12, further comprising: using the named object to determine a beginning and an end of the content within the response data.

5. Claims 1-10 are rejected on the ground of nonstatutory double patenting as being unpatentable over claims 1-7 and 14-15 of U.S. Patent No. 9,451,084. Although the claims at issue are not identical, they are not patentably distinct from each other because the claims of the instant application are similar in scope and content of the patented claims issued to the same applicant.

Application No: 16/155,523	Patent No: 9,451,084
1. A system comprising: (a) at least one data processor, the at least one data processor operatively coupled to a plurality of communication data networks; (b) at least one speaker-independent speech-recognition engine operatively coupled to the data processor; (c) memory accessible to the at least one data processor and storing at least:	1. A system for acquiring information from one or more sources maintaining a listing of web sites by receiving speech commands uttered by users into a voice-enabled device and for providing information retrieved from the web sites to the users in an audio form via the voice-enabled device, the system comprising: at least one computing device,

(i) an instruction set for querying of information to be retrieved from a plurality of sources coupled to the plurality of communication data networks, the instruction set comprising: an indication of the plurality of sources, each identified by a source identifier, and each identifying certain information to be retrieved from the source identifier, and (ii) at least one recognition grammar executable code corresponding to each instruction set and corresponding to data characterizing audio containing a naturally-spoken-speech command including an information request, (d) wherein the at least one speaker-independent-speech-recognition engine is adapted to: (1) receive the data characterizing audio containing the naturally-spoken-speech command from a voice-enabled device via a first of the plurality of communication data networks; (2) to recognize phenomes in the data characterizing audio containing naturally-spoken-speech commands to understand spoken words; and (3) to generate recognition results data, (e) wherein the at least one data processor is adapted to: (1) select the corresponding at least one recognition grammar executable code upon receiving the data characterizing the audio containing the naturally-spoken-speech command and to convert the data characterizing the audio containing the naturally-spoken-speech command into a data message for transmission to a network interface adapted to access a second of the plurality of communication data networks; and (2) retrieve the instruction set corresponding to the recognition grammar executable codes provided by the at least one speaker-independent-speech-recognition engine and to access the information source queried by the instruction set to obtain at least a part of the information to be retrieved, and (f) at least	the computing device operatively coupled to one or more networks; at least one speaker-independent speech-recognition device, the speaker-independent speech-recognition device operatively connected to the computing device and configured to receive the speech commands; at least one speech-synthesis device, the speech-synthesis device operatively connected to the computing device; memory operatively associated with the computing device with at least one instruction set for identifying the information to be retrieved, the instruction set being associated with the computing device, the instruction set comprising: a plurality of web site addresses for the listing of web sites, each web site address identifying a web site containing the information to be retrieved; at least one recognition grammar associated with the computing device, each recognition grammar corresponding to each instruction set and corresponding to a speech command, the speech command comprising an information request provided by the user, the speaker-independent speech-recognition device configured to receive the speech command from the users via the voice-enabled device and to select the corresponding recognition grammar upon receiving the speech command; the computing device configured to retrieve the instruction set corresponding to the recognition grammar provided by the speaker-independent speech-recognition device; the computing device further configured to access at least one of the plurality of web sites identified by the instruction set to obtain the information to be retrieved, wherein the computing device is further configured to periodically search via the one or more networks to identify new web sites and to add the new web sites to the plurality of web sites, the computing device configured to access a first web site of

one speech-synthesis device operatively coupled to the at least one data processor, the at least one speech-synthesis device configured to produce an audio message relating to any resulting information retrieved from the plurality of information sources including a text-to-speech conversion of at least certain data in said any resulting information retrieved from the plurality of information sources, and to convey the audio message via the voice-enabled device.	the plurality of web sites and, if the information to be retrieved is not found at the first web site, the computer configured to access the plurality of web sites remaining in an order defined for accessing the listing of web sites until the information to be retrieved is found in at least one of the plurality of web sites or until the plurality of web sites have been accessed; the speech synthesis device configured to produce an audio message containing any retrieved information from the plurality of web sites, and the speech synthesis device further configured to transmit the audio message to the users via the voice-enabled device.
2. The system of claim 1, wherein the plurality of communication data networks includes the Internet.	2. The system of claim 1, wherein the one or more networks include the Internet.
3. The system of claim 1, wherein the plurality of communication data networks include a local-area network.	3. The system of claim 1, wherein the one or more networks include a local-area network.
4. The system of claim 1, wherein the voice-enabled device is a home device.	4. The system of claim 1, wherein the voice-enabled device is at least one of a standard telephone, an IP telephone, a cellular phone, a PDA, a personal computer, a DVD player, a television or other video display device, a CD player, a MP3 player, and any other device capable of transmitting the audio message.
5. The method of claim 1, wherein the voice-enabled device is at least one of a group of an IP telephone, a cellular phone, a personal computer, a media player appliance, and a television or other video display device.	4. The system of claim 1, wherein the voice-enabled device is at least one of a standard telephone, an IP telephone, a cellular phone, a PDA, a personal computer, a DVD player, a television or other video display device, a CD player, a MP3 player, and any other device capable of transmitting the audio message.
6. The system of claim 1, wherein the speaker-independent-speech-recognition engine is adapted to analyze the phonemes to recognize conversational naturally-spoken-speech commands.	5. The system of claim 1, wherein the speaker-independent speech recognition device is configured to analyze phonemes to recognize the speech commands.
7. The system of claim 1, wherein the speaker-independent-speech-recognition	6. The system of claim 1, wherein the speaker-independent speech-recognition

engine is adapted to recognize the naturally-spoken-speech commands.	device is configured to recognize naturally spoken speech commands.
8. The system of claim 1, wherein the instruction set executable code further comprises: a content descriptor associated with each information-source identifier, the content descriptor pre-defining a portion of the information source containing the information to be retrieved.	7. The system of claim 1, wherein the instruction set further comprises: a content descriptor associated with each web site address, the content descriptor pre-defining a portion of the web site containing the information to be retrieved.
9. The system of claim 1, further comprising: a database operatively connected to the data processor, the database adapted to store the information gathered from the information sources in response to the information requests.	14. The system of claim 1, further comprising: a database operatively connected to the computing device, the database configured to store the information gathered from the web sites in response to the information requests.
10. The system of claim 8, wherein each recognition grammar executable code and each instruction set for querying of information to be retrieved are stored in the database.	15. The system of claim 14, wherein each recognition grammar and each instruction set are stored in the database.

6. Claims 1-3, 8-9, 11-12, and 17-18 are rejected on the ground of nonstatutory double patenting as being unpatentable over claims 1-3, 5-8, and 9-15 of U.S. Patent No. 8,185,402. Although the claims at issue are not identical, they are not patentably distinct from each other because the claims of the instant application are similar in scope and content of the patented claims issued to the same applicant.

Application No: 16/155,523	Patent No: 8,185,402
1. A system comprising: (a) at least one data processor, the at least one data processor operatively coupled to a plurality of communication data networks; (b) at least one speaker-independent speech-recognition	9. A system for retrieving information from web sites by uttering speech commands into a phone and for providing to users retrieved information in an audio form via said phone, said system comprising: a computer, said

engine operatively coupled to the data processor; (c) memory accessible to the at least one data processor and storing at least: (i) an instruction set for querying of information to be retrieved from a plurality of sources coupled to the plurality of communication data networks, the instruction set comprising: an indication of the plurality of sources, each identified by a source identifier, and each identifying certain information to be retrieved from the source identifier, and (ii) at least one recognition grammar executable code corresponding to each instruction set and corresponding to data characterizing audio containing a naturally-spoken-speech command including an information request, (d) wherein the at least one speaker-independent-speech-recognition engine is adapted to: (1) receive the data characterizing audio containing the naturally-spoken-speech command from a voice-enabled device via a first of the plurality of communication data networks; (2) to recognize phenomes in the data characterizing audio containing naturally-spoken-speech commands to understand spoken words; and (3) to generate recognition results data, (e) wherein the at least one data processor is adapted to: (1) select the corresponding at least one recognition grammar executable code upon receiving the data characterizing the audio containing the naturally-spoken-speech command and to convert the data characterizing the audio containing the naturally-spoken-speech command into a data message for transmission to a network interface adapted to access a second of the plurality of communication data networks; and (2) retrieve the instruction set corresponding to the recognition grammar executable codes provided by the at least one speaker-independent-speech-recognition engine and to access the	computer operatively connected to the internet and to at least one phone; at least one speaker-independent speech recognition engine, said speaker-independent speech recognition engine operatively connected to said computer; at least one speech synthesis engine, said speech synthesis engine operatively connected to said computer; a database, said database operatively connected to said computer, said database containing a plurality of web site addresses; a content descriptor associated with each said web site address, said content descriptor pre-defining a portion of said web site containing said information to be retrieved; a ranking from highest to lowest associated with each said web site address, said ranking indicating the order in which the plurality of web sites are accessed; said speaker-independent speech recognition engine configured to receive from users via said phone a speech command; said computer configured to access at least one of said plurality of web sites associated with said speech command to obtain said information to be retrieved, said computer configured to first access said web site having the highest ranking and, if said information to be retrieved is not found at said web site having the highest ranking, said computer configured to subsequently access said plurality of web sites in order of rankings until said information to be retrieved is found or until said plurality of web sites has been accessed; said computer further configured to establish or adjust said rankings associated with said plurality of web sites such that said web site having said information to be retrieved is assigned the highest ranking and any web sites not having said information to be retrieved are assigned lower rankings; said speech synthesis engine configured to produce an audio message containing any retrieved information from said web sites, and said speech synthesis

information source queried by the instruction set to obtain at least a part of the information to be retrieved, and (f) at least one speech-synthesis device operatively coupled to the at least one data processor, the at least one speech-synthesis device configured to produce an audio message relating to any resulting information retrieved from the plurality of information sources including a text-to-speech conversion of at least certain data in said any resulting information retrieved from the plurality of information sources, and to convey the audio message via the voice-enabled device.	engine further configured to transmit said audio message to said users via said phone.
2. The system of claim 1, wherein the plurality of communication data networks includes the Internet.	13. The system of claim 9 wherein said internet is the Internet.
3. The system of claim 1, wherein the plurality of communication data networks include a local-area network.	11. The system of claim 9 wherein said internet is a local area network. 12. The system of claim 9 wherein said internet is a wide area network.
4. The system of claim 1, wherein the voice-enabled device is a home device.	
5. The method of claim 1, wherein the voice-enabled device is at least one of a group of an IP telephone, a cellular phone, a personal computer, a media player appliance, and a television or other video display device.	10. The system of claim 9 wherein said phone comprises a standard telephone, a cellular phone, or an IP phone.
6. The system of claim 1, wherein the speaker-independent-speech-recognition engine is adapted to analyze the phonemes to recognize conversational naturally-spoken-speech commands.	
7. The system of claim 1, wherein the speaker-independent-speech-recognition engine is adapted to recognize the naturally-spoken-speech commands.	
8. The system of claim 1, wherein the instruction set executable code further comprises: a content descriptor associated with each information-source identifier, the content descriptor pre-defining a portion of	14. The system of claim 9 wherein said computer is configured to establish or adjust said rankings associated with said plurality of web sites when instructed by said user to

the information source containing the information to be retrieved.	access said plurality of web sites to retrieve said information. 15. The system of claim 9 wherein said computer is configured to establish or adjust said rankings associated with said plurality of web sites based on periodic polling of each of said web sites without being instructed by said user to determine the availability of each said web site, the duration of time for each said web site to respond to a request from said computer, and changes to the location of said information to be retrieved from each said web site.
9. The system of claim 1, further comprising: a database operatively connected to the data processor, the database adapted to store the information gathered from the information sources in response to the information requests.	14. The system of claim 9 wherein said computer is configured to establish or adjust said rankings associated with said plurality of web sites when instructed by said user to access said plurality of web sites to retrieve said information. 15. The system of claim 9 wherein said computer is configured to establish or adjust said rankings associated with said plurality of web sites based on periodic polling of each of said web sites without being instructed by said user to determine the availability of each said web site, the duration of time for each said web site to respond to a request from said computer, and changes to the location of said information to be retrieved from each said web site.
10. The system of claim 8, wherein each recognition grammar executable code and each instruction set for querying of information to be retrieved are stored in the database.	
11. A method comprising: (a) providing at least one data processor, the data processor operatively coupled to a plurality of communication data networks; (b) providing at least one speaker-independent-speech-recognition engine operatively coupled to the at least one data processor (c) providing	1. A method for retrieving information from web sites by uttering speech commands into a voice enabled device and for providing to users retrieved information in an audio form via said voice enabled device, said method comprising the steps of: providing a computer operatively connected to the

memory accessible to the data processor storing at least: (i) an instruction set for querying of the information to be retrieved from a plurality of sources coupled to the plurality of communication data networks, the instruction set comprising: an indication of the plurality of sources, each identified by an information-source identifier, and each identifying certain information to be retrieved from the information-source identifier, and (ii) at least one recognition grammar executable code corresponding to each instruction set and corresponding to data characterizing audio containing a naturally-spoken-speech command including an information request, (d) the at least one speaker-independent-speech-recognition engine: (i) receiving the data characterizing audio containing the naturally-spoken-speech command from the voice-enabled device via a first of the communication data networks, (ii) recognizing phenomes in the data characterizing audio containing the naturally-spoken-speech commands to understand spoken words, and (iii) generating recognition-results data, (e) the least one data processor programmed to: (i) select the corresponding at least one recognition grammar executable code upon receiving the data characterizing audio containing the naturally-spoken-speech command and convert the data characterizing audio containing naturally-spoken-speech command into a data message for transmission to a network interface adapted to access a second of the one communication networks; and (ii) retrieve the instruction set corresponding to the recognition grammar executable provided by the at least one speaker-independent-speech-recognition device and access the information source identified by the instruction set to obtain at least a part of the information to be retrieved; and (f)	internet, said computer further being operatively connected to at least one speaker-independent speech recognition engine and to at least one speech synthesis engine; providing a voice enabled device operatively connected to said computer, said voice enabled device configured to receive speech commands from users; providing a speech command to said speaker-independent speech recognition engine, said computer accessing at least one of a plurality of web sites associated with said speech command to obtain an information to be retrieved, said computer first accessing a first web site of said plurality of web sites and, if said information to be retrieved is not found at said first web site, said computer sequentially accessing said plurality of web sites until said information to be retrieved is found or until said plurality of web sites has been accessed; said speech synthesis engine producing an audio message containing any retrieved information from said web sites; and said speech synthesis engine transmitting said audio message to said users via said voice enabled device.

providing at least one speech-synthesis device operatively connected to the at least one data processor, and by the at least one speech-synthesis device adapted to: (i) produce an audio message relating to any resulting information retrieved from the plurality of information sources including text-to-speech conversion of said any resulting information retrieved from the plurality of information sources, and (ii) transmit the audio message via the voice-enabled device.	
12. The method of claim 11, wherein the plurality of communication data networks includes the Internet.	2. The method of claim 1 wherein said speech command is further associated with a content descriptor associated with each said web site address, said content descriptor pre-defining a portion of said web site containing said information to be retrieved. 3. The method of claim 1 wherein said speech command is further associated with a ranking from highest to lowest associated with each said web site, said ranking indicating the order in which the plurality of web sites are accessed.
17. The method of claim 11, wherein the instruction set executable code further comprises: a content descriptor associated with each information-source identifier, the content descriptor pre-defining a portion of the information source containing the information to be retrieved.	5. The method of claim 4 further comprising the step of adjusting said rankings associated with said plurality of web sites such that said web site having said information to be retrieved is assigned the highest ranking and any web sites not having said information to be retrieved are assigned lower rankings. 6. The method of claim 1 further comprising the step of periodically polling each said web site to determine whether said web site contains said information to be retrieved. 7. The method of claim 6 wherein the computer periodically polls each said web site without being instructed by said user to determine the availability of each said web site, the duration of time for each said web site to respond to a request from said

	computer, and changes to the location of said information to be retrieved from each said web site, said computer creating a ranking of said plurality of web sites based on said periodic polling. 8. The method of claim 1 further comprising the step of periodically searching said internet to find new web sites containing said information to be retrieved, and adding said new web sites to said plurality of web sites.
18. The method of claim 11, further comprising: providing a database and operatively connecting the database to the data processor and storing the information gathered from the information sources in response to the information requests in the database.	5. The method of claim 4 further comprising the step of adjusting said rankings associated with said plurality of web sites such that said web site having said information to be retrieved is assigned the highest ranking and any web sites not having said information to be retrieved are assigned lower rankings. 6. The method of claim 1 further comprising the step of periodically polling each said web site to determine whether said web site contains said information to be retrieved. 7. The method of claim 6 wherein the computer periodically polls each said web site without being instructed by said user to determine the availability of each said web site, the duration of time for each said web site to respond to a request from said computer, and changes to the location of said information to be retrieved from each said web site, said computer creating a ranking of said plurality of web sites based on said periodic polling. 8. The method of claim 1 further comprising the step of periodically searching said internet to find new web sites containing said information to be retrieved, and adding said new web sites to said plurality of web sites.

7. Claims 1-5, 8-10, and 17-19 are rejected on the ground of nonstatutory double patenting as being unpatentable over claims 1, 6-15 of U.S. Patent No. 7,881,941 Although the claims at issue are not identical, they are not patentably distinct from each other because the claims of the instant application are similar in scope and content of the patented claims issued to the same applicant.

Application No: 16/155,523	Patent No: 7,881,941
1. A system comprising: (a) at least one data processor, the at least one data processor operatively coupled to a plurality of communication data networks; (b) at least one speaker-independent speech-recognition engine operatively coupled to the data processor; (c) memory accessible to the at least one data processor and storing at least: (i) an instruction set for querying of information to be retrieved from a plurality of sources coupled to the plurality of communication data networks, the instruction set comprising: an indication of the plurality of sources, each identified by a source identifier, and each identifying certain information to be retrieved from the source identifier, and (ii) at least one recognition grammar executable code corresponding to each instruction set and corresponding to data characterizing audio containing a naturally-spoken-speech command including an information request, (d) wherein the at least one speaker-independent-speech-recognition engine is adapted to: (1) receive the data characterizing audio containing the naturally-spoken-speech command from a voice-enabled device via a first of the plurality of communication data networks; (2) to recognize phenomes in the data characterizing audio containing naturally-	9. A system for retrieving information from pre-selected web sites by uttering speech commands into a phone and for providing to users retrieved information in an audio form via said phone, said system comprising: a computer, said computer operatively connected to the internet and to at least one phone; at least one speaker-independent speech recognition engine, said speaker-independent speech recognition engine operatively connected to said computer; at least one speech synthesis engine, said speech synthesis engine operatively connected to said computer; a database, said database operatively connected to said computer; at least one instruction set stored in said database for identifying said information to be retrieved, said instruction set comprising: a plurality of pre-selected web site addresses, each said web site address identifying a web site containing said information to be retrieved; a content descriptor associated with each said web site address, said content descriptor pre-defining a portion of said web site containing said information to be retrieved; a ranking from highest to lowest associated with each said web site address, said ranking indicating the order in which the plurality of pre-selected web sites are accessed; at least one

spoken-speech commands to understand spoken words; and (3) to generate recognition results data, (e) wherein the at least one data processor is adapted to: (1) select the corresponding at least one recognition grammar executable code upon receiving the data characterizing the audio containing the naturally-spoken-speech command and to convert the data characterizing the audio containing the naturally-spoken-speech command into a data message for transmission to a network interface adapted to access a second of the plurality of communication data networks; and (2) retrieve the instruction set corresponding to the recognition grammar executable codes provided by the at least one speaker-independent-speech-recognition engine and to access the information source queried by the instruction set to obtain at least a part of the information to be retrieved, and (f) at least one speech-synthesis device operatively coupled to the at least one data processor, the at least one speech-synthesis device configured to produce an audio message relating to any resulting information retrieved from the plurality of information sources including a text-to-speech conversion of at least certain data in said any resulting information retrieved from the plurality of information sources, and to convey the audio message via the voice-enabled device.	recognition grammar stored in said database, each said recognition grammar corresponding to each said instruction set and corresponding to a speech command; said speaker-independent speech recognition engine configured to receive from users via said phone a speech command and to select the corresponding recognition grammar upon receiving said speech command; said computer configured to retrieve said instruction set corresponding to said recognition grammar selected by said speaker-independent speech recognition device; said computer further configured to access at least one of said plurality of web sites identified by said instruction set to obtain said information to be retrieved, said computer configured to first access said web site having the highest ranking and, if said information to be retrieved is not found at said web site having the highest ranking, said computer configured to subsequently access said plurality of web sites in order of rankings until said information to be retrieved is found or until said plurality of web sites has been accessed; said computer further configured to establish or adjust said rankings associated with said plurality of web sites such that said web site having said information to be retrieved is assigned the highest ranking and any web sites not having said information to be retrieved are assigned lower rankings; said speech synthesis engine configured to produce an audio message containing any retrieved information from said pre-selected web sites, and said speech synthesis engine further configured to transmit said audio message to said users via said phone.
2. The system of claim 1, wherein the plurality of communication data networks includes the Internet.	11. The system of claim 9 wherein said internet is a local area network. 12. The system of claim 9 wherein said internet is a wide area network.

	13. The system of claim 9 wherein said internet is the Internet.
3. The system of claim 1, wherein the plurality of communication data networks include a local-area network.	11. The system of claim 9 wherein said internet is a local area network. 12. The system of claim 9 wherein said internet is a wide area network. 13. The system of claim 9 wherein said internet is the Internet.
4. The system of claim 1, wherein the voice-enabled device is a home device.	10. The system of claim 9 wherein said phone comprises a standard telephone, a cellular phone, or an IP phone.
5. The method of claim 1, wherein the voice-enabled device is at least one of a group of an IP telephone, a cellular phone, a personal computer, a media player appliance, and a television or other video display device.	10. The system of claim 9 wherein said phone comprises a standard telephone, a cellular phone, or an IP phone.
6. The system of claim 1, wherein the speaker-independent-speech-recognition engine is adapted to analyze the phonemes to recognize conversational naturally-spoken-speech commands.	
7. The system of claim 1, wherein the speaker-independent-speech-recognition engine is adapted to recognize the naturally-spoken-speech commands.	
8. The system of claim 1, wherein the instruction set executable code further comprises: a content descriptor associated with each information-source identifier, the content descriptor pre-defining a portion of the information source containing the information to be retrieved.	8. The method of claim 1 further comprising the step of periodically searching said internet to find new web sites containing said information to be retrieved, and adding said new web sites to said plurality of web sites. 14. The system of claim 9 wherein said computer is configured to establish or adjust said rankings associated with said plurality of web sites when instructed by said user to access said plurality of web sites to retrieve said information. 15. The system of claim 9 wherein said computer is configured to establish or adjust said rankings associated with said plurality of web sites based on periodic polling of each of

	said web sites without being instructed by said user to determine the availability of each said web site, the duration of time for each said web site to respond to a request from said computer, and changes to the location of said information to be retrieved from each said web site.
9. The system of claim 1, further comprising: a database operatively connected to the data processor, the database adapted to store the information gathered from the information sources in response to the information requests.	8. The method of claim 1 further comprising the step of periodically searching said internet to find new web sites containing said information to be retrieved, and adding said new web sites to said plurality of web sites. 14. The system of claim 9 wherein said computer is configured to establish or adjust said rankings associated with said plurality of web sites when instructed by said user to access said plurality of web sites to retrieve said information. 15. The system of claim 9 wherein said computer is configured to establish or adjust said rankings associated with said plurality of web sites based on periodic polling of each of said web sites without being instructed by said user to determine the availability of each said web site, the duration of time for each said web site to respond to a request from said computer, and changes to the location of said information to be retrieved from each said web site.
10. The system of claim 8, wherein each recognition grammar executable code and each instruction set for querying of information to be retrieved are stored in the database.	8. The method of claim 1 further comprising the step of periodically searching said internet to find new web sites containing said information to be retrieved, and adding said new web sites to said plurality of web sites. 14. The system of claim 9 wherein said computer is configured to establish or adjust said rankings associated with said plurality of web sites when instructed by said user to access said plurality of web sites to retrieve said information.

	15. The system of claim 9 wherein said computer is configured to establish or adjust said rankings associated with said plurality of web sites based on periodic polling of each of said web sites without being instructed by said user to determine the availability of each said web site, the duration of time for each said web site to respond to a request from said computer, and changes to the location of said information to be retrieved from each said web site.
11. A method comprising: (a) providing at least one data processor, the data processor operatively coupled to a plurality of communication data networks; (b) providing at least one speaker-independent-speech-recognition engine operatively coupled to the at least one data processor (c) providing memory accessible to the data processor storing at least: (i) an instruction set for querying of the information to be retrieved from a plurality of sources coupled to the plurality of communication data networks, the instruction set comprising: an indication of the plurality of sources, each identified by an information-source identifier, and each identifying certain information to be retrieved from the information-source identifier, and (ii) at least one recognition grammar executable code corresponding to each instruction set and corresponding to data characterizing audio containing a naturally-spoken-speech command including an information request, (d) the at least one speaker-independent-speech-recognition engine: (i) receiving the data characterizing audio containing the naturally-spoken-speech command from the voice-enabled device via a first of the communication data networks, (ii) recognizing phenomes in the data characterizing audio containing the naturally-spoken-speech commands to understand spoken words, and (iii) generating recognition-results data, (e) the	1. A method for retrieving information from pre-selected web sites by uttering speech commands into a voice enabled device and for providing to users retrieved information in an audio form via said voice enabled device, said method comprising the steps of: providing a computer operatively connected to the internet, said computer further being operatively connected to at least one speaker-independent speech recognition engine and to at least one speech synthesis engine; providing a voice enabled device operatively connected to said computer, said voice enabled device configured to receive speech commands from users; providing at least one instruction set stored in a database operatively connected to said computer, said instruction set comprising: a plurality of pre-selected web site addresses, each said web site address identifying a web site containing said information to be retrieved; providing a speech command to said speaker-independent speech recognition engine, said speech command corresponding to said instruction set; said speaker-independent speech recognition engine assigning said speech command to a recognition grammar, said speech command and said recognition grammar corresponding to said instruction set; transmitting said speech command to said speaker-independent speech recognition engine; said speaker-independent speech recognition engine receiving said speech

least one data processor programmed to: (i) select the corresponding at least one recognition grammar executable code upon receiving the data characterizing audio containing the naturally-spoken-speech command and convert the data characterizing audio containing naturally-spoken-speech command into a data message for transmission to a network interface adapted to access a second of the one communication networks; and (ii) retrieve the instruction set corresponding to the recognition grammar executable provided by the at least one speaker-independent-speech-recognition device and access the information source identified by the instruction set to obtain at least a part of the information to be retrieved; and (f) providing at least one speech-synthesis device operatively connected to the at least one data processor, and by the at least one speech-synthesis device adapted to: (i) produce an audio message relating to any resulting information retrieved from the plurality of information sources including text-to-speech conversion of said any resulting information retrieved from the plurality of information sources, and (ii) transmit the audio message via the voice-enabled device.	command and selecting the corresponding recognition grammar upon receiving said speech command; said computer retrieving said instruction set corresponding to said recognition grammar selected by said speaker-independent speech recognition engine; said computer accessing at least one of said plurality of web sites identified by said instruction set to obtain said information to be retrieved, said computer first accessing said first web site of said plurality of web sites and, if said information to be retrieved is not found at said first web site, said computer sequentially accessing said plurality of web sites until said information to be retrieved is found or until said plurality of web sites has been accessed; said speech synthesis engine producing an audio message containing any retrieved information from said pre-selected web sites; and said speech synthesis engine transmitting said audio message to said users via said voice enabled device.
12. The method of claim 11, wherein the plurality of communication data networks includes the Internet.	
13. The method of claim 11, wherein the plurality of communication data networks include a local-area network.	
14. The method of claim 11, wherein the voice-enabled device is a telephone.	
15. The method of claim 11, wherein the speaker-independent-speech-recognition engine is adapted to analyze the phonemes to recognize conversational naturally-spoken-speech commands.	

16. The method of claim 11, wherein the speaker-independent-speech-recognition engine is adapted to recognize the naturally-spoken-speech commands.	
17. The method of claim 11, wherein the instruction set executable code further comprises: a content descriptor associated with each information-source identifier, the content descriptor pre-defining a portion of the information source containing the information to be retrieved.	6. The method of claim 1 further comprising the step of periodically polling each said web site to determine whether said web site contains said information to be retrieved. 7. The method of claim 6 wherein the computer periodically polls each said web site without being instructed by said user to determine the availability of each said web site, the duration of time for each said web site to respond to a request from said computer, and changes to the location of said information to be retrieved from each said web site, said computer creating a ranking of said plurality of web sites based on said periodic polling.
18. The method of claim 11, further comprising: providing a database and operatively connecting the database to the data processor and storing the information gathered from the information sources in response to the information requests in the database.	6. The method of claim 1 further comprising the step of periodically polling each said web site to determine whether said web site contains said information to be retrieved. 7. The method of claim 6 wherein the computer periodically polls each said web site without being instructed by said user to determine the availability of each said web site, the duration of time for each said web site to respond to a request from said computer, and changes to the location of said information to be retrieved from each said web site, said computer creating a ranking of said plurality of web sites based on said periodic polling.
19. The method of claim 18, wherein each recognition grammar executable code and each instruction set are stored in the database.	6. The method of claim 1 further comprising the step of periodically polling each said web site to determine whether said web site contains said information to be retrieved. 7. The method of claim 6 wherein the computer periodically polls each said web site without being instructed by said user to

	determine the availability of each said web site, the duration of time for each said web site to respond to a request from said computer, and changes to the location of said information to be retrieved from each said web site, said computer creating a ranking of said plurality of web sites based on said periodic polling.
20. The method of claim 18, wherein the voice-enabled device is at least one of a group of an IP telephone, a cellular phone, a personal computer, a media player appliance, and a television or other video display device.	

Conclusion

8. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure. Please see attached form PTO-892.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to VIJAY B CHAWAN whose telephone number is (571)272-7601. The examiner can normally be reached on 7-5 Monday thru Thursday.

Examiner interviews are available via telephone, in-person, and video conferencing using a USPTO supplied web-based collaboration tool. To schedule an interview, applicant is encouraged to use the USPTO Automated Interview Request (AIR) at http://www.uspto.gov/interviewpractice.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Richemond Dorvil can be reached on 571-272-7602. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

/VIJAY B CHAWAN/
Primary Examiner, Art Unit 2658

vbc
1/17/2020

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Notice of References Cited	16/155,523	Kurganov, Alexander	
	Examiner	Art Unit	
	VIJAY B CHAWAN	2658	Page 1 of 2

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*	B	US-20040193427-A1	09-2004	Kurganov, Alexander	H04M3/4938	704/275
*	C	US-20080189113-A1	08-2008	Kurganov; Alexander	H04M3/4938	704/270.1
*	D	US-20020104025-A1	08-2002	Wrench, Edwin H. JR.	H04L9/3271	726/3
*	E	US-20060190265-A1	08-2006	Kurganov; Alexander	H04M3/4938	704/270.1
*	F	US-20110091023-A1	04-2011	Kurganov; Alexander	H04M3/4938	379/88.17
*	G	US-10096320-B1	10-2018	Kurganov; Alexander	H04M3/4938	1/1
*	H	US-10320981-B2	06-2019	Kurganov; Alexander	G06F3/167	1/1
*	I	US-9377992-B2	06-2016	Kurganov; Alexander	G06F3/167	1/1
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*		Document Number Country Code-Number-Kind Code	Date MM-YYYY	Country	Name	CPC Classification
	N					
	O					
	P					
	Q					
	R					
	S					
	T					

NON-PATENT DOCUMENTS

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Notice of References Cited	Application/Control No. 16/155,523	Applicant(s)/Patent Under Reexamination Kurganov, Alexander	
	Examiner VIJAY B CHAWAN	Art Unit 2658	Page 2 of 2

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*		Document Number Country Code-Number-Kind Code	Date MM-YYYY	Name	CPC Classification	US Classification
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*	B	US-7185197-B2	02-2007	Wrench, Jr.; Edwin H.	H04L9/32	713/168
*	C	US-6721705-B2	04-2004	Kurganov; Alexander	H04M3/4938	704/270.1
*	D	US-6144991-A	11-2000	England; Paul	H04L29/06	709/205
*	E	US-7974875-B1	07-2011	Quilici; Alexander E.	G06Q30/0241	705/14.4
*	F	US-7386455-B2	06-2008	Kurganov; Alexander	H04M3/4938	704/270.1
*	G	US-7881941-B2	02-2011	Kurganov; Alexander	H04M3/4938	704/275
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*	I	US-5926789-A	07-1999	Barbara; Daniel	G10L15/22	704/270.1
	J					
	K					
	L					
	M					

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*		Document Number Country Code-Number-Kind Code	Date MM-YYYY	Country	Name	CPC Classification
	N					
	O					
	P					
	Q					
	R					
	S					
	T					

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	U	
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	X	

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Dates in MM-YYYY format are publication dates. Classifications may be US or foreign.

Search Notes	Application/Control No. 16/155,523	Applicant(s)/Patent Under Reexamination Kurganov, Alexander
	Examiner VIJAY B CHAWAN	Art Unit 2658

CPC - Searched*		
Symbol	**Date**	**Examiner**
H04M 2201/40, 3/4938, 2201/39, 2203/609. 3/382, 2201/60, 2203/105, 3/4878, 3/493, 3/4931, 7/12	01/13/2020	vbc
G10L 15/26, 15/265, 15/22, 15/222, 25/48	01/13/2020	vbc
G06F 3/16, 17/30873, 2216/15	01/13/2020	vbc

CPC Combination Sets - Searched*		
Symbol	**Date**	**Examiner**

US Classification - Searched*			
Class	**Subclass**	**Date**	**Examiner**
704	270.1, 275, 200, 270, 246, 272	01/13/2020	vbc
379	88.17, 88.01	01/13/2020	vbc
707	999.101, 999.102, 999.01, 707	01/13/2020	vbc
709	205, 219, 227	01/13/2020	vbc
715	733, 310, 3	01/13/2020	vbc
713	168, 182, 186	01/13/2020	vbc

* See search history printout included with this form or the SEARCH NOTES box below to determine the scope of the search.

Search Notes		
Search Notes	**Date**	**Examiner**
See attached PLUS and WEST searches	01/13/2020	vbc
Inventor/Assignee search in PALM and Pe2e	01/09/2020	vbc
Ip.com, Global Dossier search	01/09/2020	vbc
Google Patents/Scholar search	01/09/2020	vbc

Search Notes	Application/Control No. 16/155,523	Applicant(s)/Patent Under Reexamination Kurganov, Alexander
	Examiner VIJAY B CHAWAN	Art Unit 2658

Interference Search			
US Class/CPC Symbol	US Subclass/CPC Group	Date	Examiner

Index of Claims	Application/Control No. 16/155,523	Applicant(s)/Patent Under Reexamination Kurganov, Alexander
	Examiner VIJAY B CHAWAN	Art Unit 2658

✓	Rejected	-	Cancelled	N	Non-Elected	A	Appeal
=	Allowed	÷	Restricted	I	Interference	O	Objected

CLAIMS

☑ Claims renumbered in the same order as presented by applicant ☐ CPA ☐ T.D. ☐ R.1.47

CLAIM		DATE								
Final	Original	01/14/2020								
	1	✓								
	2	✓								
	3	✓								
	4	✓								
	5	✓								
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	15	✓								
	16	✓								
	17	✓								
	18	✓								
	19	✓								
	20	✓								

Bibliographic Data

Application No: 16/155,523

Foreign Priority claimed: ○ Yes ◉ No

35 USC 119 (a-d) conditions met: ☐ Yes ☑ No ☐ Met After Allowance

Verified and Acknowledged: /VIJAY B CHAWAN/

Examiner's Signature | Initials

Title: Acquiring Information from Sources Responsive to Naturally-Spoken-Speech Commands Provided by a Voice-Enabled Device

FILING or 371(c) DATE	CLASS	GROUP ART UNIT	ATTORNEY DOCKET NO.
10/09/2018	704	2658	10115-07594 US
RULE			

APPLICANTS

Parus Holdings, Inc., Bannockburn, IL,

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Alexander Kurganov Buffalo Grove, IL, UNITED STATES

CONTINUING DATA

This application is a CON of 15269776 09/19/2016 PAT 10096320

15269776 is a CON of 13462819 05/03/2012 PAT 9451084

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Titles of most frequently occurring classifications of patents returned from a search of 16155523 on Dec 23 , 2019

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  8  709/219       (3 OR, 5 XR)
         Class 709     ELECTRICAL COMPUTERS AND DIGITAL PROCESSING SYSTEMS:
MULTICOMPUTER DATA TRANSFERRING
         709/217       .REMOTE DATA ACCESSING
         709/219       ..Accessing a remote server

  8  370/352       (3 OR, 5 XR)
         Class 370     MULTIPLEX COMMUNICATIONS
         370/351       .PATHFINDING OR ROUTING
         370/352       ..Combined circuit switching and packet switching

  6  704/270       (3 OR, 3 XR)
         Class 704     DATA PROCESSING: SPEECH SIGNAL PROCESSING, LINGUISTICS,
LANGUAGE TRANSLATION, AND AUDIO COMPRESSION/DECOMPRESSION
         704/200       .SPEECH SIGNAL PROCESSING
         704/270       ..Application

  6  709/217       (1 OR, 5 XR)
         Class 709     ELECTRICAL COMPUTERS AND DIGITAL PROCESSING SYSTEMS:
MULTICOMPUTER DATA TRANSFERRING
         709/217       .REMOTE DATA ACCESSING

  5  379/88.17     (3 OR, 2 XR)
         Class 379     TELEPHONIC COMMUNICATIONS
         379/67.1      .AUDIO MESSAGE STORAGE, RETRIEVAL, OR SYNTHESIS
         379/88.17     ..Interaction with an external nontelephone network (e.g.,
Internet)

  5  709/203       (1 OR, 4 XR)
         Class 709     ELECTRICAL COMPUTERS AND DIGITAL PROCESSING SYSTEMS:
MULTICOMPUTER DATA TRANSFERRING
         709/201       .DISTRIBUTED DATA PROCESSING
         709/203       ..Client/server

  5  704/270.1     (3 OR, 2 XR)
         Class 704     DATA PROCESSING: SPEECH SIGNAL PROCESSING, LINGUISTICS,
LANGUAGE TRANSLATION, AND AUDIO COMPRESSION/DECOMPRESSION
         704/200       .SPEECH SIGNAL PROCESSING
         704/270       ..Application
         704/270.1     ...Speech assisted network

  4  715/234       (0 OR, 4 XR)
         Class 715     DATA PROCESSING:  PRESENTATION PROCESSING OF DOCUMENT,
OPERATOR INTERFACE PROCESSING, AND SCREEN SAVER DISPLAY PROCESSING
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        715/200       .PRESENTATION PROCESSING OF DOCUMENT
        715/234       ..Structured document (e.g., HTML, SGML, ODA, CDA, etc.)

  4  709/229       (3 OR, 1 XR)
        Class 709     ELECTRICAL COMPUTERS AND DIGITAL PROCESSING SYSTEMS:
MULTICOMPUTER DATA TRANSFERRING
        709/227       .COMPUTER-TO-COMPUTER SESSION/CONNECTION ESTABLISHING
        709/229       ..Network resources access controlling

  4  704/275       (3 OR, 1 XR)
        Class 704     DATA PROCESSING: SPEECH SIGNAL PROCESSING, LINGUISTICS,
LANGUAGE TRANSLATION, AND AUDIO COMPRESSION/DECOMPRESSION
        704/200       .SPEECH SIGNAL PROCESSING
        704/270       ..Application
        704/275       ...Speech controlled system

  4  704/E15.045   (0 OR, 4 XR)
        Class 704     DATA PROCESSING: SPEECH SIGNAL PROCESSING, LINGUISTICS,
LANGUAGE TRANSLATION, AND AUDIO COMPRESSION/DECOMPRESSION
        704/E15.001   .SPEECH RECOGNITION (EPO)
        704/E15.043   ..Speech to text systems (EPO)
        704/E15.045   ...Systems using speech recognizers (EPO)

  3  709/225       (0 OR, 3 XR)
        Class 709     ELECTRICAL COMPUTERS AND DIGITAL PROCESSING SYSTEMS:
MULTICOMPUTER DATA TRANSFERRING
        709/223       .COMPUTER NETWORK MANAGING
        709/225       ..Computer network access regulating

  3  370/401       (1 OR, 2 XR)
        Class 370     MULTIPLEX COMMUNICATIONS
        370/351       .PATHFINDING OR ROUTING
        370/389       ..Switching a message which includes an address header
        370/400       ...Having a plurality of nodes performing distributed
switching
        370/401       ....Bridge or gateway between networks

  3  709/226       (0 OR, 3 XR)
        Class 709     ELECTRICAL COMPUTERS AND DIGITAL PROCESSING SYSTEMS:
MULTICOMPUTER DATA TRANSFERRING
        709/223       .COMPUTER NETWORK MANAGING
        709/226       ..Network resource allocating

  3  707/E17.108   (0 OR, 3 XR)
        Class 707     DATA PROCESSING:  DATABASE AND FILE MANAGEMENT OR DATA
STRUCTURES
        707/E17.001   .INFORMATION RETRIEVAL; DATABASE STRUCTURES THEREFORE (EPO)
        707/E17.107   ..Retrieval from the Internet, e.g., browsers, etc. (EPO)
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707/E17.108 ...By querying, e.g., search engines or meta-search engines, crawling techniques, push systems, etc. (EPO)

2 379/93.25 (2 OR, 0 XR)
Class 379 TELEPHONIC COMMUNICATIONS
379/90.01 .TELEPHONE LINE OR SYSTEM COMBINED WITH DIVERSE ELECTRICAL SYSTEM OR SIGNALLING (E.G., COMPOSITE)
379/93.01 ..Having transmission of a digital message signal over a telephone line
379/93.17 ...Having station display
379/93.25 ....Having remote database (e.g., videotex system)

2 379/93.17 (0 OR, 2 XR)
Class 379 TELEPHONIC COMMUNICATIONS
379/90.01 .TELEPHONE LINE OR SYSTEM COMBINED WITH DIVERSE ELECTRICAL SYSTEM OR SIGNALLING (E.G., COMPOSITE)
379/93.01 ..Having transmission of a digital message signal over a telephone line
379/93.17 ...Having station display

2 707/E17.115 (0 OR, 2 XR)
Class 707 DATA PROCESSING: DATABASE AND FILE MANAGEMENT OR DATA STRUCTURES
707/E17.001 .INFORMATION RETRIEVAL; DATABASE STRUCTURES THEREFORE (EPO)
707/E17.107 ..Retrieval from the Internet, e.g., browsers, etc. (EPO)
707/E17.112 ...By using information identifiers, e.g., encoding URL in specific indicia, browsing history, etc. (EPO)
707/E17.115 ....URL specific, e.g., using aliases, detecting broken or misspelled links, etc. (EPO)

2 379/93.12 (0 OR, 2 XR)
Class 379 TELEPHONIC COMMUNICATIONS
379/90.01 .TELEPHONE LINE OR SYSTEM COMBINED WITH DIVERSE ELECTRICAL SYSTEM OR SIGNALLING (E.G., COMPOSITE)
379/93.01 ..Having transmission of a digital message signal over a telephone line
379/93.12 ...Sales, ordering, or banking system

2 379/90.01 (0 OR, 2 XR)
Class 379 TELEPHONIC COMMUNICATIONS
379/90.01 .TELEPHONE LINE OR SYSTEM COMBINED WITH DIVERSE ELECTRICAL SYSTEM OR SIGNALLING (E.G., COMPOSITE)

2 709/227 (2 OR, 0 XR)
Class 709 ELECTRICAL COMPUTERS AND DIGITAL PROCESSING SYSTEMS: MULTICOMPUTER DATA TRANSFERRING
709/227 .COMPUTER-TO-COMPUTER SESSION/CONNECTION ESTABLISHING

2 704/E15.04 (0 OR, 2 XR)
Class 704 DATA PROCESSING: SPEECH SIGNAL PROCESSING, LINGUISTICS, LANGUAGE TRANSLATION, AND AUDIO COMPRESSION/DECOMPRESSION
704/E15.001 .SPEECH RECOGNITION (EPO)
704/E15.04 ..Procedures used during a speech recognition process, e.g., man-machine dialogue, etc. (EPO)

2 707/E17.119 (0 OR, 2 XR)
Class 707 DATA PROCESSING: DATABASE AND FILE MANAGEMENT OR DATA STRUCTURES
707/E17.001 .INFORMATION RETRIEVAL; DATABASE STRUCTURES THEREFORE (EPO)
707/E17.107 ..Retrieval from the Internet, e.g., browsers, etc. (EPO)
707/E17.119 ...Browsing optimization (EPO)

2 709/202 (0 OR, 2 XR)
Class 709 ELECTRICAL COMPUTERS AND DIGITAL PROCESSING SYSTEMS: MULTICOMPUTER DATA TRANSFERRING
709/201 .DISTRIBUTED DATA PROCESSING
709/202 ..Processing agent

2 370/466 (0 OR, 2 XR)
Class 370 MULTIPLEX COMMUNICATIONS
370/464 .COMMUNICATION TECHNIQUES FOR INFORMATION CARRIED IN PLURAL CHANNELS
370/465 ..Adaptive
370/466 ...Converting between protocols

2 709/205 (1 OR, 1 XR)
Class 709 ELECTRICAL COMPUTERS AND DIGITAL PROCESSING SYSTEMS: MULTICOMPUTER DATA TRANSFERRING
709/204 .COMPUTER CONFERENCING
709/205 ..Cooperative computer processing

2 715/733 (1 OR, 1 XR)
Class 715 DATA PROCESSING: PRESENTATION PROCESSING OF DOCUMENT, OPERATOR INTERFACE PROCESSING, AND SCREEN SAVER DISPLAY PROCESSING
715/700 .OPERATOR INTERFACE (E.G., GRAPHICAL USER INTERFACE)
715/733 ..For plural users or sites (e.g., network)

2 705/14.73 (1 OR, 1 XR)
Class 705 DATA PROCESSING: FINANCIAL, BUSINESS PRACTICE, MANAGEMENT, OR COST/PRICE DETERMINATION
705/1.1 .AUTOMATED ELECTRICAL FINANCIAL OR BUSINESS PRACTICE OR MANAGEMENT ARRANGEMENT
705/14.4 ..Advertisement
705/14.73 ...Online advertisement

2 370/230 (2 OR, 0 XR)

Class 370 MULTIPLEX COMMUNICATIONS
370/229 .DATA FLOW CONGESTION PREVENTION OR CONTROL
370/230 ..Control of data admission to the network

2 370/252 (0 OR, 2 XR)
Class 370 MULTIPLEX COMMUNICATIONS
370/241 .DIAGNOSTIC TESTING (OTHER THAN SYNCHRONIZATION)
370/252 ..Determination of communication parameters

2 704/E13.011 (0 OR, 2 XR)
Class 704 DATA PROCESSING: SPEECH SIGNAL PROCESSING, LINGUISTICS, LANGUAGE TRANSLATION, AND AUDIO COMPRESSION/DECOMPRESSION
704/E13.001 .SPEECH SYNTHESIS; TEXT TO SPEECH SYSTEMS (EPO)
704/E13.011 ..Text analysis, generation of parameters for speech synthesis out of text, e.g., grapheme to phoneme translation, prosody generation, stress, or intonation determination, etc. (EPO)

16155523_CLS

Most frequently occurring classifications of patents returned from a search Of 16155523 on Dec 23 , 2019

Original Classifications

3	379/88.17
3	709/219
3	704/270.1
3	704/270
3	709/229
3	370/352
3	704/275
2	379/93.25
2	709/227
2	370/230

Cross-Reference Classifications

5	709/219
5	709/217
5	370/352
4	709/203
4	715/234
4	704/E15.045
3	709/225
3	704/270
3	709/226
3	707/E17.108
2	379/93.17
2	379/88.17
2	707/E17.115
2	370/401
2	379/93.12
2	379/90.01
2	704/270.1
2	704/E15.04
2	707/E17.119
2	709/202
2	370/466
2	370/252
2	704/E13.011

Combined Classifications

8	709/219
8	370/352
6	704/270
6	709/217
5	379/88.17
5	709/203

5	704/270.1
4	715/234
4	709/229
4	704/275
4	704/E15.045
3	709/225
3	370/401
3	709/226
3	707/E17.108
2	379/93.25
2	379/93.17
2	707/E17.115
2	379/93.12
2	379/90.01
2	709/227
2	704/E15.04
2	707/E17.119
2	709/202
2	370/466
2	709/205
2	715/733
2	705/14.73
2	370/230
2	370/252
2	704/E13.011

WEST Search History for Application 16155523

Creation Date: 2020011409:45

Prior Art Searches

Query	DB	Hits	Op.	Plur.	Thes.	Date
(6104790 6141413 5764906 5799063 5838682 5898839 5926789 5946633 5945989 5950165 5963908 5963217 5983351 6088731 6101537 6108406 6112233 6119155 6134235 6144991 6185194 6188762 6282515 6317594 6363337 6438601 6473097 6484263 6510417 6515964 6546005 6657957 6687341 6687734 6708153 6721705 6735632 6859776 6922726 6950946 6952414 6963556 6982733 6999916 7016848 7054818 7076431 7092370 7103563 7143042).pn.	PGPB, USPT, USOC, EPAB, JPAB, DWPI, TDBD	132	OR	YES		01-14-2020
(6104790 6141413 5764906 5799063 5838682 5898839 5926789 5946633 5945989 5950165 5963908 5963217 5983351 6088731 6101537 6108406 6112233 6119155 6134235 6144991 6185194 6188762 6282515 6317594 6363337 6438601 6473097 6484263 6510417 6515964 6546005 6657957 6687341 6687734 6708153 6721705 6735632 6859776 6922726 6950946 6952414 6963556 6982733 6999916 7016848 7054818 7076431 7092370 7103563 7143042).pn.	USPT	50	OR	YES		01-14-2020
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(kurganov-$.in.)	USPT	32	OR	YES		01-14-2020
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((kurganov-$.in.) and voice)	PGPB, USPT	48	OR	YES		01-14-2020
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information)) and (recogni$6 near2 grammar))	EPAB, JPAB, DWPI, TDBD, FPRS					
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(((kurganov-$.in.) and voice))	USPT	21	OR	YES		01-14-2020
(((kurganov-$.in.) and voice))	PGPB	27	OR	YES		01-14-2020
(((kurganov-$.in.) and voice)) McFadden	PGPB, USPT	8226	OR	YES		01-14-2020
(((kurganov-$.in.) and voice)) and s$-McFadden.xa.	PGPB, USPT	0	OR	YES		01-14-2020
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		19501	OR	YES		01-14-2020

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<tr><td>((((704/270.1))) | (((704/275))) | (((704/200))) | (((704/270))) | (((704/246))) | (((704/272)))).CCLS.</td><td>PGPB, USPT</td><td>10865</td><td>OR</td><td>YES</td><td></td><td>01-14-2020</td></tr>
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<tr><td>((((707/707707))) | (((707/999.101))) | (((707/999.102)))).CCLS.</td><td>PGPB, USPT</td><td>13057</td><td>OR</td><td>YES</td><td></td><td>01-14-2020</td></tr>
<tr><td>((((709/205))) | (((709/219))) | (((709/227)))).CCLS.</td><td>PGPB, USPT</td><td>31639</td><td>OR</td><td>YES</td><td></td><td>01-14-2020</td></tr>
<tr><td>((((713/168))) | (((713/182))) | (((713/186)))).CCLS.</td><td>PGPB, USPT</td><td>13551</td><td>OR</td><td>YES</td><td></td><td>01-14-2020</td></tr>
<tr><td>((((715/733))) | (((719/310))) | (((726/3)))).CCLS.</td><td>PGPB, USPT</td><td>9672</td><td>OR</td><td>YES</td><td></td><td>01-14-2020</td></tr>
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<tr><td>(((((707/707))) | (((707/999.01))) | (((707/999.101))) | (((707/999.102)))).CCLS.) and @pd > 20161208</td><td>PGPB, USPT</td><td>0</td><td>OR</td><td>YES</td><td></td><td>01-14-2020</td></tr>
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<tr><td>kurganov-Al$.in.</td><td>PGPB</td><td>28</td><td>OR</td><td>YES</td><td></td><td>01-14-2020</td></tr>
<tr><td>kurganov-Al$.in.</td><td>USPT</td><td>22</td><td>OR</td><td>YES</td><td></td><td>01-14-2020</td></tr>
<tr><td>mcfadden-Su$.xp.</td><td>USPT</td><td>1145</td><td>OR</td><td>YES</td><td></td><td>01-14-2020</td></tr>
<tr><td>(mcfadden-Su$.xp.) and kurganov-Al$.in.</td><td>USPT</td><td>6</td><td>OR</td><td>YES</td><td></td><td>01-14-2020</td></tr>
<tr><td>mcfadden-Su$.xa.</td><td>USPT</td><td>62</td><td>OR</td><td>YES</td><td></td><td>01-14-2020</td></tr>
<tr><td>(mcfadden-Su$.xa.) and kurganov-Al$.in.</td><td>USPT</td><td>0</td><td>OR</td><td>YES</td><td></td><td>01-14-2020</td></tr>
<tr><td>(mcfadden-Su$.xa. and kurganov-Al$.in.) and @pd > 20200109</td><td>USPT</td><td>0</td><td>OR</td><td>YES</td><td></td><td>01-14-2020</td></tr>
<tr><td>kurganov-Al$.in.</td><td>USPT</td><td>22</td><td>OR</td><td>YES</td><td></td><td>01-14-2020</td></tr>
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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

U.S.PATENTS Remove

Examiner Initial*	Cite No	Patent Number	Kind Code1	Issue Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear
/V.B.C/	1	6501966		2002-12-31	Bareis et al.	
	2	6505163		2003-01-07	Zhang et al.	
	3	6529948		2003-03-04	Bowman-Amuah	
	4	6532444		2003-03-11	Weber	
	5	6539359		2003-03-25	Ladd et al.	
	6	6546393		2003-04-08	Khan	
	7	6560604		2003-05-06	Fascenda	
/V.B.C/	8	6584439		2003-06-24	Geilhufe et al.	

ALL REFERENCES CONSIDERED EXCEPT WHERE LINED THROUGH. /V.B.C/

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	Examiner Name	
	Attorney Docket Number	10115-07594 US

/V.B.C/	9	6587822		2003-07-01	Brown et al.	
	10	6593944		2003-07-15	Nicolas et al.	
	11	6594348		2003-07-15	Bjurstrom et al.	
	12	6594692		2003-07-15	Reisman	
	13	6606611		2003-08-12	Khan	
	14	6618039		2003-09-09	Grant et al.	
	15	6618726		2003-09-09	Colbath et al.	
	16	6618763		2003-09-09	Steinberg	
	17	6633846		2003-10-14	Bennett et al.	
	18	6636831		2003-10-21	Profit Jr. et al.	
/V.B.C/	19	6654814		2003-11-25	Britton et al.	

EFS Web 2.1.18

ALL REFERENCES CONSIDERED EXCEPT WHERE LINED THROUGH. /V.B.C/

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	Application Number	
	Filing Date	2018-10-09
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	Attorney Docket Number	10115-07594 US

/V.B.C/	20	6665640		2003-12-16	Bennett et al.	
	21	6687341		2004-02-03	Koch et al.	
	22	6704024		2004-03-09	Robotham et al.	
	23	6718015		2004-04-06	Berstis	
	24	6721705		2004-04-13	Kurganov et al.	
	25	6724868		2004-04-20	Pradhan et al.	
	26	6732142		2004-05-04	Bates et al.	
	27	6763388		2004-07-13	Tsimelzon	
	28	6771732		2004-08-03	Xiao et al.	
	29	6775264		2004-08-10	Kurganov	
/V.B.C/	30	6785266		2004-08-31	Swartz	

ALL REFERENCES CONSIDERED EXCEPT WHERE LINED THROUGH. /V.B.C/

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	Examiner Name	
	Attorney Docket Number	10115-07594 US

/V.B.C/	31	6807257		2004-10-19	Kurganov	
	32	6812939		2004-11-02	Flores et al.	
	33	6823370		2004-11-23	Kredo et al.	
	34	6859776		2005-02-22	Cohen et al.	
	35	6888929		2005-05-03	Saylor et al.	
	36	6922733		2005-07-26	Kuiken et al.	
	37	6941273		2005-09-06	Loghmani et al.	
	38	6964012		2005-11-08	Zirngibl et al.	
	39	6964023		2005-11-08	Maes et al.	
	40	6965864		2005-11-15	Thrift et al.	
/V.B.C/	41	6996609		2006-02-07	Hickman et al.	

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	Filing Date	2018-10-09
	First Named Inventor	Alexander Kurganov
	Art Unit	
	Examiner Name	
	Attorney Docket Number	10115-07594 US

/V.B.C/	42	6999804		2006-02-14	Engstrom et al.	
	43	7003463		2006-02-21	Maes et al.	
	44	7024464		2006-04-04	Lusher et al.	
	45	7050977		2006-05-23	Bennett	
	46	7075555		2006-07-11	Flores et al.	
	47	7076431		2006-07-11	Kurganov et al.	
	48	7089307		2006-08-08	Zintel et al.	
	49	7145898		2006-12-05	Elliott	
	50	7146323		2006-12-05	Guenther et al.	
	51	7185197		2007-02-27	Wrench Jr.	
/V.B.C/	52	7327723		2008-02-05	Kurganov	

EFS Web 2.1.18

ALL REFERENCES CONSIDERED EXCEPT WHERE LINED THROUGH. /V.B.C/

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	First Named Inventor	Alexander Kurganov
	Art Unit	
	Examiner Name	
	Attorney Docket Number	10115-07594 US

/V.B.C/	53	7386455		2008-06-10	Kurganov et al.	
	54	7506022		2009-03-17	Wang et al.	
	55	7516190		2009-04-07	Kurganov	
	56	7881941		2011-02-01	Kurganov et al.	
	57	7974875		2011-07-05	Quilici et al.	
	58	8098600		2012-01-17	Kurganov	
	59	8131267		2012-03-06	Lichorowic et al.	
	60	8131555		2012-03-06	Carriere et al.	
	61	8185402		2012-05-22	Kurganov et al.	
	62	8380505		2013-02-19	Konig et al.	
/V.B.C/	63	8775178		2014-07-08	Miki et al.	

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	Examiner Name	
	Attorney Docket Number	10115-07594 US

/V.B.C/	64	8838074		2014-09-16	Kurganov	
	65	8843141		2014-09-23	Kurganov	
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	First Named Inventor	Alexander Kurganov
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	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):

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A signature of the applicant or representative is required in accordance with CFR 1.33, 10.18. Please see CFR 1.4(d) for the form of the signature.

Signature	/Reena Kuyper/	Date (YYYY-MM-DD)	2018-10-09
Name/Print	Reena Kuyper	Registration Number	33,830

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	22	4635253		1987-01-06	Urui et al.	
	23	4652700		1987-03-24	Matthews et al.	
	24	4696028		1987-09-22	Morganstein et al.	
	25	4713837		1987-12-15	Gordon	
	26	4747127		1988-05-24	Hansen et al.	
	27	4748656		1988-05-31	Gibbs et al.	
	28	4755932		1988-07-05	Diedrich	
	29	4757525		1988-07-12	Matthews et al.	
/V.B.C/	30	4761807		1988-08-02	Matthews et al.	

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/V.B.C/	31	4763317		1988-08-09	Lehman et al.	
	32	4769719		1988-09-06	Endo	
	33	4771425		1988-09-13	Baran et al.	
	34	4776016		1988-10-04	Hansen	
	35	4782517		1988-11-01	Bernardis et al.	
	36	4792968		1988-12-20	Katz	
	37	4799144		1989-01-17	Parruck et al.	
	38	4809321		1989-02-28	Morganstein et al.	
	39	4811381		1989-03-07	Woo et al.	
	40	4837798		1989-06-06	Cohen et al.	
/V.B.C/	41	4847891		1989-07-11	Kotani	

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/V.B.C/	42	4850012		1989-07-18	Mehta et al.	
	43	4852149		1989-07-25	Zwick et al.	
	44	4852170		1989-07-25	Bordeaux	
	45	4866758		1989-09-12	Heinzelmann	
	46	4873719		1989-10-10	Reese	
	47	4879743		1989-11-07	Burke et al.	
	48	4893333		1990-01-09	Baran et al.	
	49	4893335		1990-01-09	Fuller et al.	
	50	4903289		1990-02-20	Hashimoto	
	51	4903291		1990-02-20	Tsurufuji et al.	
/V.B.C/	52	4905273		1990-02-27	Gordon et al.	

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/V.B.C/	53	4907079		1990-03-06	Turner et al.	
	54	4918722		1990-04-17	Duehren et al.	
	55	4922518		1990-05-01	Gordon et al.	
	56	4922520		1990-05-01	Bernard et al.	
	57	4922526		1990-05-01	Morganstein et al.	
	58	4926462		1990-05-15	Ladd et al.	
	59	4930150		1990-05-29	Katz	
	60	4933966		1990-06-12	Hird et al.	
	61	4935955		1990-06-19	Neudorfer	
	62	4935958		1990-06-19	Morganstein et al.	
/V.B.C/	63	4941170		1990-07-10	Herbst	

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/V.B.C/	64	4942598		1990-07-17	Davis	
	65	4953204		1990-08-28	Cuschleg Jr. et al.	
	66	4955047		1990-09-04	Morganstein et al.	
	67	4956835		1990-09-11	Grover	
	68	4959854		1990-09-25	Cave et al.	
	69	4967288		1990-10-30	Mizutori et al.	
	70	4969184		1990-11-06	Gordon et al.	
	71	4972462		1990-11-20	Shibata	
	72	4974254		1990-11-27	Perine et al.	
	73	4975941		1990-12-04	Morganstein et al.	
/V.B.C/	74	4985913		1991-01-15	Shalom et al.	

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/V.B.C/	75	4994926		1991-02-19	Gordon et al.	
	76	4996704		1991-02-26	Brunson	
	77	5003575		1991-03-26	Chamberlin et al.	
	78	5003577		1991-03-26	Ertz et al.	
	79	5008926		1991-04-16	Misholi	
	80	5020095		1991-05-28	Morganstein et al.	
	81	5027384		1991-06-25	Morganstein	
	82	5029196		1991-07-02	Morganstein	
	83	5036533		1991-07-30	Carter et al.	
	84	5054054		1991-10-01	Pessia et al.	
/V.B.C/	85	5065254		1991-11-12	Hishida	

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/V.B.C/	86	5086385		1992-02-04	Launey et al.	
	87	5095445		1992-03-10	Sekiguchi	
	88	5099509		1992-03-24	Morganstein et al.	
	89	5109405		1992-04-28	Morganstein	
	90	5128984		1992-07-07	Katz	
	91	5131024		1992-07-14	Pugh et al.	
	92	5133004		1992-07-21	Heileman Jr. et al.	
	93	5145452		1992-09-08	Chevalier	
	94	5146452		1992-09-08	Pekarske	
	95	5166974		1992-11-24	Morganstein et al.	
/V.B.C/	96	5179585		1993-01-12	MacMillan Jr. et al.	

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/V.B.C/	97	5193110		1993-03-09	Jones et al.	
	98	5195086		1993-03-16	Baumgartner et al.	
	99	5233600		1993-08-03	Pekarske	
	100	5243643		1993-09-07	Sattar et al.	
	101	5243645		1993-09-07	Bissell et al.	
	102	5249219		1993-09-28	Morganstein et al.	
	103	5255305		1993-10-19	Sattar	
	104	5263084		1993-11-16	Chaput et al.	
	105	5276729		1994-01-04	Higuchi et al.	
	106	5287199		1994-02-15	Zoccolillo	
/V.B.C/	107	5291302		1994-03-01	Gordon et al.	

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/V.B.C/	108	5291479		1994-03-01	Vaziri et al.	
	109	5303298		1994-04-12	Morganstein et al.	
	110	5307399		1994-04-26	Dai et al.	
	111	5309504		1994-05-03	Morganstein	
	112	5325421		1994-06-28	Hou et al.	
	113	5327486		1994-07-05	Wolff et al.	
	114	5327529		1994-07-05	Fults et al.	
	115	5329578		1994-07-12	Brennan et al.	
	116	5333266		1994-07-26	Boaz et al.	
	117	5347574		1994-09-13	Morganstein	
/V.B.C/	118	5355403		1994-10-11	Richardson Jr. et al.	

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/V.B.C/	119	5359598		1994-10-25	Steagall et al.	
	120	5365524		1994-11-15	Hiller et al.	
	121	5365574		1994-11-15	Hunt et al.	
	122	5375161		1994-12-20	Fuller et al.	
	123	5384771		1995-01-24	Isidoro et al.	
	124	5404231		1995-04-04	Bloomfield	
	125	5408526		1995-04-18	McFarland et al.	
	126	5414754		1995-05-09	Pugh et al.	
	127	5416834		1995-05-16	Bales et al.	
	128	5426421		1995-06-20	Gray	
/V.B.C/	129	5432845		1995-07-11	Burd et al.	

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/V.B.C/	130	5436963		1995-07-25	Fitzpatrick et al.	
	131	5459584		1995-10-17	Gordon et al.	
	132	5463684		1995-10-31	Morduch et al.	
	133	5475791		1995-12-12	Schalk et al.	
	134	5479487		1995-12-26	Hammond	
	135	5495484		1996-02-27	Self et al.	
	136	5497373		1996-03-05	Hulen et al.	
	137	5499288		1996-03-12	Hunt et al.	
	138	5515427		1996-05-07	Carlsen et al.	
	139	5517558		1996-05-14	Schalk	
/V.B.C/	140	5526353		1996-06-11	Henley et al.	

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/V.B.C/	141	5533115		1996-07-02	Hollenbach et al.	
	142	5537461		1996-07-16	Bridges et al.	
	143	5555100		1996-09-10	Bloomfield et al.	
	144	5559611		1996-09-24	Bloomfield et al.	
	145	5559859		1996-09-24	Dai et al.	
	146	5566236		1996-10-15	MeLampy et al.	
	147	5603031		1997-02-11	White et al.	
	148	5608786		1997-03-04	Gordon	
	149	5610910		1997-03-11	Focsaneanu et al.	
	150	5610970		1997-03-11	Fuller et al.	
/V.B.C/	151	5611031		1997-03-11	Hertzfeld et al.	

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/V.B.C/	152	5630079		1997-05-13	McLaughlin	
	153	5652789		1997-07-29	Miner et al.	
	154	5657376		1997-08-12	Espeut et al.	
	155	5659597		1997-08-19	Bareis et al.	
	156	5666401		1997-09-09	Morganstein et al.	
	157	5675507		1997-10-07	Bobo II	
	158	5675811		1997-10-07	Broedner et al.	
	159	5689669		1997-11-18	Lynch et al.	
	160	5692187		1997-11-25	Goldman et al.	
	161	5699486		1997-12-16	Tullis et al.	
/V.B.C/	162	5712903		1998-01-27	Bartholomew et al.	

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/V.B.C/	163	5719921		1998-02-17	Vysotsky et al.	
	164	5721908		1998-02-24	Lagarde et al.	
	165	5724408		1998-03-03	Morganstein	
	166	5737395		1998-04-07	Irribarren	
	167	5742596		1998-04-21	Baratz et al.	
	168	5742905		1998-04-21	Pepe et al.	
	169	5752191		1998-05-12	Fuller et al.	
	170	5758322		1998-05-26	Rongley	
	171	5761294		1998-06-02	Shaffer et al.	
	172	5764639		1998-06-09	Staples et al.	
/V.B.C/	173	5764736		1998-06-09	Shachar et al.	

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/V.B.C/	174	5764910		1998-06-09	Shachar	
	175	5774860		1998-06-30	Bayya et al.	
	176	5787298		1998-07-28	Broedner et al.	
	177	5793993		1998-08-11	Broedner et al.	
	178	5794205		1998-08-11	Walters et al.	
	179	5796791		1998-08-18	Polcyn	
	180	5799063		1998-08-25	Krane	
	181	5799065		1998-08-25	Junqua et al.	
	182	5809282		1998-09-15	Cooper et al.	
	183	5809481		1998-09-15	Baron et al.	
/V.B.C/	184	5812796		1998-09-22	Broedner et al.	

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/V.B.C/	185	5819220		1998-10-06	Sarukkai et al.	
	186	5819306		1998-10-06	Goldman et al.	
	187	5822727		1998-10-13	Garberg et al.	
	188	5823879		1998-10-20	Goldberg et al.	
	189	5832063		1998-11-03	Vysotsky et al.	
	190	5832440		1998-11-03	Woodbridge et al.	
	191	5835570		1998-11-10	Wattenbarger	
	192	5838682		1998-11-17	Dekelbaum et al.	
	193	5867494		1999-02-02	Krishnaswamy et al.	
	194	5867495		1999-02-02	Elliott et al.	
/V.B.C/	195	5870550		1999-02-09	Wesinger Jr. et al.	

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/V.B.C/	196	5873080		1999-02-16	Coden et al.	
	197	5881134		1999-03-09	Foster et al.	
	198	5881135		1999-03-09	Watts et al.	
	199	5884032		1999-03-16	Bateman et al.	
	200	5884262		1999-03-16	Wise et al.	
	201	5884266		1999-03-16	Dvorak	
	202	5890123		1999-03-30	Brown et al.	
	203	5905476		1999-05-18	McLaughlin et al.	
	204	5914951		1999-06-22	Bentley et al.	
	205	5915001		1999-06-22	Uppaluru	
/V.B.C/	206	5917817		1999-06-29	Dunn et al.	

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/V.B.C/	207	5926789		1999-07-20	Barbara et al.	
	208	5940598		1999-08-17	Strauss et al.	
	209	5943399		1999-08-24	Bannister et al.	
	210	5946389		1999-08-31	Dold	
	211	5953392		1999-09-14	Rhie et al.	
	212	5974124		1999-10-26	Schlueter Jr. et al.	
	213	5974413		1999-10-26	Beauregard et al.	
	214	5991292		1999-11-23	Focsaneanu et al.	
	215	5995615		1999-11-30	Miloslavsky	
	216	5999525		1999-12-07	Krishnaswamy et al.	
/V.B.C/	217	5999611		1999-12-07	Tatchell et al.	

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/V.B.C/	218	5999965		1999-12-07	Kelly	
	219	6012088		2000-01-04	Li et al.	
	220	6014437		2000-01-11	Acker et al.	
	221	6014626		2000-01-11	Cohen	
	222	6018710		2000-01-25	Wynblatt et al.	
	223	6021181		2000-02-01	Miner et al.	
	224	6021190		2000-02-01	Fuller et al.	
	225	6031904		2000-02-29	An et al.	
	226	6038305		2000-03-14	McAllister et al.	
	227	6044107		2000-03-28	Gatherer et al.	
/V.B.C/	228	6047053		2000-04-04	Miner et al.	

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/V.B.C/	229	6052372		2000-04-18	Gittins et al.	
	230	6067516		2000-05-23	Levay et al.	
	231	6078580		2000-06-20	Mandalia et al.	
	232	6081518		2000-06-27	Bowman-Amuah	
	233	6081782		2000-06-27	Rabin	
	234	6091808		2000-07-18	Wood et al.	
	235	6101472		2000-08-08	Giangarra et al.	
	236	6104803		2000-08-15	Weser et al.	
	237	6115737		2000-09-05	Ely et al.	
	238	6115742		2000-09-05	Franklin et al.	
/V.B.C/	239	6130933		2000-10-10	Miloslavsky	

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/V.B.C/	240	6131095		2000-10-10	Low et al.	
	241	6137863		2000-10-24	Brown et al.	
	242	6144991		2000-11-07	England	
	243	6157705		2000-12-05	Perrone	
	244	6161128		2000-12-12	Smyk	
	245	6178399		2001-01-23	Takebayashi et al.	
	246	6185535		2001-02-06	Hedin et al.	
	247	6188683		2001-02-13	Lang et al.	
	248	6195357		2001-02-27	Polcyn	
	249	6199076		2001-03-06	Logan et al.	
/V.B.C/	250	6201814		2001-03-13	Greenspan	

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/V.B.C/	251	6201863		2001-03-13	Miloslavsky	
	252	6208638		2001-03-27	Rieley et al.	
	253	6215858		2001-04-10	Bartholomew et al.	
	254	6230132		2001-05-08	Class et al.	
	255	6233318		2001-05-15	Picard et al.	
	256	6243373		2001-06-05	Turock	
	257	6252944		2001-06-26	Hansen II et al.	
	258	6269336		2001-07-31	Ladd et al.	
	259	6285745		2001-09-04	Bartholomew et al.	
	260	6327572		2001-12-04	Morton et al.	
/V.B.C/	261	6330538		2001-12-11	Breen	

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/V.B.C/	262	6349132		2002-02-19	Wesemann et al.	
	263	6353661		2002-03-05	Bailey III	
	264	6366575		2002-04-02	Barkan et al.	
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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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See attached certification statement.

The fee set forth in 37 CFR 1.17 (p) has been submitted herewith.

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A signature of the applicant or representative is required in accordance with CFR 1.33, 10.18. Please see CFR 1.4(d) for the form of the signature.

Signature	/Reena Kuyper/	Date (YYYY-MM-DD)	2018-10-09
Name/Print	Reena Kuyper	Registration Number	33,830

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/V.B.C/	8	WO199107838	WO			Fuller Research and Development Company		
/V.B.C/	9	WO199118466	WO			Messager Partners		
/V.B.C/	10	WO199609710	WO			Xantel Corporation		
/V.B.C/	11	WO1997034401	WO			Xantel Corporation		
/V.B.C/	12	WO1997037481	WO			Northern Telecom Ltd		
/V.B.C/	13	WO1998023058	WO			Premiere Communications Inc		

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/V.B.C/	1	Brief of Appellees, submitted on June 20, 2016, to the United States Court of Appeals for the Federal Circuit, 53 pages.	
/V.B.C/	2	COLE et al., "An Architecture for a Mobile OSI Mail Access System", IEEE Journal on Selected Areas in Communications, vol. 7 (2), February 1989, pp 249-256.	
/V.B.C/	3	DAX Systems, Inc., Press Release, "Speech Recognition Success in DAXs Grasp", Nov. 22, 1995, pp. 1-2, Pine Brook, NJ.	

ALL REFERENCES CONSIDERED EXCEPT WHERE LINED THROUGH. /V.B.C/

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

/V.B.C/	4		Defendants Answer to the Amended Complaint and Demand for Jury Trial, Parus Holdings, Inc. v. Web Telephone LLC Robert Swartz, Case no. 06-cv-01146 (N.D. Ill.), August 10,2006, 14 pages.	
/V.B.C/	5		Examples: Abstract Ideas, 2014, 20 pages.	
/V.B.C/	6		GARCIA ET AL., "Issue in Multimedia Computer-Based Message Systems Design and Standardization", NATO ASI Series, vol. 1-6, 1984, 10 pgs.	
/V.B.C/	7		HEMPHILL et al. "Speech-Aware Multimedia," IEEE Multimedia, Spring 1996, Vol. 3, No. 1. pp. 74-78, IEEE.	
/V.B.C/	8		HEMPHILL et al., "Surfing the Web by Voice," ACM Multimedia 95 - Electronic Proceedings, November 5-9, 1995. San Francisco, CA.	
/V.B.C/	9		HUNT et al., "Long-Distance Remote Control to the Rescue", Chicago Tribune, June 15,2002, Section 4, page 15.	
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/V.B.C/	12		IBM AIX DirectTalk/6000 Version 1 Release 6 Improves Your Voice Processing Services to Callers and Customers, Announcement Number: 295-489, November 28, 1995, 27 pages.	
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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	

/V.B.C/	15		Joint Appendix, submitted on September 16, 2016, to the United States Court of Appeal for the Federal Circuit, 406 pages.*	
/V.B.C/	16		Judgment without Opinion for Parus Holdings Inc., v. Sallie Mae Bank, Navient Solutions Inc., PNC Bank, N.A., Suntrust Bank, Suntrust Mortgage Inc., 2016-1179, 2016-1180, 2016-1181, entered February 27, 2017 (2 pages).*	
/V.B.C/	17		KUBALA et al., "BYBLOS Speech Recognition Benchmark Results:, Workshop on Speech Natural Language, February 19-22, 1991.	
/V.B.C/	18		LY, "Chatter: A Conversational Telephone Agent", submitted to Program in Media Arts, Sciences, MIT, 1993, pp.1-130.	
/V.B.C/	19		MAEDA, et al., "An Intelligent Customer-Controlled Switching System", IEEE Global Telecommunications Conference, Hollywood, Florida, Nov. 28-Dec. 1, 1988, pp. 1499-1503.	
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/V.B.C/	21		MARX et al., "Mail Call: Message Presentation and Navigation in a Nonvisual Environment," SIGCHI Conference on Human Factors in Computing Systems, Vancouver, B.C. Canada, April 13-18, 1996.	
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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

/V.B.C/	26		OYE, Phil, "Juggler", p. 1, available at http://www.philoye.comiwork/jugglec2.shtml (accessed on December 8, 2006).	
/V.B.C/	27		OYE, Phil, "Juggler", p. 1, available at http://www.philoye.comlwork/jugglec3.shtml (accessed on December 8, 2006).	
/V.B.C/	28		OYE, Phil, "Juggler", p. 1, available at http://www.philoye.comlwork/juggler/index.shtml (accessed on December 8,2006).	
/V.B.C/	29		Paper No. 10, Denying Institution of Covered Business Method Patent Review CBM2015-00109 and CBM2015-00149, Nov. 9, 2015, 19 pages.	
/V.B.C/	30		Paper No. 10, Denying Institution of Covered Business Method Patent Review CBM2015-00110 and CBM2015-00150, Nov. 9, 2015, 20 pages.	
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/V.B.C/	33		PERDUE et al., "Conversant® 1 Voice System: Architecture and Applications", July 17, 1986, AT&T Technical Journal, pp. 1-14.	
/V.B.C/	34		Plaintiff Parus Holdings, Inc.' s Supplemental Response to Defendant Web Telephony LLC's First Set of Interrogatories (Nos. 1-12), Parus Holdings, Inc. v. Web Telephony LLC & Robert Swartz, Case No. 06-cv-01146 (N.D. III.), October 31,2006,32 pages.	
/V.B.C/	35		Plaintiff Parus Holdings, Inc.' s Supplemental Response to Defendant Web Telephony LLC's Second Set of Interrogatories (Nos. 13-17), Parus Holdings, Inc. v. Web Telephony LLC & Robert Swartz, Case No. 06-cv- 01146 (N. D. III.), October 31, 2006,31 pages.	
/V.B.C/	36		"Print outs of Internet web site, ""Wildfire Communications, Inv.,"", November 5, 1997, including print outs of the following web pages: http://www.wildfire.com; http://www.wildfire.comlconsumerhome.html; http://www.wildfire.comll06.html;http://www.wildfire.comlcarrierhome.html; http://www.wildfire.comlsfandb.html; http://www.wildfire.comlabout.html; http://www.wildfire.comlabtmgmt.html; http://www.wildfire.comlscoop.html; http://www.wildfire.comiintel.html; and http://www.wildfire.comlmsft.html."	

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	Filing Date	2018-10-09
	First Named Inventor	Alexander Kurganov
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	Examiner Name	
	Attorney Docket Number	10115-07594 US

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/V.B.C/	38		PUTZ, Steve, "Interactive Information Services Using World-Wide Web Hypertext, First Int'l Conference on World-Wide Web (May 25-27, 1994), 10 pages.	
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/V.B.C/	41		SARTORI, M., "Speech Recognition", April 1995, pp. 1-9, Mercury Communications, available at www.gar.co.ukltechnology watch/speech.htm (accessed Mar. 15, 2005).	
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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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Examiner Signature	/VIJAY B CHAWAN/	Date Considered	01/09/2020

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	First Named Inventor	Alexander Kurganov
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CERTIFICATION STATEMENT

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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☐ That no item of information contained in the information disclosure statement was cited in a communication from a foreign patent office in a counterpart foreign application, and, to the knowledge of the person signing the certification after making reasonable inquiry, no item of information contained in the information disclosure statement was known to any individual designated in 37 CFR 1.56(c) more than three months prior to the filing of the information disclosure statement. See 37 CFR 1.97(e)(2).

See attached certification statement.

The fee set forth in 37 CFR 1.17 (p) has been submitted herewith.

X A certification statement is not submitted herewith.

SIGNATURE

A signature of the applicant or representative is required in accordance with CFR 1.33, 10.18. Please see CFR 1.4(d) for the form of the signature.

Signature	/Reena Kuyper/	Date (YYYY-MM-DD)	2018-10-09
Name/Print	Reena Kuyper	Registration Number	33,830

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APPLICATION NUMBER	FILING OR 371(C) DATE	FIRST NAMED APPLICANT	ATTY. DOCKET NO./TITLE
16/155,523	10/09/2018	Alexander Kurganov	10115-07594 US

CONFIRMATION NO. 9544

PUBLICATION NOTICE

OC000000105711047

93219
Patent Law Works, LLP
310 East 4500 South, Suite 400
Salt Lake City, UT 84107

Title: Acquiring Information from Sources Responsive to Naturally-Spoken-Speech Commands Provided by a Voice-Enabled Device

Publication No.US-2019-0043505-A1
Publication Date:02/07/2019

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PATENT APPLICATION FEE DETERMINATION RECORD Substitute for Form PTO-875	Application or Docket Number 16/155,523

APPLICATION AS FILED - PART I

	(Column 1)	(Column 2)	SMALL ENTITY		OR	OTHER THAN SMALL ENTITY	
FOR	NUMBER FILED	NUMBER EXTRA	RATE($)	FEE($)		RATE($)	FEE($)
BASIC FEE (37 CFR 1.16(a), (b), or (c))	N/A	N/A	N/A	75		N/A	
SEARCH FEE (37 CFR 1.16(k), (i), or (m))	N/A	N/A	N/A	330		N/A	
EXAMINATION FEE (37 CFR 1.16(o), (p), or (q))	N/A	N/A	N/A	380		N/A	
TOTAL CLAIMS (37 CFR 1.16(i))	20 minus 20 =	*	x 50 =	0.00	OR		
INDEPENDENT CLAIMS (37 CFR 1.16(h))	2 minus 3 =	*	x 230 =	0.00			
APPLICATION SIZE FEE (37 CFR 1.16(s))	If the specification and drawings exceed 100 sheets of paper, the application size fee due is $310 ($155 for small entity) for each additional 50 sheets or fraction thereof. See 35 U.S.C. 41(a)(1)(G) and 37 CFR 1.16(s).			0.00			
MULTIPLE DEPENDENT CLAIM PRESENT (37 CFR 1.16(j))				0.00			
* If the difference in column 1 is less than zero, enter "0" in column 2.			TOTAL	785		TOTAL	

APPLICATION AS AMENDED - PART II

AMENDMENT A		(Column 1)		(Column 2)	(Column 3)	SMALL ENTITY		OR	OTHER THAN SMALL ENTITY	
		CLAIMS REMAINING AFTER AMENDMENT		HIGHEST NUMBER PREVIOUSLY PAID FOR	PRESENT EXTRA	RATE($)	ADDITIONAL FEE($)		RATE($)	ADDITIONAL FEE($)
	Total (37 CFR 1.16(i))	*	Minus	**	=	x =		OR	x =	
	Independent (37 CFR 1.16(h))	*	Minus	***	=	x =		OR	x =	
	Application Size Fee (37 CFR 1.16(s))									
	FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j))							OR		
						TOTAL ADD'L FEE		OR	TOTAL ADD'L FEE	

AMENDMENT B		(Column 1)		(Column 2)	(Column 3)					
		CLAIMS REMAINING AFTER AMENDMENT		HIGHEST NUMBER PREVIOUSLY PAID FOR	PRESENT EXTRA	RATE($)	ADDITIONAL FEE($)		RATE($)	ADDITIONAL FEE($)
	Total (37 CFR 1.16(i))	*	Minus	**	=	x =		OR	x =	
	Independent (37 CFR 1.16(h))	*	Minus	***	=	x =		OR	x =	
	Application Size Fee (37 CFR 1.16(s))							OR		
	FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j))									
						TOTAL ADD'L FEE		OR	TOTAL ADD'L FEE	

* If the entry in column 1 is less than the entry in column 2, write "0" in column 3.
** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 20, enter "20".
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APPLICATION NUMBER	FILING or 371(c) DATE	GRP ART UNIT	FIL FEE REC'D	ATTY.DOCKET.NO	TOT CLAIMS	IND CLAIMS
16/155,523	10/09/2018	2447	785	10115-07594 US	20	2

CONFIRMATION NO. 9544

FILING RECEIPT

OC000000103476277

93219
Patent Law Works, LLP
310 East 4500 South, Suite 400
Salt Lake City, UT 84107

Date Mailed: 11/01/2018

Receipt is acknowledged of this non-provisional patent application. The application will be taken up for examination in due course. Applicant will be notified as to the results of the examination. Any correspondence concerning the application must include the following identification information: the U.S. APPLICATION NUMBER, FILING DATE, NAME OF APPLICANT, and TITLE OF INVENTION. Fees transmitted by check or draft are subject to collection. Please verify the accuracy of the data presented on this receipt. **If an error is noted on this Filing Receipt, please submit a written request for a Filing Receipt Correction. Please provide a copy of this Filing Receipt with the changes noted thereon. If you received a "Notice to File Missing Parts" for this application, please submit any corrections to this Filing Receipt with your reply to the Notice. When the USPTO processes the reply to the Notice, the USPTO will generate another Filing Receipt incorporating the requested corrections**

Inventor(s)

Alexander Kurganov, Buffalo Grove, IL;

Applicant(s)

Parus Holdings, Inc., Bannockburn, IL;

Power of Attorney: The patent practitioners associated with Customer Number 93219

Domestic Priority data as claimed by applicant

This application is a CON of 15/269,776 09/19/2016 PAT 10096320
which is a CON of 13/462,819 05/03/2012 PAT 9451084
which is a CON of 12/973,475 12/20/2010 PAT 8185402
which is a CON of 12/030,556 02/13/2008 PAT 7881941
which is a CON of 11/409,703 04/24/2006 PAT 7386455
which is a CON of 10/821,690 04/09/2004 PAT 7076431
which is a CON of 09/776,996 02/05/2001 PAT 6721705
which claims benefit of 60/233,068 09/15/2000
and claims benefit of 60/180,344 02/04/2000

Foreign Applications for which priority is claimed (You may be eligible to benefit from the **Patent Prosecution Highway** program at the USPTO. Please see http://www.uspto.gov for more information.) - None.
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Projected Publication Date: 02/07/2019
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**** SMALL ENTITY ****
Title

Acquiring Information from Sources Responsive to Naturally-Spoken-Speech Commands Provided by a Voice-Enabled Device

Preliminary Class

709

Statement under 37 CFR 1.55 or 1.78 for AIA (First Inventor to File) Transition Applications: No

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PTO/AIA/82A (07-13)
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Application Number	
Filing Date	October 9, 2018
First Named Inventor	Alexander Kurganov
Title	Acquiring Information from Sources Responsive to Naturally-Spoken-Speech Commands Provided by a Voice-Enabled Device
Art Unit	
Examiner Name	
Attorney Docket Number	10115-07594 US

SIGNATURE of Applicant or Patent Practitioner

Signature	/Reena Kuyper/	Date (Optional)	2018-10-09
Name	Reena Kuyper	Registration Number	33,830
Title (if Applicant is a juristic entity)	Agent for Applicant		
Applicant Name (if Applicant is a juristic entity)	Parus Holding, Inc.		

NOTE: This form must be signed in accordance with 37 CFR 1.33. See 37 CFR 1.4(d) for signature requirements and certifications. If more than one applicant, use multiple forms.

☑ *Total of One(1) forms are submitted.

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I hereby revoke all previous powers of attorney given in the application identified in the attached statement under 37 CFR 3.73(c).

I hereby appoint:

☒ Practitioners associated with Customer Number: 93219

OR

☐ Practitioner(s) named below (if more than ten patent practitioners are to be named, then a customer number must be used):

Name	Registration Number	Name	Registration Number

As attorney(s) or agent(s) to represent the undersigned before the United States Patent and Trademark Office (USPTO) in connection with any and all patent applications assigned only to the undersigned according to the USPTO assignment records or assignments documents attached to this form in accordance with 37 CFR 3.73(c).

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Firm or Individual Name			
Address			
City		State	Zip
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Assignee Name and Address: Parus Holdings, Inc.
3000 Lakeside Drive, Suite 300N
Bannockburn, IL 60015

A copy of this form, together with a statement under 37 CFR 3.73(c) (Form PTO/AIA/96 or equivalent) is required to be Filed in each application in which this form is used. The statement under 37 CFR 3.73(c) may be completed by one of The practitioners appointed in this form, and must identify the application in which this Power of Attorney is to be filed.

SIGNATURE of Assignee of Record

The individual whose signature and title is supplied below is authorized to act on behalf of the assignee

Signature	RCMcConnell	Date	5/21/2013
Name	Robert C. McConnell	Telephone	888-387-3481
Title	Chief Financial Officer, Senior Vice President and General Counsel of Parus Holdings, Inc.		

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DECLARATION (37 CFR 1.63) FOR UTILITY OR DESIGN APPLICATION USING AN APPLICATION DATA SHEET (37 CFR 1.76)

Title of Invention	Robust Voice Browser System and Voice Activated Device Controller

As the below named inventor, I hereby declare that:

This declaration is directed to:

☐ The attached application, or

☒ United States application or PCT international application number 15/269,776

filed on September 19, 2016.

The above-identified application was made or authorized to be made by me.

I believe that I am the original inventor or an original joint inventor of a claimed invention in the application.

I hereby acknowledge that any willful false statement made in this declaration is punishable under 18 U.S.C. 1001 by fine or imprisonment of not more than five (5) years, or both.

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LEGAL NAME OF INVENTOR

Inventor: Alexander Kurganov Date (Optional) :

Signature:

Note: An application data sheet (PTO/SB/14 or equivalent), including naming the entire inventive entity, must accompany this form or must have been previously filed. Use an additional PTO/AIA/01 form for each additional inventor.

ACQUIRING INFORMATION FROM SOURCES RESPONSIVE TO NATURALLY-SPOKEN-SPEECH COMMANDS PROVIDED BY A VOICE-ENABLED DEVICE

CROSS REFERENCE TO RELATED APPLICATIONS

[0001] This application is a continuation of Application Serial No. 15/269,776, entitled "Acquiring Information from Sources Responsive to Naturally-Spoken-Speech Commands Provided by a Voice-Enabled Device," filed September 19, 2016, which is a continuation of 13/462,819, entitled "Robust Voice Browser System and Voice Activated Device Controller," filed May 3, 2012, which is a continuation of Application Serial No. 12/973,475, entitled "Robust Voice Browser System And Voice Activated Device Controller," filed December 20, 2010, which is a continuation of Application Serial No. 12/030,556, entitled "Robust Voice Browser System And Voice Activated Device Controller," filed February 13, 2008, now U.S. Patent No. 7,881,941, which is a continuation application of Application Serial No. 11/409,703, entitled "Robust Voice Browser System And Voice Activated Device Controller," filed April 24, 2006, now US Patent No. 7,386,455, which is a continuation application of Application Serial No. 10/821,690, entitled "Robust Voice Browser System And Voice Activated Device Controller," filed April 9, 2004, now US Patent No. 7,076,431, which is a continuation application of Application Serial No. 09/776,996, entitled "Robust Voice Browser System And Voice Activated Device Controller," filed February 5, 2001, now US Patent No. 6,721,705, which claims priority to U.S. Provisional Application Serial No. 60/180,344, entitled "Voice Activated Information Retrieval System," filed February 4, 2000 and U.S. Provisional Application No. 60/233,068, filed September 15, 2000, entitled "Robust Voice Browser System and Voice Activated Device Controller, all assigned to the assignee of the present application.

The subject matter in the above-identified co-pending and commonly owned applications is incorporated herein by reference.

FIELD OF THE INVENTION

[0002] The present invention relates to a robust and highly reliable system that allows users to browse web sites and retrieve information by using conversational voice commands. Additionally, the present invention allows users to control and monitor other systems and devices that are connected the Internet or any other network by using voice commands.

BACKGROUND OF THE INVENTION

[0003] Currently, three options exist for a user who wishes to gather information from a web site accessible over the Internet. The first option is to use a desktop or a laptop computer connected to a telephone line via a modem or connected to a network with Internet access. The second option is to use a Personal Digital Assistant (PDA) that has the capability of connecting to the Internet either through a modem or a wireless connection. The third option is to use one of the newly designed web-phones or web-pagers that are now being offered on the market. Although each of these options provide a user with access to the Internet to browse web sites, each of them have their own drawbacks.

[0004] Desktop computers are very large and bulky and are difficult to transport. Laptop computers solve this inconvenience, but many are still quite heavy and are inconvenient to carry. Further, laptop computers cannot be carried and used everywhere a user travels. For instance, if a user wishes to obtain information from a remote location where no electricity or communication lines are installed, it would be nearly impossible to use a laptop computer. Oftentimes, information is needed on an immediate basis where a computer is not accessible. Furthermore, the use of laptop or desktop computers to access the Internet requires either a direct or a dial-up connection tan an Internet Service Provider (ISP). Oftentimes, such connections are not available when a user desires to connect to the Internet to acquire information.

[0005] The second option for remotely accessing web sites is the use of PDAs. These devices also have their own set of drawbacks. First, PDAs with the ability to connect to the Internet and access web sites are not readily available. As a result, these PDAs tend to be very expensive. Furthermore, users are usually required to pay a special service fee to enable the web browsing feature of the PDA. A further disadvantage of these PDAS is that web sites must be specifically designed to allow these devices to access information on the web site. Therefore, a limited number of web sites are available that are accessible by these web-enabled PDAs. Finally, it is very common today for users to carry cell phones, however, users must also carry a separate PDA if they require the ability to gather information from various web sites. Users are therefore subjected to added expenses since they must pay for both cellular telephone service and also for the web-enabling service for the PDA. This results in a very expensive alternative for the consumer.

[0006] The third alternative mentioned above is the use of web-phones or web-pagers. These devices suffer many of the same drawbacks as PDAs. First, these devices are expensive to purchase. Further, the number of web sites accessible to these devices is limited since web sites must be specifically designed to allow access by these devices. Furthermore, users are often required to pay an additional fee in order to gain wireless web access. Again, this service is expensive. Another drawback of these web-phones or web-pagers is that as technology develops, the methods used by the various web sites to allow access by these devices may change. These changes may require users to purchase new web-phones or web-pagers or have the current device serviced in order to upgrade the firmware or operating system stored within the device. At the least, this would be inconvenient to users and may actually be quite expensive.

[0007] Therefore, a need exists for a system that allows users to easily access and browse the Internet from any location. Such a system would only require users to have access to any type of telephone and would not require users to subscribe to multiple services.

[0008] In the rapidly changing area of Internet applications, web sites change frequently. The design of the web site may change, the information required by the web site in order to perform searches may change, and the method of reporting search results may change. Web browsing applications that submit search requests and interpret responses from these web sites based upon expected formats will produce errors and useless responses when such changes occur. Therefore, a need exists for a system that can detect modifications to web sites and adapt to such changes in order to quickly and accurately provide the information requested by a user through a voice enabled device, such as a telephone.

[0009] When users access web sites using devices such as personal computers, delays in receiving responses are tolerated and are even expected, however, such delays are not expected when a user communicates with a telephone. Users expect communications over a telephone to occur immediately with a minimal amount of delay time. A user attempting to find information using a telephone expects immediate responses to his search requests. A system that introduces too much delay between the time a user makes a request and the time of response will not be tolerated by users and will lose its usefulness. Therefore, it is important that a voice browsing system that uses telephonic communications selects web sites that provide rapid responses since speed is an important factor for maintaining the system's desirability and usability. Therefore, a need exists for a system that accesses web sites based upon their speed of operation.

SUMMARY OF THE INVENTION

[0010] It is an object of an embodiment of the present invention to allow users to gather information from web sites by using voice enabled devices, such as wireline or wireless telephones.

[0011] An additional object of an embodiment of the present invention is to provide a system and method that allows the searching and retrieving of publicly available information by controlling a web browsing server using naturally spoken voice commands.

[0012] It is an object of another embodiment of the present invention to provide a robust voice browsing system that can obtain the same information from several web sites based upon a ranking order. The ranking order is automatically adjusted if the system detects that a given web site is not functioning, is too slow, or has been modified in such a way that the requested information cannot be retrieved any longer.

[0013] A still further object of an embodiment of the present invention is to allow users to gather information from web sites from any location where a telephonic connection can be made.

[0014] Another object of an embodiment of the present invention is to allows users to browse web sites on the Internet using conversational voice commands spoken into wireless or wireline telephones or other voice enabled devices.

[0015] An additional object an embodiment of the present invention is to provide a system and method for using voice commands to control and monitor devices connected to a network.

[0016] It is an object of an embodiment of the present invention to provide a system and method which allows devices connected to a network to be controlled by conversational voice commands spoken into any voice enabled device interconnected with the same network.

[0017] The present invention relates to a system for acquiring information from sources on a network, such as the Internet. A voice browsing system maintains a database containing a list of information sources, such as web sites, connected to a network. Each of the information sources is assigned a rank number which is listed in the database along with the record for the information source. In response to a speech command received from a user, a network interface system accesses the information source with the highest rank number in order to retrieve information requested by the user.

[0018] The a preferred embodiment of the present invention allows users to access and browse web sites when they do not have access to computers with Internet access. This is accomplished by providing a voice browsing system and method that allows users to browse web sites using conversational voice commands spoken into any type of voice enabled device (i.e., any type of wireline or wireless telephone, LP phone, wireless PDA, or other wireless device). These spoken commands are then converted into data messages by a speech recognition software engine running on a user interface system. These data messages are then sent to and processed by a network interface system. This network interface system then generates the proper requests that are transmitted to the desired web site over the Internet. Responses sent from the web site are received and processed by the network interface system and then converted into an audio message via a speech synthesis engine or a pre-recorded audio concatenation application and finally transmitted to the user's voice enabled device.

[0019] A preferred embodiment of the voice browser system and method uses a web site polling and ranking methodology that allows the system to detect changes in web sites and adapt to those changes in real-time. This enables the voice browser system of a preferred embodiment to deliver highly reliable information to users over any voice enabled device. This ranking system also enables the present invention to provide rapid responses to user requests. Long delays before receiving responses to requests are not tolerated by users of voice-based systems, such as telephones. When a user speaks into a telephone, an almost immediate response is expected. This expectation does not exist for non-voice communications, such as email transmissions or accessing a web site using a personal computer. In such situations, a reasonable amount of transmission delay is acceptable. The ranking system of implemented by a preferred embodiment of the present invention ensures users will always receive the fastest possible response to their request.

[0020] An alternative embodiment of the present invention allows users to control and monitor the operation of a variety of household devices connected to a network using speech commands spoken into a voice enabled device.

BRIEF DESCRIPTION OF THE DRAWINGS

[0021] FIG. l is a depiction of the voice browsing system of the first embodiment of the present invention;

[0022] FIG. 2 is a block diagram of a database record used by the first preferred embodiment of the present invention;

[0023] FIG. 3 is a block diagram of a media server used by the preferred embodiment;

[0024] FIG. 4 is a block diagram of a web browsing server used by the preferred embodiment; and

[0025] FIG. 5 is a depiction of the device browsing system of the second embodiment of the present invention.

DETAILED DESCRIPTION OF THE INVENTION

[0026] A first embodiment of the present invention is a system and method for allowing users to browse information sources, such as web sites, by using naturally spoken, conversational voice commands spoken into a voice enabled device. Users are not required to learn a special language or command set in order to communicate with the voice browsing system of the present invention. Common and ordinary commands and phrases are all that is required for a user to operate the voice browsing system. The voice browsing system recognizes naturally spoken voice commands and is speaker-independent; it does not have to be trained to recognize the voice patterns of each individual user. Such speech recognition systems use phonemes to recognize spoken words and not predefined voice patterns.

[0027] The first embodiment allows users to select from various categories of information and to search those categories for desired data by using conversational voice commands. The voice browsing system of the first preferred embodiment includes a user interface system referred to as a media server. The media server contains a speech recognition software engine. This speech recognition engine is used to recognize natural, conversational voice commands spoken by the user and converts them into data messages based on the available recognition grammar. These data messages are then sent to a network interface system. In the first preferred embodiment, the network interface system is referred to as a web browsing server. The web browsing server then accesses the appropriate information source, such as a web site, to gather information requested by the user.

[0028] Responses received from the information sources are then transferred to the media server where speech synthesis engine converts the responses into audio messages that are transmitted to the user. A more detailed description of this embodiment will now be provided.

[0029] Referring to FIG. 1, a database 100 designed by Webley Systems Incorporated is connected to one or more web browsing servers 102 as well as to one or more media servers 106.

The database may store information on magnetic media, such as a hard disk drive, or it may store information via other widely acceptable methods for storing data, such as optical disks. The database 100 contains a separate set of records for each web site accessible by the system. An example of a web site record is shown in FIG. 2. Each web site record 200 contains the rank number of the web site 202, the associated Uniform Resource Locator (URL) 204, and a command that enables the appropriate "extraction agent" 206 that is required in order to generate proper requests sent to and to format data received from the web site. The database record 200 also contains the timestamp 208 indicating the last time the web site was accessed. The extraction agent is described in more detail below. The database 100 categorizes each database record 200 according to the type of information provided by each web site. For instance, a first category of database records 200 may correspond to web sites that provide "weather" information. The database 100 may also contain a second category of records 200 for web sites that provide "stock" information. These categories may be further divided into subcategories. For instance, the "weather" category may contain subcategories depending upon type of weather information available to a user, such as "current weather" or "extended forecast". Within the "extended forecast" subcategory, a list of web site records may be stored that provide weather information for multiple days. The use of subcategories may allow the web browsing feature to provide more accurate, relevant, and up-to-date information to the user by accessing the most relevant web site. The number of records contained in each category or subcategory is not limited. In the preferred embodiment, three web site records are provided for each category.

[0030] Table 1 below depicts two database records 200 that are used with the preferred embodiment. These records also contain a field indicating the "category" of the record, which is "weather" in each of these examples.

[0031]

TABLE 1

category: weather
URL:URL=http://cgi.cnn.com/cgi-bin/weather/redirect?zip=_zip
rank: 1
command: web_dispatch.pl weather_cnn
browsingServer: wportal1
browsingServerBackup: wportal2
dateTime: Dec 21 2000 2:15PM

category: weather
URL: URL= http: //weather.lycos.com/wcfiveday.asp?city=zip
rank: 2
command: web_dispatch.pl weather_lycos
browsingServer: wportal1
browsingServerBackup: wportal2
dateTime: Dec 21 2000 1:45PM

[0032] The database also contains a listing of pre-recorded audio files used to create concatenated phrases and sentences. Further, database 100 may contain customer profile information, system activity reports, and any other data or software servers necessary for the testing or administration of the voice browsing system.

[0033] The operation of the media servers 106 will now be discussed in relation to FIG. 3. The media servers 106 function as user interface systems. In the preferred embodiment, the media servers 106 contain a speech recognition engine 300, a speech synthesis engine 302, an Interactive Voice Response (IVR) application 304, a call processing system 306, and telephony and voice hardware 308 required to communicate with the Public Switched Telephone Network (PSTN) 116. In the preferred embodiment, each media server is based upon Intel's Dual Pentium III 730 MHz microprocessor system.

[0034] The speech recognition function is performed by a speech recognition engine 300 that converts voice commands received from the user's voice enabled device 112 (i.e., any type of wireline or wireless telephone, Internet Protocol (IP) phones, or other special wireless units) into data messages. In the preferred embodiment, voice commands and audio messages are transmitted using the PSTN 116 and data is transmitted using the TCP/IP communications protocol. However, one skilled in the art would recognize that other transmission protocols may be used for either voice or data. Other possible transmission protocols would include SIP/VoIP (Session Initiation Protocol/Voice over IP), Asynchronous Transfer Mode (ATM) and Frame Relay. A preferred speech recognition engine is developed by Nuance Communications of 1380 Willow Road, Menlo Park, Calif. 94025 (www.nuance.com). The Nuance engine capacity is measured in recognition units based on CPU type as defined in the vendor specification. The natural speech recognition grammars (i.e., what a user can say that will be recognized by the speech recognition engine) were developed by Webley Systems.

[0035] Table 2 below provides a partial source code listing of the recognition grammars used by the speech recognition engine of the preferred embodiment for obtaining weather information.

[0036]

TABLE 2

```
?WHAT_IS ?the weather?[info information report conditions]
                    ?( (?like in)
                    ]
UScities:n
                    {<paraml $n.zip> <param2    $n.city> <param3
                    $n.state>}
                     (  (area code)  AREA_CODE:n ) {<paraml $n>}
                     (  AREA_CODE:n (area code) ) {paraml $n>}
                     ( (ZIP_CODE:n (zip ?code) ) {<paraml $n>}
                     ]
                     )
                              ) {<mem 194>}
```

[0037] The media server 106 uses recognition results generated by the speech recognition engine 300 to retrieve a web site record 200 stored in the database 100 that can provide the information requested by the user. The media server 106 processes the recognition result data identifying keywords that are used to search the web site records 200 contained in the database 100. For instance, if the user's request was "What is the weather in Chicago?", the keywords "weather" and "Chicago" would be recognized. A web site record 200 with the highest rank number from the "weather" category within the database 100 would then be selected and transmitted to the web browsing server 102 along with an identifier indicating that Chicago weather is being requested.

[0038] The media servers 106 also contain a speech synthesis engine 302 that converts the data retrieved by the web browsing servers 102 into audio messages that are transmitted to the user's voice enabled device 112. A preferred speech synthesis engine is developed by Lernout and Hauspie Speech Products, 52 Third Avenue, Burlington, Mass. 01803 (www.lhsl.com).

[0039] A further description of the web browsing server 102 will be provided in relation to FIG. 4. The web browsing servers 102 provide access to any computer network such as the Internet 110. These servers are also capable of accessing databases stored on Local Area Networks (LANs) or Wide Area Networks (WANs). The web browsing servers receive responses from web sites and extract the data requested by the user. This task is also known as "content extraction." The web browsing servers 102 also perform the task of periodically polling or "pinging" various web sites and modifying the ranking numbers of these web sites depending upon their response and speed. This polling feature is further discussed below. The web browsing server 102 is comprised of a content extraction agent 400, a content fetcher 402, a polling and ranking agent 404, and the content descriptor files 406. Each of these are software applications and will be discussed below.

[0040] Upon receiving a web site record 200 from the database 100 in response to a user request, the web browsing server 102 invokes the "content extraction agent" command 206 contained in the record 200. The content extraction agent 400 allows the web browsing server 102 to properly format requests and read responses provided by the web site 114 identified in the URL field 204 of the web site record 200. Each content extraction agent command 206 invokes the content extraction agent and identifies a content description file associated with the web page identified by the URL 204. This content description the directs the extraction agent where to extract data from the accessed web page and how to format a response to the user utilizing that data. For example, the content description for a web page providing weather information would indicate where to insert the "city" name or ZIP code in order to retrieve Chicago weather information. Additionally, the content description file for each supported URL indicates the location on the web page where the response information is provided. The extraction agent 400 uses this information to properly extract from the web page the information requested by the user.

[0041] Table 3 below contains source code for a content extraction agent 400 used by the preferred embodiment.

[0042]

TABLE 3

```
#!/usr/local/www/bin/sybper15
#$Header:
/usr/local/cvsroot/webley/agents/service/web_dispatch.pl,v
1.6
# Dispatches all web requests
#http://wcorp.itn.net/cgi/flstat?carrier=ua&flight_no=155&mo
n_abbr=jul&date=
6&stamp=OhLN~PdbuuE*itn/ord,itn/cb/sprint_hd
#http://cgi.cnnfn.com/flightview/rlm?airline=amt&number=300
require “config_tmp.pl” ;
#check parameters
die “Usage: $0 service [params]\n” if $#ARGV < 1;
#print STDERR @ARGV;
#get parameters
my %Services = (
            weather_cnn => ‘webget.pl weather_cnn’,
            weather_lycos => ‘webget.pl
weather_lycos’,
            weather_weather => ‘webget.pl
weather_weather’,
            weather_snap => ‘webget.pl
weather_snap’,
            weather_infospace => ‘webget.pl
weather_infospace’,
            stockQuote_yahoo => ‘webget.pl stock’,
            flightStatus_itn => ‘webget.pl
flight_delay’,
            yellowPages_yahoo => ‘yp_data.pl’,
            yellowPages_yahoo => ‘yp_data.pl’,
            newsHeaders_newreal => ‘news.pl’,
            newsArticle_newsreal => ‘news.pl’,
            ) ;
# test param
my $date = ‘date’;
```

TABLE 3-continued

```
chop ( $date );
my ( $short_date ) = $date =~ / \s+ (\w{3}\s+\d{1,2}\s+/;
my %Test = (
                weather_cnn => '60053',
                weather_lycos => '60053',
                weather_weather => '60053',
                weather_snap => '60053',
                weather_infospace => '60053',
                stockQuote_yahoo => 'msft',
                flightStatus_itn => 'ua 155' .
$short_date,
                yellowPages_yahoo => 'tires 60015',
                newsHeaders_newsreal => '1' ,
                newsArticle_newsreal => '1 1',
                );
die "$date: $0: error: no such service: $service (check this
script)\n"
unless $Services {$service };

# prepare absolute path to run other scripts
my ( $path, $script ) = $0 =~ ml^ (*/) ([^/]*) | ;

# store the service to compare against datatable
my $service_stored = $service;

# run service
While ( !( $response = '$path$Services [$service }@param' ) )
{
        # response failed
        # check with test parameters
        $response = '$path$Services{service }$Test{$service
}" ;
        if (    $response ) {
                $service = &switch_service ( $service );
#               print "Wrong parameter values were supplied:
$service –
@param\n";
#               die "$date: $0: error: wrong parameters: $service
-
@param\n";
        }
        else    {
                # change priority and notify
                $service = &increase_attempt ( $service ) ;
        }
```

```
}

# output the response
print $response;
sub increase_attempt {
          my ( $service ) = @_;
          my ( $service_name ) + split ( /_/, $service );
          print STDERR "$date: $0: attn.: changing priority for
service:
$service\n";
          # update priority
          &db_query( "update mcServiceRoute "
                    / "set priority = ( select max ( priority )
from
mcServiceRoute "
                    . "where service = '$service_name' ) + 1
"
                    . " date = getdate( ), "
                    . "attempt = attempt + 1 "
                    . "where route = '$script $service' " );
          #         print "---$route===\n";
                    # find new route
                    my $route = @{&db_query ( "select route from
          mcServiceRoute "
                                        . "where service =
          '$service_name' "
                                        . "and attempt < 5
          "
                                        . "order by
          priority ")
                                        } -> [0] {route };
                    &db_query ( "update mcServiceRoute "
                              . "set attempt = ) "
                              . "where route = '$script $service' " )
                         if ( $route eq "$script $service_stored" ) ;
                                   or $route eq "$script $service_stored" ) ;
                    ( $service_name, $service ) = split ( / \ s + /, $route );
                    die "$date: $0: error: no route for the service:
          $service (add
          more) \n"
                              unless $service;
                    return $service;
          }
          sub switch_swervice {
                    my ( $service ) = @_;
```

TABLE 3-continued

```
                my ( $service_name ) = split (  /_/, $service );
                print STDERR “$date: $0: attn.: changing priority for

        service:
        $service\n”;
                # update priority
                &db_query ( “update mcServiceRoute “
                                . “set priority = ( select max ( priority )

from
mcServiceRoute ”
                        . “ where service = ‘$service_name’ ) + 1,
“
                        . “ date = getdate ( ) ”
                        . “ where route = ‘$script $service’ ” ) ;
#       print “---$route===\n”;
        # find new route
        my $route = @ {$db_query ( “select route from
mcServiceRoute ”
                                        . “where service =
‘$service_name’ ”
                                        . “ and attempt < 5
“
                                        . “ order by priority ”)
                                } - > [0] {route };
        die “ $date: $0: error: there is the only service:
$route (add
more) \n”
                if ( $route eq “$script $service”
                        or $route eq “$script $service_stored” ) ;
        ( $service_name, $service ) = split ( / \s+/, $route );
        die “$date: $0: error: no route for the service:
$service        (add
more) \n”
                        nless $service;
                return $service;
}       -
```

[0043] Table 4 below contains source code of the content fetcher 402 used with the content extraction agent 400 to retrieve information from a web site.

TABLE 4

```
# !/usr/local/www/bin/sybper15
#-T
# -w
# $Header:
/usr/local/cvsroote/Webley/agents/service/webget.pl,v 1.4
# Agent to get info from the web.
# Parameters: service_name [service_parameters], i.e. stock
msft or weather
60645
# Configuration stored in files service_name.ini
# if this file is absent the configuration is received from
mcServices table
# This script provides autoupdate to datatable if the .ini
file is newer.
$debug= 1;
use URI : : URL;
use LWP : : UserAgent;
use HTTP : : Request : : Common;
use Vail : : VarList;
use Sybase : : CTlib;
use HTTP : : Cookies;
#print "Sybase: : CTlib $DB_USER, $DB_PWD, $DB_SRV;" ;
open (STDERR, ">>$0.log" ) if $debug;
#open ( STDERR, ">&STDOUT" ) ;
$log = 'date';
#$response = './url.pl
http://cgi.cnn.com/cgi-bin/weather/redirect?zip=60605" ';
#$response = 'pwd';
#print STDERR " ls = $response\n" ;
#$response = 'ls';
#print STDERR "ls = $response\n";
chop ( $log ) ;
$log .= "pwd=" . 'pwd';
chop ( $log );
#$debug2 = 1;
my $service = shift;
$log .= " $service: ". Join( ' : ', @ARGV ) . "\n" ;
Print STDERR $log if $debug;
#$response = './url.pl
"/http://cgi.cnn.com/cgi-bin/weather/redirect?zip=60605" ';
my @ini = &read_ini ( $service );
chop ( @ini );
my $section = " " ;
do ($section = &process_section ( $section ) }while $section;
```

```
#$response = './url/pl
"http://cgi.cnn.com/cgi-bin/weather/redirect?zip=60605" ';
exit;
sub read_ini {
        my ( $service ) = @_;
        my @ini= ( );
        # first, try to read file
        $0 =~ m|^ (.*/) [^/]*|;
        $service = $1 . $service;
        if ( open ( INI, "$service.ini" ) ) {
                @ ini = ( <INI> );
                return @ ini unless ( $DB_SRV );
                # update datatable
                My $file_time = time – int ( ( -M "$service.ini" )
* 24*
3600 ) ;
#               print "time $file_time\n";
                my $dbh = new Sybase: : CTlib $DB_USR, $DB_PWD,
$DB_SRV;
                unless ( $dbh ) {
                        print STDERR "webget.pl: Cannot connect to
dataserver $DB_SRV:$DB_USR:$DB_PWD\n";
                        return @ini;
                }
                my @row_refs = $dbh ->ct_sql ( "select lastUpdate
from
mcServices where service = '$service' ", undef, 1 );
        if ($dbh -> RC } == CS_FAIL ) {
                print STDERR "webget.pl: DB select from
mcServices
failed\n";
                return @ini;
        }
        unless (defined @ row_refs ) {

                # have to insert
                my ( @ini_escaped ) = map {
                        ( my $x = $_ ) =~ s / \' / \ ' \' / g;
                        $x;
                }@ini;
                $dbh-> ct_sql ( "insert mcServices values (
'$service',
'@ini_escaped', $file_time )" );
                if ( $dbh-> {RC } =- CS_FAIL ) {
                        print STDERR "webget,pl: DB insert to
```

TABLE 4-continued

```
mcServices failed\n";
                }
                return @ini;
        }
        Print "time $file_time:".$row_refs [ 0 ] –
>{'lastUpdate'
} . " \n" ;
                if ( $file_time > $row_refs [0] => {'lastUpdate' })
{
                        # have to update
                        my ( @ini_escaped ) = map {
                                (my $x = $_ ) =~s/ \'/ \'\'/g;
                                $x;
                        }@ini;
                        $dbh->ct_sql ( "update mcServices set config =
'@ini_escaped', lastUpdate = $file_time where service =
'$service' " );
                        if ( $dbh -> {RC } ==CS_FAIL ) {
                                print STDERR "webget.pl: DB update to
mcServices failed\n";
                        }
                }
                return @ini;

        }
        else {
                print STDERR "$0: WARNING: $service.ini n/a in" .
'pwd'
                        . "Try to read DB\n";
        }
        # then try to read datatable
        die "webget.pl: Unable to find service $service\n"
unless ( $DB_SRV)
) ;
        my $ dbh = new Sybase : : CTlib $DB_USR, $DB_PWD, $DB_SRV;
        die "webgetlpl: Cannot connect to dataserver
$DB_SRV : $DB_USR: $DB_PWD\n" unless ( $dbh );
        my @row_refs = $dbh -> ct_sql ( "select config from
mcServices where
service = '$service' ", undef, 1 );
        die "webget.pl: Unable to find service $service\n"
unless ( defined
@row_refs ) ;
        $row_refs [0] -> {'config' } =~ s/\n /\n\r/g;
        @ini = split ( / \ r /, $row_refs [0] -> {'config' } );
```

TABLE 4-continued

```
        return @ini;
}       -
################################################################################
sub process_section {
        my ( $prev_section ) = @_;
        my ( $section, $output, $content );
        my %Param;
        my %Content;
#       print" #####################\n";
         foreach ( @ini ) {
#               print;
#               chop;
                s/ \s+ $ / / ;
                s/ ^ \s + / /;
                # get section name
                if ( /^\ [ (.*) / ] ) {
#                       print "$_: $section:$prev_section\n";
                        last if $section;
                        next if $1 eq "print" ;
                        next if $prev_section ne " " and $prev_section
ne $1;
                        if ( $prev_section eq $1 ) {
                             $prev_section = " "
                             next;
                        }
                        $section = $1;
                }
                # get parameters
                push ( @{$Param{$1 }}, $2 ) if $section and
/ ([^=]+) – (.*) / ;
        }
#       print" ++++++++++++++++++++++++++++++++++++\n" ;
        return 0 unless $section;
#       print "section $section\n";
        # substitute parameters with values
        map {$Param{URL } -> [ 0 ] =~ s/$Param{ Input } -> [ $_
] / $ARGV [ $_
] / g
        }0 . . $#{$Param{Input } };

        # get page content
        ( $Content{'TIME' }, $content ) = &get_url_content (
${$ Param {URL
} } [ 0 ] ) ;
```

TABLE 4-continued

```
        # filter it
        map {
                if ( / \" ([^\"] + ) \" ([^\"]*) \" / or
/ \ / ( [^ \ / ] + ) \ / ( [ ^ \ / ] *) \ / / )
{
                        my $out = $2; $content =~ s/$1/$out/g;
                }
        -}@{$Param {"Pre-filter" } };

#print STDERR $content;
        #do main regular expression
        unless ( @values = $content =~
/$ {$Param {Regular_expression } } [ 0
] / ) {
                &die_hard ( ${$Param{Regular_expression } } [0 ],
$content
);
                return $section;
        }
        %Content = map { ( $Param[Output } -> [ $_ ], $values [ $_
] )
        }0 . . $#{$Param{Output }} ;

        # filter it
        map {
                if ( / ([^\"]+) \ " ([^\"] +) \ = ( [^\"] +) \ " /
                  or / ([^\ /]+) \ / ([^ \ / ] + ) \ / ( [ ^ \ / ] * ) \ / / ) {
                   my $out = $3;
                  $Content {$1 } =~ s/$2/$out/g;
                }
        _ }@{Param {"Pre-filer" } };

#print STDERR $content;
        # do main regular expression
        unless ( @values = $content =~
/${$Param{Regular_expression } } [ 0
] / ) {
                &die_hard( ${$Param{Regular_expression}} [ 0 ],
$content
);
                return $section;
        }
        %Content= map { ( $Param{Output }->[ $_ ], $values [ $_
] )
        }0 .. $#{$Param{Output }};
```

TABLE 4-continued

```
        #filter it
        map {
                if ( / ( [ ^ \"] +) \ " ( [^\"]+) \ " ( [ ^ \ " ] * ) \ " /
                     or / ( [^\ / ] + ) \ / ( [ ^ \ / + ) \ / ( [ ^ \ / ] * ) \ / / ) {
                      my $out = $3;
                     $Content {$1 } =~ s/$2/$out/g;

                }
        }@{$Param{"Post-filter" } };

        # calculate it
        map {
                if { / ( [ ^ =] + ) = (.*) / ) {
                        my $eval = $2;
                        map {$eval =~ s/$_/$Content {$_ } / g
                        } keys %Content;
                  $Content {$1 } = eval ($eval ) ;
                }

        }@{$Param{Calculate }};

# read section [print]
foreach $i ( 0 .. $#ini ) {
        next unless $ini [ $i ] =~ /^\ [print \ ] /;
         foreach ( $i + 1 . . $#ini ) {
                last if $ini [ $_ ] =~ /^\ [.+\]/;
           $output .= $ini [ $_] . "\n";
        }
        last;
}

# prepare output
map {$output =~ s/$_/$Content{$_} / g
}keys %Content;
print $output;
-return 0;
}
#######################################################
sub get_url_content {
        my ( $url ) = @_;
        print STDERR $url if $debug;
        # $response= './url.pl '$url' ';
        $response= './url.pl '$url' ';
        return( $time - time, $response );
        my $ua = LWP: :UserAgent->new;
        $ua->agent ( 'Mozilla/4.0 [en] (X11; I; freeBSD 2.2.8-
```

TABLE 4-continued

```
STABLE i386)'
) ;
#	$ua->proxy( ['http', 'https'],
'http://proxy.webley:3128/' ) ;
#	$ua->no_proxy( 'webley', 'vail' );
	my $cookie= HTTP::Cookies->new;
	$ua->cookie_jar( $cookie ) ;
	$url = url $url;
		print "$url\n" if $debug2;
	my $time = time;
	my $res= $ua->request( GET $url ) ;
	print "Response: " . ( time - $time "sec\n" if
$debug2;
	return ( $time - time, $res->content ) ;
}]
#####################################################
sub die_hard {
	my ( $re, $content ) = @_;
	my ( $re_end, $pattern ) ;
	while ( $content !~ /$re/ ) {
		if ($re=~ s/(\(["\(\))+\) ["\(\) | *$) / / ) {
			$re_end = $1 . $re_end;
		}
		else {
			$re_end = $re;
			last;
		}
	}
	$content =~ /$re/;
	Print STDERR “The regular expression did not match:\n
$ re\n
Possible misuse:
$re end:\n
Matched:
$&\n
Mismatched:
$'\n
" if $debug;
	if (	$debug ) {
		print STDERR "Content:\n $content\n" unless
$’; -
	}
}

######################################################
```

[0044] Table 5 below contains the content descriptor file source code for obtaining weather information from the web site www.cnn.com that is used by the extraction agent 400 of the preferred embodiment.

TABLE 5

```
[cnn]
Input=_zip
URL=http://cgi.cnn.com/cgi-bin/weather/redirect?zip=_zip
Pre-filter="\n" "
Pre-filter="<[^< >]+>""
Pre-filter=/ \s+/ /
Pre-filter=" [ \ ( \) \ | ] " !"
Output=_location
Output=first_day_name
Output=first_day_weather
Output=first_day_high_F
Output=first_day_high_C
Output=first_day_low_F
Output=first_day_low_C
Output=second_day_name
Output=second_day_weather
Output=second_day_high_F
Output=second_day_high_C
Output=second_day_low_F
Output=second_day_low C
Output=third_day_name
Output=third_day_weather
Output=third_day_high_F
Output=third_day_high_C
Output=third_day_low_F
Output=third_day_low_C
Output=fourth_day_name
Output=fourth_day_weather
Output=fourth_day_high_F
Output=fourth_day_ high_C
Output=fourth_day_low_F
Output=fourth_day_low_C
Output=undef
Output=_current_time
Output=_current_month
```

TABLE 5-continued

```
Output=_current_day
Output=_current_weather
Output=_current_temperature_F
Output=_current_temperature_C
Output=_humidity
Output=_wind
Output=_pressure
Output=_sunrise
Output=_sunset
Regular_expression=Author   (.+) Four Day Forecast
(\S+) (\S+) HIGH
(\S+) F (\S+) C LOW (\S+) F (\S+) C (\S+) (\S+) HIGH (\S+) F
( \S+) C LOW
(\S+) F (\S+) C (\S+) (\S+) HIGH (\S+) F (\S+) C LOW (\S+) F
(\S+) C (\S+)
(\S+) HIGH (\S+) F (\S+J C LOW (\S+) F (\S+) C ( .+) Current
Conditions (. +)
! local ! , (\S+) (\S+) (. t) Temp: (\S+) F, (\S+) C Rel.
Humidity: ( \S+) Wind:
(. +) Pressure: ( .+) Sunrise: ( .+) Sunset: ( .+) Related Links
Post-filter=_ current_ weather" p/" partly "
Post-filter=_current weather"l/"little "
Post-filter=_current_weather"m/"mostly "
Post-filter=_current weather"t-/"thunder" -
Post-filter=_wind"N"North"
Post-filter=_wind"E"East "
Post-filter=_wind"S"South " --
Post-filter=_wind"W"West fl
Post-filter=_wind/mph/miles per hour/
Post-filter=_wind/kph!/kilometers per hour/
Post-filter=_wind" \s+ !", ”
[print]
Current weather in_location is_current_weather.
Temperature is _current_temperature_F Fahrenheit,
_current_temperature_C
Celsium.
Humidity is_humidity.
Wind from the _wind.
```

[0045] Table 6 below contains the content descriptor file source code for obtaining weather information from the web site www.lycos.com that is used by the extraction agent 400 of the preferred embodiment.

TABLE 6

```
[lycos]
Input=zip
Input=_city
URL=http://weather.lycos.com/wcfiveday.asp?city=zip
Pre-filter="\n” ”
Pre-filter="</TD>”td"
Pre-filter="<!.*?->""
Pre-filter="<br>"_br_"
Pre-filter=/alt="/>alt=/
Pre-filter="<[^< >]+>""
Pre-filter=" " "
Pre-filter=/ \s+/ /
Output=_location
Output=_current_weather
Output=_current_temperature_F
Output=_humidity
Output=_winddir
Output=_windspeed
Output=_windmeasure
Output=_pressure
```

TABLE 6-continued

```
Output=first_day_name
Output=second_day_name
Output=third_day_name
Output=fourth_day_name
Output=fifth_day_name
Output=first_day_weather
Output=second_day_weather
Output=third_day_weather
Output=fourth_day_weather
Output=fifth_day_weather
Output=first_day_high_F
Output=first_day_low_F
Output=second_day_high_F
Output=second_day_low_F
Output=third_day_high_F
Output=third_day_low_F
Output=fourth_day_high_F
Output=fourth_day_low_F
Output=fifth_day_high_F
Output=fifth_day_low_F
Output=_windkmh
Regular expression=Guide My Lycos (.+) Click image to
enlarge
alt=([^"]+) " (?:.+) Temp: (\(d+) (?:.+)F _br_ Humidity:
(\S+) (?: .+) Wind: ( .+?)
_br_
Output=_current_temperature_C
Post-filter=_location"_br_" "
Post-filter=_current_weather"p/"partly "
Post-filter=_current_weather"m/"mostly "
Post-filter=_current weather"t-/"thunder "
Post-filter=_winddir"@" at"
Post-filter=_winddir/mph/miles per hour/
Post-filter=_wind/kph!/kilometers per hour/
Calculate=_current _temperature_C=int ( (_current_temperature_F
-32) * 5/9)
Calculate=_windkmh=int(_windspeed*l.6)
[print]
The current weather in _location is _current_weather.
The current temperature is _current_temperature_F_Farenheit
_current_temperature_C Celcius.
Humidity is _humidity.
Winds _winddir.
```

[0046] Once the web browsing server 102 accesses the web site specified in the URL 204 and retrieves the requested information, the information is forwarded to the media server 106. The media server uses the speech synthesis engine 302 to create an audio message that is then transmitted to the user's voice enabled device 112. In the preferred embodiment, each web browsing server 102 is based upon Intel's Dual Pentium III 730 MHz microprocessor system.

[0047] Referring to FIG. 1, the operation of the robust voice browser system will be described. A user establishes a connection between his voice enabled device 112 and a media server 106. This may be done using the Public Switched Telephone Network (PSTN) 116 by calling a telephone number associated with the voice browsing system 118. Once the connection is established, the media server 106 initiates an interactive voice response (WR) application 304. The IVR application plays audio messages to the user presenting a list of options, such as, "stock quotes", "flight status", "yellow pages", "weather", and "news". These options are based upon the available web site categories and may be modified as desired. The user selects the desired option by speaking the name of the option into the voice enabled device 112.

[0048] As an example, if a user wishes to obtain restaurant information, he may speak into his telephone the phrase "yellow pages". The FIR application would then ask the user what he would like to find and the user may respond by stating "restaurants". The user may then be provided with further options related to searching for the desired restaurant. For instance, the user may be provided with the following restaurant options, "Mexican Restaurants", "Italian Restaurants", or "American Restaurants". The user then speaks into the telephone 112 the restaurant type of interest. The IVR application running on the media server 106 may also request additional information limiting the geographic scope of the restaurants to be reported to the user. For instance, the IVR application may ask the user to identify the zip code of the area where the restaurant should be located. The media server 106 uses the speech recognition engine 300 to interpret the speech commands received from the user. Based upon these commands, the media server 106 retrieves the appropriate web site record 200 from the database 100. This record and any additional data, which may include other necessary parameters needed to perform the user's request, are transmitted to a web browsing server 102. A firewall 104 may be provided that separates the web browsing server 102 from the database 100 and media server 106. The firewall provides protection to the media server and database by preventing unauthorized access in the event the firewall for web browsing server 108 fails or is compromised. Any type of firewall protection technique commonly known to one skilled in the art could be used, including packet filter, proxy server, application gateway, or circuit-level gateway techniques.

[0049] The web browsing server 102 then uses the web site record and any additional data and executes the extraction agent 400 and relevant content descriptor file 406 to retrieve the requested information.

[0050] The information received from the responding web site 114 is then processed by the web browsing server 102 according to the content descriptor file 406 retrieval by the extraction agent. This processed response is then transmitted 30 to the media server 106 for conversion into audio messages using either the speech synthesis software 302 or selecting among a database of prerecorded voice responses contained within the database 100.

[0051] As mentioned above, each web site record contains a rank number 202 as shown in FIG. 2. For each category searchable by a user, the database 100 may list several web sites, each with a different rank number 202. As an example, three different web sites may be listed as searchable under the category of "restaurants". Each of those web sites will be assigned a rank number such as 1, 2, or 3. The site with the highest rank (i.e., rank=l) will be the first web site accessed by a web browsing server 102. If the information requested by the user cannot be found at this first web site, then the web browsing server 102 will search the second ranked web site and so forth down the line until the requested information is retrieved or no more web sites left to check.

[0052] The web site ranking method and system of the present invention provides robustness to the voice browser system and enables it to adapt to changes that may occur as web sites evolve. For instance, the information required by a web site 114 to perform a search or the format of the reported response data may change. Without the ability to adequately monitor and detect these changes, a search requested by a user may provide an incomplete response, no response, or an error. Such useless responses may result from incomplete data being provided to the web site 114 or the web browsing server 102 being unable to recognize the response data messages received from the searched web site 114.

[0053] The robustness and reliability of the voice browsing system of the present invention is further improved by the addition of a polling mechanism. This polling mechanism continually polls or "pings" each of the sites listed in the database 100. During this polling function, a web browsing server 102 sends brief data requests or "polling digital data" to each web site listed in database 100. The web browsing server 102 monitors the response received from each web site and determines whether it is a complete response and whether the response is in the expected format specified by the content descriptor file 406 used by the extraction agent 400. The polled web sites that provide complete responses in the format expected by the extraction agent 400 have their ranking established based on their "response lime". That is, web sites with faster response times will be will be assigned higher rankings than those with slower response times. If the web browsing server 102 receives no response from the polled web site or if the response received is not in the expected format, then the rank of that web site is lowered. Additionally, the web browsing server contains a warning mechanism that generates a warning message or alarm for the system administrator indicating that the specified web site has been modified or is not responsive and requires further review.

[0054] Since the web browsing servers 102 access web sites based upon their ranking number, only those web sites that produce useful and error-free responses will be used by the voice browser system to gather information requested by the user. Further, since the ranking numbers are also based upon the speed of a web site in providing responses, only the most time efficient sites are accessed. This system assures that users will get complete, timely, and relevant responses to their requests. Without this feature, users may be provided with information that is not relevant to their request or may not get any information at all. The constant polling and reranking of the web sites used within each category allows the voice browser of the present invention to operate efficiently. Finally, it allows the voice browser system of the present invention to dynamically adapt to changes in the rapidly evolving web sites that exist on the Internet.

[0055] It should be noted that the web sites accessible by the voice browser of the preferred embodiment may use any type of mark-up language, including Extensible Markup Language (XML), Wireless Markup Language (WML), Handheld Device Markup Language (HDML), Hyper Text Markup Language (HTML), or any variation of these languages.

[0056] A second embodiment of the present invention is depicted in FIG. 5. This embodiment provides a system and method for controlling a variety of devices 500 connected to a network 502 by using conversational speech commands spoken into a voice enabled device 504 (i.e., wireline or wireless telephones, Internet Protocol (EP) phones, or other special wireless units). The networked devices may include various household devices. For instance, voice commands may be used to control household security systems, VCRs, TVs, outdoor or indoor lighting, sprinklers, or heating and air conditioning systems.

[0057] Each of these devices 500 is connected to a network 502. These devices 500 may contain embedded microprocessors or may be connected to other computer equipment that allow the device 500 to communicate with network 502. In the preferred embodiment, the devices 500 appear as "web sites" connected to the network 502. This allows a network interface system, such as a device browsing server 506, a database 508, and a user interface system, such as a media server 510, to operate similar to the web browsing server 102, database 100 and media server 106 described in the first preferred embodiment above. A network 502 interfaces with one or more network interface systems, which are shown as device browsing servers 506 in FIG. 5. The device browsing servers perform many of the same functions and operate in much the same way as the web browsing servers 102 discuss above in the first preferred embodiment. The device browsing servers 506 are also connected to a database 508.

[0058] Database 508 lists all devices that are connected to the network 502. For each device 500, the database 508 contains a record similar to that shown in FIG. 2. Each record will contain at least a device identifier, which may be in the form of a URL, and a command to "content extraction agent" contained in the device browsing server 506. Database 508 may also include any other data or software necessary to test and administer the device browsing system.

[0059] The content extraction agent operates similarly to that described in the first embodiment. A device descriptor file contains a listing of the options and functions available for each of the devices 500 connected on the network 502. Furthermore, the device descriptor file contains the information necessary to properly communicate with the networked devices 500. Such information would include, for example, communication protocols, message formatting requirements, and required operating parameters.

[0060] The device browsing server 506 receives messages from the various networked devices 500, appropriately formats those messages and transmits them to one or more media servers 510 which are part of the device browsing system. The user's voice enabled devices 504 can access the device browsing system by calling into a media server 510 via the Public Switched Telephone Network (PSTN) 512. In the preferred embodiment, the device browsing server is based upon Intel's Dual Pentium III 730 MHz microprocessor system.

[0061] The media servers 510 act as user interface systems and perform the functions of natural speech recognition, speech synthesis, data processing, and call handling. The media server 510 operates similarly to the media server 106 depicted in FIG. 3. When data is received from the device browser server 506, the media server 510 will convert the data into audio messages via a speech synthesis engine that are then transmitted to the voice enabled device of the user 504. Speech commands received from the voice enabled device of the user 504 are converted into data messages via a speech recognition engine running on the media server 510. A preferred speech recognition engine is developed by Nuance Communications of 1380 Willow Road, Menlo Park, Calif. 94025 (www.nuance.com). A preferred speech synthesis engine is developed by Lernout and Hauspie Speech Products, 52 Third Avenue, Burlington, Mass. 01803 (www.lhsl.com). The media servers 510 of the preferred embodiment are based on Intel's Dual Pentium III 730 MHz microprocessor system. A specific example for using the system and method of this embodiment of the invention will now be given.

[0062] First, a user may call into a media server 510 by dialing a telephone number associated with an established device browsing system. Once the user is connected, the IVR application of the media server 510 will provide the user with a list of available systems that may be monitored or controlled based upon information contained in database 508.

[0063] For example, the user may be provided with the option to select "Home Systems" or "Office Systems". The user may then speak the command "access home systems". The media server 510 would then access the database 508 and provide the user with a listing of the home subsystems or devices 500 available on the network 502 for the user to monitor and control. For instance, the user may be given a listing of subsystems such as "Outdoor Lighting System", "Indoor Lighting System", "Security System", or "Heating and Air Conditioning System". The user may then select the indoor lighting subsystem by speaking the command "Indoor Lighting System". The IVR application would then provide the user with a set of options related to the indoor lighting system. For instance the media server 510 may then provide a listing such as "Dining Room", "Living Room", "Kitchen", or "Bedroom". After selecting the desired room, the IVR application would provide the user with the options to hear the "status" of the lighting in that room or to "turn on", "turn off", or "dim" the lighting in the desired room. These commands are provided by the user by speaking the desired command into the users voice enabled device 504. The media server 510 receives this command and translates it into a data message. This data message is then forwarded to the device browsing server 506 which routes the message to the appropriate device 500.

[0064] The device browsing system 514 of this embodiment of the present invention also provides the same robustness and reliability features described in the first embodiment. The device browsing system 514 has the ability to detect whether new devices have been added to the system or whether current devices are out-of-service. This robustness is achieved by periodically polling or "pinging" all devices 500 listed in database 508. The device browsing server 506 periodically polls each device 500 and monitors the response. If the device browsing server 506 receives a recognized and expected response from the polled device, then the device is categorized as being recognized and in-service. However, if the device browsing server 506 does not receive a response from the polled device 500 or receives an unexpected response, then the device 500 is marked as being either new or out-of-service. A warning message or a report may then be generated for the user indicating that a new device has been detected or that an existing device is experiencing trouble.

[0065] Therefore, this embodiment allows users to remotely monitor and control any devices that are connected to a network, such as devices within a home or office. Furthermore, no special telecommunications equipment is required for users to remotely access the device browser system. Users may use any type of voice enabled device (i.e., wireline or wireless telephones, IP phones, or other wireless units) available to them. Furthermore, a user may perform these functions from anywhere without having to subscribe to additional services. Therefore, no additional expenses are incurred by the user.

[0066] The descriptions of the preferred embodiments described above are set forth for illustrative purposes and are not intended to limit the present invention in any manner. Equivalent approaches are intended to be included within the scope of the present invention. While the present invention has been described with reference to the particular embodiments illustrated, those skilled in the art will recognize that many changes and variations may be made thereto without departing from the spirit and scope of the present invention. These embodiments and obvious variations thereof are contemplated as falling within the scope and spirit of the claimed invention.

CONTINUATION CLAIMS

What is claimed:

1. A system comprising:

(a) at least one data processor, the at least one data processor operatively coupled to a plurality of communication data networks;

(b) at least one speaker-independent speech-recognition engine operatively coupled to the data processor;

(c) memory accessible to the at least one data processor and storing at least:

(i) an instruction set for querying of information to be retrieved from a plurality of sources coupled to the plurality of communication data networks, the instruction set comprising:

an indication of the plurality of sources, each identified by a source identifier, and each identifying certain information to be retrieved from the source identifier, and

(ii) at least one recognition grammar executable code corresponding to each instruction set and corresponding to data characterizing audio containing a naturally-spoken-speech command including an information request,

(d) wherein the at least one speaker-independent-speech-recognition engine is adapted to:

(1) receive the data characterizing audio containing the naturally-spoken-speech command from a voice-enabled device via a first of the plurality of communication data networks;

(2) to recognize phenomes in the data characterizing audio containing

naturally-spoken-speech commands to understand spoken words; and

(3) to generate recognition results data,

(e) wherein the at least one data processor is adapted to:

(1) select the corresponding at least one recognition grammar executable code upon receiving the data characterizing the audio containing the naturally-spoken-speech command and to convert the data characterizing the audio containing the naturally-spoken-speech command into a data message for transmission to a network interface adapted to access a second of the plurality of communication data networks; and

(2) retrieve the instruction set corresponding to the recognition grammar executable codes provided by the at least one speaker-independent-speech-recognition engine and to access the information source queried by the instruction set to obtain at least a part of the information to be retrieved, and

(f) at least one speech-synthesis device operatively coupled to the at least one data processor, the at least one speech-synthesis device configured to produce an audio message relating to any resulting information retrieved from the plurality of information sources including a text-to-speech conversion of at least certain data in said any resulting information retrieved from the plurality of information sources, and to convey the audio message via the voice-enabled device.

2. The system of claim 1, wherein the plurality of communication data networks includes the Internet.

3. The system of claim 1, wherein the plurality of communication data networks include

a local-area network.

4. The system of claim 1, wherein the voice-enabled device is a home device.

5. The method of claim 1, wherein the voice-enabled device is at least one of a group of an IP telephone, a cellular phone, a personal computer, a media player appliance, and a television or other video display device.

6. The system of claim 1, wherein the speaker-independent-speech-recognition engine is adapted to analyze the phonemes to recognize conversational naturally-spoken-speech commands.

7. The system of claim 1, wherein the speaker-independent-speech-recognition engine is adapted to recognize the naturally-spoken-speech commands.

8. The system of claim 1, wherein the instruction set executable code further comprises: a content descriptor associated with each information-source identifier, the content descriptor pre-defining a portion of the information source containing the information to be retrieved.

9. The system of claim 1, further comprising:

a database operatively connected to the data processor, the database adapted to store the information gathered from the information sources in response to the information requests.

10. The system of claim 8, wherein each recognition grammar executable code and each instruction set for querying of information to be retrieved are stored in the database.

11. A method comprising:

(a) providing at least one data processor, the data processor operatively coupled to a plurality of communication data networks;

(b) providing at least one speaker-independent-speech-recognition engine operatively coupled to the at least one data processor

(c) providing memory accessible to the data processor storing at least:

(i) an instruction set for querying of the information to be retrieved from a plurality of sources coupled to the plurality of communication data networks, the instruction set comprising: an indication of the plurality of sources, each identified by an information-source identifier, and each identifying certain information to be retrieved from the information-source identifier, and

(ii) at least one recognition grammar executable code corresponding to each instruction set and corresponding to data characterizing audio containing a naturally-spoken-speech command including an information request,

(d) the at least one speaker-independent- speech-recognition engine:

(i) receiving the data characterizing audio containing the naturally-spoken-speech command from the voice-enabled device via a first of the communication data networks,

(ii) recognizing phenomes in the data characterizing audio containing the naturally-spoken-speech commands to understand spoken words, and

(iii) generating recognition-results data,

(e) the least one data processor programmed to:

(i) select the corresponding at least one recognition grammar executable code upon receiving the data characterizing audio containing the naturally-spoken-speech command and convert the data characterizing audio containing naturally-spoken-speech command into a data message for transmission to a network interface adapted to access a second of the one communication networks; and

(ii) retrieve the instruction set corresponding to the recognition grammar executable provided by the at least one speaker-independent-speech-recognition device and access the information source identified by the instruction set to obtain at least a part of the information to be retrieved; and

(f) providing at least one speech-synthesis device operatively connected to the at least one data processor, and by the at least one speech-synthesis device adapted to:

(i) produce an audio message relating to any resulting information retrieved from the plurality of information sources including text-to-speech conversion of said any resulting information retrieved from the plurality of information sources, and

(ii) transmit the audio message via the voice-enabled device.

12. The method of claim 11, wherein the plurality of communication data networks includes the Internet.

13. The method of claim 11, wherein the plurality of communication data networks include a local-area network.

14. The method of claim 11, wherein the voice-enabled device is a telephone.

15. The method of claim 11, wherein the speaker-independent-speech-recognition engine is adapted to analyze the phonemes to recognize conversational naturally-spoken-speech commands.

16. The method of claim 11, wherein the speaker-independent-speech-recognition engine is adapted to recognize the naturally-spoken-speech commands.

17. The method of claim 11, wherein the instruction set executable code further comprises: a content descriptor associated with each information-source identifier, the content descriptor pre-defining a portion of the information source containing the information to be retrieved.

18. The method of claim 11, further comprising:

providing a database and operatively connecting the database to the data processor and storing the information gathered from the information sources in response to the information requests in the database.

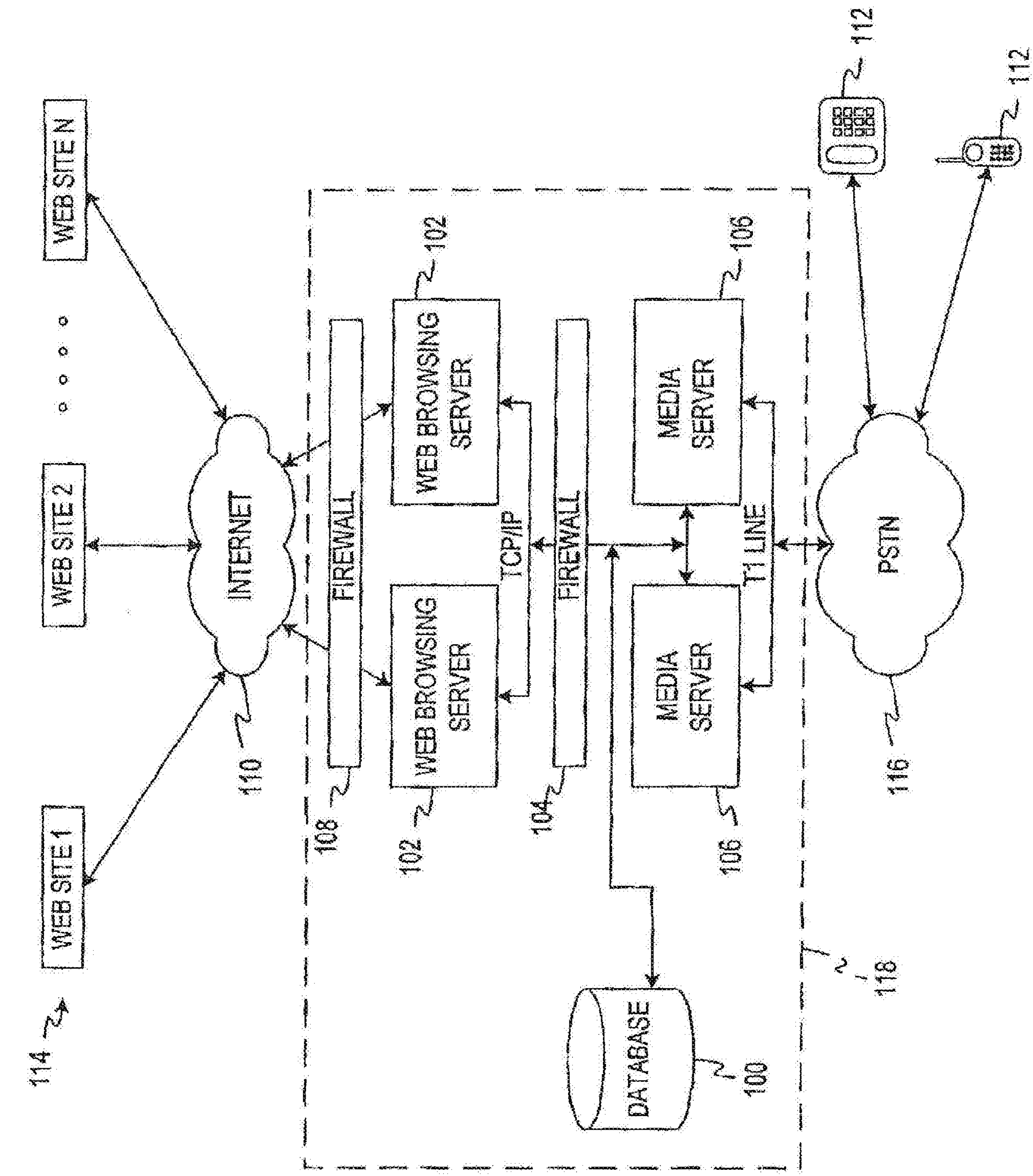
19. The method of claim 18, wherein each recognition grammar executable code and each instruction set are stored in the database.

20. The method of claim 18, wherein the voice-enabled device is at least one of a group of an IP telephone, a cellular phone, a personal computer, a media player appliance, and a television or other video display device.

ABSTRACT OF THE INVENTION

The present invention relates to a system for acquiring information from sources on a network, such as the Internet. A voice browsing system maintains a database containing a list of information sources, such as web sites, connected to a network. Each of the information sources is assigned a rank number which is listed in the database along with the record for the information source. In response to a speech command received from a user, a network interface system accesses the information source with the highest rank number in order to retrieve information requested by the user.

FIG. 1


WEB SITE 1
WEB SITE 2
WEB SITE N
114
INTERNET
110
FIREWALL
108
WEB BROWSING SERVER
WEB BROWSING SERVER
102
102
TCP/IP
FIREWALL
104
MEDIA SERVER
MEDIA SERVER
106
106
T1 LINE
PSTN
116
112
112
DATABASE
100
118

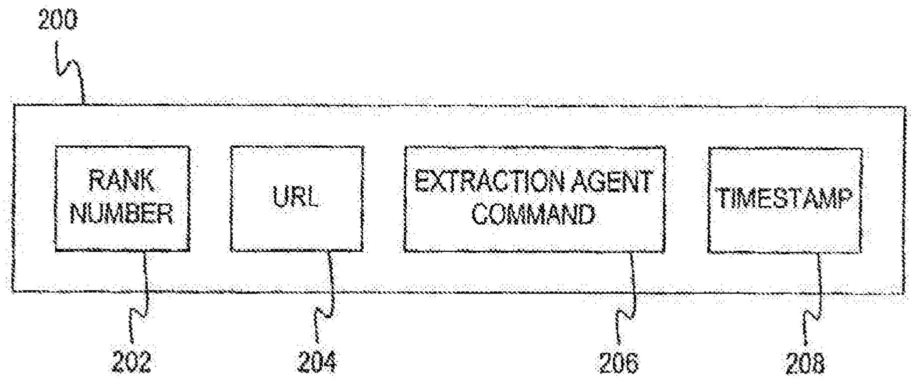

200
RANK NUMBER
URL
EXTRACTION AGENT COMMAND
TIMESTAMP
202
204
206
208

FIG. 2

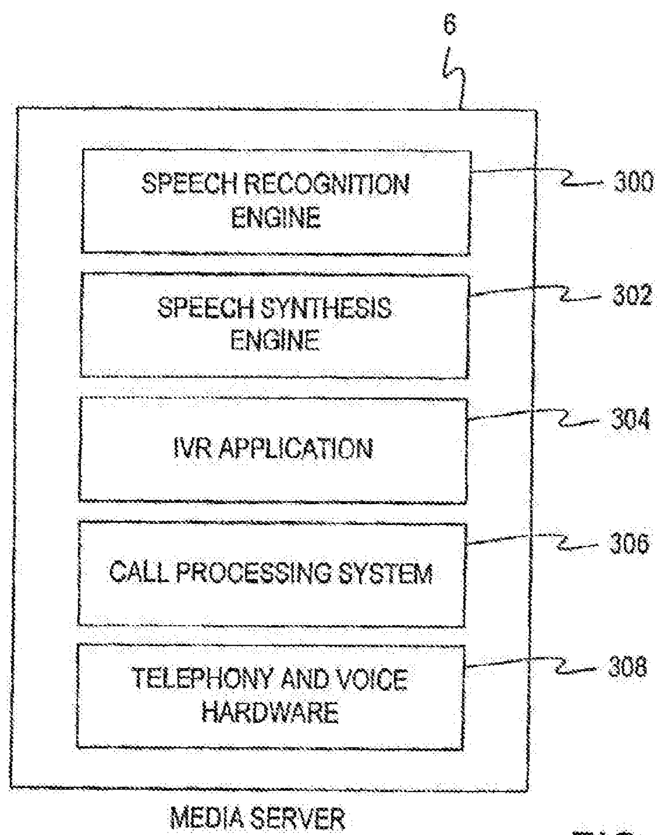

6
SPEECH RECOGNITION ENGINE
300
SPEECH SYNTHESIS ENGINE
302
IVR APPLICATION
304
CALL PROCESSING SYSTEM
306
TELEPHONY AND VOICE HARDWARE
308
MEDIA SERVER

FIG. 3

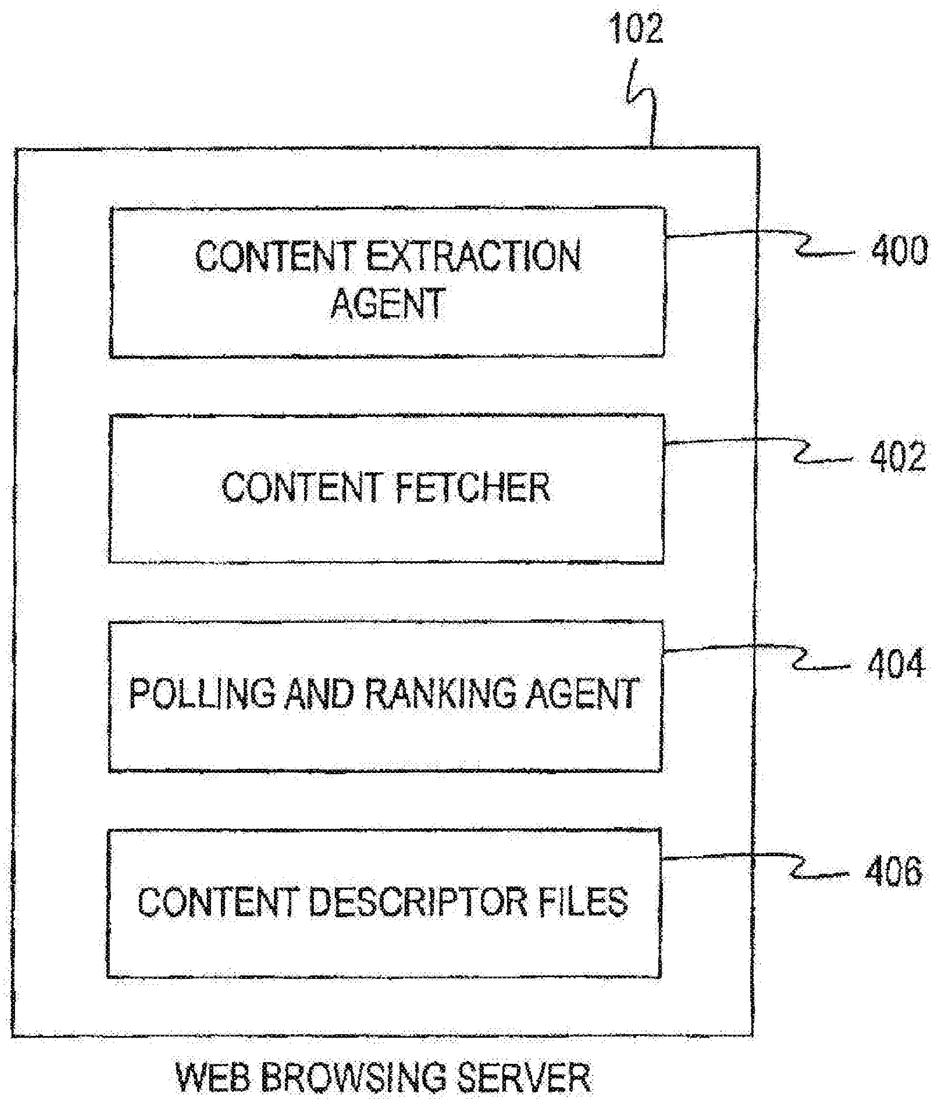

102
CONTENT EXTRACTION AGENT
400
CONTENT FETCHER
402
POLLING AND RANKING AGENT
404
CONTENT DESCRIPTOR FILES
406
WEB BROWSING SERVER

FIG. 4

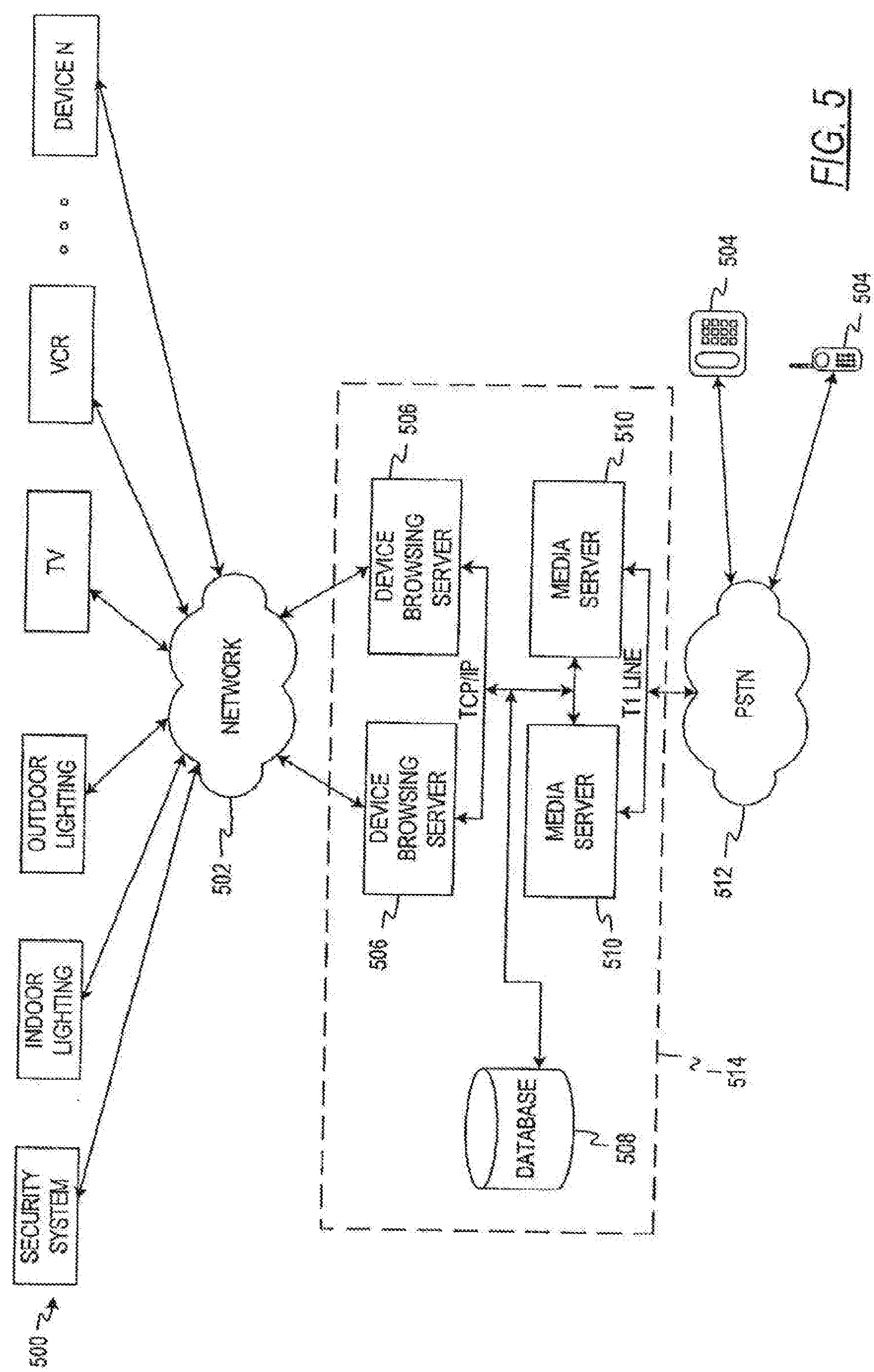

500
SECURITY SYSTEM
INDOOR LIGHTING
OUTDOOR LIGHTING
TV
VCR
DEVICE N
NETWORK
502
DEVICE BROWSING SERVER
506
DEVICE BROWSING SERVER
506
TCP/IP
MEDIA SERVER
510
MEDIA SERVER
510
T1 LINE
DATABASE
508
514
PSTN
512
504
504

FIG. 5

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	First Named Inventor	Alexander Kurganov
	Art Unit	
	Examiner Name	
	Attorney Docket Number	10511-07594 US

U.S.PATENTS

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	21	4602129		1986-07-22	Matthews et al.	
	22	4635253		1987-01-06	Urui et al.	
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	24	4696028		1987-09-22	Morganstein et al.	
	25	4713837		1987-12-15	Gordon	
	26	4747127		1988-05-24	Hansen et al.	
	27	4748656		1988-05-31	Gibbs et al.	
	28	4755932		1988-07-05	Diedrich	
	29	4757525		1988-07-12	Matthews et al.	
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	Attorney Docket Number	10511-07594 US

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	35	4782517		1988-11-01	Bernardis et al.	
	36	4792968		1988-12-20	Katz	
	37	4799144		1989-01-17	Parruck et al.	
	38	4809321		1989-02-28	Morganstein et al.	
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	44	4852170		1989-07-25	Bordeaux	
	45	4866758		1989-09-12	Heinzelmann	
	46	4873719		1989-10-10	Reese	
	47	4879743		1989-11-07	Burke et al.	
	48	4893333		1990-01-09	Baran et al.	
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	52	4905273		1990-02-27	Gordon et al.	

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	54	4918722		1990-04-17	Duehren et al.	
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	57	4922526		1990-05-01	Morganstein et al.	
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	59	4930150		1990-05-29	Katz	
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	Art Unit	
	Examiner Name	
	Attorney Docket Number	10511-07594 US

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	69	4967288		1990-10-30	Mizutori et al.	
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	92	5133004		1992-07-21	Heileman Jr. et al.	
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	112	5325421		1994-06-28	Hou et al.	
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CERTIFICATION STATEMENT

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).

OR

☐ That no item of information contained in the information disclosure statement was cited in a communication from a foreign patent office in a counterpart foreign application, and, to the knowledge of the person signing the certification after making reasonable inquiry, no item of information contained in the information disclosure statement was known to any individual designated in 37 CFR 1.56(c) more than three months prior to the filing of the information disclosure statement. See 37 CFR 1.97(e)(2).

See attached certification statement.

The fee set forth in 37 CFR 1.17 (p) has been submitted herewith.

☒ A certification statement is not submitted herewith.

SIGNATURE

A signature of the applicant or representative is required in accordance with CFR 1.33, 10.18. Please see CFR 1.4(d) for the form of the signature.

Signature	/Reena Kuyper/	Date (YYYY-MM-DD)	2018-10-09
Name/Print	Reena Kuyper	Registration Number	33,830

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CERTIFICATION STATEMENT

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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☐ That no item of information contained in the information disclosure statement was cited in a communication from a foreign patent office in a counterpart foreign application, and, to the knowledge of the person signing the certification after making reasonable inquiry, no item of information contained in the information disclosure statement was known to any individual designated in 37 CFR 1.56(c) more than three months prior to the filing of the information disclosure statement. See 37 CFR 1.97(e)(2).

See attached certification statement.

The fee set forth in 37 CFR 1.17 (p) has been submitted herewith.

☒ A certification statement is not submitted herewith.

SIGNATURE

A signature of the applicant or representative is required in accordance with CFR 1.33, 10.18. Please see CFR 1.4(d) for the form of the signature.

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(54) Facsimile Telecommunications System and Method

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Abstract of The Disclosure

A system and method for facilitating facsimile transmissions has one or more store and forward facilities, each associated with a plurality of subscriber facsimile machines, typically coupled over the switched telephone network. The store and forward facilities include a computer for controlling operations and mass data storage equipment. A subscriber to the system delivers an outgoing facsimile message to the store and forward facility with which it is associated, which records the fax message together with data as to originating facsimile machine and destination facsimile machine. The store and forward facility then delivers the facsimile message to the intended receiver facsimile machine, either directly or through another store and forward facility. If unsuccessful on an initial attempt, the store and forward facility periodically retrys to send the facsimile message. The system also provides spooling of all facsimile messages for an intended receiver machine, which are all transmitted upon making connection with the receiver machine. Subscriber mailboxes are provided as part of the mass storage, which can be accessed by a subscriber to have his messages delivered to any facsimile machine he designates. Secure facsimile transmission is achieved through use of subscriber PIN numbers. Broadcasting, redirecting messages and cost accounting can also be achieved by the system and method.

Facsimile Telecommunications System and Method

Field of the Invention

The field of this invention is telecommunications systems used in connection with facsimile transmissions. More specifically, this invention relates to a system and method for enhancing ease of facsimile transmissions and providing features relative to facsimile transmissions not heretofore available.

Background of the Invention

The electronic transmission of documents by way of facsimile (fax) systems employing public and private switched telephone networks has become both commonplace and, often, an essential component in many business activities. In such a setting, it is very common for the fax terminals (fax machines) to be kept quite busy during a major fraction of the business day. Moreover, where sender and recipient are in different time zones, the "business day" can approach 24 hours, particularly in international activities. It is common for fax users to "broadcast" documents to a number of different recipients, that is, send the same message to several different fax machines. It is also true that the contents of some faxed documents are of such a sensitive nature that the originator or addressee would like to have a measure of control over who might see those documents as they move from the receiving machine to the hands of the actual addressee.

These circumstances present a number of practical problems for a fax user. In order to make a successful fax transmission it is necessary that the receiving machine be available at the time that the transmitting machine attempts to contact it. If the receiving machine is already in use handling another message, the transmitter will receive a "busy" signal. The originator's only recourse is to continue initiating telephone calls until contact can be established. This is a "hit or miss" process at best and can be very wasteful of the originating operator's time.

Some, rather expensive, fax machines have digital memories which will allow them to memorize the document to be transmitted and to be programmed to make multiple redials in an effort to establish contact in an automatic way. However, this is limited to only one or two documents and, more importantly, it ties up the transmitting machine until the effort is successful or abandoned. This is hardly an acceptable solution if that machine has other documents to send or receive.

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There are other conditions which can result in a failure to transmit even though a telephone connection has been established. Perhaps the most common of these is the absence of paper in the receiving machine. In such situations, repeated attempts to "redial" will lead to repeated toll charges with each attempt, with no actual success until the receiving machine is serviced (which may be some time if the machine is operating unattended because it is nighttime half-way around the world).

Busy machines which are destined to receive messages are affected by the converse problem. Since they and the prospective transmitting machines must engage in (perhaps, automated) "telephone tag", they are used very inefficiently. When a transmitting machine gets a busy signal, even if it automatically redials, it can only guess at when the receiving machine will be available. Thus, the receiving machine will likely remain idle for some fraction of the time until such an attempt is made.

The practice of broadcasting documents to a number of addressees obviously compounds these problems and adds still others of its own. Even if one does not encounter busy signals or impaired machines, convenient broadcasting demands an expensive memory-type fax machine on the transmitting end. Such machines read in the document once and then proceed to automatically dial the various recipient machines. This process ties up the sending machine and its telephone line and makes them unavailable for incoming calls. This, of course, exacerbates the busy signal problem for those units trying to contact the sending machine.

The security of sensitive documents is still another problem. Once contact is established between two fax machines, the transmission of the document proceeds automatically, irrespective of who may be standing by the receiving machine at the time. In a busy office, the contents of these documents are accessible to the fax operator and anyone else who happens to be in the vicinity.

It is also common for individuals to wish to deliver fax documents to a recipient who is not currently available through a known machine (eg. a person on a business trip). This is a very inconvenient situation in that it requires that the paper documents be held until the traveler phones in from a remote machine. It further requires that there be someone available at that time who has knowledge of and access to the documents intended for the recipient.

Still another concern is adequate accounting control over the billing of calls. Typically, many businesses wish to be able to track the costs of both fax machine use and the associated telephone charges. While telephone charges can be ascribed from telephone company records, in the present environment these must be related to records of the number of pages transmitted per call and so forth, separately maintained by the fax machine or its attendants.

Summary of the Invention

The objects of this invention are to address these many shortcomings of present fax systems and to provide an integrated system for their solution. Furthermore, the intention is to achieve this in a way which is fundamentally compatible with existing fax terminal machines. The basic approach is to provide special computer-based fax Store And Forward Facilities (SAFF's) as an integral part of a switched telephone network system. All fax transmissions entered into the network are routed to such a facility, typically geographically near the originating machine, where they are temporarily stored or "spooled" by the computer in a mass storage buffer, such as a magnetic disk.

The fax message from the originating machine is intended for a destination machine, which may or may not be in a position to immediately answer the call. If the destination machine is within the service region of that SAFF, the system then proceeds to attempt to call the destination fax machine. If the destination machine is within the service area of a different SAFF, the system forwards the fax document data to that facility by long-distance lines, in which case this second facility attempts to call the destination machine. In either case, if contact is established and the message is delivered immediately, the system directs a printed report back to the originating fax machine confirming delivery to the destination machine, and other pertinent data.

If, on the other hand, the delivery cannot be completed immediately due to a busy signal, a machine fault (eg, receiving machine out of paper) or any other reason, the spooled document is saved and the system makes periodic attempts to contact the destination machine and complete the transmission.

In the meantime, the system sends a printed report back to the originating machine acknowledging that the message has been entered into the system, indicating the reason the delivery is being delayed, stating the protocol the system will take to deliver the message, and providing a reference number or "Message Code" which identifies the message and may be used at a later time to trace the status of the document.

Placing the delivering spooling system geographically near the destination machine has the advantage of more economical use of any long-distance lines that may be involved. These lines are used only to move the message from the originator to the spooling system in the vicinity of the destination, which is virtually certain to be successful on the first try. Subsequent attempts to contact the destination machine can be handled more or less locally and need not tie up the bulk of the long-distance facilities.

If the delayed delivery is ultimately successful, the system will send a printed delivery report to the originating machine. On the other hand, if the delivery attempt protocol has gone through its whole cycle without success, a report will be sent to the originator indicating that the delivery procedure has failed and requesting instructions as to how to proceed (eg. try again, redirect the message to an alternate number, or delete the message).

An important feature of the system is that it recognizes all of the documents that are spooled in the system at a given time for a given destination machine. These are identified and linked together to form a message queue for that machine. In this way, once contact is established, all of the waiting messages can be "dumped" to that machine in a continuous batch. Furthermore, if new messages arrive while that dump is occurring, they are simply appended to the end of the active queue and are transmitted when their turn comes. This has the advantage of greatly enhancing the utilization efficiency of a busy destination machine.

Since all outgoing fax documents are temporarily stored at the facility near the originating machine, it is also practical to provide for automatic broadcasting of documents to multiple destinations. Lists of "broadcast groups" of phone numbers can be programmed into the facility by users, or a list of destination phone numbers entered "by hand" at the time of a call. The SAFF can then broadcast the message to every machine of the selected list. This is a great advantage to broadcast users in that they need only tie up their machines for one outgoing transmission, the one to the SAFF. The SAFF copies the message to all of the destination machines as outlined above. In the meantime, the originating machine is available for receiving or transmitting other documents.

Similarly, since the documents are stored near the originator, the system can permit messages which have already been sent to be copied to other destinations after the fact, without the necessity of resending the message to the SAFF. Likewise, since the messages are also spooled in a facility near the destination, the system also provides the recipient with the option of forwarding or redirecting documents to still other destinations, as if the recipient were the originator. The system can also accept and store messages destined for a fictitious destination or "Mail Box". Thus, individuals who are traveling can, at their convenience, dial into the system and pick up any waiting documents.

Closely akin to these features is the ability to have the originator of a transmission include the requirement that the recipient provide a security code, such as a PIN number, in order to release the document from the spool to the destination machine. In this case, the SAFF sends a written report to the destination machine advising that a secure message is waiting for a particular recipient and the fax identification of the originating machine. The recipient must then call in to the SAFF and key in the security code to initiate the delivery of the document. Since the document is spooled, the delivery easily may be delayed until the recipient is available to supply the code.

Finally, since the documents and their delivery are both under the control of the telephone system, as a special service the telephone call accounting system can provide both time and charges for the telephone services rendered and fax information, such as pages transmitted, sorted according to the originator's clients. This can greatly facilitate the fax user who wishes to do cost accounting or to bill clients for costs incurred.

Brief Description of the Drawings

Other objects and advantages of the present invention will be apparent from the following Detailed Description of the preferred embodiments thereof and from the attached Drawings of which:

Fig. 1 illustrates the inter-relationships of the principal elements of a connection between two SAFFs.

Fig. 2 shows a more detailed view of the various systems within a single SAFF, such as those shown in Figure 1.

Fig. 3 illustrates the major components of the Originate Function in the SAFFs shown in Figures 1 and 2.

Fig. 4 illustrates the major components of the Answer Function in the SAFFs shown in Figures 1 and 2.

Figs. 5a and b show a flow chart describing the general processing steps required to handle a fax or voice message incoming to the Originate Function of a SAFF, as described particularly in Figures 2 and 3.

Figs. 6a and b show a flow chart describing the general processing steps required to handle the delivery of a fax message incoming to the Answer Function of a SAFF, as described particularly in Figures 2 and 4.

Fig. 7 shows a flow chart of the general processing steps required to handle a service request in the General Service unit of a SAFF, as described particularly in Figure 2.

Detailed Description

Introduction

The preferred embodiment of this invention is a multi-function, interactive facsimile transmission system which is integrated into a switched telephone distribution network, where "network" is taken broadly to mean the entire system required to complete a communication from an originator to an answerer. This embodiment provides a comprehensive computerized fax message management system based on automated fax Store And Forward Facilities (SAFF) embedded in the network. This system requires no modifications to existing facsimile machines, but rather, relies on the network to provide the enhanced services.

The system contains several components which actually transmit the fax messages and related information, provide written fax reports to users about the status of messages within the system, allow user intervention in the sequence of automatic actions of the system, provide an accounting of services rendered for both the customer and the telephone company, and control and supervise all of these activities.

In the preferred embodiment, it is presumed that the SAFF's are placed at the interface between the local telephone delivery system and the long-distance delivery system, as indicated in Figures 1 and 2. In this setting, the SAFF system can be controlled and its services offered by either one. However, it is obvious that useful systems can be constructed where the SAFF exists as close to the user as a component of his or her own in-house telephone system (such as a PBX or Centrex) or as remotely as a single, independent, stand-alone SAFF serving a wide geographical area. It is also obvious that commercially viable systems can be constructed which provide subsets of the features of the preferred embodiment. The choice of site/control setting and service features might be driven by any number of economic, market, or legal considerations, which would militate toward offering the system at an alternate location in the network, or in a "stripped down" form.

To more clearly understand the present invention, it is useful to consider the manner in which a fax transmission occurs in the traditional setting. Here the communication between two machines is initiated when the destination machine answers a telephone call directly from the originating machine. Typically, there is an exchange of digital data identifying the sending and receiving machines to each other and establishing the fax mode or format to be used. If this exchange is satisfactory, then the actual image transmission takes place. Otherwise, the call is terminated, usually with some form of written diagnostic to the respective users.

Message Interception

In the present invention, all fax transmissions initiated by a subscriber to the fax management system are first intercepted by an "originator" SAFF; that is, the SAFF which directly services the originating fax machine. Figure 1 shows two exemplary SAFFs 8 and 18, with interconnections between the SAFFs and with subscriber fax machines being diagrammatically indicated. Thus in Figure 1, the SAFF 8 includes an originate function 9 coupled over telephone lines 4 to originating fax machines 1. Likewise, the SAFF 18 includes an originate function 22 coupled over telephone lines 26 to originating fax machines 30. Each of the SAFFs 8 and 18 also includes respective answer function blocks 12 and 19 respectively connected over telephone lines 6, 24 to fax machines 3, 28. Each of the SAFFs 8, 18 also includes service interfaces 10, 21 coupled via telephone lines 5, 25 to telephones 2, 29. The function and purpose of the service interfaces is more fully explained hereafter, and they are under control of status and control blocks 11 and 21.

Access to the system of Figure 1 can be obtained much the same as access to a specific long-distance company's network. That is, subscribers such as 1 in Figure 1 can dial a unique access code at the time a call is initiated, or a telephone line dedicated to a fax terminal may be permanently routed to the SAFF system, in this case the SAFF 8 of Figure 1. Either way, one accesses SAFF Directed Lines 4 and the SAFF 8 itself in the process of dialing the destination fax machine.

The SAFF 8 then answers the phone in place of the destination machine, such as one of 28 shown in Figure 1 as serviced by SAFF 18. For the moment, this SAFF 8 near the originator becomes the proxy for the destination machine 28. While noting the actual destination telephone number, the SAFF 8 engages the originating machine in the same digital dialogue that would have occurred if a direct connection to the destination machine had actually been made. Thus, it echoes back the destination telephone number, to identify the intended destination machine, and agrees to accept the fax format requested by the originating machine.

This causes the originating machine 1 to respond by transmitting the fax document image data. The originating machine's identification, the destination machine's telephone number, the fax format, and the document image data are all stored on a mass storage device 67 (in Figure 3), such as a computer magnetic disk unit. Furthermore, a unique alphameric Message Code is assigned to the block of data to identify it while it is resident in the SAFF system. This Message Code is related to the file name for the stored data.

Delivery

At this point the SAFF 8 initiates two actions. The first is to generate an "Acceptance Record" of the transaction to this point. This record, in one form or another, will be returned to the originator as will be described below. The second step is to begin to deliver the fax message to the destination machine 28.

The details of the delivery process depend to some degree on the geographic location of the destination within the network. A single SAFF can, in principle, service a broad geographical area. However, in the preferred embodiment, communications beyond a certain limiting distance involve at least two SAFFs, one 8 near the originator 1 and the other, a "destination SAFF", 18 near the recipient 28 of the document. The choice of one, two, or more SAFFs is determined by network economics, or other considerations, and is not essential to the invention.

For the sake of this discussion, we will define a "local" message to imply that the originating and the destination machines are serviced by the same SAFF. (Although, this does not preclude the possibility that the two machines are some considerable distance apart and connected by a toll call.) On the other hand, we will define a "long-distance" message to mean that the originating and destination fax machines are serviced by different SAFFs and, thus, one SAFF must exchange data with the other, perhaps through intermediaries. Similarly, the term "near" used in connection with a SAFF refers to being within the service area of that SAFF.

Each SAFF 8, 18 has two clearly defined roles: the "Originate Function" 9, 22 for handling data with an originating machine, and the "Answer Function" 19, 12 for handling data with a destination machine. The details of these two subsystems are illustrated in Figures 3 and 4 respectively. In the local message mode, the connection between the Originate Function, such as 9, and the Answer Function, such as 12, is linked within the single SAFF 8 by way of a Local Call Loop-back connection 13, between the two Functions. In the long-distance mode, the Originate Function 9 of SAFF 8, near the originator, is linked to the Answer Function 19 of another SAFF 18, near the destination, by long-distance lines, such as 14, or 16 for SAFF 18. Thus, processing a long-distance message involves the same basic steps as a local message, except that the activity is shared interactively between at least two different SAFFs.

Originate Function

With this understanding of SAFF functions, the following detailed discussion will illustrate the operation of the system in the long-distance case, since it is the more complex, and therefore provides a more comprehensive example. Figures 1, 2, 3, and 4 all show elements of the SAFF system in varying degrees of detail and all will be referred to in the following. It will be noted that some critical elements are shown in more than one of the Figures.

As an example, it is assumed that one of the subscribers 1 attached to SAFF 8 wishes to send a fax message to one of the subscribers 28 attached to SAFF 18. The subscriber 1 places the call to the destination machine 28 which is routed over SAFF Directed Lines 4 to the Originate Function 9 of SAFF 8. These signals originate within the SAFF system and they are picked up by the On-net Interface 64 which is part of the Originate Function, as shown in Figure 3. This Interface signals the Originate Host Computer 70 of the incoming call and the Host responds by directing the incoming data to a Mass Storage Unit 67 where it is stored in a file 68.

During this storage process the Host directs two other activities. It creates a call status record file 69 (Figure 3) in mass storage, recording the time and date of the origination, the telephone number of the calling machine, the telephone number of the destination machine, any security or other special services requested, various housekeeping information, and it assigns the Message Code number which locates not only the status file but also the fax data file associated with it. The Host also passes the destination machine's telephone number to the Outbound Control unit 74 which proceeds to connect the originating SAFF 8 with the nearest available SAFF 18 to the destination through a long-distance interface 75 over long-distance circuits 79 (14 in Figure 1). In the process of establishing this connection, the Outbound Control unit employs an algorithm which examines the number and kind of available trunk resources and chooses the most efficient combination of these lines for the task required.

Answer Function

The originating SAFF 8 then proceeds to transmit the originator and destination telephone numbers, the stored fax image, the Message Code, and other housekeeping data to the destination SAFF 18. These data are sent by the most expedient mechanism offered by the long-distance service. For example, if this service employs digital communications, the fax data may well be transmitted at a significantly higher rate than it was originally received into the system.

The fax data is received by the Long-distance Interface 95 (Figure 4) in the Answer Function 19 of the destination SAFF. This unit signals the Answer Host Computer 85 of the incoming data. The Host then routes these data to its Mass Storage facility 87. (It should be noted for later reference that the originator SAFF and the destination SAFF now both have a copy of these data.) The Host notes whether other messages are pending for the destination machine and either opens a Delivery Queue file 88, or appends the new message to the existing Queue File.

The Host also records the arrival time and other pertinent information about the fax message in a Call Status file 90 in Mass Storage unit 87, and sends a status update back to the originating SAFF 8 by way of the Status and Control Interface 84, and the System Status and Control Unit 11 via Long-distance Trunks 15.

It then signals the Local Interface 83 to dial the destination machine's (81 in Figure 4) telephone number on ordinary outgoing local lines 24, 82. If the destination fax's line is available, the destination SAFF now becomes the proxy for the originating fax machine and engages the destination machine in the necessary preliminary digital dialogue.

If this is successful, the document image, including the source and destination identification information, the Message Code, and the entry and delivery times, is played back from storage and delivered to the destination. A "Delivery Record" is then created by the Answer Host 85 which indicates the date and time of delivery, and any other pertinent data. The

Delivery Record is sent back to the originating SAFF 8, again by way of the Status and Control Interface 84, and the System Status and Control Unit 11, via Long-distance Trunks 15. The originator SAFF 8 then appends this information to the Acceptance Record to form a complete "Transaction File". The originating SAFF 8 then sends this file, as a delivery receipt or report, back to the originating machine 1, 60, as a fax document.

If the destination machine's line is busy, or the contact fails for some other reason, the destination SAFF's Host Computer 85 will enter a sequence whereby it will attempt to contact the destination machine and transmit the document on a predetermined schedule for a specific period of time or number of tries. As this sequence is entered, a "Retry Record" is generated documenting the situation and the system's response to it. This record contains the reason that the delivery was delayed and it indicates which protocol the system will use to attempt to deliver the message. This is transmitted back to the originating SAFF 8, as described above, and appended to the previously described Acceptance Record to form a Transaction File which is then sent as a fax message back to the originator. The assigned Message Code is a part of every transaction report and may be used at any time to trace the status of undelivered documents, as will be described shortly.

If the retry effort is ultimately successful, a Delivery Record is appended to the Transaction File which is sent back to the originating machine. If the effort fails after reaching the predetermined limit, this is also recorded, appended, and sent back to the originator. In this case, the originator is given the option of dialing back into the system within a certain length of time (typically several hours) and instructing the destination SAFF as to how to dispose of the document (eg. repeat retry sequence, forward to a different telephone number, or delete the message).

This process is handled by using an ordinary touchtone phone to dial a multipurpose (perhaps, toll free) fax system "Service Number"; which will be referred to here and in later sections. This might be a unique number for every SAFF, or it might be a standardized number common to many localities, except perhaps for area code, such as is 555-1212 for calling "Information". This Service Number is answered by the General Service Control units (10 in Figure 1, 50 in Figure 2) of the SAFF to which the call is directed. This unit contains an automated voice response system that presents a menu of the available services and prompts the user to select the desired choices by pressing particular numbers on the touchtone keypad. In an advanced embodiment, a computer-based voice recognition system replaces the keypad and accepts verbal commands in a conversational way.

The General Service Control unit 50 can communicate with its own System Status and Control unit 11, and through that unit, any other such unit 11, 20 via Long-distance Trunks 15. Through these connections, both inquiries relating to messages in the system and instructions as to their disposition may be addressed to the entire SAFF system.

Having selected the "failed-connection message disposition" choice, the user is prompted to key in the Message Code. The system verbally repeats the code and the delivery discrepancy for verification, and then presents a menu of disposition options for the user to select with the keypad.

If the user does not take advantage of this "what to do now" opportunity within the time limit, the message is retransmitted back to the originator with a report. It is then erased from both the originator and destination SAFF files after a suitable delay (typically six hours). If the originator wishes to resend the message during this "grace" period, it may be recovered and resent to the original destination or forwarded to another destination(s), as will be described later.

In each of the various cases where the SAFFs automatically direct fax message status reports (such as, the Acceptance, Delivery, or Retry records above), the system can be programmed to accumulate records from all calls over a period of time (eg. an hour) at the originator SAFF and deliver them as a single fax document at the end of the period or upon request by the originator. This has the advantage of reducing the number of report calls and the subsequent burden on the originating fax machine. The originator SAFF will enter a retry sequence if it finds the originator's line busy or the machine unavailable when it attempts to deliver reports. This is a persistent sequence which it will continue trying for direct contact at intervals of an hour or so for a considerable length of time (eg. 72 hours). It also places a copy of the report in the originator's Mail Box (described below) so that the originator may recover it in between SAFF delivery attempts.

It should also be noted that the originator has the option of dialing the Service Number at any time and inquiring about the status of a given message. Here again, the voice response system prompts, presents menus, and uses the Message Code to locate and report on the current location and condition of the message. A written record can be directed to the originating or destination fax machine, if desired.

Another feature of the system is that the act of accepting and storing an incoming message at the originator SAFF, and the act of dialing and forwarding that message to the destination by the destination SAFF, can overlap in time. That is, if the originator SAFF has lines available, once the initial connection dialogue between the originator and the SAFF is complete, the SAFF may immediately make its first attempt to contact the destination SAFF and, thus, the destination machine, while it is beginning to spool the document.

If this immediate contact is successful, then the message is passed from the originator SAFF 8 to the destination SAFF 18 to the destination machine 28 directly from the Originate Host Computer's memory 70 while the two SAFFs are still in the process of spooling the document to disk. This is facilitated by a "write-through pipeline" whereby the Originate Host 70 passes the incoming fax data through directly to the Outbound Control unit 74 at the same time it is being written to mass storage. It is held in a temporary memory buffer in the Outbound unit until it is clear whether or not an immediate connection to the destination machine is possible. At that point the temporary buffer fax data is either sent and then deleted, or merely deleted. The net effect is that the spooling process only adds a few seconds delay in the message delivery over the traditional direct machine-to-machine contact when the destination machine is readily available.

On the other hand, if lines are limited, the originating SAFF can choose to delay until suitable lines are available. This has the advantage of improving communications resource management and enhancing the efficiency of the telephone system's line usage over the direct contact scheme.

The foregoing describes the basic fax SAFF message handling system and from this discussion several advantages should be apparent. The originating machine always functions as if it makes contact and delivers documents on the first try, thus immediately freeing the machine and the attendant personnel for sending or receiving other transmissions. Likewise, the telephone system only handles one call across its local and long-distance lines from the originating machine to the destination SAFF, since the state of the destination machine has no impact on the call. This significantly improves the efficiency of line usage when messages are addressed to busy fax terminals.

Although some additional calls are needed to deliver the various reports, these require very little long-distance time, as they are transmitted over the circuits as highly compressed coded messages. It is the nearby originating SAFF that translates them into "plain language" for fax delivery as a local message. As pointed out, additional savings in these local messages can be gained by compiling multiple reports and delivering them in bulk as a single call. It should be noted that the delivery of reports to an originator is a cooperative process between the Originate Function and the Answer Function of the originate SAFF. The Originate Function 9 actually generates these reports and passes them through the Local Call Loop-back 13 (76 in Figure 3) to the Answer Function 12 for delivery as an ordinary fax message.

In addition to these basic features, the design of the system also provides for a number of additional services and advantages which are described below.

Message Queuing

As pointed out, all fax messages directed to a particular telephone number are spooled by the Answer Function of the destination SAFF, as detailed in Figure 4. The Host Computer 85 controlling this function monitors the incoming messages and links all undelivered messages for a given telephone number into a message Delivery Queue file 88. The computer also compiles a constantly updated, ordered catalog of the file names of the messages waiting for each fax machine.

Consequently, when messages arrive at a rate faster than they can be delivered, for whatever reason, they are held in this queue for delivery. As soon as the destination SAFF establishes contact with the destination machine, it begins sending the entire queue of messages in a single, essentially uninterrupted transmission. Messages that arrive while the transmission is in progress are appended to the end of the queue.

This scheme eliminates the "trial and error" dial and redial attempts that result from a number of independent incoming calls competing in an uncoordinated way for the single destination line. It can significantly enhance the efficiency of the destination fax machine and the long-distance and local telephone circuits connected to a busy machine.

When the queue exceeds a certain limiting size, the destination SAFF will periodically insert and send a "Queue Report" (as a fax document) to the destination machine showing a list of the waiting messages. This list shows the originating machine identification, the time entered into the originator SAFF, the number of pages in the document, and the approximate time that the message will be delivered based on its position in the queue.

The user can advance a particular message to the head of the queue by calling the fax Service Number and supplying the desired message number, by using the voice response menus. The General Service unit 50 directs these instruction to the System Status and Control Unit 11, which in turn directs them to the Answer Function Host 85 through its Status and Control Interface 84.

Alternately, the originator can designate a priority level to a given fax message at the time it is dialed in (eg. by using a different access code). In this case, the destination SAFF will insert higher priority messages ahead of lower priority messages in the queue as they are received. The originator would normally pay a premium price for this service.

Another originator option is the time of delivery. If desired, the originator can specify the time of day which the message should be delivered. In this case the message is forwarded to the destination SAFF directly, but is not entered into the queue until the specified time. This can be used in combination with an assigned high priority to insert the message at the head of the queue at the appointed time.

When messages are finally delivered to the destination machine they are not immediately erased from the spool file 88 at the destination SAFF. Rather, they are maintained in a "Delivered Message" directory 90 for a period of time (typically six hours). A feature offered by this action is the opportunity for the subscribing recipient of a message to make additional copies, redirect, or forward copies of selected messages to other destinations. This is accomplished by calling the Service Number and selecting the appropriate choices from the voice response menus.

Security and Mail Boxes

It is not uncommon for documents of a sensitive nature to be sent by facsimile from place to place. This is often a problem, especially in a busy office or where a machine is nominally unattended during the transmission, in that the originator has no control over who may be standing by the machine when the document prints out, or who may leaf through a stack of faxes piled up in a hopper right after lunch.

This is a problem which others have attempted to deal with in a variety of ways. For example, Bond, U.S. Pats. 3,594,495 and 3,641,432, discloses a "radio facsimile postal system" which features the direct delivery of documents to specific addressees by facsimile via communications satellites. In this system, intended as a replacement for or supplement to the ordinary "paper" postal system, fax messages were directed from special public fax terminals operated by the post office to a central satellite earth-station. Here the messages were sorted according to their geographical destination for concentration and uplinking to a satellite servicing that area. The satellite then broadcasts all of the uplinked messages back to Earth.

In principle, anyone with a radio receiver in the satellite's service area could access any of the messages, so Bond built in a "privacy code" which operated with the receiver to allow the message to print out only on the desired machine. In reality, this privacy code was nothing more than an addressing signal which enables the selected fax receiving system. Thus, Bond's system is merely a restricted version of the services presently provided to fax users by the telephone networks. His privacy code function is the same as a telephone number: it selects which of a plurality of fax machines will actually receive the message. Unfortunately, his approach leads to exactly the security dilemma facing telephone fax users.

Chapman, U.S. Pat. 4,106,060, has approached the problem in a somewhat different way. He too discloses a facsimile-based mail system. However, in his system, the messages are directed by whatever means to a "paper" post office near the addressee, rather than the addressee's home or place of business. This post office then makes a paper copy of the the fax message, places it in an envelope, and delivers it to the addressee as ordinary mail. This is a reasonably effective solution to the security problem, but it can only be relied upon to provide "next day" delivery, and there are a number of other, competing alternatives for document delivery service on that time scale.

In the present invention the security problem is addressed by a control variation of the destination SAFF queuing system. Messages which the originator wishes to designate as secure are temporarily directed to a auxiliary storage file 54, 89 in the Answer Function of the destination SAFF called a "Mail Box". Instead of being delivered to the destination machine, a report is sent to that machine indicating that a secure message is waiting for a particular addressee. Optionally, a voice message may be directed to a designated telephone number by the General Service Control 50.

This feature works in the following way. Each individual SAFF is assigned its own unique telephone exchange code or codes (typically indicated by the first three digits of a seven digit local number). Thus, the SAFF appears to the world as if it were a distinct telephone exchange(s), separate from all other exchanges in that area code region. All subscriber's to a given SAFF are assigned their fax telephone numbers with that exchange prefix. Subscribing individuals wishing Mail Boxes (typically associated with a "default" fax machine) are issued "fictitious" telephone numbers which actually terminate in fax Mail Boxes, rather than in an actual telephone line.

Mail Box numbers are published so that correspondents may use them. In addition, each individual is also given a secret security code or PIN number which will access his or her box. The host computer managing the SAFF maintains a list that relates each fictitious number with the individual's name, the security code, and the real telephone number of the default destination machine. This default machine is the one to which messages and reports will normally be sent, when appropriate.

An originator wishing to send a secure message merely dials the (fictitious) Mail Box telephone number at the time the document is sent. The system directs the message to the Mail Box file 89 in the destination SAFF associated with that number, and the Answer Host 85 sends a "Message Waiting" report to the default destination fax machine through the Local Interface 83. If more than one message is in the Mail Box queue, then this report lists them all.

In order to get the fax document actually sent to the destination, the security code must be sent back to the destination SAFF. Typically, this would be done by the addressee dialing his or her own Mail Box number. Since this call originates from a "normal" telephone 34 over Ordinary Local Lines 40, rather than the fax's SAFF Directed Lines 38, the call is directed to the Off-net Incoming Screener 48 in the (destination) SAFF which functions in conjunction with a mailbox service control 49. This unit recognizes that the call is not a fax transmission and thus treats it as a voice service request. A voice response system then prompts the caller to key in the security code. When the correct code is supplied, the SAFF system announces the number of messages waiting and, if desired, the message codes of each. Mail Box contents are maintained in a queue 89 just as are "regular" spool files. Thus, the user is also given the opportunity to reorder the messages within a Mail Box Delivery queue, through the System Status and Control units 11, 20 in the same way as other messages.

The system finally permits the addressee to make a selection of messages for immediate release, and provides an opportunity to "redirect" them to a fax machine 3 other than the default machine over ordinary local lines 39. The SAFF then releases the selected documents and moves them to the head of the appropriate destination Delivery Queue 88 for immediate delivery.

When messages are accepted into the SAFF system and arrive at a mail box, The Answer Function of the destination SAFF issues a "Posting Report" which is directed back to the Originator in the manner described for other reports. The report is similar to a Delivery Report, except that it indicates that the message has been received by the mail box. When the Mail Box Queue is actually read by the addressee the Destination SAFF sends an actual Delivery Report to the originator indicating the date and time of delivery and so forth.

Another advantage of the Mail Box system is that it can provide a convenient way for individuals who are away from their "home" machine to still have access to their documents. Such individuals may call in to their Mail Box number to hear from the voice response unit whether they have any messages waiting. By use of the redirection feature, messages sent to a fax Mail Box can be accessed by an individual with the security code from any telephone with a fax machine.

For example, a person on a business trip can have all his or her fax documents directed to their Mail Box. Upon arriving at a hotel that has a fax machine, the traveler places a call to the Mail Box number and supplies the information outlined above, including the telephone number of the hotel fax machine. The SAFF then calls the hotel machine and dumps the queue of waiting documents.

Broadcasting

The queuing, Mail Boxes, and security codes are all derivative benefits of the spooling of messages at the destination SAFF. There is a counterpart advantage to the originator SAFF's spooling as well. Since the originator SAFF maintains a copy of each message, that copy can be used to broadcast messages to multiple destinations.

This can be initiated in a number of ways. For example, the user can dial in a code prefix indicating that a list of destination numbers is to follow. The numbers are then entered and finally another code is entered to signal "end of list". The Originate Host 70 recognizes these inputs and attaches them to the message which follows. As an alternative, the user can store different numbered broadcast telephone lists in the Originate SAFF mass storage files 69 (entered much as described above) and invoke them simply by dialing a two or three digit "short-cut" code. In either case, from there the fax transmission to the originator SAFF proceeds normally.

Upon reception of the list and the document, the originator SAFF proceeds to open as many local loop-back or long-distance lines as it can to deliver the broadcast message to the various destinations, essentially simultaneously. Although the originator is billed for making a number of different calls, in fact the originating machine is only tied up for the time required to make one call. Furthermore, the full power of the delivery system is asserted for each destination machine, including reporting, redials, queuing, and so forth.

A feature related to broadcasting is the redirection of messages by the originator. Since fax messages are spooled at the originator SAFF and held for a period of time even after delivery (typically six hours), the originator can dial the Service Number any time during this period and direct a copy of the spooled message to be sent to other destination machines.

Communications With Non-subscribers

Thus far, the discussion has presumed that both the originator and answerer were subscribers to the SAFF system. It is quite reasonable to assume that subscribers will wish to send or receive fax messages with non-subscribers, as well. While the services provided by the SAFF are more limited in such cases, nevertheless, the system both anticipates and enhances communications with non-subscribers for the benefit of the subscribers.

When a subscriber originates a call to a non-subscriber the delivery process is almost identical to subscriber-to-subscriber calls. The fax data is forwarded to the Answer Function of the appropriate destination SAFF and delivery is pursued, all in the usual way. For the benefit of the subscribing originator, the message is stored in the usual way at the destination SAFF until delivery is completed. If multiple SAFF-processed messages arrive before the delivery is complete, a temporary Delivery Queue will be created and used as required. However, since the non-subscriber will have no account in the system, attempts to use the Service Number to manipulate the queue, forward messages, make multiple copies, and use the other special services available to a subscribing answerer, will be unsuccessful.

Calls originated by a non-subscriber directed to a subscribing answerer move by a somewhat different mechanism. As noted, each SAFF appears to the world as a distinct telephone exchange and all subscriber's to a given SAFF are assigned their fax telephone numbers with that exchange prefix. Consequently, all calls directed to a SAFF subscriber eventually end up at the subscriber's SAFF, whether they originated from within the SAFF system network or not. Messages originating "off-network" can arrive by any route. For example, they may be truly local calls, or they may be long-distance calls which arrive over any available long-distance network.

In any case, messages originating from a non-subscriber 33 are delivered to the answering fax machine's SAFF by the local lines 39 provided by the local telephone company. They are answered by the SAFF's Off-net Incoming Screener 48, which, upon noting that they are fax transmissions, directs the calls to the Originate Function 9 of that SAFF. From that point, the call is treated as if it were a local fax call and it is passed over to the Answer Function 12 via the Local Call Loop-back 13 for delivery to the subscriber.

In this situation an Acceptance Record will be returned to the originating machine, but no further originator services are provided. On the other hand, the answering subscriber has the full range of Answer Function available.

Charges and Detailed Billing

Normally, the Originate Function of the originator SAFF has ultimate responsibility for the management of outgoing messages. It initiates all connections to the Answer Functions of the various SAFFs with which it must communicate. It is the node to which all reports concerning message status and disposition must flow. It interrogates Answer SAFFs when extraordinary updates are required. Consequently, the Originate Function is also the focus of charging data.

The telephone company presumably charges for all of the various services provided by this system. The method, algorithm, and rates are determined by actual costs and applicable regulations. Typically, the user would be billed for telephone connect time, toll charges, extraordinary services, such as those provided by calling the Service Number, the amount of mass storage space consumed as a function of time, and so forth.

One of the user services for which a special charge might be made is a subscriber's customer specific billing system. In this option the user can "flag" each fax transmission with a keyed-in prefix which contains a user customer, client, or project number. This number is stored as a key field in the

Transaction File for that call. Thus, when the telephone bill is prepared, the billing computer can sort the subscriber's bill on this field and present the user with a list of all fax messages, total usage time, number of pages, and related charges, all grouped by the subscriber's own customers, clients, or projects. Furthermore, it can accept the subscriber's particular algorithm for billing calls to customers or clients and generate a column showing what the subscriber will bill for the service (as a separate matter from what the SAFF system and the telephone company have billed the subscriber). This can be of great assistance in attributing costs and billing customers for services rendered.

Software Control

In the preferred embodiment, each of the principal units of the SAFF such as described in Figures 2,3, and 4 is controlled by its own computer processing unit or units. These units are interrupt-driven computers which are connected together by the System Status and Control unit 11. This unit is an electronic switch yard for control communications between the Originate, Answer, and other units within a given SAFF, as well as the the other SAFFs in the system through the control long-distance trunks 15. While there are many tasks which the various control processors must perform to handle fax operations, the primary ones are intercepting incoming calls, either for fax forwarding or service requests, and delivering the fax messages to their destinations. The general software organization of these principal activities is shown in Fig. 5, 6, and 7. It should be noted that these figures are simplified and intended to be generally descriptive. For example, some procedures illustrated here as sequential (for the sake of simplicity) can actually be performed concurrently. Likewise, not every function of the system is represented in detail. Generally speaking, similar results also can be obtained with a number of other obvious arrangements of the functional blocks.

Broadly speaking, fax messages addressed to the Originate Function of a SAFF arise either through the special SAFF Directed Local Lines 4 (Figures 2 and 3) as a result of direct connection or dialing a special access code, or they arise from Ordinary Local Lines (off-net lines) 39, 40, 63. Those which arrive via off-net lines are processed first by the Off-net Screener 48, which may direct them to either the Originate Function 9 or to Mail Box Service 49. Figure 5, therefore combines all three of these related functions.

At the outset one of the two incoming call interfaces 64 and 65 signals the Host Computer 70 that it is beginning to process a call at 100 in Figure 5a. These units have their own buffer capability and can tolerate some delay before the Host responds. Ultimately the Host must decide whether it is responding to an on-net or off-net call 101. If it is an off-net call there are two possibilities (excluding wrong numbers) 102: it may either be a fax call, in which case it is from a non-subscriber to a subscriber, or it is a mail box service call. If it is a fax call then the billing for services must be directed to the subscribing destination addressee 112. From that point it is handled like an on-net call as will be described shortly.

If it is not a fax call then it is presumed to be a mail box service call 103, and the caller is presented with the voice response menu 104 for such service. The user responds to these prompts with a touchtone keypad, or verbally, 105 and a decision ladder, shown succinctly as 107 selects the desired implementation routine 108, 109, 110 (for brevity only three typical choices are shown, and this element is actually a loop which will permit multiple commands). The chosen routine passes parameters to a command parser 121 (Figure 5b) which prepares an command statement which is then sent 122 to the System Status and Control unit 11, through the interface 72. This command will be passed to the Answer Host 85 through its interface 84 for actual action on the Mail Box Queue 89. If the service requires a response to the caller the transmission path is reversed. When the operation is completed 123 the call is terminated.

If on the other hand, the original call is found at 101 to be an on-net call, billing is generally directed at the originator 113 and the Host 70 begins the opening digital dialogue 114 with the calling machine, acting in place of the destination machine. This dialogue includes gathering and storing the fax identifications, originating and destination telephone numbers and so forth 116. The Host opens a Transaction File and links it to a data file 117 for the expected data, and then stores all of the call and file information 118 keyed to the Message Code. The destination telephone number and other information are passed almost immediately 119 to the Outbound Controller 74, which then opens a temporary buffer to hold the fax message in case immediate contact can be established, and it attempts to establish that contact through the destination SAFF.

In pursuing this contact, the Outbound Controller 74 examines the status of available trunks. If trunks are available, it will immediately attempt to connect with the destination SAFF, otherwise it will defer the call until a trunk is available. In the event of a broadcast message, the Outbound Controller will select the number of trunks to use simultaneously based on the percentage of the trunks already in use, in order to avoid tying up all of the SAFF's outgoing capacity with a single message task. Other considerations can affect these usage choices depending on the details of the setting of the system.

The Host then enters a loop which gets the incoming fax data 125 (Figure 5b) from the On-net 64 or Off-net 65 Interface's buffer and stores each byte in the fax data file 126 while sending another copy 127 to the Outbound controller 74 until the incoming data is complete 128. The Host then checks 129 with the Outbound controller to see if it was successful in making immediate connection with the destination machine. If it was successful and a satisfactory transfer occurred, then a Delivery Report is sent back 132 to the originating machine before it leaves the line. Otherwise, an Acceptance Report is sent 131, and in either case the outcome is reported 133 to the Transaction File and the call is terminated 134.

A complementary set of activities occurs in the Answer Function of the destination SAFF as described in Figure 6a and b. Here an incoming call is detected 136 by the Inbound Control 92 (Figure 4). The Answer Host Computer 85 then opens a new fax data and Transaction file for the message if there is no current queue for that destination machine, or it prepares to append the data to an existing queue 137. The various call and file parameters are linked and stored 138 and the call parameters are passed through 139 to the Local Interface 83, which then decides 141 whether the call is addressed to a "real" fax number, or a fictitious number terminating in a mail box. If the number is real the Local Interface attempts to contact the destination machine for immediate delivery.

The Host then enters a loop where it gets the incoming data 147, stores it 148 in the fax Delivery Queue, and passes it through 149 to the Local Interface buffer. When the Host determines that the fax transfer is complete 150, it then checks 152 (Figure 6b) to see if the Local Interface has been able to make immediate delivery. If it has, the Host initiates the transmission of the Delivery Report 167 back through its Status and Control Interface 84 to the System Control and Status unit 11, which in turn updates the Transaction File and sends it back to the originator SAFF over Trunk 15. It is this communication which ultimately results in the immediate Delivery Report described previously. The transaction in then terminated 169.

If immediate connect is not established a Retry Report is sent 153 back through the System Status and Control unit and the Retry sequence begins. The Retry criteria can be varied

154, both in place and with the SAFF setting. For example, if the SAFF is integrated into a local exchange, the SAFF can actually monitor the desired line and simply wait for it to become available. In other settings it will be necessary for the SAFF to actually redial at prescribed intervals. In any case attempts to connect are made 155 and if they are not successful 156 a counter or timer is checked 159 to see if the retry limit has been exceeded. If not, the process is repeated and if so, a Failed Delivery Report 160 is sent back through the system and the effort terminated 170.

If the retry effort is successful the Delivery Queue is retrieved 158 and message by message 162 the queue is dumped, with a pause 163 after each message to confirm receipt, send a Delivery Report 164 and to check for end of queue 165. If a message fails during the queue dump the retry sequence at 154 is resumed at the failure point and the process repeated to a conclusion. When the last message has been received satisfactorily, the transaction is terminated 168.

If it is determined at 141 (Figure 6a) that this is a mail box call, a loop is entered which gets the fax data 142 and stores it 143 in the appropriate Mail Box Queue. When the end of message is detected 144, a Posting Report 145 is sent back through the system and a Message Waiting Report 146 is sent forward through the system to the default destination machine.

General Service calls always arrive on Ordinary Local Lines 5. Upon detection and answering 172, the voice response menu is presented 173 to the user. As with the Mail Box Service, the user keys in responses or gives them orally 174 and a decision ladder 175 identifies the desired service routine such as 177, 178, or 179. Here again only a few of the possible choices have been shown for sake of illustration and looping for multiple service requests is provided. The selected service routine generates command parameters which are parsed 181 as system commands and sent 182 to the System Status and Control unit 11 for execution. Upon completion of all requests the call is terminated 183.

What has been described are the presently preferred embodiments of a system and method for providing a comprehensive interactive facsimile message management system embedded in a switched telephone network. It should be apparent that many modifications to the system and the method are possible without departing from the true spirit and scope of the invention.

CLAIMS

1. A system for facilitating facsimile communications between a transmitting facsimile machine and at least one intended receiving facsimile machine, comprising at least one store and forward facility, means coupling the at least one store and forward facility to the switched telephone network for receiving transmissions from a transmitting facsimile machine, said store and forward facility including computer means for controlling its operation and including mass storage means for storing facsimile transmissions together with information identifying the transmitting facsimile machine and the at least one intended receiving facsimile machine under control of said computer means, said store and forward facility also including means coupling it to the switched telephone network for transmitting facsimile messages stored in the mass storage means to at least one intended receiving facsimile machine.

2. The system of claim 1 wherein said computer means is programmed such that if the at least one intended receiving facsimile machine is busy or otherwise unable to receive a transmission at the time the store and forward facility attempts to transmit a facsimile message stored in the mass storage means, the store and forward facility periodically retries transmitting the facsimile message to the at least one intended receiving facsimile machine.

3. The system of claim 2 wherein said computer means is additionally programmed to establish a linked queue in said mass storage means spooling all stored facsimile messages intended for a particular receiving facsimile machine, and transmitting all the spooled facsimile messages intended for that particular receiving facsimile machine upon successfully making contact with the intended receiving facsimile machine.

4. The system of claim 1 wherein said computer means of said at least one store and forward facility is programmed, upon successful completion of a facsimile transmission to an intended receiving facsimile machine, to transmit a message to the transmitting facsimile machine confirming delivery of the transmission to the intended receiving facsimile machine.

5. The system of claim 2 wherein said computer means of said at least one store and forward facility is programmed, upon being unsuccessful in making a transmission to an intended receiving facsimile machine, to transmit a message to the transmitting facsimile machine indicating that the message has been entered into the mass storage means at the store and forward facility, and at least also indicating the reason for

a delay in transmitting the message to the intended receiving facsimile machine.

6. The system of claim 1 wherein the at least one store and forward facility includes means for receiving broadcast instructions from a user at a transmitting facsimile machine and associating those broadcast instructions with a facsimile message received from the transmitting facsimile machine and stored in the mass storage means, and for transmitting the stored facsimile message to a plurality of receiving facsimile machines in accordance with the broadcast instructions.

7. A system in accordance with claim 1 wherein said mass storage means additionally includes mailboxes associated with particular system subscribers and wherein facsimile messages received and stored by the mass storage means and intended for receiving facsimile machines associated with those subscribers are stcred in the respective mailboxes, said store and forward facility being responsive to instructions received from a subscriber to transmit the facsimile messages stored in that subscriber's mailbox to any particular facsimile machine designated in the instructions by the subscriber, whereby a subscriber who is traveling or otherwise away from the fixed location of his facsimile machine may have facsimile messages intended for receipt by his facsimile machine collected, and retrieve them from any location where any other facsimile machine is situated.

8. A system in accordance with claim 1 wherein said computer means of said at least one store and forward facility is programmed to retain a facsimile message in the mass storage means for a predetermined time period even after successful transmission of the facsimile message to an intended receiving facsimile machine, and wherein the store and forward facility is responsive to instructions received from either originating or receiving subscribers to retransmit the facsimile message to another intended receiving facsimile machine.

9. A system in accordance with claim 1 for use in system operation wherein individual subscribers may be provided with unique PIN numbers, wherein individual subscriber PIN numbers are stored in the mass storage means, and wherein the store and forward facility recognizes an incoming facsimile message that is security coded by a transmitting facsimile machine, and wherein the security coded facsimile message is sent to an intended receiving facsimile machine only upon receipt from the intended receiving facsimile machine of the appropriate subscriber PIN number.

10. A system in accordance with claim 9 where said computer means is programmed such that, upon receipt by the store

and forward facility of a security coded facsimile message from a transmitting facsimile machine, the store and forward facility sends a transmission to an intended receiving facsimile machine indicating that the store and forward facility is holding a security coded facsimile message, whereby a subscriber at the intended receiving facsimile machine is prompted to input to the store and forward facility his PIN in order to have the facsimile message transmitted to the intended receiving facsimile machine.

11. A method for facilitating facsimile communications between a transmitting facsimile machine and at least one intended receiving facsimile machine, comprising the steps of providing at least one store and forward facility having computer means for controlling its operation and having mass storage means for storing facsimile messages, coupling the at least one store and forward facility to the switched telephone network for receiving facsimile messages from transmitting facsimile machines, recording received facsimile messages in the mass storage means together with information indicating the transmitting facsimile machine and the intended receiving facsimile machine, and transmitting facsimile messages stored in the mass storage means to intended receiving facsimile machines.

12. A method in accordance with claim 11 including the step that if an intended receiving facsimile machine is busy or otherwise unavailable to receive at the time the at least one store and forward facility attempts contact to transmit a facsimile message, of periodically retrying to transmit the facsimile message to the intended receiving facsimile machine.

13. A method in accordance with claim 11 including the step of establishing a linked queue in the mass storage means spooling all stored facsimile messages intended for a particular receiving facsimile machine, and transmitting all the spooled facsimile messages intended for that particular receiving facsimile machine upon successfully making contact with the intended receiving facsimile machine.

14. A method in accordance with claim 11 including the step, upon successful completion of a facsimile transmission to an intended receiving facsimile machine, of transmitting a message to the transmitting facsimile machine confirming delivery of the transmission to the intended receiving facsimile machine.

15. A method in accordance with claim 11 including the step, upon being unsuccessful in making a transmission to an intended receiving facsimile machine, of transmitting a message to the transmitting facsimile machine indicating that the message has been entered into the mass storage means at the store and forward facility, and at least also indicating in the message the reason for a delay in successfully transmitting the message to the intended receiving facsimile machine.

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16. A method in accordance with claim 11 including the step of providing the at least one store and forward facility with means for receiving broadcast instructions from a user at a transmitting facsimile machine and associating those broadcast instructions with a facsimile message received from the transmitting facsimile machine and stored in the mass storage means, and including the step of transmitting the stored facsimile message to a plurality of receiving facsimile machines in accordance with the broadcast instructions.

17. A method in accordance with claim 11 including the step of defining mailboxes in the mass storage means associated with particular system subscribers, and including the step of storing facsimile messages intended for those particular system subscribers in their respective mailboxes, and further including the step, in response to instructions received from a system subscriber, of transmitting facsimile messages stored in that subscriber's mailbox to a facsimile machine designated by that subscriber in the instructions.

18. A method in accordance with claim 11 including the step of retaining facsimile messages in the mass storage means for a predetermined time period after successful delivery of the facsimile messages to intended receiving facsimile machines, and, in response to instructions received from either the transmitting or receiving facsimile machines with respect to a particular facsimile message, the step of retransmitting that particular facsimile message to additional intended receiving facsimile machines.

19. A method in accordance with claim 11 including the step of providing subscribers with unique individual PIN numbers, storing the individual PIN numbers in the mass storage means, recognizing an incoming facsimile message from a transmitting facsimile machine which has been security coded, transmitting to the intended receiving facsimile machine for the security coded message a message indicating that the store and forward facility is holding a security coded message, and transmitting to the intended receiving facsimile machine the security coded message only after receipt by the store and forward facility from the intended receiving facsimile machine of the unique PIN number of a subscriber associated with that intended receiving facsimile machine.

20. A method for facilitating facsimile communications between a transmitting facsimile machine and at least one intended receiving facsimile machine, comprising the steps of providing a plurality of store and forward facilities at geographically spaced locations each having computer means for controlling its operation and having mass storage means for storing facsimile messages, coupling each store and forward

facility to the switched telephone network for both receiving from and transmitting to a plurality of facsimile machines associated with each store and forward facility facsimile messages, recording in the mass storage means each facsimile message transmitted from an associated facsimile machine together with information indicating the transmitting facsimile machine and the intended receiving facsimile machine, and transmitting facsimile messages stored in the mass storage means to intended receiving facsimile machines if those intended receiving facsimile machines are associated with the store and forward facility which received the facsimile message from a transmitting facsimile machine, or to another of the plurality of store and forward facilities if the intended receiving facsimile machine is associated with the another store and forward facility.

21. A method in accordance with claim 20 including the step that if an intended receiving facsimile machine is busy or otherwise unavailable to receive at the time a store and forward facility attempts contact to transmit a facsimile message, or periodically retrying to transmit the facsimile message to the intended receiving facsimile machine.

22. A method in accordance with claim 21 including the step of establishing a linked queue in each mass storage means spooling all stored facsimile messages intended for a particular receiving facsimile machine, and transmitting all the spooled facsimile messages intended for that particular receiving facsimile machine upon successfully making contact with the intended receiving facsimile machine.

23. A method in accordance with claim 22 including the step, upon successful completion of a facsimile transmission to an intended receiving facsimile machine, of transmitting a message to the transmitting facsimile machine, either directly or through another store and forward facility associated with that particular transmitting facsimile machine, confirming delivery of the transmission to the intended receiving facsimile machine.

24. A method in accordance with claim 23 including the step, upon being unsuccessful in making a transmission to an intended receiving facsimile machine, of transmitting a message to the transmitting facsimile machine, either directly or through another store and forward facility associated with that particular transmitting facsimile machine, indicating that the message has been entered into the mass storage means at one of the store and forward facilities, and at least also indicating the reason for a delay in successfully transmitting the message to the intended receiving facsimile machine.

25. A method in accordance with claim 24 including the step of providing the store and forward facilities with means for receiving broadcast instructions from a user at a transmitting facsimile machine and associating those broadcast instructions with a facsimile message received from the transmitting facsimile machine and stored in the mass storage means, and including the step of transmitting the stored facsimile message to a plurality of receiving facsimile machines in accordance with the broadcast instructions, either directly or through additional store and forward facilities associated with particular ones of the plurality of intended receiving facsimile machines.

26. A method in accordance with claim 25 including the step of defining mailboxes in the mass storage systems of each store and forward facility associated with particular system subscribers associated with particular store and forward facilities, and including the step of storing facsimile messages intended for those particular system subscribers in their respective mailboxes, and further including the step, in response to instructions received from a system subscriber, of transmitting facsimile messages stored in that subscriber's mailbox to a facsimile machine designated by that subscriber in the instructions.

27. A method in accordance with claim 26 including the step of retaining facsimile messages in the mass storage means for a predetermined time period after successful delivery of the facsimile messages to intended receiving facsimile machines, and, in response to instructions received from either the transmitting or receiving facsimile machines with respect to a particular facsimile message, the step of retransmitting that particular facsimile message to additional intended receiving facsimile machines.

28. A method in accordance with claim 27 including the step of providing subscribers with unique individual PIN numbers, storing the individual Pin number in the mass storage means of a store and forward facility associated with a particular subscriber, recognizing an incoming facsimile message from a transmitting facsimile machine which has been security coded, transmitting to the intended receiving facsimile machine for the security coded message a message indicating that the store and forward facility is holding a security coded message, and transmitting to the intended receiving facsimile machine the security coded message only after receipt by the store and forward facility from the intended receiving facsimile machine of the unique PIN number of a subscriber associated with that intended receiving facsimile machine.

29. A system in accordance with claim 1 wherein said computer means is programmed to store in the mass storage means relevant charging parameters including number of pages, destination and special system feature options provided for each facsimile message sent by a subscriber and received by a subscriber from a non-subscriber, and to generate charging summaries for subscribers periodically from the stored charging parameters.

30. A method in accordance with claims 11 or 20 including the step of storing in the mass storage means relevant charging parameters including number of pages, destination and special system feature options provided for each facsimile message sent by a subscriber and received by a subscriber from a non-subscriber, and generating charging summaries for subscribers periodically from the stored charging parameters.

31. A method in accordance with claims 11 or 20 including the step, upon receipt of a facsimile message from a transmitting facsimile machine, of immediately attempting delivery of the facsimile message to an intended receiving machine at the same time the message is being recorded in the mass storage means.

32. A method in accordance with claims 11 or 20 including the step that when an additional facsimile message intended for a particular receiving facsimile machine is received by a store and forward facility while that facility is in communication with that particular facsimile machine, the additional facsimile message is immediately appended to a message queue for the particular facsimile machine and delivered as part of the communication with that particular facsimile machine.

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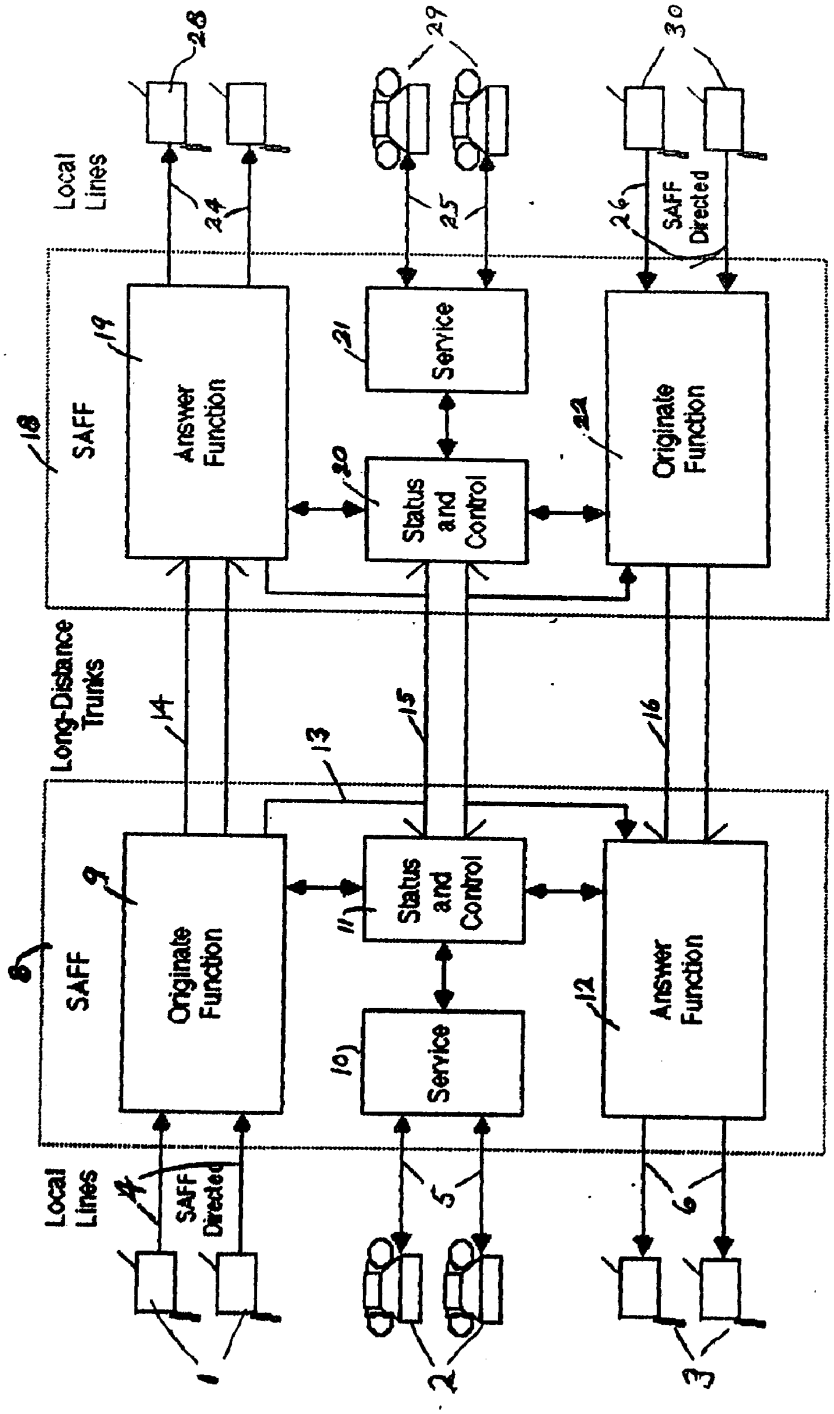

Fig 1

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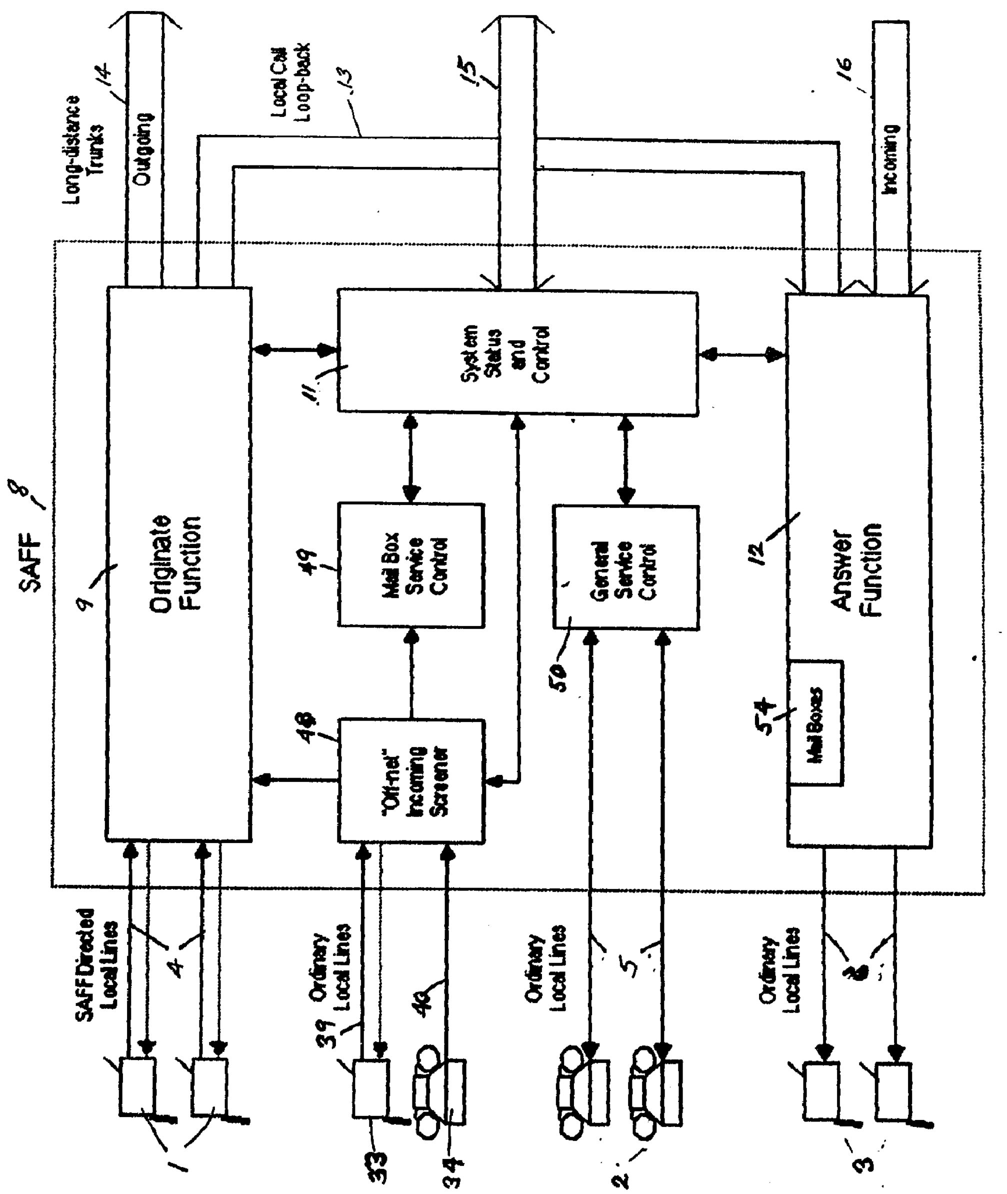

Fig. 2

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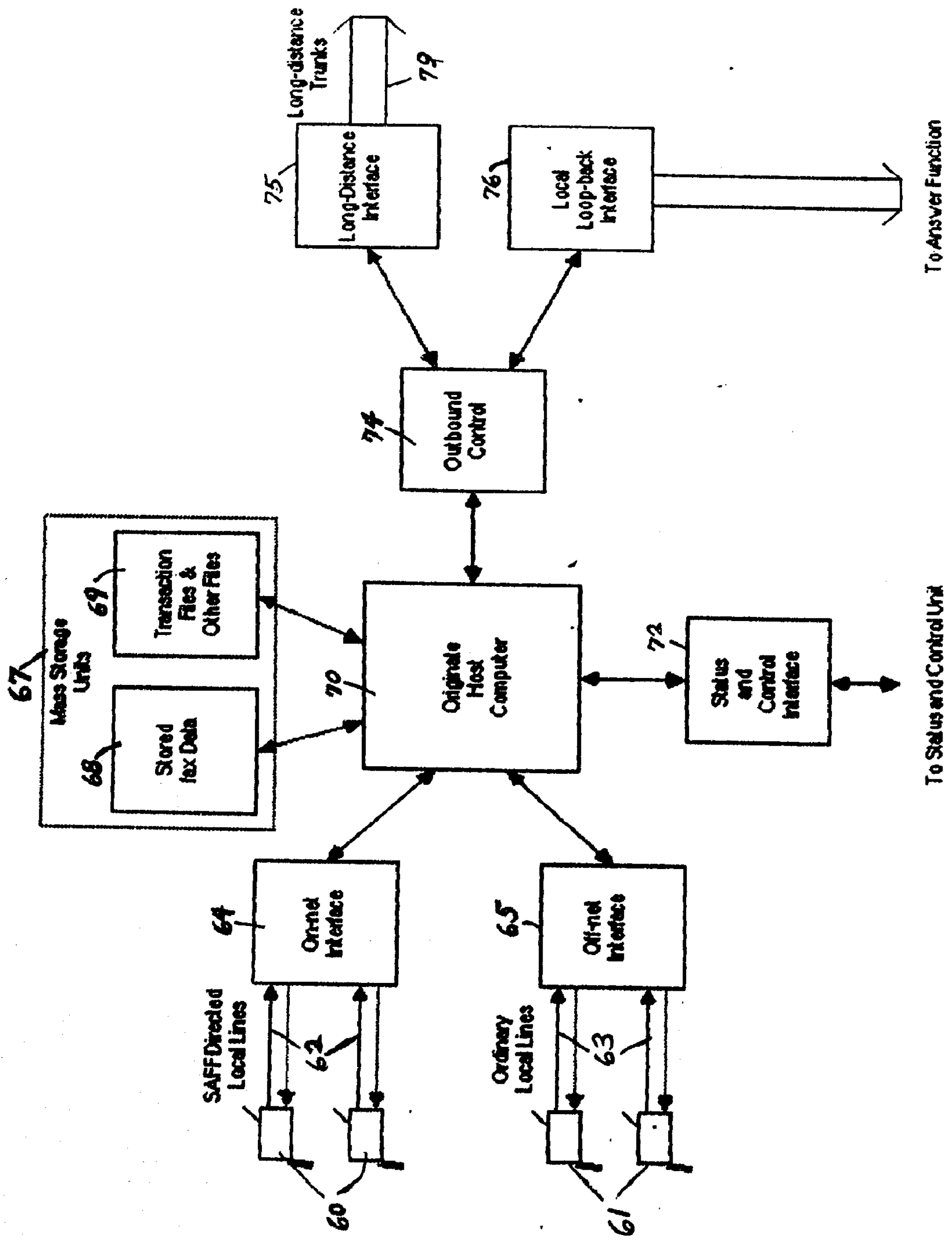

Fig. 3

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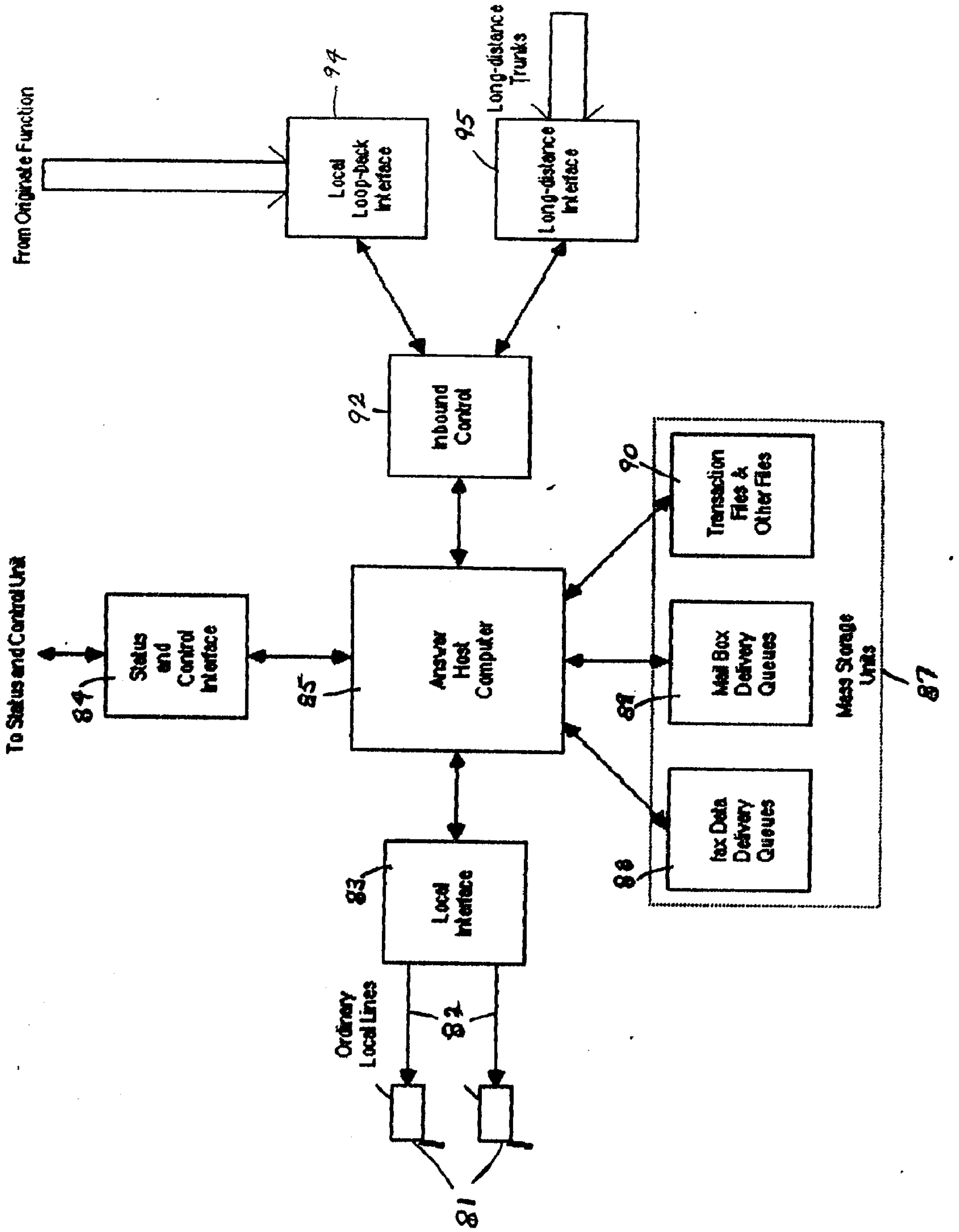

Fig 4

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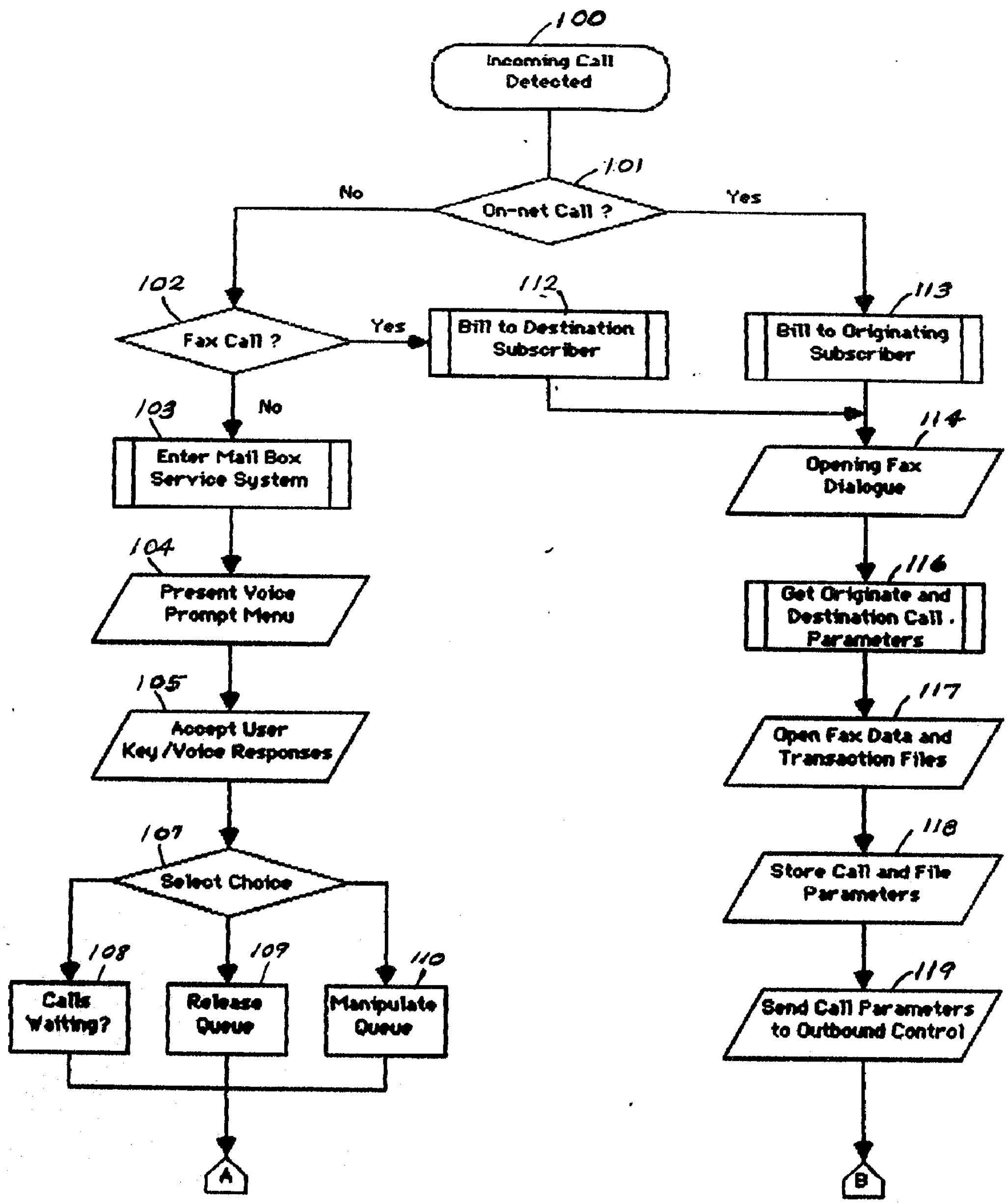

Fig. 5a

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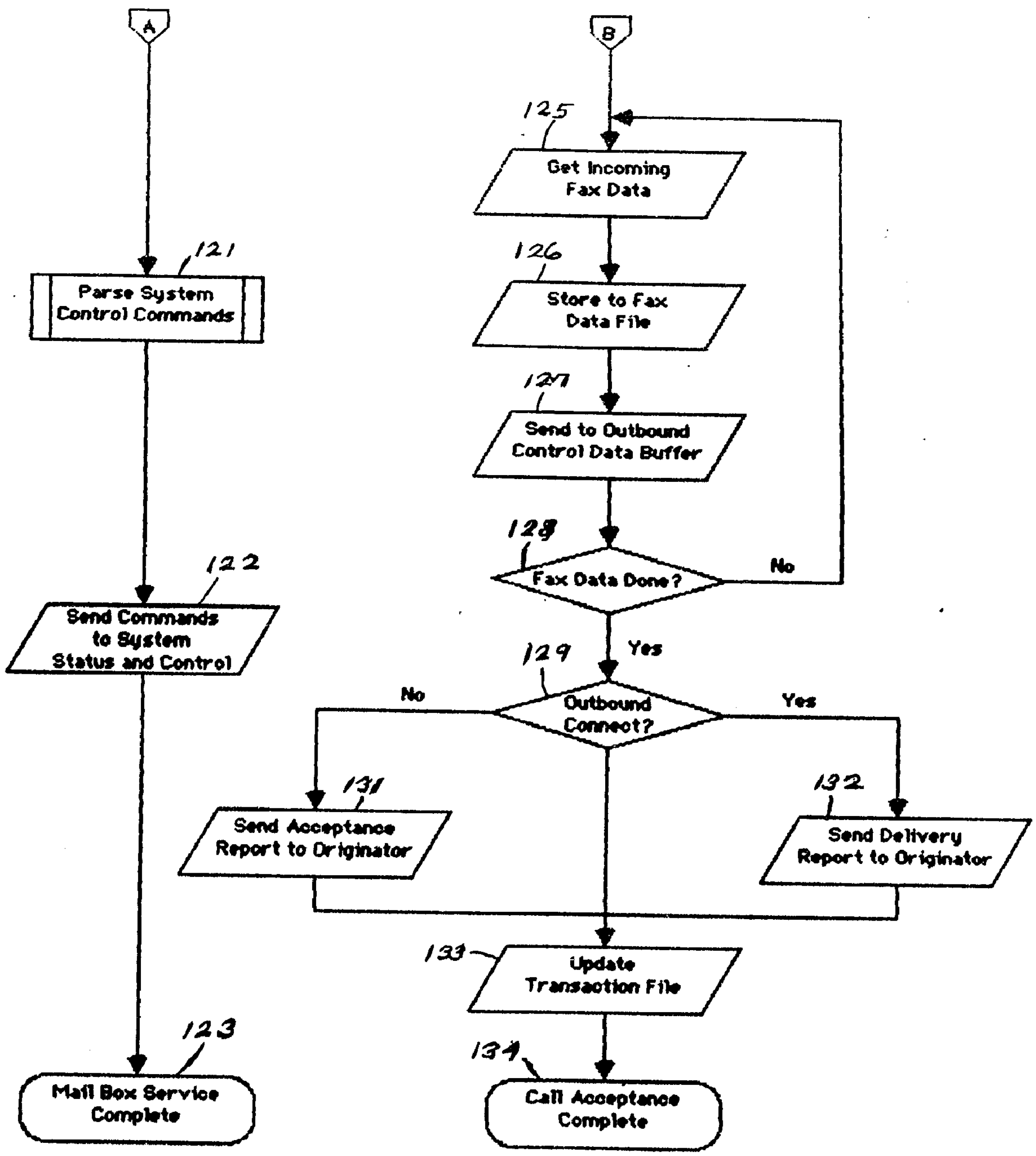

Fig. 5b

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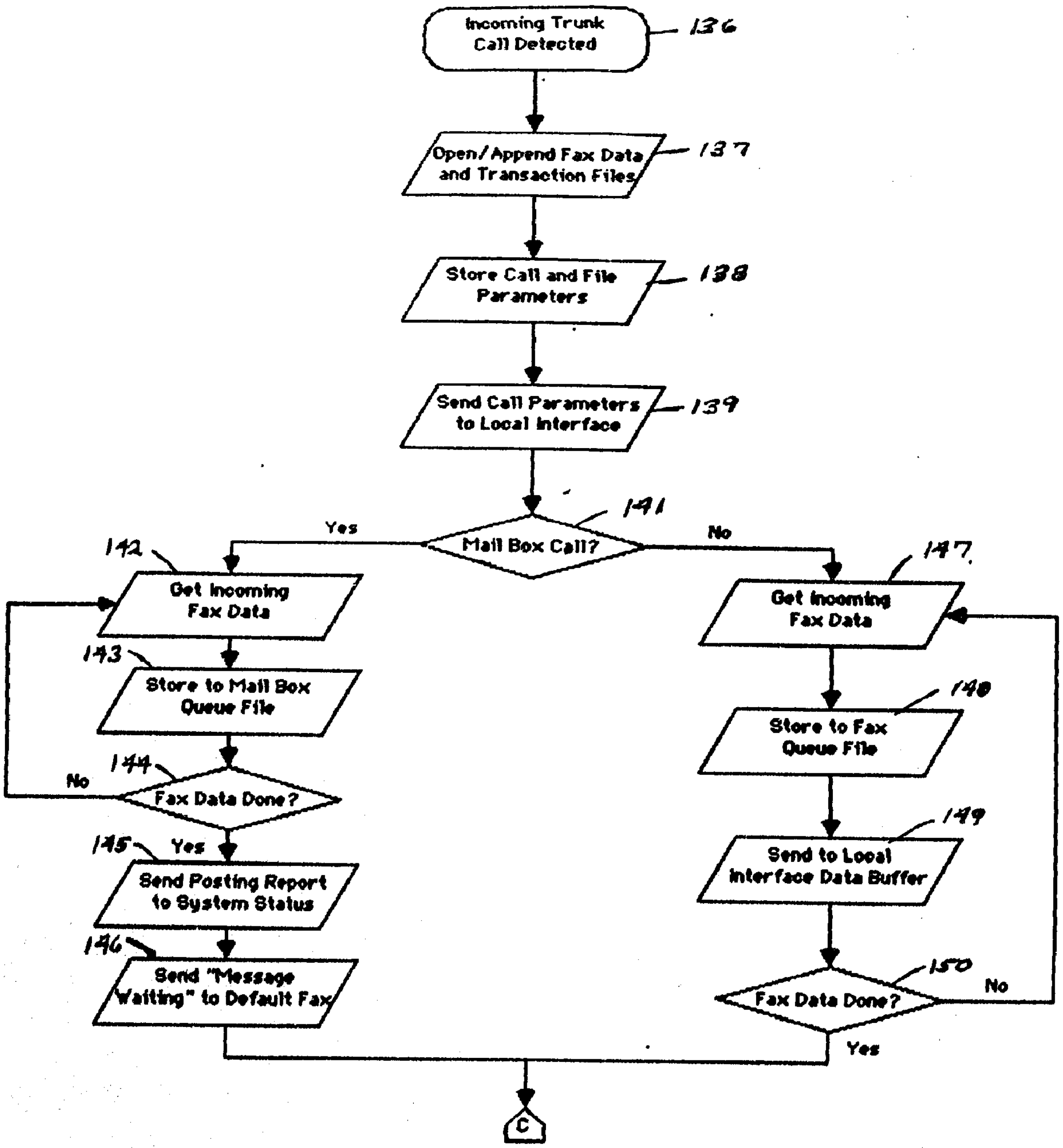

Fig. 6a

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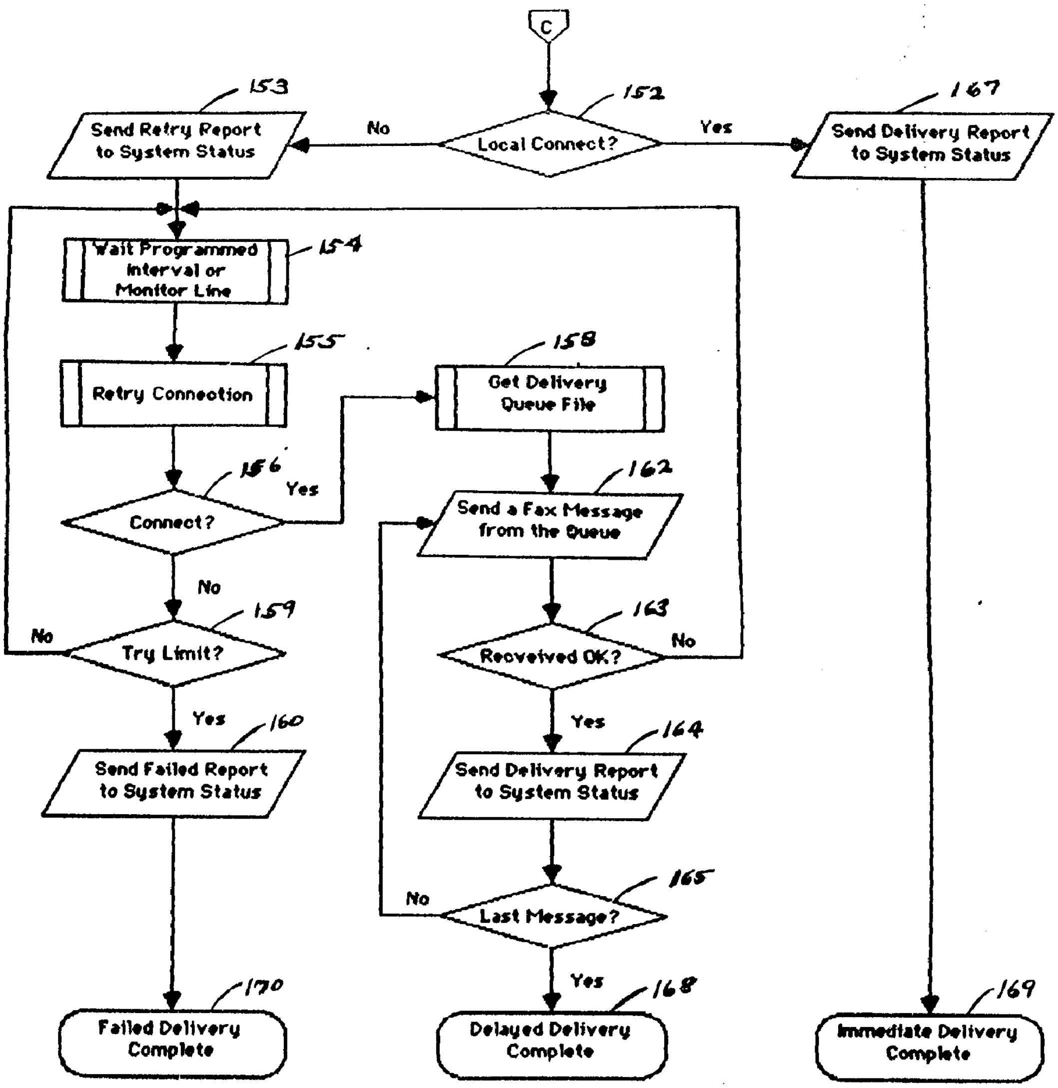

Fig. 6b

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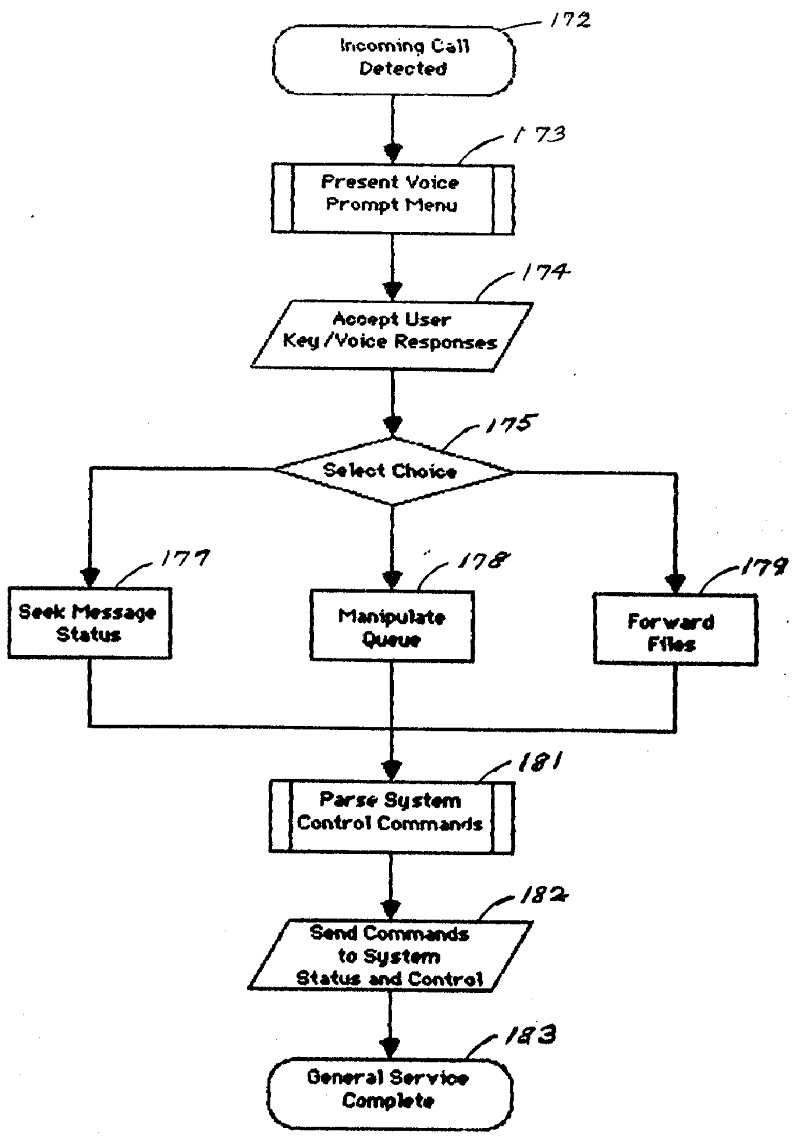

Fig. 7

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(54) INTEGRATED APPLICATION CONTROLLED CALL PROCESSING AND MESSAGING SYSTEM

INTEGRIERTES ANWENDUNGGESTEUERTES ANRUFVERARBEITUNGS- UND NACHRICHTENSYSTEM

SYSTEME INTEGRE DE MESSAGERIE ET DE TRAITEMENT D'APPEL GERE PAR APPLICATIONS

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(56) References cited:
WO-A-87/00375
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US-A- 4 759 056
US-A- 4 878 240
US-A- 4 926 462

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EP 0 572 544 B1

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Description

FIELD OF THE INVENTION

The present invention relates to the field of call processing and messaging systems. Specifically, the present invention relates to call processing and voice messaging systems for a telephone network.

BACKGROUND OF THE INVENTION

An important aspect of most any business or organization's daily operations is the ability to handle incoming telephone calls in an efficient and timely manner. One well known way to handle a multitude of incoming telephone lines is the use of a private branch exchange (PBX) or central exchange (Centrex). A PBX or Centrex is a telephone exchange system serving an organization, which may be coupled with multiple incoming and outgoing trunk lines and multiple telephone sets at the organization's premises. PBX or Centrex systems provide a variety of functions such as switching of calls from the incoming trunk lines to any of the extensions, switching calls between two extensions, and switching calls between extensions and outgoing trunk lines. Numerous PBX and Centrex systems are well known and commercially available.

A number of call processing and voice messaging systems are known in the art and are commercially available which may be coupled with a PBX or Centrex and used to automate the answering of incoming calls from the outside telephone network and the taking of messages when the extensions are not answered by the called parties. Such voice messaging systems incorporate features such as the recording of voice messages for users in what are known as user's "mailboxes". Such voice messaging systems may be accessed by users calling from PBX extensions or from the telephone network over incoming trunks.

These prior systems can be categorized into four main types of systems: 1) call processing systems sometimes called automated attendant systems for directing incoming calls to an extension, 2) call processing systems usually called voice messaging sustems for handling a call that does not complete connection to an extension, 3) two-way voice messaging systems or voice store and forward systems for speaking messages to a caller from fixed address mailboxes, and 4) interactive voice response systems for retrieving data from a database of information in response to a caller request and speaking messages to a caller.

In the first category of prior art systems, the automated attendant system answers incoming trunk calls by instructing the PBX or Centrex to direct the incoming calls to a group of extensions. Voice ports of the system are coupled with this group of extensions and appear to the PBX or Centrex simply as single line telephone sets. Typically, the automated attendant system will answer a call directed to it and provide a prerecorded voice message asking the caller to enter the extension number to which he/she desires to be connected. Depending on the specific automated attendant system, the caller may be offered the option of being transferred to a PBX or Centrex attendant. Examples of such automated attendant systems include Dytel, Inc. and a call processing and voice messaging system called Direct Access Link (D.I.A.L) manufactured by the assignee of the present invention.

In the second category of prior art systems, an important voice messaging function is included for the handling of calls which do not successfully complete connection to the originally intended extension (the extension was busy, did not answer, or had been intentionally places in a mode in which it was not accepting calls). Such a function may be accomplished in known voice messaging systems by instructing the PBX or Centrex to forward all such unanswered calls to a group of extensions coupled with the voice ports of the voice messaging system. The voice messaging system may then answer the uncompleted calls. Various methods are known in the art whereby the PBX or Centrex systems provide information to the voice messaging system regarding the identification of the caller and the called party. See U.S. Patent No. 4,926,462, Ladd et al. Depending on the specific voice messaging system, the caller may be allowed to leave a voice message or call another extension. Known prior art systems of this kind include a call processing system manufactured by Digital Sound, Inc., Octel, Inc., AT&T, Inc. and a call processing and voice messaging system called Direct Access Link (D.I.A.L) manufactured by the assignee of the present invention.

The third category of prior art systems include two-way voice messaging systems for storing and forwarding voice messages from preassigned fixed mailbox addresses between users of such systems. Each user of such a two-way voice messaging system is assigned a "voice mailbox" which he/she uses to record and send messages to other users and to listen to messages received from other users. These fixed mailbox addresses are typically configured and installed in the voice messaging system when a new "user" is added to the system. Such a prior art system is disclosed in U.S. Patent No. 4,602,129, Matthews et al.

The fourth category of prior systems include interactive voice response systems for retrieving data from a database of information in response to a caller request. The retrieved data is then converted to audible form and spoken as a message to a caller. For example, a prior art system may be used by a caller to audibly receive the balance in a bank account. These interactive voice response systems are typically limited to a specific set of applications where transactions between a caller and the system are highly predictable. Such an interactive voice response system is manufactured by Intervoice Inc. of Dallas, Texas.

Several problems have arisen with these prior art approaches for a call processing, voice messaging, or interactive voice response system. The main drawback with prior art call processing and messaging systems is a lack of flexibility, or in the case of an interactive voice response system, the inability to design applications which take advantage of the capabilities of a call processing and messaging system. Such systems are limited in their ability to be customized for the various applications in which a call processing and messaging system might be used. For example, the prior art automated attendant systems can be improved to provide a means for directing a call to different extensions depending upon many application and caller-specific conditions. Such conditions include, for example, the identity of the caller, account status of the caller, and a variety of other conditions specific to a particular application. If calls to a desired extension are not completed, stored application-related voice information such as customer-specific instructions can be spoken to the called party. Prior art systems are unsuited to the unpredictable and variant needs of these diverse call processing applications.

It is often convenient for an organization to have a single incoming telephone number used for general customer assistance or general information. It would be desirable to provide more flexibility in handling such calls. In prior art systems, incoming calls to such a general information telephone number would be routed to the fixed address associated with the telephone line which could be routed to one available (i.e. not busy) called party of a group of called parties. Thus, a fixed mailbox for each called party in the group of called parties would be used to inconveniently store messages for a general assistance line. Callers using prior art systems and the organization receiving incoming telephone calls are not able to communicate in an application-specific, individualized, and efficient manner. Such systems can be improved by allowing any message to be routed to any available mailbox. Prior art systems often require intervention by a live operator in order to properly handle incoming telephone calls. In addition, applications using prior art systems are often slaved to the capabilities provided by the system and not to the needs of the particular application.

Prior art interactive voice response systems are not flexible enough to respond in various ways to the content of information retrieved from a database in a particular application. Such systems can be improved by incorporating processing logic into the system for tailored handling of the vast number of different cases and situations presented by a database of information in many diverse applications. In addition, as situations or conditions change, prior art systems can be improved by allowing call and message processing logic modification without disrupting normal operation of the system.

It is therefore an object of the present invention to provide a call processing and messaging system that integrates PBX or Centrex call information with application-specific and database information to create a more flexible and adaptable system customized to the needs of a panicular application using the call processing and messaging system. It is a further object of the present invention to provide a call processing and messaging system that may handle calls differently depending on the source of the call. It is a further object of the present invention to provide a call processing and messaging system that does not require a fixed mailbox address to be associated with calling or called parties. It is a further object of the present invention to provide a call processing and messaging system that uses customized logic to handle incoming calls and messages. It is a further object of the present invention to provide a call processing and messaging system that allows customized call processing and messaging logic to be modified while the system is still operating.

Prior art document GB-A-2 225 916 discloses a voice messaging apparatus. In response to a received message the apparatus allocates a temporary mail box to enable any reply to a message to be left. The apparatus provides a caller with an address such as a telephone number or a password to enable the caller to access that temporary mail box at a later time. When the user of the messaging apparatus retrieves the initial message, he is alerted by the messaging apparatus to the fact that a storage location has been allocated to the caller. The user can then respond by recording a reply to the initial message in the allocated temporary mail box. The subsequent access by the original caller of that location enables the caller to hear the reply.

WO-A-87/00375 discloses that during the course of a telephonic communication, a computer formulates identification data for the caller which includes the chronological sequence of the call, the assigned designation of the call, and a set of acknowledgement digits for the call. Such data identification is registered in a specific storage section. Moreover, the acknowledgement digits may be related to the call record sequence. The computer fetches the call record sequence number, assigns a designation and encodes the sequence number as the acknowledgement digits. Then the computer queues the voice generator to provide information to the caller.

US-A-4,759,056 discloses a personal-servicing communication system which requires a portable memory device in which personal information is stored and which renders the service available to the person in accordance with the information stored on that memory device upon having read the information on the memory device. This prior art system is provided with a mail system, which is unlocked by a password. Moreover, on the portable memory device an identification number is recorded and when a calling person takes a mail read-out action through a button or dial, the identification number is read by the system and a central controller connects the calling telephone instrument to a vacant mail trunk so as to establish a connection to a mail box corresponding to the caller's identification number and the caller's identification number is transferred to a controller in a mail system.

SUMMARY OF THE PRESENT INVENTION

An integrated application controlled call processing and messaging system for improved call processing and voice messaging is disclosed. The disclosed system comprises a PBX or Centrex to which a prior art voice messaging system is coupled. An applications processor operates in conjunction with the voice messaging system and, in some cases, a host computer. Using information from the PBX or Centrex, the voice messaging system, optionally the host computer, and tables internal to the application's processor and the voice messaging system, the disclosed system integrates information from multiple sources and controls call and message processing in a seamless applications environment. The disclosed system uses voice messages as input and output permitting them to be created and controlled by applications free from the restrictions of a prior art fixed voice mailbox addressing methods.

The disclosed system includes a Call Flow programming language. The Call Flow Language is a collection of commands and statements used by the applications processor for handling an incoming telephone call. Using the call flow language, call handling capabilities allow the development and customization of call handling, message processing and voice response applications. These call and message handling capabilities include call processing, voice messaging, interactive voice response, host data base access, call routing features, and local data base access. Application call flows are derived from the call language to develop and customize call processing and messaging applications. An application flow is the programming code based on the call flow language. A flow is comprised of a series of statements that follow the call flow language syntax. The applications processor interprets and executes the flows into usable applications. A particular application flow or program flow is made up of a series of flow statements to be performed in sequential order.

A flow program consists of three sections: VARIABLES, DEFAULTS, and COMMANDS. Flow statements are written and organized following the order of these sections. VARIABLES are items used for information storage and retrieval. The current value of a variable can be changed during flow execution. The call flow language also makes direct use of some system configuration tables and application specific tables. These tables are used to acquire information needed to process a call. Dynamic variables may also be defined in a call flow statement. The DEFAULTS section of a flow program is used to specify default handlers that could occur within a flow. The COMMANDS section, which is the body of the flow program, contains the actual flow statements that control how a call is handled. Statements are executed sequentially unless a GO TO or GO SUB statement transfers control to a different place in the program, or a condition transfers execution to a default handler. The body of a flow program begins with the first executable statement and terminates whenever an end flow statement is encountered.

The present invention comprises a method for processing messages in a call processing and messaging system having the features of claim 1 and a call processing and messaging system having the features of claim 13. Other functions and features of the present invention will become apparent from the detailed description of the preferred embodiment.

BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 is a block diagram illustrating a prior art call processing and messaging system.

Figure 2 is a block diagram illustrating a call processing and messaging system as may be utilized by the preferred embodiment.

Figure 3 is a block diagram of the internal architecture of the applications processor.

Figure 4 illustrates a typical architecture of the configuration applications terminal of the present invention.

Figures 5 and 6 illustrate the linkage of tables of the applications processor.

Figure 7 is a flow chart illustrating a method of handling incoming calls as may be utilized by the preferred embodiment.

Figures 8-10e are flow charts illustrating examples of call flow applications serviced by the preferred embodiment.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

An integrated application controlled call processing and messaging system is described. In the following description, numerous specific details are set forth such as specific prompts and menus, specific codes, etc. in order to provide a thorough understanding of the present invention. It will be apparent, however, to one of ordinary skill in the art, that the present invention may be practiced without the specific details. In other instances, well known circuits, structures and techniques have not been shown in detail in order not to unnecessarily obscure the present invention.

The present invention is an integrated application controlled call processing and messaging system for controlling the processing of calls and messages received on incoming telephone lines. Although the following description makes specific reference to PBX systems, the present invention anticipates the use of the disclosed application controlled call processing and messaging system with other telephone switching systems such as a key system, central office or Centrex system, or hybrid system. Further, the methods for processing calls and messages described herein may be practiced in a system utilizing an external voice messaging system or may be equally practiced in a telephone switching system which provides the features of a voice messaging system as in integral function of the switching system. The methods for processing calls and messages described herein may be applied in a variety of systems in which the required call information may be obtained through the methods and apparatus detailed herein or through other methods and apparatus. It will be appreciated that the methods of processing calls and messages are not intended to be limited to use of the disclosed methods and apparatus for obtaining call information, except as specifically provided in the claims.

Referring to Figure 1, a prior art call processing and messaging system is illustrated. As shown, trunk lines 101 connect with PBX 100. Individual telephone lines 151 are provided for direct connection of telephone handsets. Individual telephone lines 102 connect PBX 100 with a voice messaging system (VMS) 105. One such voice messaging system is the Direct Access Link (D.I.A.L) manufactured by the assignee of the present invention. A voice message storage device 109 connects to voice messaging system 105. Control line 103 connects PBX 100 to VMS 105. Control line 103 provides VMS 105 with useable information regarding the source of the call.

Using the voice messaging system as illustrated in Figure 1, calls coming in on trunk lines 101 may be transferred to one of a plurality of telephones connected to telephone lines 151. If the telephone thus connected is busy or does not answer, the call may be forwarded to VMS 105 for further processing. Alternatively, calls coming in on trunk lines 101 may be transferred directly to the appropriate line of telephone lines 102. VMS 105 may thereafter handle the incoming telephone call using an automated attendant capability or a voice messaging or two-way voice messaging capability with a voice mailbox stored on voice message storage device 109. Storage device 109 contains a plurality of mailboxes each for storing a plurality of voice messages. In the prior art system, each mailbox stored on storage device 109 is associated with one of the extension numbers associated with telephone lines 151 or a predetermined party.

Referring now to Figure 2, the integrated application controlled call processing and messaging system of the present invention is illustrated. As shown, the preferred embodiment is coupled to components of the prior art. Specifically, PBX 100, VMS 105, and voice message storage 109 are prior art components as illustrated in Figures 1 and 2. The hardware components of the present invention include applications processor 110, an optional configuration applications terminal (CAT) 112, CAT data storage component 113, an optional host computer 106 and host data storage unit 107 as illustrated in Figure 2. In the preferred embodiment of the present invention, applications processor 110 is a circuit board that is inserted into an available slot in VMS 105. It will be apparent to those skilled in the art that the applications processor 110 need not reside on a separate circuit board. For example, an applications processor may be included as an internal component of a prior art call and message processing system. A prior art call and message processing system and associated operating system may also be modified to perform the application processing function of the present invention. The present invention is thus not limited to the specific architecture of the preferred embodiment.

Referring to Figure 3, applications processor 110 includes a microprocessor CPU 120, random access memory (RAM) 121 and a disk storage device 122 for storing local data. Applications processor 110 also includes a VMS interface 123, host interface 124 and a CAT interface 125. VMS interface 123 is a high speed internal command link permitting command signals and events and status information to be transferred between VMS 105 and applications processor 110. In the preferred embodiment, host interface 124 is an IBM 3270 SNA/SDLC compatible interface. Such an interface protocol is well known in the art. Host interface 124 is optionally used for coupling applications processor 110 to a host computer in order to access external data residing thereon. It will be apparent to those of ordinary skill in the art that alternative interface protocols can be used for interfacing with a host computer. CAT interface 125 provides means for configuring the operation of applications processor 110 and for generating and modifying call flows. In the preferred embodiment, a personal computer is used as CAT 112 to which a disk storage device 113 is coupled. In this embodiment, CAT interface 125 is an RS232 interface well known in the art. In alternative embodiments, CAT interface 125 may be implemented using other interface protocols. Similarly, alternative embodiments may use other than a personal computer for a configuration application terminal. For example, a standard ASCII data terminal may be used for this purpose. Once configured, the operation of the present invention does not require the connection of the CAT 112.

The present invention integrates the messaging capabilities of the VMS 105 with the flexible control of a call flow processing language to provide a powerful call and message handling environment. The call flow language of the present invention is a programming language for specifying an application-specific sequence of events and operations in response to an incoming telephone call or system event. The call flow language allows a flow programmer to design and build custom applications incorporating various call and messaging features. In the preferred embodiment, the applications processor 110 uses the voice message handling and storage capabilities of VMS 105 while allowing the operation of VMS 105 to be directed under control of the applications processor 110 and the call flow language commands executing therein. It will be apparent to those skilled in the art that the scope of the present invention is not limited to an architecture using the voice message handling and storage capabilities of VMS 105. Equivalent alternative embodiments may be implemented where message handling and storage is controlled directly by the applications processor.

Using the call flow language, call flows for various applications of an organization may be generated and stored for access by applications processor 110. Each flow is a script defining the operations for handling an incoming telephone call. Using this method, incoming calls and associated messages are controlled by an application flow and not by a mailbox. Moreover, more than one call flow can be used for handling a particular incoming call. Also, a particular call flow can be used simultaneously by more than one incoming call.

In operation, incoming telephone calls are received by VMS 105 on phone lines 102. The initial task of the present invention is to assign a call flow to a particular incoming call. The call is thereafter processed depending upon the command sequence coded in the particular call flow. In the preferred embodiment, incoming calls are assigned to a call flow using a set of system configuration tables residing in applications processor 110.

When an incoming call is received by VMS 105, VMS 105 can be configured to pass call control to applications processor 110 via interface 111. Using PBX integration information passed to VMS 105 via control line 103, call control information is passed to applications processor 110 by VMS 105. This call control information includes, for example, the trunk number of the incoming call, the port number, extension number or the mailbox number associated with the incoming call, and in some cases, status information indicating a reason why the call was passed (i.e. the extension was busy). The VMS 105 may also pass to applications processor 110 the called party ID and/or the calling party ID number. Once the applications processor 110 receives the call information, the call is processed using information in the system configuration tables.

Referring now to Figure 5, applications processor 110 first checks a CALLID table 600 to determine the class of service (COS) to be assigned to the call. If VMS 105 passes a called party ID to applications processor 110, applications processor 110 searches CALLID table 600 for an entry that matches the called party ID number. If the applications processor 110 references CALLID table 600 and does not find an entry corresponding to the number associated with the call, the applications processor 110 references PORTS table 601. PORTS table 601 is used to determine the class of service (COS) to be assigned to the call where a called party ID is not available.

After a COS is assigned to the call, the applications processor selects one of a plurality of schedule tables 604, each corresponding to a particular COS. The specific schedule table 604 that is selected is based on day of week and time of day when the call is received. Each schedule can have as many as eight time periods in the preferred embodiment. The applications processor 110 uses the selected schedule table number to locate the information table number to be assigned to the call. Applications processor 110 then checks the information table 606 corresponding to the information table number to determine the name of the flow to be used to process the call. The information table 606 associates an information table number with a flow name, which is then assigned to the call. If the information table 606 does not contain a flow name corresponding to the call, a user error is logged. After a flow name is assigned to the call using information table 606, the applications processor 110 checks flow table 608 to find the appropriate flow to execute. Flow table 608 associates the assigned flow name with a flow containing a set of instructions. At this point, applications processor 110 begins to execute the instructions in the assigned flow. Figure 7 illustrates the sequence of operations taken from the receipt of an incoming call to the execution of an associated call flow.

After a call flow begins to process a call, additional application-specific tables are provided for handling an incoming call. These additional tables include: a HOST table, a PROMPT table, a SCREEN table, a SESSION table, a TRANSLATION table, a LOCAL DATA table, and a SYSTEM PARAMETER table. A particular application may not require all of the configuration tables listed above. In fact, a flow may not reference a table at all. For example, if a flow does not require a host application, the flow for the application does not reference the HOST table, SCREEN table, or the SESSION table. Figure 6 illustrates the linkage between the available configuration tables and commands within flow table 608.

The HOST table 713 is used to describe how the applications processor communicates with the host computer to which it is optionally connected. There are two aspects to each HOST table entry: the physical link between the applications processor and the host, and the data link of the host connection. The physical link describes the characteristics of the connection between the host computer and the applications processor. This information consists of: host name, number of sessions, line mode, data encoding method, and connection type. Host name is the name assigned to the host connected to the applications processor. The number of sessions represents the total sessions configured in the SESSION table. Sessions are described in more detail below. Line mode specifies whether the data communications line 108 is operating in a full duplex or half duplex mode. The data encoding method refers to the way bits 0 and 1 are encoded. Connection type specifies whether the connection between the applications processor and the host is switched through a modem or non-switched through direct lines.

The second aspect of each HOST table entry contains information about the data link level of the host connection.

This portion of the HOST table entry describes the type of device being emulated by the applications processor. The data link portion of a HOST table entry describes the applications processor connection to a host computer. This portion of the HOST table entry contains the following information: station address, line configuration, and XID (exchange identification). The station address specifies origin and destination stations connected via the host computer data communications line. Line configuration specifies whether the data communications line is connected between two or more stations. For example, if one line connects two stations, the line configuration is set to a value of 2. If one line connects more than two stations, the line configuration is set to a value corresponding to a multipoint connection. XID is an acronym meaning exchange identification. XID consists of product identification and installation identification. In the preferred embodiment, the applications processor emulates an IBM 3274 cluster controller. The product identification in this case is 017.

The PROMPT table 712 contains information about voice prompts. The PROMPT table includes prompt names and up to 128 characters of prompt text. When a call flow references a prompt name, the PROMPT table is used to find the voice mailbox where the prompt is stored.

The SCREEN table 710 is used to identify each host application screen accessed by the applications processor. A host applications screen, or host screen is generated by the host computer. A host screen contains fields in which data is entered for processing or retrieving information associated with an incoming telephone call. Some of the fields on a host screen are used only by the host; others are used by both the applications processor and the host computer. Each SCREEN table entry corresponds to one host screen. The SCREEN table contains a list of field definitions for all the fields relevant to a particular application. Each screen table contains the following field definitions: field name, offset base, field offset, field length, protection level, and data type. The field name is a name assigned to a field in the host screen. This field name is used in call flows to specify the location for data entry or retrieval. Offset base is a parameter that specifies whether the field offset is absolute or relative to the cursor. Field offset is the exact location that specifies where the field starts. Field length specifies how many characters the field occupies. Protection level is a data item indicating when the data can be only retrieved (read only) or unprotected when the data can be entered retrieved or placed (read/write). The data type parameter is used to indicate whether the field is alphanumeric or numeric only.

The SESSION table 708 is used to group host sessions into pools. A session is a logical connection between the applications processor and a host computer. A session links the application flow in the applications processor to the host computer. Through a session, the applications processor accesses the host computer to get the information requested from the application call flow. Generally, a session pool contains all the sessions executing in a given application. By grouping the sessions into pools, multiple callers can simultaneously access an application while another group of callers can access a different application (on another pool). There is one SESSION table entry for each application. Each SESSION table entry contains the following: pool name, host name, session number range, logon flow name, clean-up flow name, idle flow name, and maximum idle time. Pool name is a name used by the application flow to refer to the session pool. A flow can get a session from this pool. Host name is a name given to the host. This name must be the same as one defined in the HOST table. Session number range is the number of sessions allocated for a particular pool. Logon flow name is the name of the flow to be used when the applications processor first communicates with the host after a session power-on event. Clean-up flow name is the name of the flow used to bring up the first screen of a particular application, when a flow has freed the session explicitly, has reached an ENDFLOW statement or has exited abruptly. Idle flow name is the name of the flow used to prevent the session from logging off when no host access is performed for a period of time (idle time). Maximum idle time is the time range between which the idle flow is run.

The TRANSLATION table 705 contains individual sets of translation items that are paired together to create a relationship in the form of a SEARCH-ITEM and a RETURN-ITEM. A translation refers to a single item in a TRANSLATION table, for example, a SEARCH-ITEM that translates into a RETURN-ITEM. In the TRANSLATION table, SEARCH-ITEMs are used to find a corresponding RETURN-ITEM. RETURN-ITEMs are used to provide information to the caller. This information corresponds to the RETURN-ITEM configured to match a SEARCH-ITEM. For example, if a call flow instructs the caller to enter a code (representing a SEARCH-ITEM), the applications processor would use that entry to check the list of SEARCH-ITEMS configured in the TRANSLATION table. Once the item is found in the table, the applications processor can deliver to the caller the matching item (RETURN-ITEM) in a spoken form.

The LOCAL DATA table 711 contains information about local databases. Each table entry serves as a definition of a data base, identifying the data base structure and type. A data base consists of a collection of records, each containing several fields of data. When a local data base is defined by making a LOCAL DATA table entry, the data base is assigned a structure that describes the format of the individual records in the data base. This structure consists of group of field names. Optionally, one field name in the group can be assigned as the index field. Each local data base has a data base type that determines which flow commands can be used to access records. Additionally, the data base type identifies whether flows can modify records or simply retrieve records. The present invention provides at least three different data base types: look-up only, read/write, and sequential. The look-up only data base allows call flows to retrieve information only. The read/write data base allows flows to retrieve and modify information stored in existing records. The sequential data base allows flows to sequentially retrieve and modify information stored in existing records.

A SYSTEM PARAMETER table contains values that pertain to all activities in the operation of the present invention. These activities can be categorized in five major areas: system specific, report generation, flow control, flow inputs, and flow conditions. System specific parameters include system name, system ID, and the default BAUD rate. Report generation parameters include the number of lines to be printed in each page of a report. Flow control parameters include the maximum duration of all flows. Flow input parameters set the format for inputting entries to be processes by the applications processor. Flow condition parameters specify the values for various time-out and error conditions encountered during the operation of the system.

The preferred embodiment of the present invention uses the voice message handling and storage capabilities of VMS 105 while allowing VMS 105 to be directed under flow language command control. It will be apparent to those skilled in the art that the scope of the present invention is not limited to an architecture using the voice message handling and storage capabilities of VMS 105. Equivalent alternative embodiments may be implemented where message handling and storage is controlled directly by the applications processor.

Voice messages manipulated by applications processor 110 are stored and identified using a message number. The message number for each message is retained by applications processor 110 for later access by a call flow. These message numbers are stored either in a host data base or in an applications processor local data base.

Having described the hardware and system environment of the present invention, the call flow programming language of the present invention is described in the following sections. The Call Flow Language is a collection of commands and statements used by the applications processor for handling an incoming telephone call and associated messages. Using the call flow language, call processing and messaging capabilities allow the development and customization of call processing and messaging applications. These call processing and message handling capabilities include call processing, voice messaging, interactive voice response, host data base access, call routing features, and local data base access. Application call flows are derived from the call flow programming language to develop and customize voice response applications. An application flow is the programming code based on the call flow language. A call flow is comprised of a series of statements that follow the call flow language syntax. The applications processor interprets and executes the call flows into usable applications. A particular application flow or program flow is made up of a series of flow statements to be performed in sequential order. Most statements start with a keyword that identifies the operation performed by the statement. Depending on the statement, the keyword is followed by additional information that must conform to the syntax or structure appropriate for the type of statement. All statements have a statement number associated with them. Statement numbers are generated by the system when the flow program is entered using. configuration applications terminal 112.

A flow program consists of three sections: VARIABLES, DEFAULTS, and COMMANDS. Call flow statements are written and organized following the order of these sections. VARIABLES are items used for information storage and retrieval. VARIABLES are conveniently identifiable through the use of descriptive names that are assigned in the variable's section of a call flow. The current value of a variable can be changed during call flow execution. The value of a VARIABLE can be changed during flow execution by a CAPTURE, GET, INPUT, LET, or TRANSLATE command. The call flow language of the present invention also allows numeric and character constants to be used in call flow statements. The call flow language also makes direct use of some system configuration tables and application-specific tables. These tables are used to acquire information needed to process a call.

All the variables used in a flow program are declared in the VARIABLES section. The syntax of the statements in the VARIABLES section include the type of the variable being declared, the length of the variable (optional depending on the type of the variable), and the variable name. Each variable has an associated type. The types of variables supported by the present invention include: NUM, CHAR, and PROMPT. The NUM variable type is used to declare a variable used in arithmetic operations. If a NUM variable needs a decimal point, the number of significant digits to the right of the decimal point must be included in the length portion of the statement. If the number of decimal places is not specified, the variable is considered an integer. The number of decimal places specified determines the range of values that can be stored in the number. The CHAR variable type is used to declare a variable that contains text strings. When declaring a CHAR variable, a length value is required. A length value specifies the maximum length of any string stored in that variable. The maximum length allowed for any CHAR declaration is 132.

CHAR variables may also be used to define dual tone multi-frequency (DTMF) variables. These character variables are used in the flow language to dial an extension. They are typically used with the CALL statement. DTMF characters can also be used to outpulse DTMF digits through the SPEAK statement. An example of this usage is DTMF signaling for potential networking applications. To use DTMF characters in a call flow, a variable must be declared as a CHAR variable, and then assigned a value using a command such LET or TRANSLATE.

The PROMPT variable type is used to declare variables that contain a prompt identifier. Prompts are used to urge a caller to take a specified action or make a valid menu selection. Variables of type PROMPT can only be assigned by a TRANSLATE or LET command, and are then spoken using the SPEAK command. The PROMPT declaration does not require a length specifier.

In the preferred embodiment of the present invention, variable names may be made up of letters, digits, and underscore characters. The first character of a variable name must be a letter. No blank spaces are allowed within variable names. A variable name consists of a maximum of 16 characters. An error is generated if a variable name of more than 16 characters is declared.

Dynamic variables may also be defined in a call flow statement. A dynamic variable contains dynamic run time data. The values in these variables are installation or call specific. The value of a dynamic variable can be assigned to a flow variable using the LET command. Dynamic variables are uniquely identified by a two character type prefix, a dollar sign, and the identifier which describes the variable. The five types of dynamic variables provided by the preferred embodiment include: a system variable, a call variable, a host specific variable, local data variable, and flow specific variable. Dynamic system variables are used to reference system-wide values required by a call flow. These values include the current day, current month/year, current time by hour, minutes and seconds, the current day of the week, a random number generator, and a value indicating the number of ports currently being used in the system. Dynamic call variables give specific information about the currently executing call flow. These variables include information such as the original called party ID, the original calling party ID, a caller digit entered on a menu, or the port currently being used by the current call.

Using information provided by the PBX integration line and the VMS, an incoming call can be classified into four different types: an internal direct, internal forwarded, external direct, and external forwarded call. An internal direct call is a call placed from an internal extension directly to the VMS. An internal forwarded call is a call placed from an internal extension to another extension forwarded by VMS on a busy, RNA condition, etc. An external direct call is a call from an external trunk that the PBX connected directly to the VMS. An external forwarded call is a call from an external trunk to an internal extension that forwarded to the VMS on a busy, RNA condition, etc.

A host specific dynamic variable gives specific information about the current host session assigned to the flow. These dynamic variables are used to retrieve the current host session number. Flow specific variables give information about the currently executing flow. This information includes the flow execution time in seconds and the flow status from the last statement executed. Local data base dynamic variables are used to obtain the current value of the current record number in use by a call flow.

The DEFAULTS section of a flow program is used to specify default handlers for conditions that could occur within a call flow. For example, if a caller is prompted with a menu of choices and asked to make a selection by entering a number using the DTMF keypad, the input can time out if no selection is made. If the timeout occurs several consecutive times, a timeout limit condition is set. Using the DEFAULTS section of the present invention, default handling of other system error conditions may be specified.

The COMMANDS section, which is the body of the flow program, contains the actual flow statements that control how a call is handled. Statements are executed sequentially unless a GOTO or GOSUB statement transfers control to a different place in the program, or a condition transfers execution to a default handler. The body of a flow program begins with the first executable statement. All statements in the COMMANDS section are considered part of the body of the program. Execution of a flow terminates whenever an end flow statement is encountered. The following section describes each of the commands provided by the call flow language of the present invention. The syntax of each command and specific examples of the use of each command is also provided.

1. CALL

The CALL statement places a call to the number specified. The port to be used is automatically assigned as the next idle VMS port that is configured for outcall. The call statement is valid only if the flow is not already connected to a port. CALL puts the flow on a port in connected mode.

The syntax for the call command is as follows:

CALL <charvar>

If the specified variable <charvar> contains more than 8 digits, it is assumed to be an external call. If the variable specified contains a valid VMS mailbox, the mailbox extension is called. Otherwise, the extension number contained in the variable specified is called. The extension number must contain valid DTMF characters.

An example of the CALL command follows:

```
LET AGENTEXT = '300'
CALL AGENTEXT
```

2. CAPTURE

The CAPTURE statement retrieves data from a host screen that has been previously received using the HOSTRCV command.

The syntax for the CAPTURE command is as follows:

CAPTURE <charvar> AT <field> ON <screen>.

CAPTURE retrieves the contents of a field on a host screen and then assigns it to a CHAR variable defined in the VARIABLES section. The screen refers to a screen name previously defined in the SCREEN Table. The field must be the name of a filed previously defined in that specific screen entry.

An example of the CAPTURE command follows:

CAPTURE PASSWORD AT PASSFIELD ON LOGONSCR

3. CONNECT

The CONNECT statement is used to complete a successful transfer of a call to a mailbox or extension. (See TRANSFER command) The syntax for the CONNECT command is as follows:

CONNECT

An example of the CONNECT command follows:

```
"Attempt transfer to agent
TRANSFER AGENTEXT
ON CN$FAIL GOTO NOAGENT
TRANSFER succeeded, so connect the caller to agent CONNECT
```

4. DESELECT

The DESELECT statement is used to release current access to a local database. A SELECTed database must be DESELECTed before a flow can successfully execute a GOFLOW statement. A local database is implicitly deselected whenever a flow is terminated.

The syntax for the DESELECT command is as follows:

DESELECT <database>

An example of the call command follows:

DESELECT TRAVEL

5. EDIT

The EDIT statement is used to remove characters and substrings from character strings that are stored in CHAR variables during flow execution.

The syntax for the EDIT command is as follows:

EDIT <charvar> REMOVE CHARS <string>

An example of the call command follows:

(1)

```
330:   LET BAD_CHARS = ',$'
340:   CAPTURE ACCT_BALANCE AT BALANCE ON ACCT_INFO
350:   EDIT ACCT_BALANCE REMOVE CHARS 'BAD_CHARS'
```

(2)

```
340:   CAPTURE SOC_SEC_NUM AT SSN ON ACCT_INFO
350:   EDIT SOC_SEC_NUM REMOVE STRING 'SSN' FIRST
```

6. ENDFLOW

The ENDFLOW statement completes the processing of a flow. It performs an implicit HANGUP of the VMS port associated with the flow.

The syntax for the ENDFLOW command is as follows:

ENDFLOW [CONDITION]

The keyword CONDITION can be added to an ENDFLOW statement to report the reason for a flow's termination, according to the last condition set in the flow.

An example of the ENDFLOW command follows:

ENDFLOW
ERR:
ENDFLOW CONDITION

7. ENDSUB

The ENDSUB statement terminates the subroutine and directs the flow to exit from a subroutine. Execution continues at the statement immediately following the GOSUB that called the subroutine. (Also see RETURN). It is the last statement in a subroutine. Each subroutine must have exactly on ENDSUB statement.

The syntax for the ENDSUB command is as follows:

ENDSUB

An example of the ENDSUB command follows:

ENDSUB

8. ERASE

The ERASE statement erases a previously recorded message.

The syntax for the ERASE command is as follows:

ERASE <numvar>

The message is a variable declared of type NUM:0 that has a message value set up by a previous RECORD command.

An example of the ERASE command follows:

SEND CUSTMSG TO AGTMBOX
ERASE CUSTMSG

9. FIND

The FIND statement searches for an item using a TRANSLATION Table, or searches for an item in a LOCAL DATA Table.

The syntax for the FIND command is as follows:

1) Finding items in a TRANSLATION Table

FIND <variable> USING <tablename>

Variable contains the data searched for in the TRANSLATION Table, which is specified by <tablename>. Tablename must be the name of a previously defined TRANSLATION Table. If the variable is not found in the TRANSLATION Table, the CN$FAIL condition is set.

2) Finding items in a LOCAL DATABASE Table

The FIND statement is also used to gain access to a specific record in the database. Once a flow FINDs a record, it can use GET to obtain information from the record, or use PUT to put the information into a record. The information in the record can then be updated by using WRITEDB to write a modified record back into the database.

Each of the FIND commands has a specific keyword (NEW, NTH, NEXT, FIRST), which specifies how the database is accessed to FIND the requested record. Only certain FIND statements are allowed on a specific type of database: SEQ, RW, or LU.

A) NEW

FIND NEW RECORD USING <database>

This keyword reserves an uninitialized record at the end of the current sequential database (SEQ). After a FIND NEW statement, the only allowable statements using the local database are PUT followed by a WRITEDB. A record number is assigned after the WRITEDB statement is executed. This record number is stored in the dynamic variable DB$RECORDNUM.

B) FIRST

FIND FIRST RECORD WITH <index> <op> <var> USING <database>

- <index> is an index field
- <op> can be =, <>, >, <, >=, or <=
- <var> is a specific variable to look for
- <database> is database name

This keyword searches the database specified by index, and matches the first record with the specified search condition. The database must also have an index defined.

C) NEXT

FIND NEXT RECORD WITH <index> <op> <var> USING <database>

This keyword operates similarly to FIND FIRST, except that it searches for the database starting from the current record rather than searching from the beginning of the database. The database must also have an index defined.

D) NTH

FIND NTH RECORD WITH RECORD = <integer> USING <database>

- <integer> is an integer variable or constant
- <database> is the database name

This keyword accesses a specific record specified by the record number. The record number is specified through the use of an integer variable or constant. This type of access is allowed on all types of databases.

An example of the FIND command follows:

A) Using FIND for TRANSLATION Tables

(1) FIND ERRCODE USING ERRTBL "If error code not found in table, it means no error ON CN$FAIL GOTO NOERR

(2) FIND ACCOUNT USING VALACNTS ON CN$FAIL GOTO INVALIDACNT

B) Using Find for Databases

(1) FIND FIRST RECORD WITH ZIPCODE = INPUTZIP USING LOCATE

(2) FIND NEW RECORD USING SAVEINFO

(3) FIND NEXT RECORD WITH BALANCE > 10000.00 USING ACCOUNTS

(4) FIND NTH RECORD WITH RECORD = RCNUM USING ORDERS

10. FREESESSION

The FREESESSION statement frees the host session currently held by the flow. If the SESSION TABLE has a Cleanup Flow associated with the pool to which this session belongs, then the Cleanup flow is run on this session in the BACKGROUND mode.

The syntax for the FREESESSION command is as follows:

FREESESSION [RESET]

If the RESET option is specified, the session is reset after freeing it. This option should be chosen if the session is in an unknown state and the flow program is not able to bring it to a known state. After a reset is complete, the session will go through the normal logon procedures and will eventually be brought to a known state. If the SESSION Table has a Logon flow associated with the session, it is run on the session in the BACKGROUND mode.

An example of the FREESESSION command follows:

FREESESSION
FREESESSION RESET

11. GET

This command retrieves data from an array, or from a local database record. "
The syntax for the GET command is as follows:

1) Retrieving Data from an Array
GET <variable> AT <index> USING <arrayname>
Variables declared as arrays can only be used in GET and PUT statements. The variable type must match the array type. Index must be either a NUM:0 variable or an integer constant.
2) Retrieving Data from a Local Database
GET <variable> AT <field> USING <database>
GET assigns the contents of the specified field to the flow variable using the currently selected record in the database. A database record must be selected for access using the FIND command. Otherwise, an error occurs if no record is selected when GET is executed. The flow variable and the field used in the GET statement must be of the same declared type.

An example of the GET command follows:

Retrieving Data from an Array
GET TEMPCHECK AT INDEX USING CHECKS
Retrieving Data from a Local Database
GET TELL_CALLER AT LOC_PROMPT USING LOCATE

12. GETSESSION

The GETSESSION statement allocates a host session to a flow from a specified session pool.
The syntax for the GETSESSION command is as follows:
GETSESSION <session pool>
If the host associated with the SESSION Table entry "session pool" is deactivated or down, the dynamic variable CN$HOSTDOWN is raised; if the host is up and if a session configured in the "session pool" is available, it is allocated to the flow; otherwise CN$FAIL is raised. A flow has to execute a GETSESSION statement successfully before it can use any other host communication statements like PLACE, CAPTURE etc.
An example of the GETSESSION command follows:

GETSESSION CHECKING
ON CN$HOSTDOWN GOTO HOSTDOWN
ON CN$FAIL GOTO NOSESSION

13. GODIAL

The GODIAL statement gives control of the call to VMS.
The syntax for the GODIAL command is as follows:
GODIAL <charvar>
The caller will be treated as a new caller who has entered the mailbox specified as a CHAR variable. The CHAR variable must contain valid DTMF digits. If the first digit of the CHAR variable contains a '#', the caller proceeds through the normal mailbox logon sequence.
An example of the GODIAL command follows:

(1) LET ASSISTEXTN = '333'
GODIAL ASSISTEXTN
(2) LET USERMBOX = '#5930'
GODIAL USERMBOX ON CN$FAIL GOTO INVALID

14. GOFLOW

The GOFLOW statement completes the processing of the flow that is currently being executed an begins the processing of the flow specified.

The syntax of the GOFLOW command is as follows:

GOFLOW <flowname>

An example of the GOFLOW command follows:

GOFLOW ORDERS

15. GOSUB

The GOSUB statement directs the flow to execute the subroutine specified. When the subroutine is completed, it returns and continues at the statement immediately following the GOSUB.

The syntax for the the GOSUB command is as follows:

GOSUB <name>

An example of the GOSUB command follows:

GOSUB INITIALIZE

16. GOTO

The GOTO statement transfers control of the flow program to another section of the program specified by a label.

The syntax for the GOTO command is as follows:

GOTO <label>

AN example of the GOTO command follows:

"Reprompt the caller
GOTO ASKAGAIN

17. HANGUP

The HANGUP statement hangs up the VMS port associated with a particular call, freeing it to handle other calls. The flow will continue until an ENDFLOW statement is executed.

The syntax and an example of the HANGUP command is as follows:

HANGUP

18. HOSTRCV

The HOSTRCV statement waits for a screen of data from the host computer.

The syntax for the HOSTRCV command is as follows:

HOSTRCV

An example of the HOSTRCV command follows:

HOSTRCV
ON CN$FAIL GOTO HOSTTIMEOUT

19. HOSTSND

The HOSTSND statement sends a screen of data to a host computer.

The syntax for the HOSTSND command is as follows:

HOSTSND [aid]

Aid is an attention identifier terminal key sent to the host to request an action. The specific aid to be used is the same key that the terminal operator would press to perform the function and is dependent upon the host application. ENTER is the default key that is used if one is not specified. The valid keys are:

Aid Key	Function
ENTER	Processes the screen. (Send the ENTER key).
CLEAR	Send the CLEAR key

(continued)

Aid Key	Function
ATTN	Send the ATTN key
PF1 through PF24	Programmable function keys. (application dependent)
PA1 through PA3	Programmable function keys. (application dependent)

An example of the HOSTSND command follows:

(1) HOSTSND ENTER
(2) HOSTSND PF11
HOSTRCV

20. IF

The IF statement compares two variables or compares a variable with a constant. The comparison operators allowed are:

- Equal =
- Not equal <>
- Greater than >
- Less than <
- Greater than or equal to >=
- Less than or equal to <=

The IF statement is immediately followed by a THEN GOTO label statement. The flow will transfer to the label if the comparison is accurate.

The syntax for the IF command is as follows:

If <variable> (<>, o, >, <, <=, >=) <variable>
THEN GOTO <label>

IF <variable> (=, <>, >, <, <=, >=) <constant>
THEN GOTO <label>

The variable or constants in the comparison must be of the same type. For comparisons of CHAR types, the ASCII values of the first characters that differ in the variables are used in the comparison. Also, if the current length of the variables is different, but all characters are the same up to the length of the shorter string, the shorter string is considered less than (<) the longer string. When comparing NUM variables of different significance, all significance is used.

An example of the IF command follows:

(1) IF ENTRDPSS = 'HOSTPSS '
THEN GOTO GOODPASS
(2) IF ACNTBAL <= 0
THEN GOTO OVERDRWN
(3) LET A = 3.1
LET B = 3.10001
"Following will fail, A < B succeeds
IF A = B
THEN GOTO ABEQUAL

21. INPUT

The INPUT statement accepts DTMF input from the person who is currently connected to the flow. The input can be placed in either a NUM or CHAR variable.

The syntax for the INPUT command is as follows:

INPUT <numvar> RANGE <numeric> TO <numeric>
INPUT <charvar> LENGTH <numeric> TO <numeric>

In the first example, the command reads numeric input from the DTMF keypad and places it in the NUM variable, verifying that the value is in the specified range. The second command is for inputting data into a CHAR variable, checking to assure that the length of the string entered is valid. There are system configuration parameters that can be set that affect how INPUT is handled. These include:

- Whether "#" is required to terminate input.
- Whether "*" should be used for a decimal point, a negative sign, or both on numeric input. (E.g.: "*", "0", "*", "3", "5" would be -0.35).
- The number of seconds to wait for a character to be entered before setting CN$TIMEOUT.
- The number of successive timeouts before setting CN$TMOLIMIT
- The number of successive invalid inputs before setting CN$INVLIMIT.

An example of the INPUT command follows:

(1) INPUT WRAPUP RANGE 0 TO 99999
(2) INPUT VISANUM LENGTH 16 TO 16

In example two, VISANUM is declared as a CHAR variable because it is too large to store as a NUM variable.

22. LET

The LET statement assigns a new value to a variable. The value assigned can be:

- A constant.
- The result of an arithmetic operation involving either two variables or a constant and a variable.
- A variable.
- A run-time variable.
- A prompt.

The syntax for the LET command is as follows:
LET assigns a value to either a CHAR variable, a NUM variable, or a PROMPT variable.

(1) LET <numvar> = <numvar> {+,-,*,/} <numeric>
NUM variables can be assigned a numeric constant, or be the result of an arithmetic operation involving variables or a variable and a constant.
(2) LET <numvar> = <numvar> MOD numeric>
NUM:0 variables are divided to create a remainder.
(3) LET <numvar> = -<numeric>
The minus sign (-) can also be used to negate a value.
(4) LET <charvar> = <charvar> CONCAT {charvar, character}
(5) LET <charvar> = {LEFTJUST, RIGHTJUST, UPPER, LOWER} <charvar>
String operators (CONCAT, LEFTJUST, RIGHTJUST, UPPER, LOWER) are valid only on CHAR variables. The CONCAT operator works on two CHAR variables, or a CHAR variable and a constant. The other four operators take a single CHAR variable as the operand.
(6) LET <charvar> = SUBSTR <charvar, numeric, numeric> The SUBSTR string operator extracts a string from a CHAR variable, where the string is specified by a numeric offset and length respectively. The offset starts at 1 for the first character in the source string.
(7) LET <variable> = {LENGTH, MAKENU;M, MAKECHAR} <variable>
This construct is used for mixed mode operations, where an operation is performed on a variable of one type and the result is of a different type. MAKENUM and LENGTH take a CHAR operand with the result being a NUM, while MAKECHAR takes a NUM operand and generates a CHAR result. A blank or empty CHAR operand has a value of 0. IN the following example, the value 0 is stored in NUM_RESULT:

```
LET EMPTY_STRING = "
LET NUM_RESULT = MAKENUM
```

EMPTY_STRING

(8) LET <prompt> = <prompmame>
This form assigns a prompt identifier to a PROMPT variable. This may be required when it is necessary to speak a series of prompts that have been determined by caller input.
(9) LET <variable> = <XX$variable> [for tablename]
Assigns the value of a dynamic run-time variable to a flow variable. The XX can either be SY, CL, HT, FL, or DB. When using DB, the FOR portion must be included to identify the local data table, with tablename being an entry in the Configuration Tables.

An example of the LET command follows:
Reassigning values to variables

(1) LET ERRFIELD = 'ERROR'
(2) LET VALUE = 99.999
(3) LET ACCOUNT = MAKENUM BALANCE
Assigning the result of an arithmetic operation
(4) LET I = I + 1
(5) LET INTEREST = ACCOUNT * .065
Assigning a remainder
(6) LET STOCK_LEFT = TOTAL_SHARES MOD SHARES_PER_BLOCK
Assigning a negative value
(7) LET NEGBAL = -BAL
Appending a string of characters
(8) LET FULLNAME = FIRST CONCAT LAST
Converting a character value to a numeric value
(9) LET SELECTION = MAKENUM ITEMNUM
Extracting a substring from a string
(10) LET MIDDLE = SUBSTR (LONGSTRING, 10, 10)
Assigning a prompt name to a prompt variable
(11) LET GREET = OFFHOURS
Assigning a dynamic run-time variable to a numeric variable
(12) LET SYSDAY = SY$DAY

23. MENU

The MENU statement branches to a choice of labels based on the caller's single-digit input.
The syntax for the MENU command is as follows:

MENU
ON [X]...
ON [Y]...
ON [Z]...
GOTO...

The MENU statement includes all of the following ON statements as part of the MENU. If there is no corresponding ON statements for the character input, or the user entered invalid input, the flow continues sequentially with the following statement, which is the GOTO TRYAGAIN in the example below.
An example of the MENU command follows:

MENU
ON CN$TIMEOUT GOTO REPROMPT
ON [0] GOTO LAB0
ON [1] GOTO LAB1
ON [*] GOTO LABSTAR
ON [#] GOTO LABPND
GOTO TRYAGAIN

24. ON

The ON statement handles conditions that can occur during execution of a flow. It is also used to check for a specific DTMF digit as part of the MENU command.

The syntax for the ON command is as follows:

(1) ON [{0,1,2,3,4,5,6,7,8,9,*,#,A,B,C,D}]GOTO<label>
(2) ON[{0,1,2,3,4,5,6,7,8,9,*,#,A,B,C,D}]GOSUB <subroutine>
(3) ON CN$condition GOSUB <subroutine>

Syntax statements 1 and 2 show the syntax after a MENU command. The characters A, B, C, D refer to DTMF tones, and not to keys on the phone. Brackets [] are required to surround the menu choices in this usage of the ON command. Syntax statements 3 and 4 show the usage of ON to handle a condition. The syntax in statement 4 is used only in the COMMANDS section of flow; it cannot be used in the DEFAULTS section.

An example of the ON command follows:

(1) ON [1] GOTO FIRSTCHOICE
(2) ON CN$FAIL GOTO ASSIST

25. PLACE

The PLACE statement places the contents of a variable onto a host screen.

The syntax for the PLACE command is as follows:

PLACE <charvar> AT <field> ON <screen>

PLACE puts the value of a CHAR variable or constant onto a screen field. The screen and the field in which to place the variable are identified by their names as specified in the SCREEN Table.

An example of the PLACE command follows:

PLACE ITEMNUM AT ITEMPOS ON SCREEN 1

26. PUT

This command places data in arrays or into a local database record.

The syntax for the PUT command is as follows:

PUT <value> AT <index> USING <arrayname>

Values declared as arrays are only used in the PUT and GET flow statements. The value must have the same data type as the array.

Value can be:

. A numeric constant.
. A character constant.
. A NUM variable name.
. A CHAR variable name.

The index value must be:

. A NUM:0 variable name.
. An integer constant.

The index value can be a variable or an integer constant. In either case, the index must be a positive non-zero value that is less than or equal to the maximum number of elements declared for the array. If <index> is a variable, it must be declared as a NUM:0.

Placing data into a local database

PUT <variable> AT <field> USING <database>

The flow variable and the field used in the PUT statement must be of the same declared type. PUT places the contents of the flow variable into the specified field using the currently selected record of the database. A database record is selected for access using the FIND command. An error occurs if no record is selected when PUT is executed. PUT places the variable into the database record, but the database is not updated until the record is written onto disk using the WRITEDB command. A PUT operation to an index field is not allowed.

An example of the PUT command follows:

Placing data in arrays
PUT TEMPCHECK AT INDEX USING CHECKS

Placing data in local database
FIND NEW RECORD USING SAVEINFO
ON CN$FAIL GOTO FULLDATABASE
PUT REQUEST AT REQ_ITEM USING
SAVEINFO
PUT MSG_NUMBER AT SAVE MSG ;USING
SAVEINFO
WRITEDB RECNUM SAVEINFO

27. RECORD

Records a new message or records a message at the end of an existing message. Flow execution continues when the recording terminates.

The syntax for the RECORD command is as follows:

Recording a new message

RECORD <numvar> NEW

Where <numvar> is the message ID and is a variable of type NUM:0. The <message id> is returned for subsequent use, after the statement is executed. However, subsequent RECORD NEW statements are illegal until the message is explicitly saved or erased using SAVE or ERASE. No other prompt can access the message until a SAVE statement is executed. If a new message is outstanding at ENDFLOW, it is automatically erased, and an error is logged.

Appending a message

RECORD <numvar> APPEND

Where <numvar> is the message ID of type NUM:0 and must have been previously used in a RECORD NEW statement. This message identifier can be used in ERASE, SPEAK, SEND, and SAVE statements. The message is recorded at the end of the existing message specified by the message ID. All messaging access by other channels to this message is locked out while RECORD...APPEND is in progress.

An example of the RECORD command follows:

RECORD CUSTMSG NEW
ON CN$FAIL GOTO NOMSG
RECORD CUSTMSG APPEND

28. RETURN

The RETURN statement directs the flow to exit from the subroutine and to continue execution at the statement immediately following the GOSUB that called the subroutine. An ENDSUB statement does an implicit RETURN.

The syntax for the RETURN command is as follows:

RETURN

An example of the RETURN command follows:

RETURN

29. SAVE

The SAVE statement allows a message created with RECORD NEW to be stored internally for access by a subsequent flow. After a RECORD NEW statement, the message created is considered "open." The message must be saved or erased (by using the SAVE or ERASE command) before the end of the flow. Otherwise, the message is automatically erased and a User Error is logged. The SAVE statement is valid only once for the message created by the RECORD NEW statement within the flow. In order for other flows to access a message, the following two steps are necessary:

(1) PLACE <numvar> on the host screen and send it to the host. or WRITEDB the <msgid> to the local database.
(2) SAVE <numvar> only after successful HOSTSND or WRITEDB.

The syntax for the SAVE command is as follows:

SAVE <numvar>

Where <numvar> is the message ID of the variable type NUM:0 that must have been previously used in a RECORD statement. This message identifier can be used in ERASE, SPEAK, SEND, and SAVE statements.

An example of the SAVE command follows:

SAVE CUSTMSG

30. SELECT

The SELECT statement is used to gain access into a local database. A database must be SELECTed before an FIND, GET, or PUT commands are introduced in the flow program.

The syntax for the SELECT command is as follows:

SELECT <database>

Where <database> must be the name of a database defined in the Local Data Table

An example of the SELECT command follows:

SELECT TRAVEL
ON CN$FAIL GOTO RECORDS_NT_AVAIL

31. SEND

This command sends a message to a destination mailbox. The message is marked as having been sent by the creator mailbox, allowing a reply or error message to automatically return to the creator as specified. If no creator is specified, the creator defaults to the error mailbox configured on VMS. The destination mailbox can be a network address. Unless explicitly ERASEd, the message remains available after SEND is executed.

The syntax for the SEND command is as follows:

SEND <numvar> TO <mbox> [FROM creator]

Where <numvar> is the message ID and is a variable declared NUM:0 that has a message identifier set up by a previous RECORD command. The destination mailbox for the message must be specified as a CHAR variable. If the optional creator mailbox is specified, a reply or failure is directed to the creator mailbox. Otherwise, the system error mailbox specified on VMS is used as the default creator. The creator mailbox must be specified as a CHAR variable.

An example of the SEND command follows:

(1) RECORD CUSTMSG NEW TRANSLATE AGENTCODE TO AGENTMBOX
USING AGTABLE
SEND CUSTMSG TO AGENTMBOX
ERASE CUSTMSG
(2) SEND INQUIRE TO SVCMBOX FROM GUESTBOX

31. SPEAK

The SPEAK statement relays information to the person connected to the flow.

The syntax for the SPEAK command is as follows:

(1) SPEAK <variable> [AS <speaktype>]
(2) SPEAK <promptname> [AS PROMPT]

SPEAK is used to output a variable or a prompt to the caller or agent. A variable may be spoken in a variety of speaktypes which are described below. If no speaktype qualifier is specified, a default is chosen based upon the type of the variable.

Speaktype	Definition
MONEY	The value 100.35 would be spoken as *"One hundred dollars and thirty-five cents."* Valid only on NUM variables.
DIGITS	All alphanumeric characters (A-Z and 0-9) are spoken as individual characters. The value 100.35 would be spoken as *"One zero zero point three five."* The value 'HELLO' would be spelled out as 'H-E-L-L-O'. DIGITS is the default speak type for CHAR variables. Valid for all NUM and CHAR variables.

NUMERIC	The value 100.35 is spoken as *"One hundred point three five."* This is the default for NUM variables and is only valid for NUM variables.
MONTH	The variable must be a NUM:0 type and have a value in the range 1012. For example, the value 3 would be spoken as "March."
DAY	The value must be a NUM:0 type and have a value in the range 1-7. For example, the value 3 would be spoken as "Tuesday."
PROMPT	The variable contains a prompt identifier previously assigned via a LET or TRANSLATE statement. This is the default value and the only allowable speak type for PROMPT variables.
MESSAGE	The variable contains a message identifier previously set using RECORD. Only valid for NUM:0 variables.
DTMF	The value is outpulsed as a DTMF string. The variable must a CHAR type and contain valid DTMF characters.

An example of the SPEAK command follows:

(1) SPEAK BALANCE AS MONEY
(2) NUM:5 VALUE
•
•
"Default speaktype is numeric when not specified SPEAK VALUE
(3) SPEAK AGNTMSG AS MESSAGE

32. STARTSUB

STARTSUB is the first statement in a subroutine. It identifies the subroutine's name. Each subroutine must have one STARTSUB statement.

The syntax for the STARTSUB command is as follows:

STARTSUB <name>

Name identifies the subroutine's name.

An example of the SUBROUTINE follows:

STARTSUB INITIALIZE

33. TRANSFER

The TRANSFER statement puts the current caller on hold and calls the mailbox or extension specified. Call progress is monitored. If TRANSFER succeeds, the CONNECT statement is required to complete the transfer. Either the standard or alternate transfer codes are selected, based on the INFORMATION Table used. The INFORMATION Table corresponds to the mailbox or to the port if the variable specified does not contain a valid mailbox.

The syntax for the TRANSFER command is as follows:

TRANSFER <charvar>

If the variable specified contains a valid VMS mailbox, the mailbox extension is used. Otherwise, the extension number contained in the variable specified is used. The extension number must contain valid DTMF characters.

An example of the TRANSFER command follows:

LET ASSISTEXT = '300'
TRANSFER ASSISTEXT

34. TRANSLATE

The TRANSLATE statement is used to convert one data value to another using a TRANSLATION Table. A previously defined TRANSLATION Table is searched to see if it contains the particular statement's value in a variable. If so, the table entry for that variable is assigned to a new variable.

The syntax for the TRANSLATE command is as follows:

TRANSLATE <search_var> TO <retum_var> USING
<tablename>

In the above statement, the first variable contains the data that will be translated to a new value and placed in the second variable based upon the contents of the table specified by the tablename. The search and return variables must match their corresponding types in the specified table. Table name must be previously defined in the TRANSLATION Table using the configuration applications terminal software.

An example of the TRANSLATE command follows:

TRANSLATE ITEMNUM TO ITEMPRMP USING
ITEMTBL

35. WAIT

The WAIT statement delays execution of the flow program until the number of specified second has elapsed. The maximum time is 65535 seconds.

The syntax for the WAIT command is as follows:

WAIT <numeric>

Numeric is either a NUM:0 variable or a constant that specifies the number of seconds to delay before reactivating the flow program.

An example of the WAIT command follows:

```
LOOP:  TRANSFER AGEXT
       ON CN$FAIL GOTO DELAY
       CONNECT
       GOTO END
DELAY: SPEAK PLSHOLD PROMPT
       WAIT 30
       GOTO LOOP
```

36. WRITEDB

The WRITEDB statement is used to output the currently selected record to the local database file and then returns the written record number.

The syntax for the WRITEDB command is as follows:

<WRITEDB <numvar> <database>

After a record is output to the file, the flow no longer has a selected record for the database and must do another FIND before any PUT or GET commands can be executed. The record number for the record just written is returned in an integer variable <numvar> specified in the WRITEDB command. This variable must be a NUM:0 variable.

An example of the WRITEDB command follows:

```
PUT REQUEST AT REQ_ITEM USING SAVEINFO
PUT MSG_NUMBER AT SAVE_MSG USING SAVEINFO
WRITEDB RECNUM SAVEINFO
ON CN$FAIL GOTO SORRY
```

The Configuration Applications Terminal (CAT) 112 is a computer used to configure the operation of the applications processor 110 and its associated call flows. In general, the CAT 112 has several functions: 1) to perform terminal emulation for communication with the applications processor 110, 2) to backup and restore system configuration tables and application-specific tables, 3) to backup and restore local databases, 4) to to transfer a call flow program for off-line editing, and 5) to update applications processor system software. A typical architecture of the configuration applications terminal 112 is shown in Figure 4.

Terminal emulation allows the CAT to communicate with applications processor 110 directly through a cabled connection or remotely through a modem. Characters typed on the CAT keyboard are sent to applications processor 110, and information from applications processor 110 is displayed on the CAT monitor. The only time the CAT is not performing terminal emulation is when menus are displayed. At this time, communication with applications processor 110 is temporarily suspended because the menus control the terminal.

The CAT uses a packet-based, error-correcting data transfer protocol at speeds up to 19200 baud to pass information between applications processor 110 and the CAT. Data transfer is initiated by menu selection and is performed by software residing in the CAT. This CAT software includes a Backup function, a Restore function and an Update function. The Backup function transfers data from applications processor 110 to the CAT hard disk drive 113. The Restore and Update options transfer data from the CAT hard disk drive 113 to Applications processor 110.

The CAT software also includes utilities to manage individual applications processor system configurations. System configurations, including the content of system configuration tables and application-specific tables, can be added, deleted, and modified. Backed-up system configurations can be archived to and recovered from a floppy diskette.

The CAT software also includes utilities to manage log files, which capture communication sessions from applications processor 110. Multiple log files can be stored, displayed printed, and deleted for each applications processor 110 system configuration defined within the CAT software.

OPERATION OF THE PRESENT INVENTION

In order to illustrate the hardware and the call flow language processing logic of the present invention, three examples of the operation of the preferred embodiment are described herein and illustrated in Figures 8, 9a-9f and 10a-10e. Referring to Figure 8, a sample application processing flow chart is illustrated. The application processing flow illustrated in Figure 8 represents a flow program activated and executed by applications processor 110 in response to an incoming telephone call received from VMS 105 in the manner described above. Once the application flow is initiated, processing flow begins at bubble 901. Initially, an announcement message is audibly transmitted to the caller using the SPEAK command of the call flow language (processing box 902). Hardware means for recording and playing back audible speech is an apparatus well known in the art. The announcement message spoken or transmitted to the caller in processing box 902 includes a menu of options provided for the caller and a prompt for the operator to enter one of the options provided. Options are entered by a caller using the numeric keypad buttons on the telephone handset. These buttons generate dual tone multi-frequency (DTMF) signals which can be used to distinguished the various options entered by a caller. The technique of using DTMF signals is well known in the art. Once the caller is prompted by a message in processing box 902, an INPUT command is executed by applications processor 110 in order to retrieve the caller's selection (processing box 903). If the caller fails to enter a selection within a predetermined timeout period, execution control returns to processing box 902 where the menu and prompt message is again transmitted. If the caller enters a 1 key on the telephone handset (decision box 904), processing path 905 is taken to processing box 907. Since the entry of a 1 key is associated with a request for interest rates in the example of Figure 8, another SPEAK command is executed to announce the current interest rates as retrieved from an internal database using the SELECT, FIND and GET commands. Once the SPEAK command is executed in processing box 907, processing flow continues at the bubble labeled A where the announcement message, the menu and prompt are again spoken in processing box 902.

If the caller enters key 2 (decision box 908), the example of Figure 8 processes a request for an account balance. First, the caller is prompted to enter his account number (processing box 911) the caller enters the account number using the telephone keypad. An INPUT command is executed to retrieve the account number entered by the caller (processing box 912). Once an account number is received from the caller, the account number is validated in decision box 913. The caller-entered account number may be invalid for several reasons. First, an inappropriate quantity of numbers may have been entered. Secondly, the entered account number is used with a FIND command in order to access a database containing valid account numbers. If the caller-entered account number is not found or is inaccessible to the caller, the account number is rendered invalid. For an invalid account number (processing path 915), a message is spoken to the caller stating that the account is invalid and prompting the caller to enter a selection corresponding to a subsequent action to be taken (processing box 917). An INPUT command is then executed to retrieve the selection made by the caller for an invalid account number (processing box 918). If the user enters a 3 key (decision box 919), a termination message is spoken to the caller (decision box 922), and the call flow processing logic of Figure 8 is terminated at termination box 924. If the user does not enter a key 3 at decision box 919, the call is transferred to a live personal representative at processing box 923 where the call is handled manually and without further control by the call flow logic of Figure 8. The call is transferred to live assistance in processing box 923 by execution of the TRANSFER command and the CONNECT command.

If at decision box 913 a valid account number is entered (processing path 914), another SPEAK command is executed to audibly convey to the caller the current balance of the account number entered earlier (processing box 916). The account balance is retrieved from an account database using the SELECT, FIND, and GET commands. Once the account balance is spoken to the caller at processing box 916, processing loops back to processing box 902 through the bubble labeled A in Figure 8.

Referring back to decision box 908, if the number 1 key or the number 2 key are not depressed by the caller in response to the prompt in processing box 902, processing path 910 is taken to decision box 925. If the caller enters the 3 key (processing path 926), processing control is transferred to the bubble labeled B where the call is terminated with a termination message (processing box 922). If the 3 key is not depressed or a timeout occurs, processing again transfers to the bubble labeled A as shown in Figure 8. Thus, a means for handling an incoming call using the applications control hardware and call flow language processing logic of the present invention is illustrated in this first example. A script for handling an incoming call as in this first example is presented in LISTING A appearing just before the claims section of this patent application.

A second example of the operation of the present invention is illustrated in Figures 9a through 9f. Figures 9a through 9f illustrate the call flow processing logic for handling service calls made to a help desk. In this application, callers with problems or questions call a general help number in order to resolve their questions. The present invention serves to facilitate fast and effective communication between callers with problems and the specialists who can help them. In addition, the present invention automatically manages and tracks information about the problems presented by the callers and the results or answers provided to them by organization specialists. In a help desk application, prior art systems are unable to effectively handle the wide array of caller problems and questions placed to a single help line telephone number. With the present invention, however, the help desk application is efficiently automated as illustrated by the logic in Figures 9a through 9f.

Referring now to Figure 9a, the call flow for the help desk application of the second example is illustrated. After the call flow is initiated in the matter described earlier, processing begins at bubble 1001. Using integration information received from PBX 100 and VMS 105, applications processor 110 determines that the called ID is that of the help desk (processing box 1002). Using the SPEAK command, an initial announcement message, a menu of options, and a prompt message is spoken to the caller (processing box 1003). The caller selection is then received (processing box 1004) using the INPUT command. Depending upon the key entered by the caller, one of several possible processing paths are taken. For example, for the entry of key 1 on the telephone keypad, processing path 1007 is taken to processing box 1008. In this second example, key 1 is associated with a request by the caller to record an inquiry for the help desk. If the caller enters key 2, processing path 1066 is taken to the bubble labeled P illustrated in Figure 9f. in this second example, key 2 is associated with a request by the caller to update an inquiry already existing from a previous transaction with the help desk. If the caller enters key 3 on the telephone keypad, processing path 1069 is taken to the bubble labeled N illustrated in Figure 9e. In this second example, key 3 is associated with a request by the caller to check on a response to a previous inquiry made to the help desk. If the zero key is entered by the caller, processing path 1073 is taken to processing box 1074 where the call is transferred to a live representative for further manual call processing. If a key other than keys 0 through 3 is entered or if a timeout occurs, processing path 1072 is taken to the bubble labeled K illustrated in Figure 9a where the initial announcement message, menu and prompt messages are repeated (processing box 1003).

Starting at processing box 1008 shown in Figure 9a, call processing in response to the operator selection of key 1 is illustrated. Processing started at processing box 1008 handles a caller request for recording a new inquiry for the help desk. In order to accomplish this task, an inquiry database is accessed using the SELECT command. A new record is opened in the inquiry database using the FIND NEW construct of the FIND command. A PUT command followed by a WRITEDB command is used to initialize the new record and to produce a record number associated with the new record (processing box 1008). Once a new inquiry record is initialized in processing box 1008, the SPEAK command is used to prompt the caller to enter a seven digit telephone number (processing box 1016). The INPUT command is then used to retrieve the telephone number entered by the caller (processing box 1017). At processing box 1018, a new voice message is created for recording an inquiry by the caller. The RECORD NEW command construct of the call flow language can be used for this purpose. Information such as time, date, caller telephone number, etc. can be integrated or recorded into the new message using the RECORD command. The RECORD command returns a message number associated with the new message for use in later call flow processing. Processing then continues at the bubble labeled L in Figure 9b.

Referring now to Figure 9b, call flow processing for a help desk application of the second example continues. Having created a new message for recording the caller's inquiry, the SPEAK command is used to prompt the caller to speak his/her name for recording into the new message (processing box 1019). The name of the caller is recorded into the new message at processing box 1020. Next, the SPEAK command is again used to prompt the operator to enter his/her inquiry at the tone and to terminate his/her inquiry by pressing the pound (#) key (processing box 1022). The RECORD command is again used to receive a voice message from the caller, the content of which represents his/her inquiry to the help desk (processing box 1023). After receiving the caller's name and inquiry, a technician database is accessed using the SELECT and FIND commands. A new record in the technician database is opened using the FIND NEW command construct. Once the new record in the technician database is created, the message number associated with the message used to receive the caller's name and inquiry is posted to the new record in the technician database using the PUT command. A unique inquiry reference number associated with the caller's inquiry is generated and put into the technician database along with the associated message number (processing box 1021). Using the RECORD command, a new message is generated which integrates the inquiry reference number into a termination message to the caller (processing box 1024). Call flow processing for the new inquiry then terminates at termination box 1025.

Referring again to Figure 9a, an inquiry update request is specified if a caller depresses key 2 of the telephone keypad (processing path 1066). In this case, call flow processing passes to the bubble labeled P as illustrated in Figure 9f. Referring now to Figure 9f, call flow processing for updating an existing inquiry is illustrated. First, the SPEAK command is used to prompt the caller to enter the existing inquiry number (processing box 1057). The INPUT command is used to retrieve the inquiry number entered by the caller (processing box 1058). Next, the inquiry database is accessed using the SELECT command and the FIND command. The inquiry message number associated with the inquiry reference number entered by the caller is retrieved (processing box 1061). The SPEAK is then used to prompt the caller to enter the inquiry update (processing box 1060). The RECORD command is used to record the update to the inquiry (processing box 1059). The RECORD APPEND command construct can be used to append a new message onto an existing message. In addition, data such as time, date, and caller ID can be integrated into the message. Once the caller completes the inquiry update, flow processing continues at bubble K illustrated in Figure 9a. At bubble K, the initial announcement message, menu of selections, and prompt to enter a selection is again spoken to the caller.

Referring again to Figure 9a, if key 3 on the telephone keypad is depressed by a caller, processing path 1069 is taken to the bubble labeled N illustrated in Figure 9e. Referring now to Figure 9e, processing in response to an operator request to check on a response to a previous inquiry is illustrated. First, the SPEAK command is used to prompt the caller to enter the inquiry number (processing box 1051). Next, the INPUT command is used to retrieve the inquiry number entered by the caller (processing box 1052). The inquiry database is then accessed. Using the inquiry number entered by the caller, the inquiry record associated with the inquiry is retrieved. Similarly, the message number associated with the original inquiry message is retrieved (processing box 1053). Using the message number of the original inquiry, the original inquiry message is spoken to the caller using the SPEAK command (processing box 1054). Next, the message number of the technician's response to the original inquiry message is retrieved from the inquiry record in the inquiry database (processing box 1055). The technician's response is spoken to the caller using the SPEAK command (processing box 1056). Current information such as date and time can also be integrated into the message. Call flow processing then is transferred back to the bubble labeled K illustrated in Figure 9a where the initial announcement message, menu of command selections, and prompt is spoken to the caller. Thus, call flow processing in a help line application for a call received by a caller making an inquiry is described.

A related call flow for the help line application is illustrated in Figures 9c and 9d. Figures 9c and 9d illustrate call flow processing for a call received by a technician placing an incoming call on a line dedicated to technicians or specialists servicing callers making inquiries to the help line. This call flow is initiated in a matter similar to that described earlier.

Referring now to Figure 9c, a call flow is illustrated for an incoming call received by the call processing and messaging system from a technician or specialist servicing inquiries to a help line. When the flow is initiated, processing starts at the bubble labeled 1030. Using information received from the PBX and the VMS, the applications processor identifies the caller ID as a technician (processing box 1031). When this occurs, the inquiry database is accessed using the SELECT command the FIND command. The inquiry database is searched for new inquiries. If a new inquiry is found, the message number associated with the new inquiry is retrieved from the database (processing box 1032). Using the SPEAK command, the inquiry found in the inquiry database is spoken to the technician (processing box 1033). In addition, the name of the caller making the inquiry, the time of day and the date is incorporated into the message spoken to the technician. Having spoken the inquiry to the technician, the technician is prompted to enter a selection corresponding to the desired action to be subsequently taken. In the example presented in Figure 9c, the technician is prompted to press a I to reply to the inquiry and forward the reply now, to press 2 to reply to the message later, or to press 3 to speak directly to the caller making the inquiry. The INPUT command is used to receive the selection made by the caller (processing box 1034). If the technician enters the 1 key (processing path 1037), a reply to the inquiry is processed immediately starting at processing box 1044. If the technician enters the 2 key (processing path 1040), the call flow terminates at termination box 1041. If the technician enters the 3 key (processing path 1070), the applications processor dials the telephone number entered by the caller when the original inquiry was recorded (processing box 1043). The call is placed using the CALL command provided by the call flow language of the present invention. Once the call is placed, subsequent automatic call flow processing is pre-empted. If the technician does not enter a 1, 2, or 3 key or a timeout occurs on the entry of one of the keys, processing transfers to the bubble labeled J illustrated in Figure 9c. At bubble J in Figure 9c, the technician is again prompted to enter a selection.

Referring again to Figure 9c at processing box 1044, the calling technician has chosen to reply to the new inquiry and to forward his response immediately. Using the RECORD NEW command construct, a new voice message is created for storing the technician's response to the inquiry. Additional information such as date and time are also integrated into the message (processing box 1044). Processing for the call flow then is transferred to the bubble labeled M illustrated starting in Figure 9d. Referring now to Figure 9d, the SPEAK command is used to prompt the technician to speak his response to the inquiry made by the original help desk caller (processing box 1045). Using the RECORD command, the voice response spoken by the technician is received and stored in the new message (processing box 1046). Once the technician completes the entry of his response, the inquiry database is accessed and the response message number is stored in the inquiry record (processing box 1047). Processing then terminates for this call flow at processing box 1048). Thus, a method and means for implementing a help desk using the present invention is described.

A third example of the operation of the present invention is illustrated in Figures 10a through 10e. The call flow illustrated in Figures 10a through 10e implements a streamlined order entry and tracking system using the applications hardware and call flow processing logic of the present invention. In a manner similar to the operation in the first two examples presented above, the call flow language commands are used for processing and order entry call at each step in the application. Using commands provided by the call flow language, various databases of information are accessed and updated as the call is being serviced. Thus, the present invention provides a flexible and dynamic call processing and messaging system providing seamless access to data and voice messages.

Thus, an integrated application controlled call processing and messaging system for improved call processing and voice messaging is disclosed.

Although the invention has been described herein with reference to a specific embodiment, many modifications and variations therein will readily occur to those skilled in the art. Accordingly, all such variations and modifications are included within the intended scope of the present invention as defined by the following claims.

<u>LISTING A</u>

```
10    NUM: 4 RATE
20    CHAR:8 ACCTNUM
30    NUM:2 ACCTBAL
40    CHAR: 3 ASSISTEXT
```

```
50   DEFAULT ON CN$FAIL GOTO ASSISTANCE
60   DEFAULT ON CN$TMOLIMIT GOTO ENDCALL
70   DEFAULT ON CN$INVLIMIT GOTO ASSISTANCE
80   DEFAULT ON CN$LOCKED GOTO ASSISTANCE
90   "INITIALIZE ASSIST EXTENSION AND LOCAL DATABASE
100  LET ASSISTEXT = '333'
110  SELECT LDINTRST
120  SELECT LDBAL
130  "BEGIN CALLER INTERACTION
140  SPEAK WELCOME AS PROMPT
150  GIVOPTS:
160  SPEAK SELECTIO AS PROMPT
170  MENU
180  ON CN$TIMEOUT GOTO GIVOPTS
190  ON [1] GOSUB RATEINFO
200  ON [2] GOTO SAVEINFO
210  ON [3] GOTO ENDCALL
220  GOTO GIVOPTS
230  "--ASK FOR ACCT NUMBER
240  SAVEINFO:
250  SPEAK INACCT AS PROMPT
260  INPUT ACCTNUM LENGTH 4 TO 8
270  ON CN$TIMEOUT GOTO SAVEINFO
280  ON CN$INVALID GOTO BADNUM
290  "USE THE LOCAL DATABASE TO VALIDATE ACCOUNT
300  "IN FIND FAILS IT IS AN INVALID ACCOUNT
310  FIND FIRST RECORD WITH ACCTFLD = ACCTNUM USING LDBAL
320  ON CN$FAIL GOTO BADNUM
330  GET ACCTBAL AT BALFLD USING LDBAL
340  SPEAK BALANCIS AS PROMPT
350  SPEAK ACCTBAL AS NUMERIC
360  GOTO GIVOPTS
370  "--INVALID ACCT ENTERED
380  BADNUM:
390  SPEAK INVACCT AS PROMPT
400  "--TRANSFER CALLER TO LIVE ASSISTANCE
450  ASSISTANCE:
460  TRANSFER ASSISTEXT
470  ON CN$FAIL GOTO XFERFAIL
480  CONNECT
490  ENDFLOW
500  XFERFAIL:
510  SPEAK PLSWAIT AS PROMPT
520  WAIT 10
530  GOTO ASSISTANCE
540  ENDCALL:
550  SPEAK THANKYOU AS PROMPT
560  ENDFLOW
570  "--GIVE CURRENT RATE INFORMATION
580  STARTSUB RATEINFO
590  FIND NTH RECORD WITH RECORD = 1 USING LDINTRST
600  GET RATE AT RATEFLD USING LDINTRST
610  SPEAK CURRATE AS PROMPT
620  SPEAK RATE AS NUMERIC
630  ENDSUB
```

Claims

1. A method for processing messages in a call processing and messaging system, comprising the steps of:

   a) receiving (1002, 1007) a first incoming call;
   b) storing (1021) a message from said first incoming call;
   c) assigning (1008) a message identifier to said message, said message identifier being assigned without caller intervention; and
   d) receiving (1002, 1069, N) a second incoming call; characterized by the steps of:
   e) dynamically generating (1021) an inquiry reference number;
   f) storing (1021) said message identifier and said inquiry reference number;
   g) speaking (1024) said inquiry reference number to a caller;
   h) receiving (1052) said inquiry reference number from said second incoming call; and
   i) retrieving (1053) said message identifier using said inquiry reference number;

   where steps e), f) and g) are performed before step d) and steps h) and i) are performed after step d).

2. The method as claimed in claim 1, further including a step of accessing (1053) said message using said retrieved message identifier.

3. The method as claimed in claim 1 or 2, further including a step of playing (1054) said message to which said message identifier is assigned.

4. The method as claimed in one of claims 1 to 3, further including the steps of:

   - connecting said call processing and messaging system to an external host computer; and
   - sending said message identifier to said host computer.

5. The method as claimed in claim 4, further including a step of retrieving said message identifier from said host computer.

6. The method as claimed in one of claims 1 to 5, further including the steps of:

   - storing a response message for said message;
   - assigning a response message identifier for said response message, said response message identifier being assigned without caller intervention; and
   - storing said response message identifier in said storing device.

7. The method as claimed in claim 6 further including the step of associating (1047) said response message identifier with said inquiry reference number.

8. The method as claimed in claims 6 or 7, further including the step of accessing (1055) said response message using said response message identifier.

9. The method as claimed in one of claims 1 to 8, wherein said retrieving step further includes a step of retrieving a plurality of message identifiers using said inquiry reference number.

10. The method as claimed in claim 9, further including a step of playing each message to which each message identifier of said plurality of message identifiers is assigned.

11. The method as claimed in one of claims 1 to 10, wherein said assigning step further includes a step of assigning a message identifier with a plurality of appended messages.

12. The method as claimed in one of claims 1 to 11, wherein said message identifier is used to reference said message in commands of a call processing script.

13. A call processing and messaging system, comprising:

- call receiving means (100, 105) having a plurality of lines (101, 151) for receiving a first incoming call from one of said lines (101, 151);
- means (107, 113) for storing a message from said first incoming call;
- means (106, 110, 112) for assigning a message identifier to said message, said message identifier being assigned without caller intervention; and
- means for receiving a second incoming call from one of said lines (101, 151);

characterized by:

- means (106, 110, 112) for dynamically generating an inquiry reference number;
- means (107, 113) for storing said message identifier and said inquiry reference number;
- means (105, 109) for speaking said inquiry reference number to a caller;
- means for receiving (105, 110, 112) said inquiry reference number from said second incoming call; and
- means for retrieving (106, 110, 112) said message identifier using said inquiry reference number.

14. The call processing and messaging system as claimed in claim 13, further including means for accessing said message using said message identifier.

15. The call processing and messaging system as claimed in claim 13 or 14, further including means for playing said message to which said message identifier is assigned.

16. The call processing and messaging system as claimed in one of claims 13 to 15, further including:

- means for connecting said call processing and messaging system to an external host computer; and
- means for sending said message identifier to said host computer.

17. The call processing and messaging system as claimed in claim 16, further including means for retrieving said message identifier from said host computer.

18. The call processing and messaging system as claimed in one of claims 13 to 17, further including:

- means for storing a response message to said message;
- means for assigning a response message identifier to said response message, said response message identifier being assigned without caller intervention; and
- means for storing said response message identifier.

19. The call processing and messaging system as claimed in claim 18, further including means for associating said response message identifier with said inquiry reference number.

20. The call processing and messaging system as claimed in claim 18 or 19, further including means for accessing said response message using said response message identifier.

21. The call processing and messaging system as claimed in one of claims 13 to 20, further including means for retrieving a plurality of message identifiers using said inquiry reference number.

22. The call processing and messaging system as claimed in claim 21, further including means for playing each message to which each message identifier of said plurality of message identifiers is assigned.

23. The call processing and messaging system as claimed in one of claims 13 to 22 further including means for assigning a message identifier with a plurality of appended messages.

24. The call processing and messaging system as claimed in one of claims 13 to 23 wherein said message identifier is used to reference said message in commands of a call processing script.

Patentansprüche

1. Verfahren zur Verarbeitung von Nachrichten in einem Anrufverarbeitungs- und Nachrichtensystem, mit den Schritten:

a) Empfangen (1002, 1007) eines ersten eingehenden Anrufs;
b) Speichern (1021) einer Nachricht aus dem ersten eingehenden Anruf;
c) Zuweisen (1008) eines Nachrichtenidentifikators zu dieser Nachricht, wobei dieser Nachrichtenidentifikator ohne Eingriff seitens des Anrufers zugewiesen wird; und
d) Empfangen (1002, 1069, N) eines zweiten eingehenden Anrufs;

gekennzeichnet durch die Schritte:

e) dynamisches Generieren (1021) einer Anfragereferenznummer;
f) Speichern (1021) des Nachrichtenidentifikators und der Anfragereferenznummer;
g) Ansagen (1024) der Anfragereferenznummer für einen Anrufer;
h) Empfangen (1052) der Anfragereferenznummer vom zweiten eingehenden Anruf; und
i) Abrufen (1053) des Nachrichtenidentifikators unter Verwendung der Anfragereferenznummer,

wobei die Schritte e), f) und g) vor dem Schritt d) und die Schritte h) und i) nach dem Schritt d) ausgeführt werden.

2. Verfahren nach Anspruch 1, das weiterhin einen Schritt des Zugreifens (1053) auf die Nachricht unter Verwendung des abgerufenen Nachrichtenidentifikators enthält.

3. Verfahren nach Anspruch 1 oder 2, das weiterhin einen Schritt des Wiedergebens (1054) der Nachricht, der dieser Nachrichtenidentifikator zugewiesen ist, enthält.

4. Verfahren nach einem der Ansprüche 1 bis 3, das weiterhin folgende Schritte enthält:

- Verbinden des Anrufverarbeitungs- und Nachrichtensystems mit einem externen Hostcomputer; und
- Senden des Nachrichtenidentifikators an den Hostcomputer.

5. Verfahren nach Anspruch 4, das weiterhin einen Schritt des Abrufens des Nachrichtenidentifikators aus dem Hostcomputer enthält.

6. Verfahren nach einem der Ansprüche 1 bis 5, das weiterhin folgende Schritte enthält:

- Speichern einer Antwortnachricht für diese Nachricht;
- Zuweisen eines Antwortnachricht-Identifikators zu dieser Antwortnachricht, wobei dieser Antwortnachricht-Identifikator ohne Eingriff seitens des Anrufers zugewiesen wird; und
- Speichern des Antwortnachricht-Identifikators in einer Speichereinrichtung.

7. Verfahren nach Anspruch 6, das weiterhin einen Schritt des Zuordnens (1047) des Antwortnachricht-Identifikators zu der Anfragereferenznummer enthält.

8. Verfahren nach Anspruch 6 oder 7, das weiterhin einen Schritt des Zugreifens (1055) auf die Antwortnachricht unter Verwendung des Antwortnachricht-Identifikators enthält.

9. Verfahren nach einem der Ansprüche 1 bis 8, bei dem der Schritt des Abrufens weiterhin einen Schritt des Abrufens einer Vielzahl von Nachrichtenidentifikatoren unter Verwendung der Anfragereferenznummer enthält.

10. Verfahren nach Anspruch 9, das weiterhin einen Schritt des Wiedergebens jeder Nachricht, der jeder Nachrichtenidentifikator der Vielzahl von Nachrichtenidentifikatoren zugewiesen ist, enthält.

11. Verfahren nach einem der Ansprüche 1 bis 10, bei dem der Zuweisungsschritt weiterhin einen Schritt des Zuweisens eines Nachrichtenidentifikators zu einer Vielzahl angefügter Nachrichten enthält.

12. Verfahren nach einem der Ansprüche 1 bis 11, bei dem der Nachrichtenidentifikator dazu verwendet wird, in den Befehlen einer Script-Datei für die Anrufverarbeitung auf die Nachricht zu verweisen.

13. Anrufverarbeitungs- und Nachrichtensystem mit:

- einer Anrufempfangseinrichtung (100, 105) mit einer Vielzahl von Leitungen (101, 151) zum Empfangen eines ersten eingehenden Anrufs von einer der Leitungen (101, 151);
- einer Einrichtung (107, 113) zum Speichern einer Nachricht aus dem ersten eingehenden Anruf;
- einer Einrichtung (106, 110, 112) zum Zuweisen eines Nachrichtenidentifikators zu der Nachricht, wobei dieser Nachrichtenidentifikator ohne Eingriff seitens des Anrufers erfolgt; und
- einer Einrichtung zum Empfangen eines zweiten eingehenden Anrufs von einer der Leitungen (101, 151);

gekennzeichnet durch:

- eine Einrichtung (106, 110, 112) zum dynamischen Generieren einer Anfragereferenznummer;
- eine Einrichtung (107, 113) zum Speichern des Nachrichtenidentifikators und der Anfragereferenznummer;
- eine Einrichtung (105, 109) zum Ansagen der Anfragereferenznummer an einen Anrufer;
- eine Einrichtung zum Empfangen (105, 110, 112) der Anfragereferenznummer von dem zweiten eingehenden Anruf; und
- eine Einrichtung zum Abrufen (106, 110, 112) des Nachrichtenidentifikators unter Verwendung der Anfragereferenznummer.

14. Anrufverarbeitungs- und Nachrichtensystem nach Anspruch 13, das weiterhin eine Einrichtung zum Zugreifen auf die Nachricht unter Verwendung des Nachrichtenidentifikators enthält.

15. Anrufverarbeitungs- und Nachrichtensystem nach Anspruch 13 oder 14, das weiterhin eine Einrichtung zum Wiedergeben der Nachricht, der dieser Nachrichtenidentifikator zugewiesen ist, enthält.

16. Anrufverarbeitungs- und Nachrichtensystem nach Anspruch 13 bis 15, das weiterhin enthält:

- eine Einrichtung zum Verbinden des Anrufverarbeitungs- und Nachrichtensystems mit einem externen Hostcomputer; und
- eine Einrichten zum Senden des Nachrichtenidentifikators an den Hostcomputer.

17. Anrufverarbeitungs- und Nachrichtensystem nach Anspruch 16, das weiterhin eine Einrichtung zum Abrufen des Nachrichtenidentifikators aus dem Hostcomputer enthält.

18. Anrufverarbeitungs- und Nachrichtensystem nach Anspruch 13 bis 17, das weiterhin enthält:

- eine Einrichtung zum Speichern einer Antwortnachricht auf die Nachricht;
- eine Einrichtung zum Zuweisen eines Antwortnachricht-Identifikators zu dieser Antwortnachricht, wobei dieser Antwortnachricht-Identifikator ohne Eingriff seitens des Anrufers zugewiesen wird; und
- eine Einrichtung zum Speichern des Antwortnachricht-Identifikators.

19. Anrufverarbeitungs- und Nachrichtensystem nach Anspruch 18, das weiterhin eine Einrichtung zum Zuordnen des Antwortnachricht-Identifikators zu der Anfragereferenznummer enthält.

20. Anrufverarbeitungs- und Nachrichtensystem nach Anspruch 18 oder 19, das weiterhin eine Einrichtung zum Zugreifen auf die Antwortnachricht unter Verwendung des Antwortnachricht-Identifikators enthält.

21. Anrufverarbeitungs- und Nachrichtensystem nach einem der Ansprüche 13 bis 20, das weiterhin eine Einrichtung zum Abrufen einer Vielzahl von Nachrichtenidentifikatoren unter Verwendung der Anfragereferenznummer enthält.

22. Anrufverarbeitungs- und Nachrichtensystem nach Anspruch 21, das weiterhin eine Einrichtung zum Wiedergeben jeder Nachricht, der jeder Nachrichtenidentifikator der Vielzahl von Nachrichtenidentifikatoren zugewiesen ist, enthält.

23. Anrufverarbeitungs- und Nachrichtensystem nach einem der Ansprüche 13 bis 22, das weiterhin eine Einrichtung zum Zuweisen eines Nachrichtenidentifikators zu einer Vielzahl angefügter Nachrichten enthält.

24. Anrufverarbeitungs- und Nachrichtensystem nach einem der Ansprüche 13 bis 23, bei dem der Nachrichtenidentifikator dazu verwendet wird, in den Befehlen einer Script-Datei für die Anrufverarbeitung auf die Nachricht zu verweisen.

Revendications

1. Procédé de traitement de messages dans un système de traitement d'appels et de messagerie, comprenant les opérations de :

   a) réception (1002, 1007) d'un premier appel entrant,
   b) mémorisation (1021) d'un message provenant de ce premier appel entrant,
   c) attribution (1008) d'un identificateur de message à ce message, cet identificateur de message étant attribué sans intervention du demandeur, et
   d) réception (1002, 1069, N) d'un deuxième appel entrant,

   caractérisé par les opérations de :

   e) production dynamique (1021) d'un numéro de référence d'interrogation,
   f) mémorisation (1021) de l'identificateur de message et du numéro de référence d'interrogation,
   g) communication orale (1024) du numéro de référence d'interrogation à un demandeur,
   h) réception (1052) du numéro de référence d'interrogation provenant du deuxième appel entrant, et
   i) récupération (1053) de l'identificateur de message au moyen du numéro de référence d'interrogation,

   les opérations e), f) et g) étant exécutées avant l'opération d), et les opérations h) et i) exécutées après l'opération d).

2. Procédé selon la revendication 1, comprenant en outre une opération d'accès (1053) au message au moyen de l'identificateur de message récupéré.

3. Procédé selon l'une des revendications 1 et 2, comprenant en outre une opération de lecture (1054) du message auquel l'identificateur de message est attribué.

4. Procédé selon l'une des revendications 1 à 3, comprenant en outre les opérations de :

   - connexion du système de traitement d'appels et de messagerie à un ordinateur hôte extérieur, et
   - envoi de l'identificateur de message à cet ordinateur hôte.

5. Procédé selon la revendication 4, comprenant en outre une opération de récupération de l'identificateur de message depuis l'ordinateur hôte.

6. Procédé selon l'une des revendications 1 à 5, comprenant en outre les opérations de :

   - mémorisation d'un message de réponse au message,
   - attribution d'un identificateur à ce message de réponse, cet identificateur de message de réponse étant attribué sans intervention du demandeur, et
   - mémorisation de cet identificateur de message de réponse dans la mémoire.

7. Procédé selon la revendication 6, comprenant en outre l'opération d'association (1047) de l'identificateur de message de réponse au numéro de référence d'interrogation.

8. Procédé selon l'une des revendications 6 et 7, comprenant en outre l'opération d'accès (1055) au message de réponse au moyen de l'identificateur de message de réponse.

9. Procédé selon l'une des revendications 1 à 8, dans lequel l'opération de récupération comprend en outre une opération de récupération d'une série d'identificateurs de messages au moyen du numéro de référence d'interrogation.

10. Procédé selon la revendication 9, comprenant en outre une opération de lecture de chaque message auquel

chaque identificateur de message de la série d'identificateurs de messages est attribué.

11. Procédé selon l'une des revendications 1 à 10, dans lequel l'opération d'attribution comprend en outre une opération d'attribution d'un identificateur de message à une série de messages annexés.

12. Procédé selon l'une des revendications 1 à 11, dans lequel l'identificateur de message est utilisé pour référencer le message dans des commandes d'un scénario de traitement d'appels.

13. Système de traitement d'appels et de messagerie comprenant :

- des moyens de réception d'appels (100, 105) ayant une série de lignes (101, 151) pour le réception d'un premier appel entrant d'une de ces lignes (101, 151),
- des moyens (107, 113) de mémorisation d'un message provenant de ce premier appel entrant,
- des moyens (106, 110, 112) d'attribution d'un identificateur de message à ce message, cet identificateur de message étant attribué sans intervention du demandeur, et
- des moyens de réception d'un deuxième appel entrant d'une des lignes (101, 151),

caractérisé par :

- des moyens (106, 110, 112) de production dynamique d'un numéro de référence d'interrogation,
- des moyens (107, 113) de mémorisation de l'identificateur de message et du numéro de référence d'interrogation,
- des moyens (105, 109) de communication orale du numéro de référence d'interrogation à un demandeur,
- des moyens (105, 110, 112) de réception du numéro de référence d'interrogation provenant du deuxième appel entrant, et
- des moyens (106, 110, 112) de récupération de l'identificateur de message au moyen du numéro de référence d'interrogation.

14. Système de traitement d'appels et de messagerie selon la revendication 13, comprenant en outre des moyens d'accès au message au moyen de l'identificateur de message.

15. Système de traitement d'appels et de messagerie selon l'une des revendications 13 et 14, comprenant en outre des moyens de lecture du message auquel l'identificateur de message est attribué.

16. Système de traitement d'appels et de messagerie selon l'une des revendications 13 à 15, comprenant en outre :

- des moyens de connexion de ce système à un ordinateur hôte extérieur, et
- des moyens d'envoi de l'identificateur de message à cet ordinateur hôte.

17. Système de traitement d'appels et de messagerie selon la revendication 16, comprenant en outre des moyens de récupération de l'identificateur de message depuis l'ordinateur hôte.

18. Système de traitement d'appels et de messagerie selon l'une des revendications 13 à 17, comprenant en outre :

- des moyens de mémorisation d'un message de réponse au message,
- des moyens d'attribution d'un identificateur à ce message de réponse, cet identificateur de message de réponse étant attribué sans intervention du demandeur, et
- des moyens de mémorisation de cet identificateur de message de réponse.

19. Système de traitement d'appels et de messagerie selon la revendication 18, comprenant en outre des moyens d'association de l'identificateur de message de réponse au numéro de référence d'interrogation.

20. Système de traitement d'appels et de messagerie selon l'une des revendications 18 et 19, comprenant en outre des moyens d'accès au message de réponse au moyen de l'identificateur de message de réponse.

21. Système de traitement d'appels et de messagerie selon l'une des revendications 13 à 20, comprenant en outre des moyens de récupération d'une série d'identificateurs de messages au moyen du numéro de référence d'interrogation.

22. Système de traitement d'appels et de messagerie selon la revendication 21, comprenant en outre des moyens de lecture de chaque message auquel chaque identificateur de message de la série d'identificateurs de messages est attribué.

23. Système de traitement d'appels et de messagerie selon l'une des revendications 13 à 22, comprenant en outre des moyens d'attribution d'un identificateur de message à une série de messages annexés.

24. Système de traitement d'appels et de messagerie selon l'une des revendications 13 à 23, dans lequel l'identificateur de message est utilisé pour référencer le message dans des commandes d'un scénario de traitement d'appels.

FIGURE 1 (PRIOR ART)

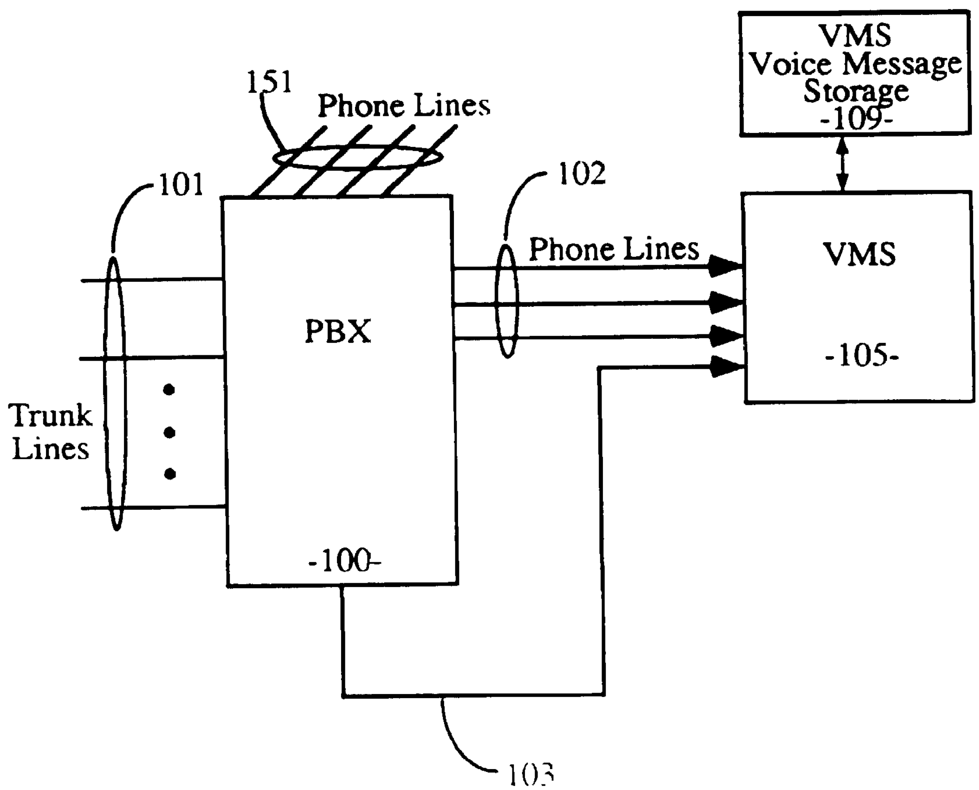


FIGURE 2

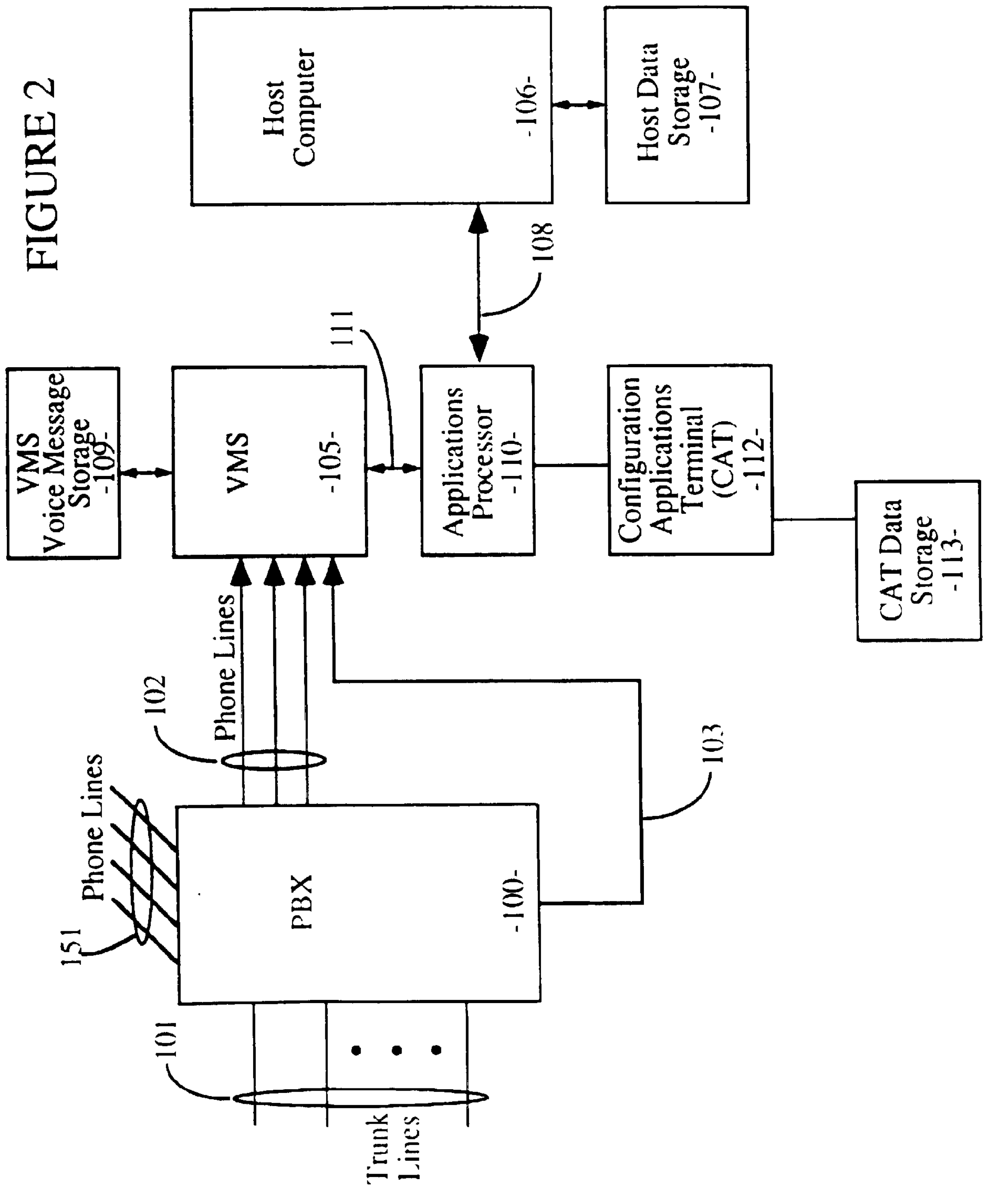
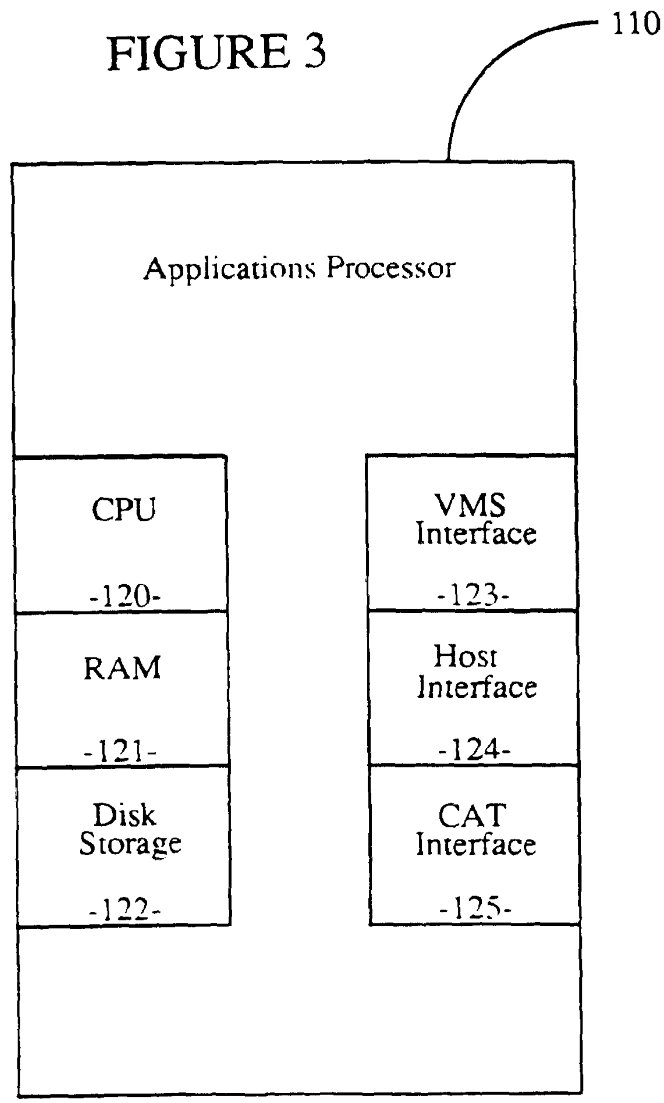

Host Computer
-106-
Host Data Storage
-107-
108
111
VMS Voice Message Storage
-109-
VMS
-105-
Applications Processor
-110-
Configuration Applications Terminal (CAT)
-112-
CAT Data Storage
-113-
Phone Lines
102
103
Phone Lines
151
PBX
-100-
101
Trunk Lines

FIGURE 3


110
Applications Processor
CPU
-120-
RAM
-121-
Disk Storage
-122-
VMS Interface
-123-
Host Interface
-124-
CAT Interface
-125-

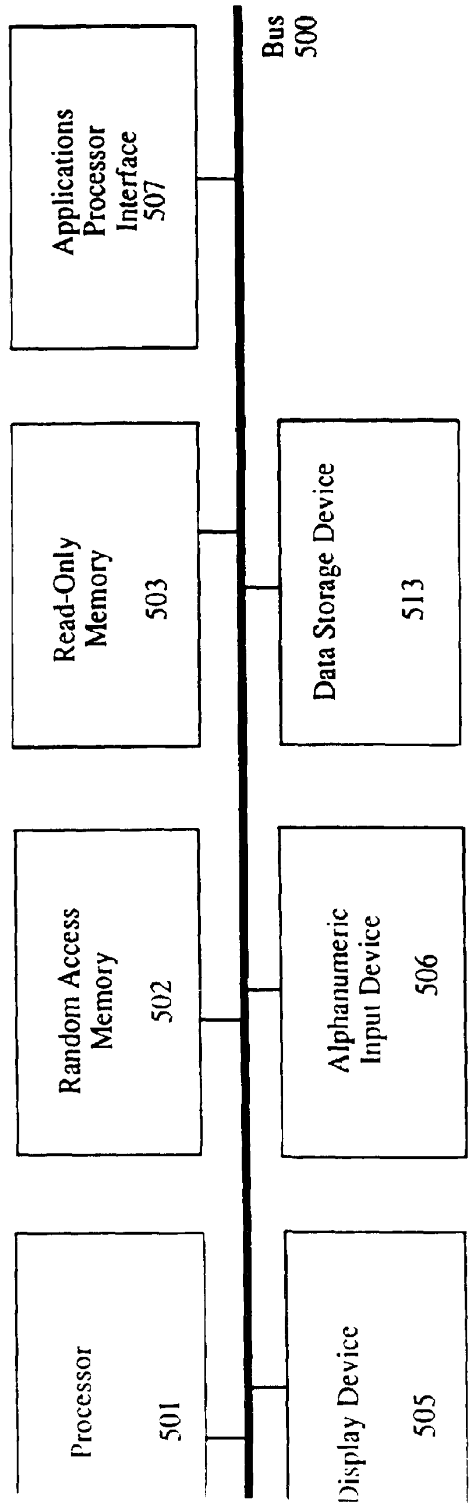

Bus
500
Applications
Processor
Interface
507
Read-Only
Memory
503
Data Storage Device
513
Random Access
Memory
502
Alphanumeric
Input Device
506
Processor
501
Display Device
505

Figure 4

FIGURE 5

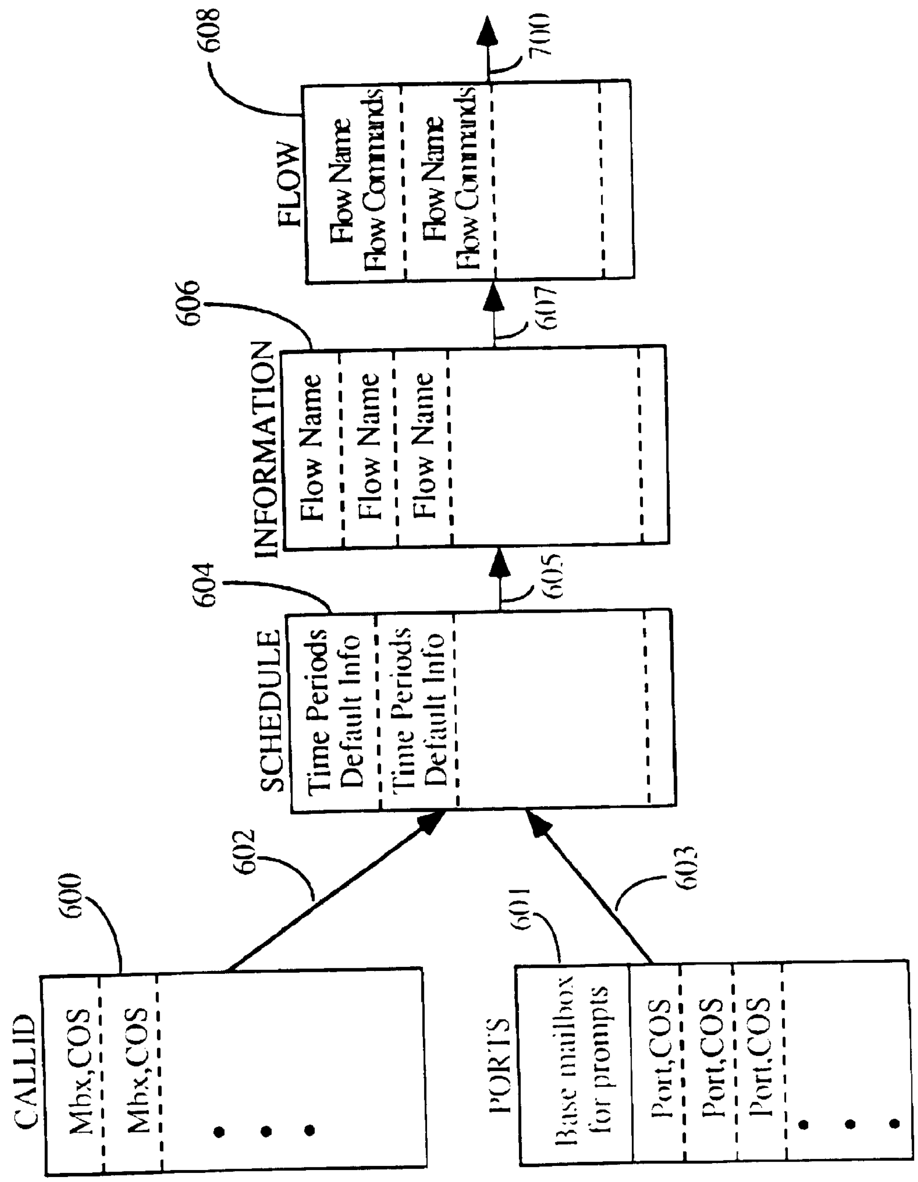
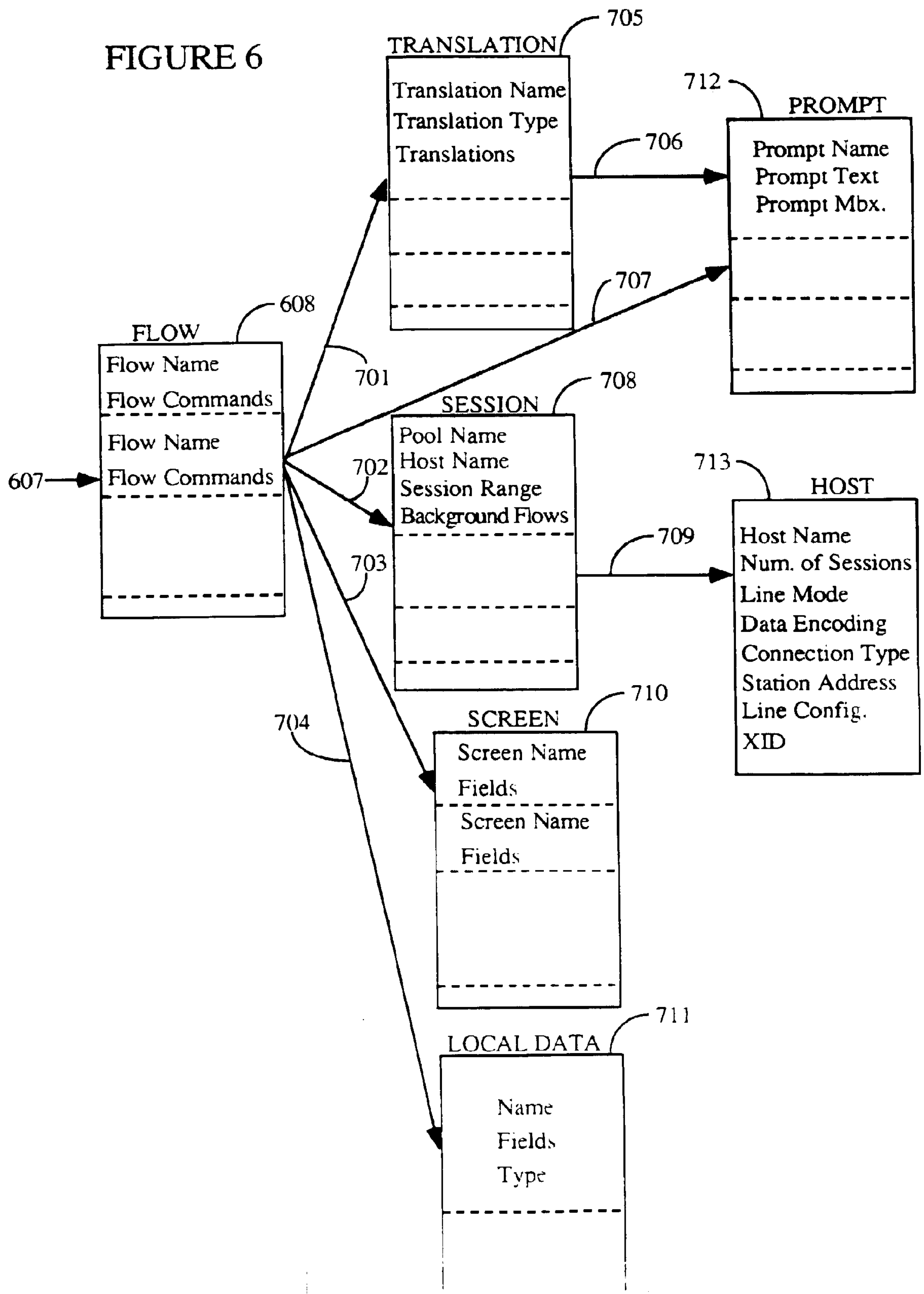
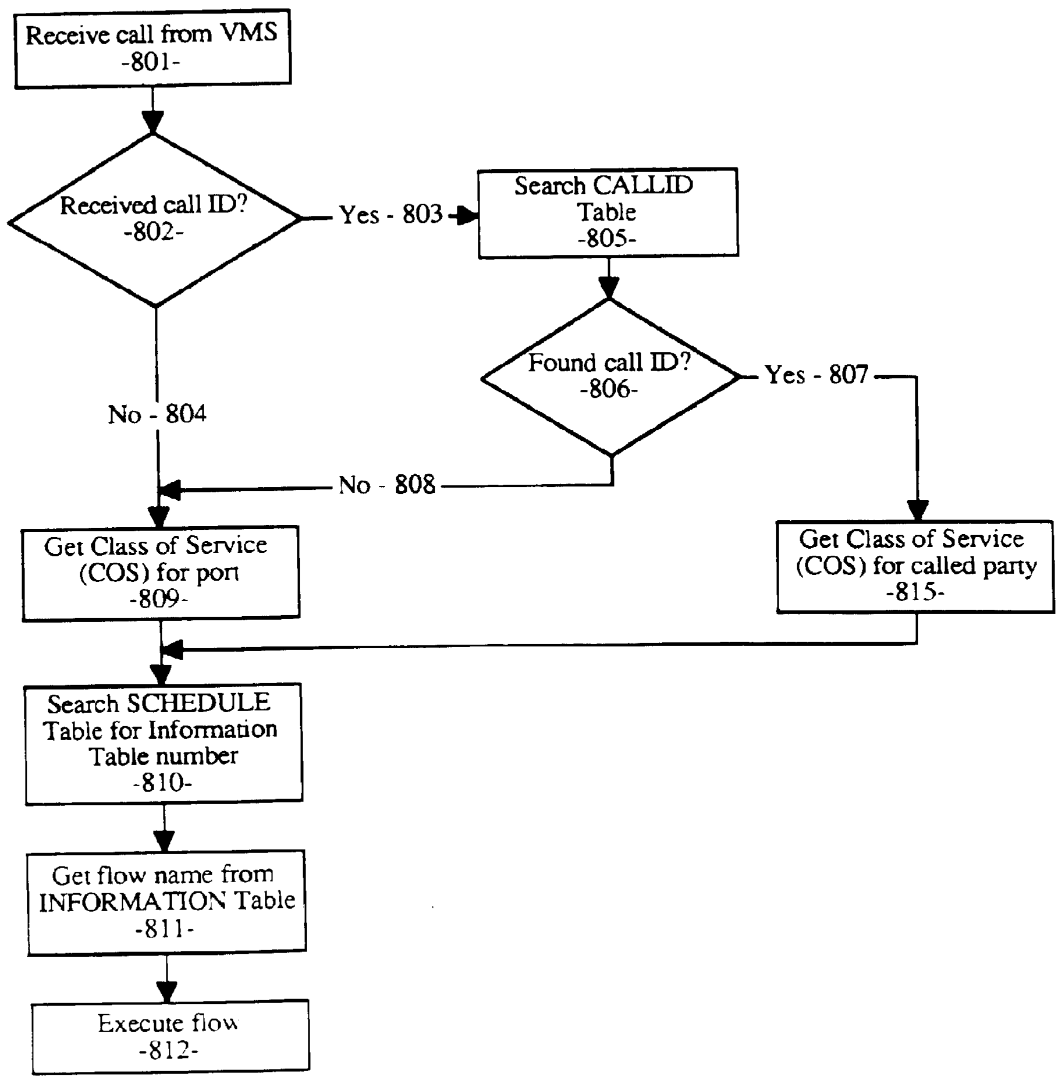

CALLID
Mbx,COS
Mbx,COS
600
602
PORTS
Base mailbox for prompts
Port,COS
Port,COS
Port,COS
601
603
SCHEDULE
Time Periods
Default Info
Time Periods
Default Info
604
605
INFORMATION
Flow Name
Flow Name
Flow Name
606
607
FLOW
Flow Name
Flow Commands
Flow Name
Flow Commands
608
700

FIGURE 6



705
TRANSLATION
Translation Name
Translation Type
Translations
712
PROMPT
706
Prompt Name
Prompt Text
Prompt Mbx.
707
608
FLOW
Flow Name
Flow Commands
Flow Name
Flow Commands
607
701
708
SESSION
Pool Name
Host Name
Session Range
Background Flows
702
713
HOST
709
Host Name
Num. of Sessions
Line Mode
Data Encoding
Connection Type
Station Address
Line Config.
XID
703
710
SCREEN
704
Screen Name
Fields
Screen Name
Fields
711
LOCAL DATA
Name
Fields
Type

FIGURE 7


Receive call from VMS
-801-
Received call ID?
-802-
Yes - 803
Search CALLID
Table
-805-
Found call ID?
-806-
Yes - 807
No - 804
No - 808
Get Class of Service
(COS) for port
-809-
Get Class of Service
(COS) for called party
-815-
Search SCHEDULE
Table for Information
Table number
-810-
Get flow name from
INFORMATION Table
-811-
Execute flow
-812-

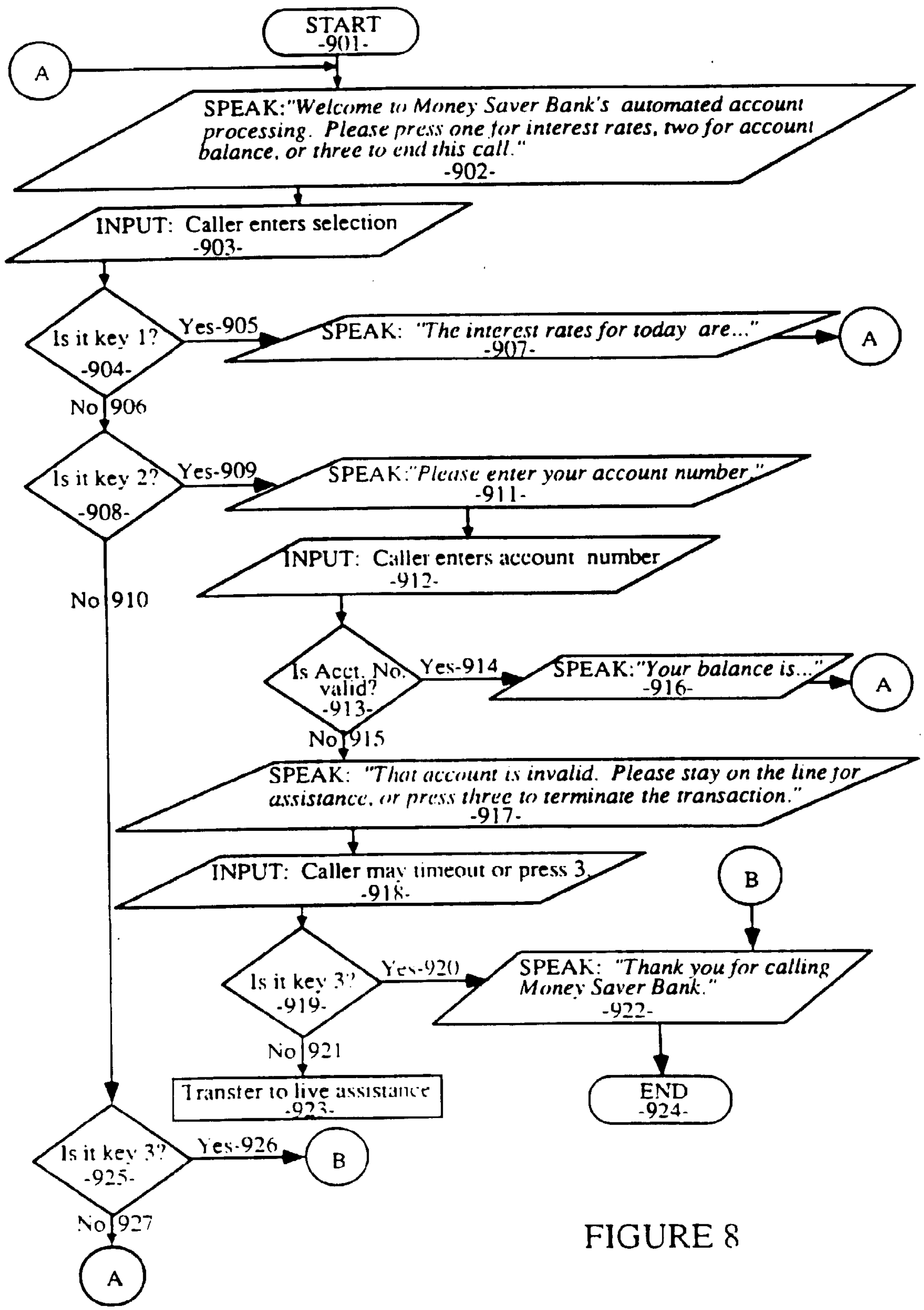

START
-901-
A
SPEAK: "Welcome to Money Saver Bank's automated account processing. Please press one for interest rates, two for account balance, or three to end this call."
-902-
INPUT: Caller enters selection
-903-
Is it key 1?
-904-
Yes-905
SPEAK: "The interest rates for today are..."
-907-
A
No 906
Is it key 2?
-908-
Yes-909
SPEAK: "Please enter your account number."
-911-
INPUT: Caller enters account number
-912-
No 910
Is Acct. No. valid?
-913-
Yes-914
SPEAK: "Your balance is..."
-916-
A
No 915
SPEAK: "That account is invalid. Please stay on the line for assistance, or press three to terminate the transaction."
-917-
INPUT: Caller may timeout or press 3
-918-
B
Is it key 3?
-919-
Yes-920
SPEAK: "Thank you for calling Money Saver Bank."
-922-
No 921
Transfer to live assistance
-923-
END
-924-
Is it key 3?
-925-
Yes-926
B
No 927
A

FIGURE 8

FIGURE 9a

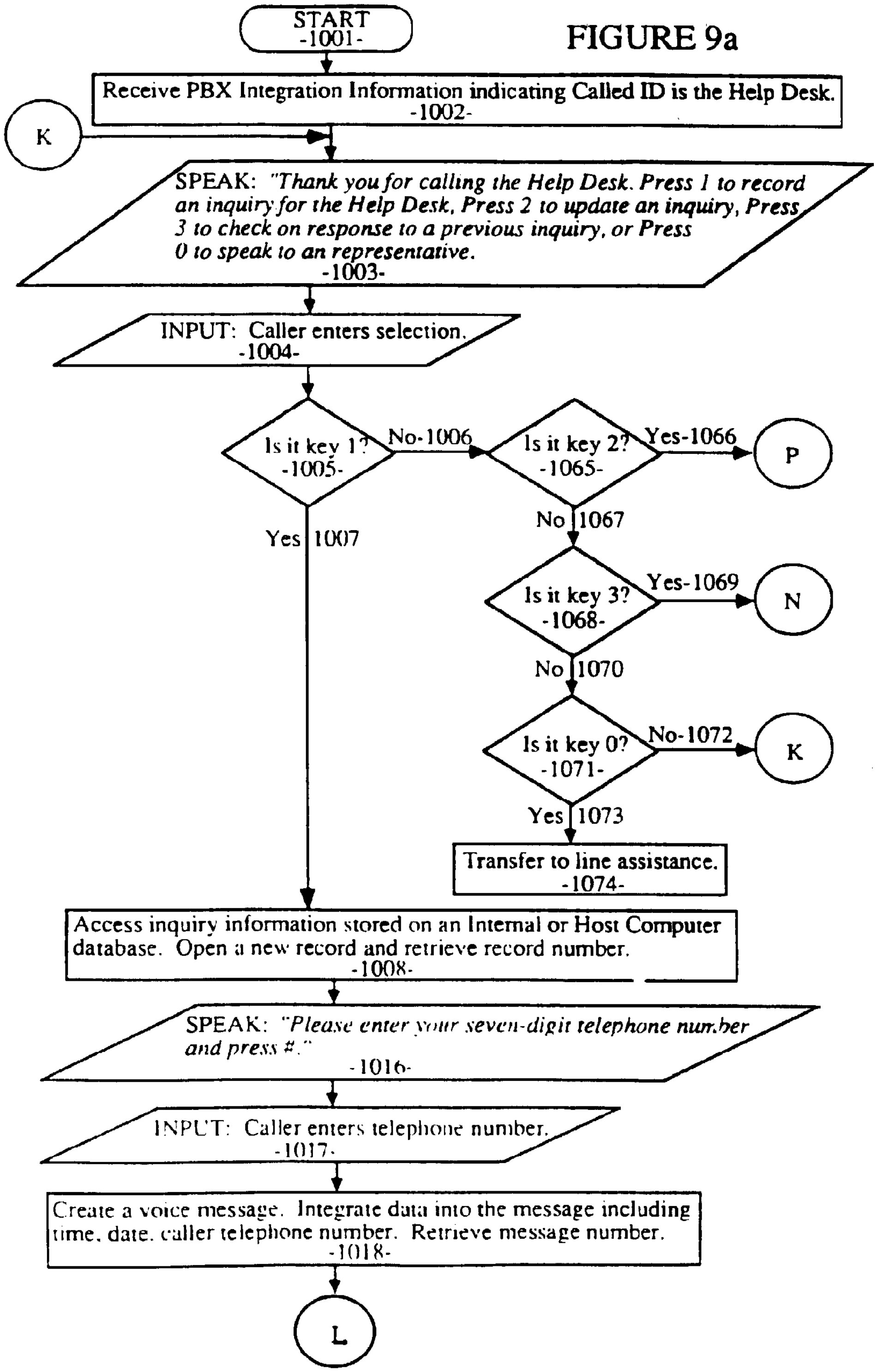

START
-1001-
Receive PBX Integration Information indicating Called ID is the Help Desk.
-1002-
K
SPEAK: "Thank you for calling the Help Desk. Press 1 to record an inquiry for the Help Desk, Press 2 to update an inquiry, Press 3 to check on response to a previous inquiry, or Press 0 to speak to an representative.
-1003-
INPUT: Caller enters selection.
-1004-
Is it key 1?
-1005-
No-1006
Is it key 2?
-1065-
Yes-1066
P
No 1067
Yes 1007
Is it key 3?
-1068-
Yes-1069
N
No 1070
Is it key 0?
-1071-
No-1072
K
Yes 1073
Transfer to line assistance.
-1074-
Access inquiry information stored on an Internal or Host Computer database. Open a new record and retrieve record number.
-1008-
SPEAK: "Please enter your seven-digit telephone number and press #."
-1016-
INPUT: Caller enters telephone number.
-1017-
Create a voice message. Integrate data into the message including time, date, caller telephone number. Retrieve message number.
-1018-
L

FIGURE 9b

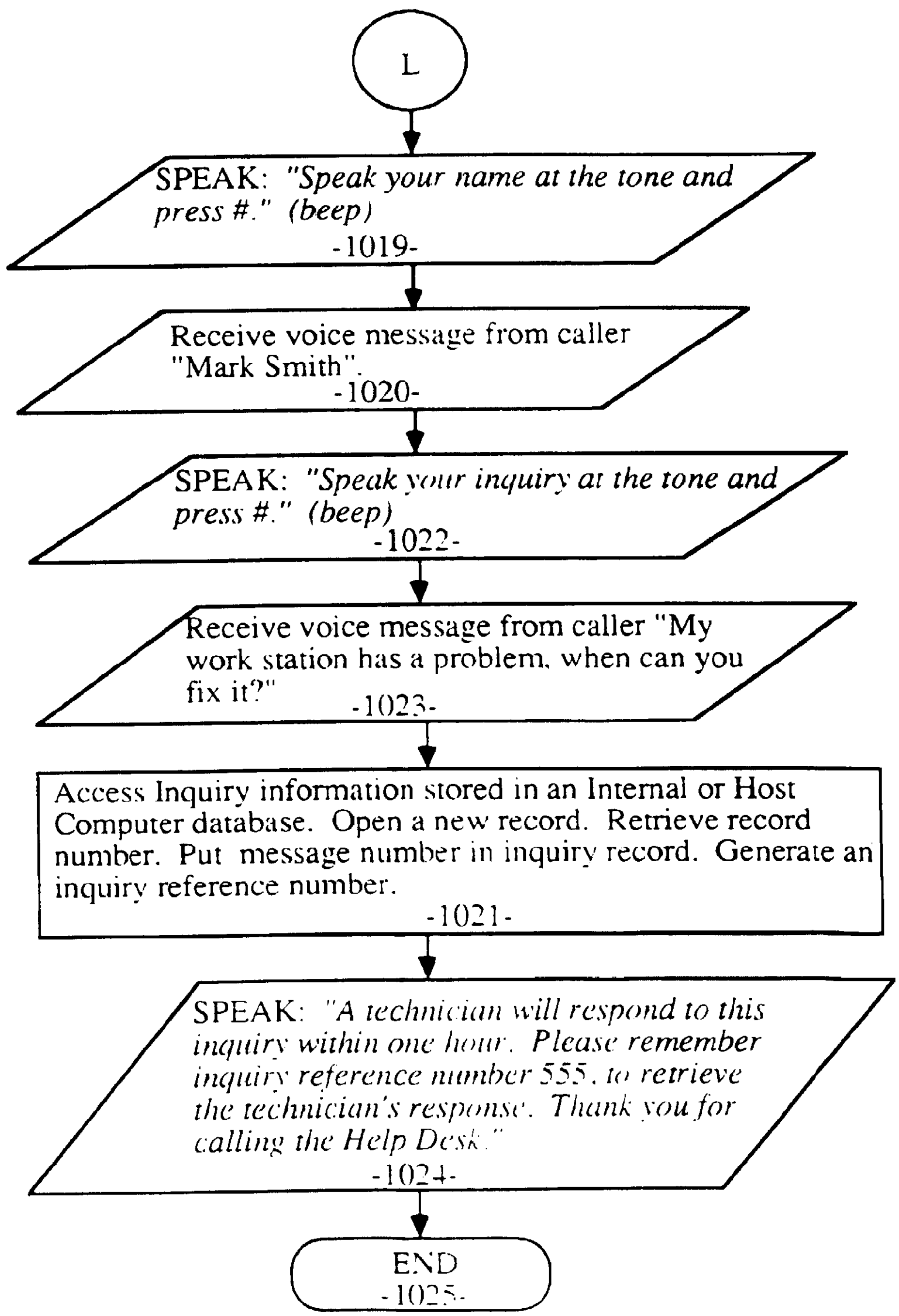

L
SPEAK: "Speak your name at the tone and press #." (beep)
-1019-
Receive voice message from caller "Mark Smith".
-1020-
SPEAK: "Speak your inquiry at the tone and press #." (beep)
-1022-
Receive voice message from caller "My work station has a problem, when can you fix it?"
-1023-
Access Inquiry information stored in an Internal or Host Computer database. Open a new record. Retrieve record number. Put message number in inquiry record. Generate an inquiry reference number.
-1021-
SPEAK: "A technician will respond to this inquiry within one hour. Please remember inquiry reference number 555, to retrieve the technician's response. Thank you for calling the Help Desk."
-1024-
END
-1025-

FIGURE 9c

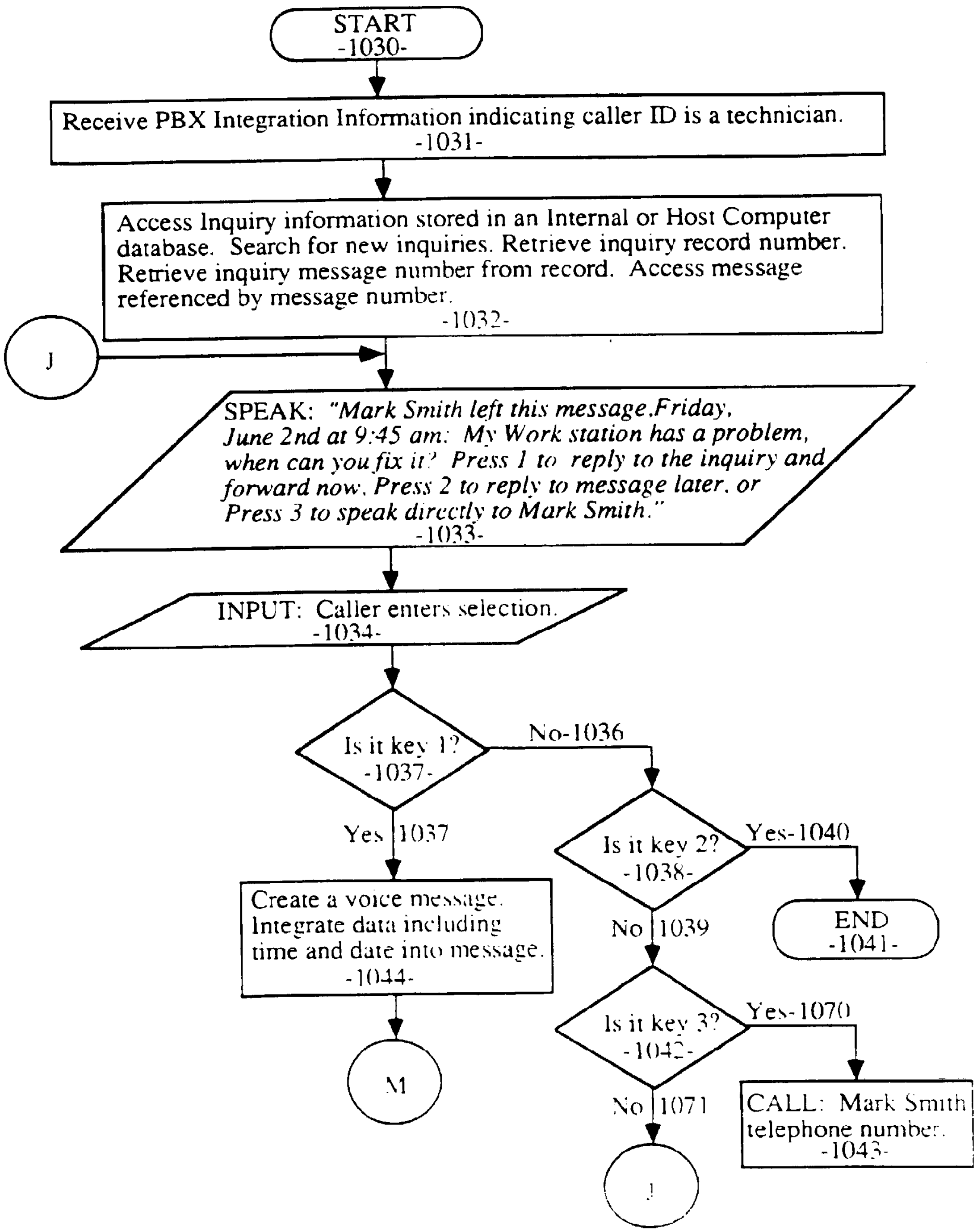

START
-1030-
Receive PBX Integration Information indicating caller ID is a technician.
-1031-
Access Inquiry information stored in an Internal or Host Computer database. Search for new inquiries. Retrieve inquiry record number. Retrieve inquiry message number from record. Access message referenced by message number.
-1032-
J
SPEAK: "Mark Smith left this message,Friday, June 2nd at 9:45 am: My Work station has a problem, when can you fix it? Press 1 to reply to the inquiry and forward now. Press 2 to reply to message later, or Press 3 to speak directly to Mark Smith."
-1033-
INPUT: Caller enters selection.
-1034-
Is it key 1?
-1037-
No-1036
Yes 1037
Create a voice message. Integrate data including time and date into message.
-1044-
M
Is it key 2?
-1038-
Yes-1040
END
-1041-
No 1039
Is it key 3?
-1042-
Yes-1070
CALL: Mark Smith telephone number.
-1043-
No 1071
J

FIGURE 9d

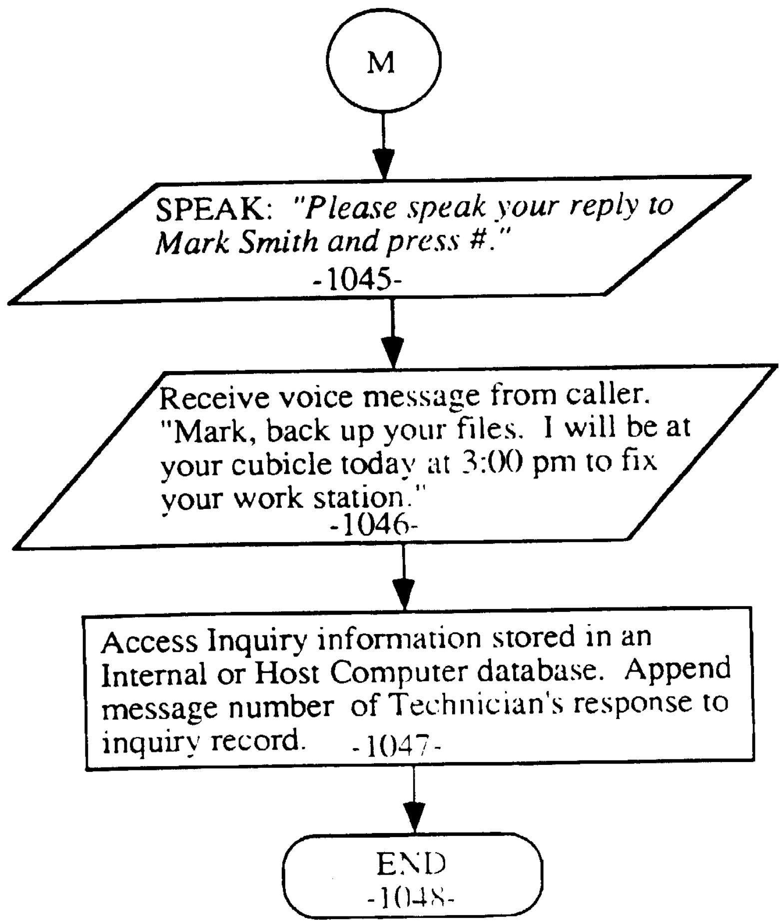

M
SPEAK: *"Please speak your reply to Mark Smith and press #."*
-1045-
Receive voice message from caller. "Mark, back up your files. I will be at your cubicle today at 3:00 pm to fix your work station."
-1046-
Access Inquiry information stored in an Internal or Host Computer database. Append message number of Technician's response to inquiry record. -1047-
END
-1048-

FIGURE 9e

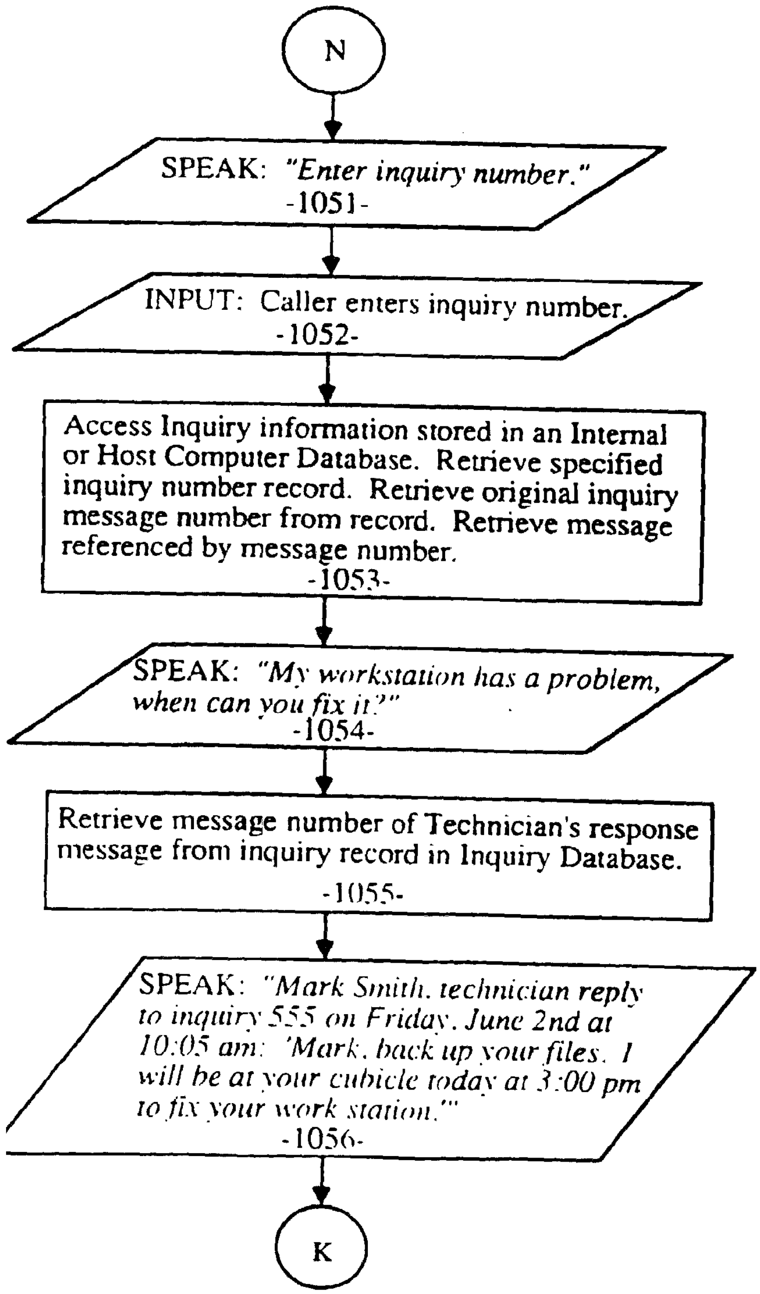

N
SPEAK: "Enter inquiry number."
-1051-
INPUT: Caller enters inquiry number.
-1052-
Access Inquiry information stored in an Internal or Host Computer Database. Retrieve specified inquiry number record. Retrieve original inquiry message number from record. Retrieve message referenced by message number.
-1053-
SPEAK: "My workstation has a problem, when can you fix it?"
-1054-
Retrieve message number of Technician's response message from inquiry record in Inquiry Database.
-1055-
SPEAK: "Mark Smith, technician reply to inquiry 555 on Friday, June 2nd at 10:05 am: 'Mark, back up your files. I will be at your cubicle today at 3:00 pm to fix your work station.'"
-1056-
K

FIGURE 9f

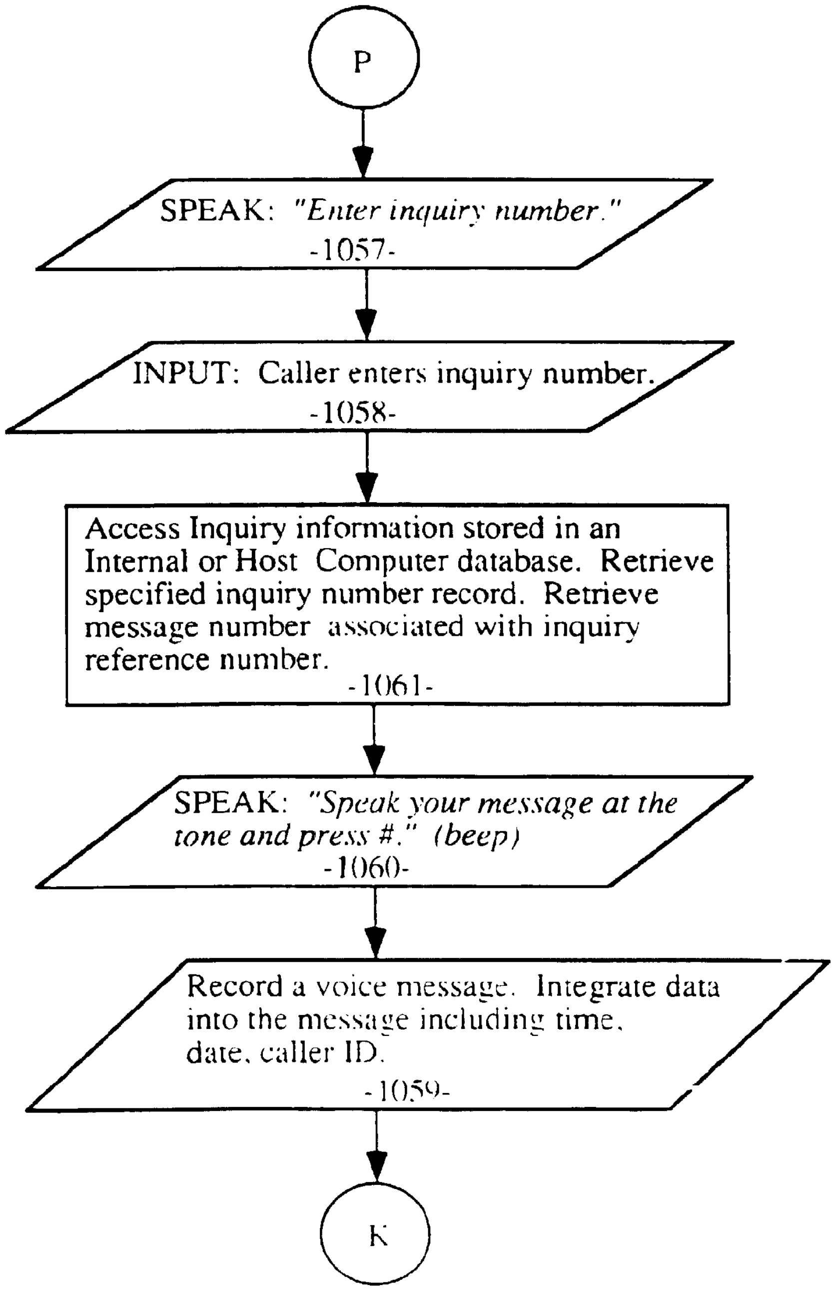

P
SPEAK: "Enter inquiry number."
-1057-
INPUT: Caller enters inquiry number.
-1058-
Access Inquiry information stored in an Internal or Host Computer database. Retrieve specified inquiry number record. Retrieve message number associated with inquiry reference number.
-1061-
SPEAK: "Speak your message at the tone and press #." (beep)
-1060-
Record a voice message. Integrate data into the message including time, date, caller ID.
-1059-
K

FIGURE 10a

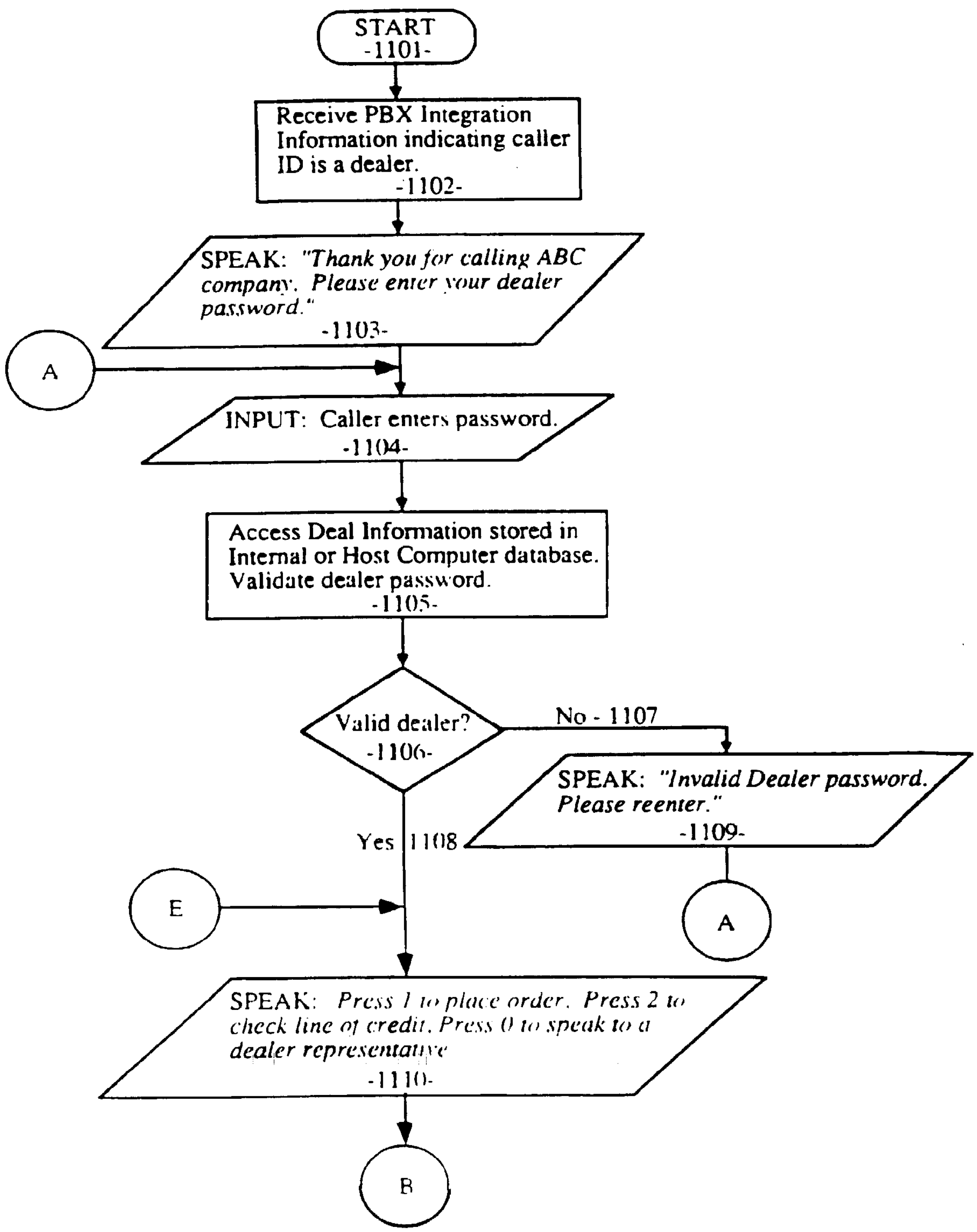

START
-1101-
Receive PBX Integration Information indicating caller ID is a dealer.
-1102-
SPEAK: "Thank you for calling ABC company. Please enter your dealer password."
-1103-
A
INPUT: Caller enters password.
-1104-
Access Deal Information stored in Internal or Host Computer database. Validate dealer password.
-1105-
Valid dealer?
-1106-
No - 1107
SPEAK: "Invalid Dealer password. Please reenter."
-1109-
A
Yes 1108
E
SPEAK: Press 1 to place order. Press 2 to check line of credit. Press 0 to speak to a dealer representative
-1110-
B

FIGURE 10b

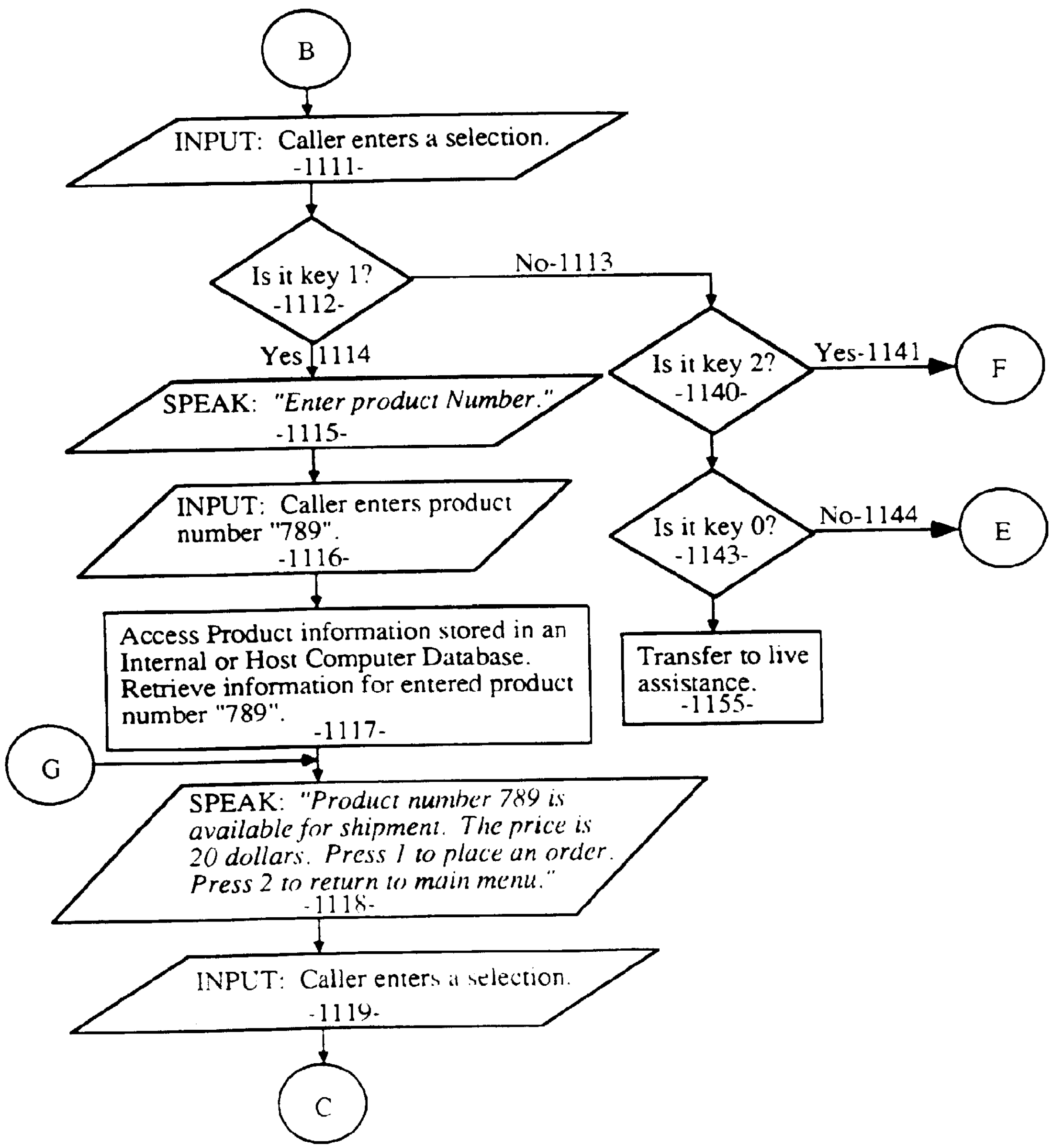

B
INPUT: Caller enters a selection.
-1111-
Is it key 1?
-1112-
No-1113
Yes 1114
SPEAK: "Enter product Number."
-1115-
INPUT: Caller enters product number "789".
-1116-
Access Product information stored in an Internal or Host Computer Database. Retrieve information for entered product number "789".
-1117-
G
SPEAK: "Product number 789 is available for shipment. The price is 20 dollars. Press 1 to place an order. Press 2 to return to main menu."
-1118-
INPUT: Caller enters a selection.
-1119-
C
Is it key 2?
-1140-
Yes-1141
F
Is it key 0?
-1143-
No-1144
E
Transfer to live assistance.
-1155-

FIGURE 10c

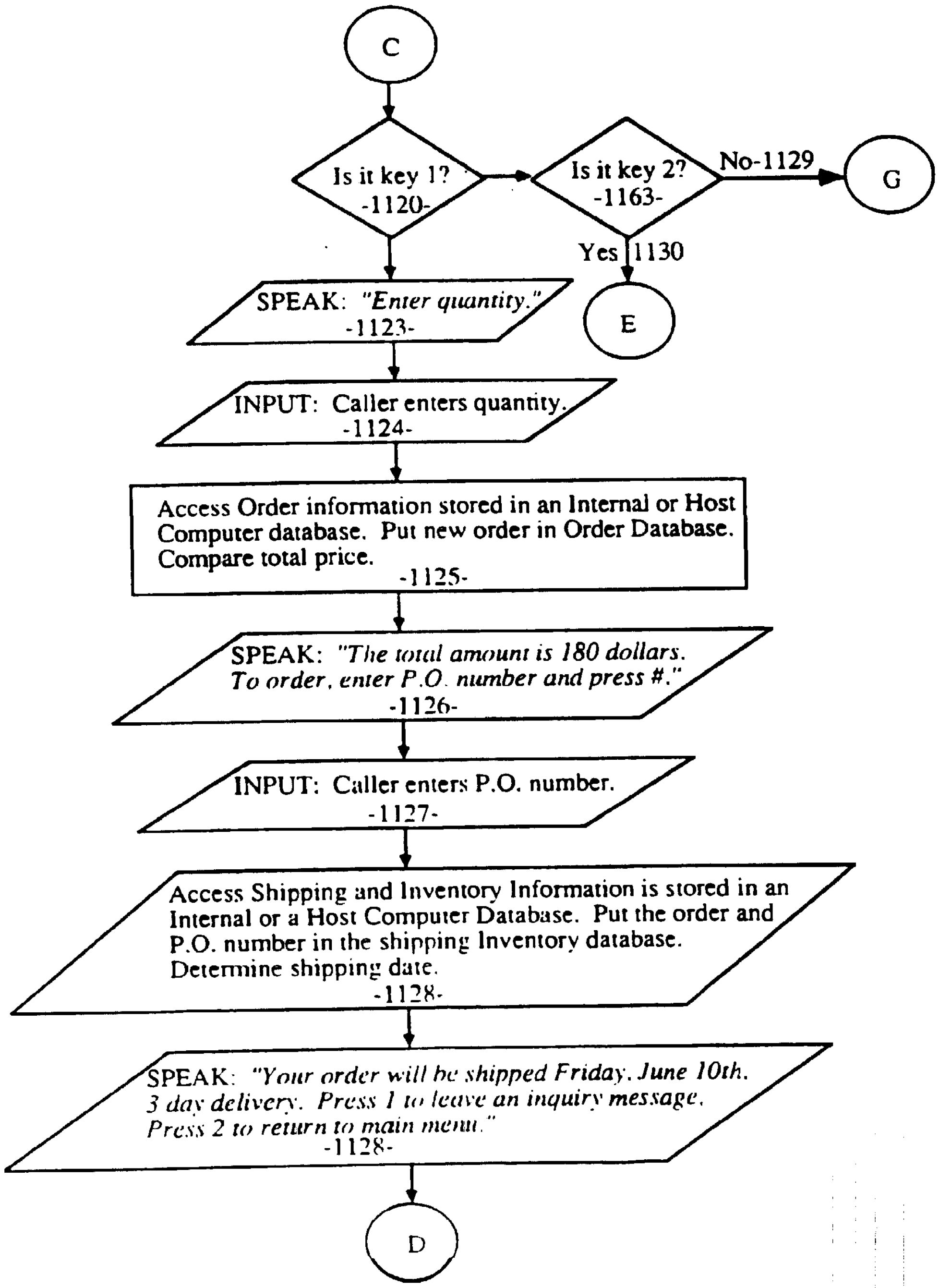

C
Is it key 1?
-1120-
Is it key 2?
-1163-
No-1129
G
Yes
1130
E
SPEAK: "Enter quantity."
-1123-
INPUT: Caller enters quantity.
-1124-
Access Order information stored in an Internal or Host Computer database. Put new order in Order Database. Compare total price.
-1125-
SPEAK: "The total amount is 180 dollars. To order, enter P.O. number and press #."
-1126-
INPUT: Caller enters P.O. number.
-1127-
Access Shipping and Inventory Information is stored in an Internal or a Host Computer Database. Put the order and P.O. number in the shipping Inventory database. Determine shipping date.
-1128-
SPEAK: "Your order will be shipped Friday, June 10th. 3 day delivery. Press 1 to leave an inquiry message. Press 2 to return to main menu."
-1128-
D

FIGURE 10d

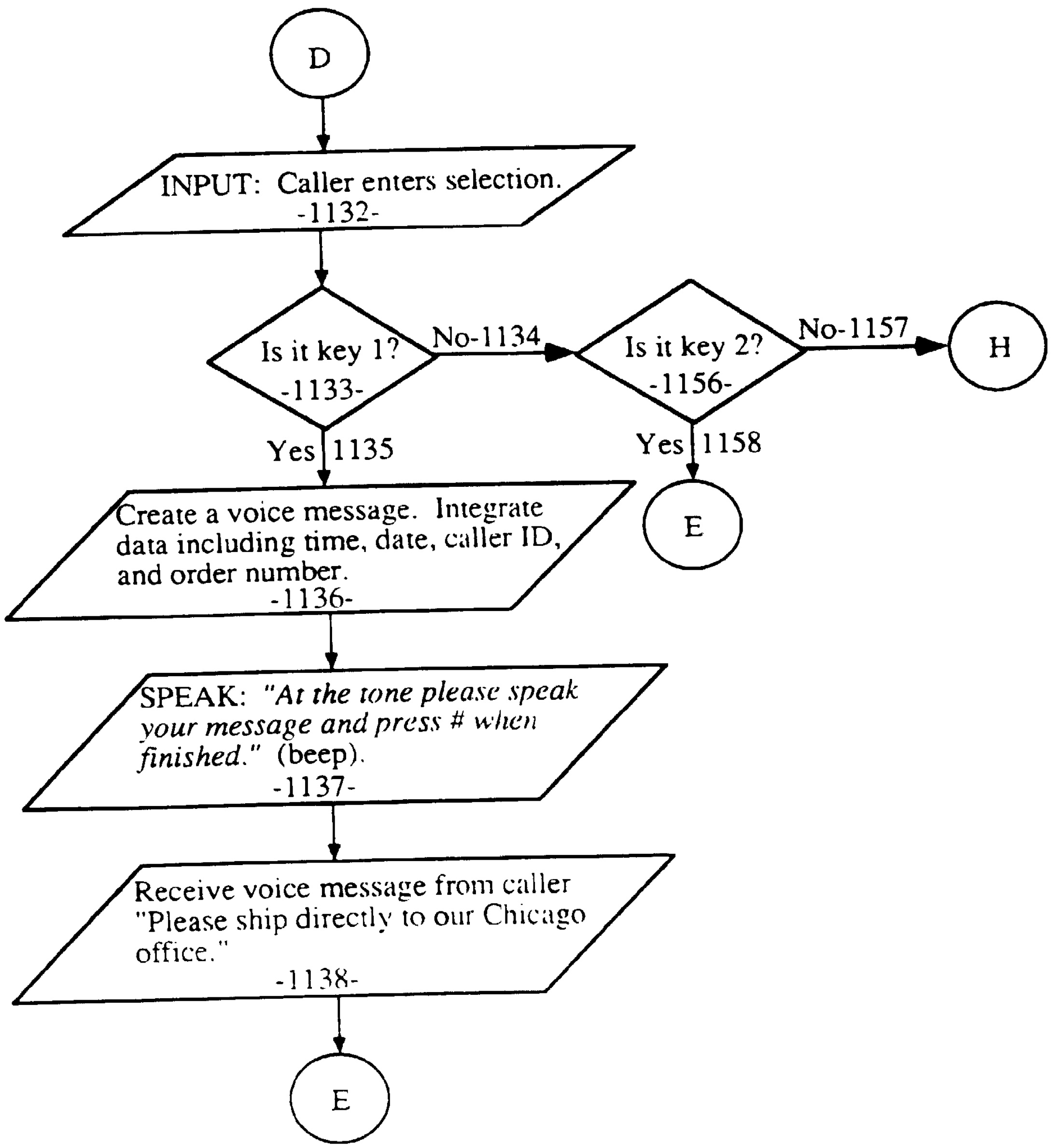
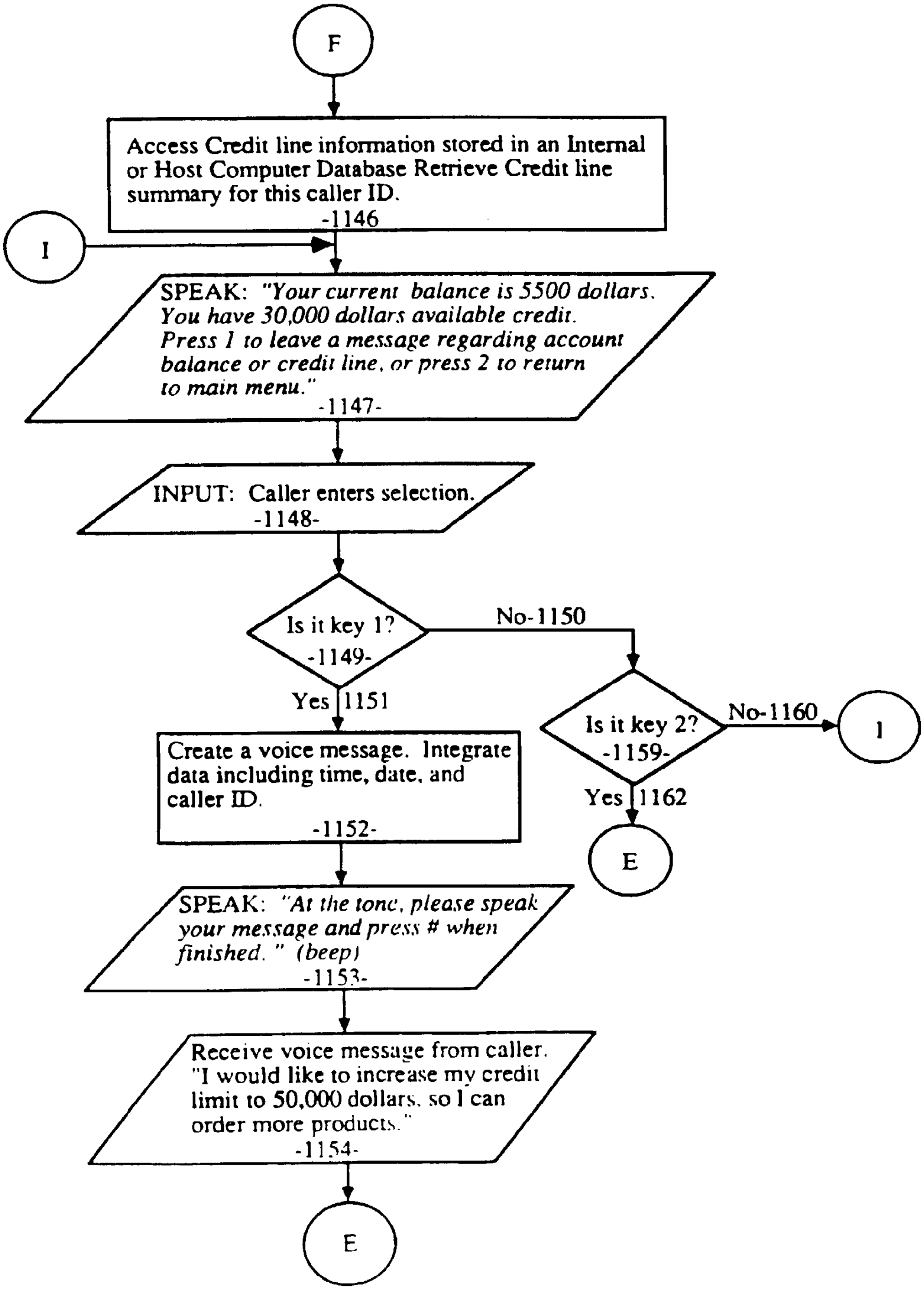

D
INPUT: Caller enters selection.
-1132-
Is it key 1?
-1133-
No-1134
Is it key 2?
-1156-
No-1157
H
Yes 1135
Yes 1158
E
Create a voice message. Integrate data including time, date, caller ID, and order number.
-1136-
SPEAK: "At the tone please speak your message and press # when finished." (beep).
-1137-
Receive voice message from caller "Please ship directly to our Chicago office."
-1138-
E

FIGURE 10e


F
Access Credit line information stored in an Internal or Host Computer Database Retrieve Credit line summary for this caller ID.
-1146
I
SPEAK: "Your current balance is 5500 dollars. You have 30,000 dollars available credit. Press 1 to leave a message regarding account balance or credit line, or press 2 to return to main menu."
-1147-
INPUT: Caller enters selection.
-1148-
Is it key 1?
-1149-
No-1150
Yes 1151
Is it key 2?
-1159-
No-1160
I
Yes 1162
E
Create a voice message. Integrate data including time, date, and caller ID.
-1152-
SPEAK: "At the tone, please speak your message and press # when finished." (beep)
-1153-
Receive voice message from caller. "I would like to increase my credit limit to 50,000 dollars, so I can order more products."
-1154-
E

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European Patent Office
Office européen des brevets

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(54) **Voice mail on the internet**

(57) A first Internet telephone system 620 attempts to call with a second Internet telephone system 630 via the Internet 600. However, the second Internet telephone system 630 is not logged onto the Internet at the time of the call. In response to the failed attempt to call, the first Internet telephone system prompts the user to send voice mail to the user of the second Internet telephone system. This results in a phone call over the Internet between a voice mail system 610 and the first Internet telephone system, allowing a greeting to be heard, and a message to be stored. This message may be subsequently retrieved, either using an Internet telephone system over the Internet, or using a standard phone over the conventional telephone network.

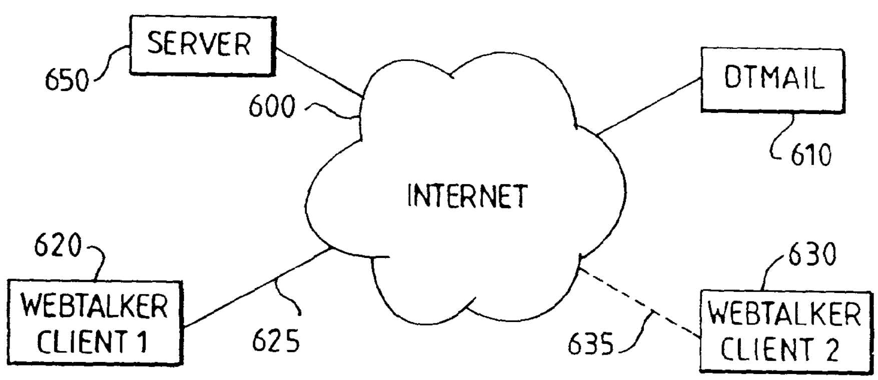

FIG. 6

EP 0 794 650 A2

Description

Background of the invention

The present invention relates to voice mail systems for use over the Internet.

Conventional voice mail systems (also termed voice messaging systems) attach to the telephone network, often via a PBX, and are used to store messages from incoming calls when the intended recipient is absent or otherwise engaged. The intended recipient can then listen to their stored messages at some future time. A voice mail system is generally implemented either on special purpose computer hardware, or else on a standard computer workstation equipped with a suitable telephony interface. Such voice mail systems are well-known; one example is the DirectTalkMail system, available from IBM Corporation, which operates in conjunction with the DirectTalk/6000 voice processing system (also available from IBM). Other examples of voice mail systems are described in US 5,136,648 and EPA 0588576.

Also known are electronic mail systems (email), which allow the transmission of text messages over a computer network. The most prominent such network over the last couple of years is the Internet, which provides a readily accessible, worldwide network for packet-based communications. Background information about the Internet and the World Wide Web can be found in "Spinning the Web" by Andrew Ford (International Thomson Publishing, London 1995) and "The World Wide Web Unleashed" by John December and Neil Randall (SAMS Publishing, Indianapolis 1994).

Although originally intended primarily for the transmission of computer data, more recently the Internet has been exploited to provide real-time telephony communications. The primary attraction of the Internet for telephony communications is the charging structure. Thus many Internet users have a dial-up connection to an access provider. This is normally over a local telephone line, so such users pay only local telephone charges when logged on. Some access providers charge a monthly description, whilst others charge on the basis of connection time (some may do both). However, there is generally no charge associated with actual data transfer over the network. As a result, the effective cost of an international call over the Internet may be no more than that of a local call of the same duration to the access provider. In addition, the fully digital nature of the Internet may potentially offer a richer functionality (eg in terms of conference calling) than conventional telephone networks. Internet telephones are surveyed in the article "Dial 1-800-Internet" in Byte Magazine, February 1996, p83-88, and in the article "Nattering On", in New Scientist, 2 March 1996, p38-40.

The transmission of voice signals over a packet network is described for example in "Using Local Area Networks for Carrying Online Voice" by D Cohen, pages 13-21, in "Voice Transmission over an Ethernet Backbone" by P Ravasio, R Marcogliese, and R Novarese, pages 39-65, both in "Local Computer Networks" (edited by P Ravasio, G Hopkins, and N Naffah; North Holland, 1982) and also in GB 2283152. The basic principles of such a scheme are that a first terminal or workstation digitally samples a voice input signal at a regular rate (eg 8 kHz). A number of samples are then assembled into a data packet for transmission over the network to a second terminal, which then feeds the samples to a loudspeaker or equivalent device for playout, again at a constant 8 kHz rate. Voice transmission over the Internet is substantially similar to transmission over a LAN (which may indeed provide part of the Internet transmission path), but there tends to be less spare bandwidth available on the Internet. As a result, Internet telephones normally compress the voice signal at the transmitting end, and then decompress it at the receiving end.

There are several well-known problems with packet-based voice communications. Firstly, there is a transmission delay over the network, which is variable, depending particularly on the utilisation of the network by other nodes at any given time. Thus the arrival of packets at a destination node is both delayed and irregular. If the packets were played out in irregular fashion, this would have an extremely adverse effect on intelligibility of the voice signal. Therefore, packet voice transmissions utilise some degree of buffering at the reception end, to absorb such irregularities. Such buffering however increases the delay between the original voice signal and the audio output at the destination end. This delay may cause problems with echos, and more importantly, can render natural interactive two-way conversation difficult (in the same way that an excessive delay on a transatlantic conventional phone call can be highly intrusive). Yet further delays are introduced by the necessity to compress/decompress the voice signal.

Some Internet telephones which are currently commercially available provide a voice mail facility, whereby the calling party can leave a message if the called party does not answer. This is somewhat analogous to the conventional answerphone. Note that such systems are limited in that generally the called party is required to be logged onto the Internet to receive a voice message. However, many users with dial-up Internet access via a modem have to pay on a time basis for a telephone call into their Internet access provider, and moreover some Internet access providers charge for connections on a time basis. Therefore most users log off from the Internet when they are not using it to avoid incurring excessive charges. In such circumstances, this implies that they are no longer able to receive a voice message.

It would of course be possible for a user to transmit a voice message in the form of a normal email (data) message to another user who is not currently logged on. The message is then queued at the Internet access provider's site for delivery at next logon by the relevant user. However, generally this approach offers few benefits over simply sending an email message, rather than the voice message.

It is an object of the invention to provide improved voice mail facilities for the users of Internet telephones.

Accordingly, the invention provides a method of providing voice mail on the Internet comprising the steps of:

selecting to make a call from a first user at a first Internet telephone system to a second user at a second Internet telephone system;
determining that said second Internet telephone system is not currently logged onto the Internet to receive said call; and
transmitting a voice message for said second user over the Internet to a voice mail system distinct from said second Internet telephone system for subsequent retrieval by said second user.

The voice mail service is therefore available when the first user cannot directly contact the second user. This may be determined as a result of actually trying to place a call to the second user, or perhaps simply by viewing the list of currently accessible parties. The voice mail system is distinct from the second Internet telephone system, and is intended to be continuously attached to the Internet. The voice mail system is therefore available when the second Internet telephone system is not logged onto the Internet, thereby avoiding the subscriber of this second system having to pay overly high Internet connection or usage charges.

In a preferred embodiment, responsive to a determination that the second Internet telephone system is not currently logged onto the Internet, the first user at the first Internet telephone system is offered a choice of whether or not to leave a voice message for the second user. Typically this choice is generated locally at the calling Internet telephone, although it may be generated at some intermediate server in the network, which is used for routing and placing calls. Assuming that the option to leave voice mail is accepted, a communications link is established over the Internet between the first Internet telephone system and the voice mail system in order to transmit the voice message for said second user. Said communications link permits two-way communications, and the voice mail system transmits to the first user one or more prompts or greetings prior to transmission of the message for the second user to the voice mail system. Typically, said one or more prompts or greetings include information concerning the current location and availablity of the second user.

In the preferred embodiment, said voice message can be retrieved by the second user either using an Internet telephone over the Internet, or by using a conventional telephone over the conventional phone network. It is generally most convenient if substantially the same prompts and/or greetings are heard irrespective of whether the voice mail system is accessed by the conventional telephone network or over the Internet, since a subscriber is then only required to keep a single set of prompts/greetings up-to-date. Note that one of the advantages of the invention is that any voice mail stored in the voice mail system can be immediately available for retrieval by the mailbox owner, either by calling the voice mail system by telephone, or by Internet access (not necessarily via the mailbox owner's normal access provider).

Preferably the method further comprises the step of the voice mail system providing said second user of said second Internet telephone with an indication that he or she has a voice mail message waiting next time said second user logs on to the Internet using an Internet telephone. This indication may be provided in response to a request from the Internet telephone, which is transmitted from the Internet telephone to the voice mail system at start-up time.

It is also preferred that this indication is provided irrespective of whether said voice mail message was received over the Internet or over the conventional telephone network. This emphasises one of the important aspects of the invention, namely that a subscriber should only need one voice mail service, and the distinction between the Internet and the conventional telephone network should be essentially transparent to the subscriber.

The invention also provides a telephone system for making a telephone call over the Internet, including:

means for allowing a first user at said system to select a second user at a second Internet telephone system to call;
means for determining that said second Internet telephone system is not currently logged onto the Internet to receive said call; and
means for transmitting a voice message for said second user over the Internet to a voice mail system distinct from said second Internet telephone system for subsequent retrieval by said second user.

The telephone system will typically include means for displaying one or more prompts or greetings to said first user in textual or graphical form, although usage of this facility may be somewhat restricted in order to maintain conformity between the access mechanisms over the Internet and over the conventional telephone network.

The invention further provides a voice mail system including:

means for receiving a voice message over the Internet;
means for storing said voice message;
means for allowing said message to be retrieved over the Internet; and means for allowing said message to be retrieved over the conventional telephone network.

In a preferred embodiment, the voice mail system further comprises means for transmitting one or more prompts or greetings prior to receipt or retrieval of a voice message over the Internet. Generally said one or more prompts or greetings transmitted prior to receipt of a voice message over the Internet may include information concerning the current location and availablity of the intended message recipient.

It is preferred that the voice mail system includes means for receiving a voice message over the conventional telephone network, which can then be accessed and processed in the same manner as messages received over the Internet. Thus a message received over either the Internet or the conventional telephone network can be retrieved over either the Internet or the conventional telephone network without restriction. Further in this vein, preferably there are means for providing an indication to a subscriber that he or she has one or more voice mail messages waiting, and this indication includes both voice messages received over the Internet, and voice messages received over the conventional telephone network.

In the preferred embodiment, said means for receiving a voice message over the Internet includes means for receiving a voice signal in compressed form split into a plurality of packets, and means for assembling the voice signal and decompressing it into said voice message; and said means for allowing a voice message to be retrieved over the Internet includes means for compressing the message and packetising it prior to transmission over the Internet.

Preferred embodiments of the invention will now be described by way of example only, with reference to the following drawings:

Figure 1 is a schematic diagram of a voice mail system;
Figure 2 is a schematic diagram of the prior art arrangement of software running on the voice mail system;
Figure 3 is a schematic diagram of an Internet telephone system;
Figure 4 is a schematic diagram of the audio processing components on the audio adapter card of the Internet telephone system of Figure 3;
Figure 5 illustrates the user interface presented by the Internet telephone system of Figure 3;
Figure 6 is a schematic diagram of two Internet telephone systems and a voice mail system connected by the Internet;
Figure 7 is a schematic diagram of the enhancements made to the voice mail system software, compared with that shown in Figure 2, in order to receive voice messages over the Internet in accordance with the present invention.

Detailed Description

Figure 1 is a simplified diagram of a voice mail system based on a conventional computer workstation comprising system unit 20, display screen 12, and keyboard 14. The system unit includes a microprocessor 22, ROM/RAM 24, and disk storage 26, connected together via bus 28. In order to operate as a voice mail unit, the computer workstation is connected to telephone line 66 via a digital trunk processor 64 and a digital trunk adapter card 62. The voice mail system also includes a network adapter card 30 to connect the voice mail system to a computer network (eg a LAN). The voice mail system may also include other known components, such as an audio capture/playback card, a CD-ROM and mouse (not shown).

The voice mail system illustrated in Figure 1 is based on the DirectTalk/6000 voice processing system, available from IBM Corporation. The hardware components of this system are an RISC System/6000 computer workstation, plus the digital trunk processor and digital trunk adapter. The DirectTalk/6000 system connects to the telephone network through a T1 or El digital trunk line 66 (and via a PBX in most installations). The digital trunk processor is used to demultiplex incoming signals and multiplex outgoing signals on the trunk line, and perform log-linear conversion as appropriate. The digital trunk processor is also used to perform compression/decompression. The digital trunk adapter effectively acts as an interface between the workstation itself and the digital trunk processor. Further details about the DirectTalk/6000 voice processing system can be found in the manual "IBM AIX DirectTalk/6000, General Information and Planning" (publication number GC33-1720-00) and the other manuals referenced therein.

Figure 2 is a simple block diagram of the main software components running on the voice mail system of Figure 1. Running on the RISC System/6000 workstation is first of all the operating system for the workstation, which in the present case is AIX 110, and then the DirectTalk/6000 software 120 itself. The latter includes a device driver for the telephony hardware (the digital trunk adapter). Also on the RISC System/6000 workstation, effectively running as an application on top of the DirectTalk/6000 system, is DirectTalkMail 130, which provides standard voice messaging capabilities.

It will be appreciated that voice mail systems such as that illustrated in Figures 1 and 2 are well-known, and that many variations on the system illustrated are also well-known. For example, there are many systems in which telephone line 66 is analog, in which case the digital trunk processor is often omitted, and its functions performed, where appropriate by a suitable telephony adapter card. The precise hardware configuration employed in the voice mail system is not relevant to an understanding of the present invention, and so will not be discussed further.

The DirectTalkMail voice messaging system itself can be considered as a form of voice database system, based on mailboxes. Thus each user or subscriber has a mailbox, which has associated with it all the information for that user, eg their extension number, their password, the number of new messages that they have, their current greeting, and so on. The mailbox also logically contains the digitised stored messages for that user (although physically the audio recording may be stored in a different location from the other information). Each mailbox has a unique identifier, such as a number or name, for example, each mailbox can be allocated the extension number of the user associated with that mailbox. The DirectTalkMail voice messaging system also contains routines to allow callers to telephone messages into the database and subscribers to extract messages from the database for listening over the telephone, as well as other functions such as forwarding messages. The operation of a voice mail system in such a manner is well-known and so will not be described further.

Figure 3 is a simplified schematic diagram of a client computer system which may be used for telephone transmission over the Internet. The computer workstation of Figure 3, which is similar to that of Figure 1, but generally less powerful, has a system unit 310 including microprocessor 322, semi-conductor memory (ROM/RAM) 324, hard disk 326, and a bus over which data is transferred 328. Other typical components of the computer are a display 312, keyboard 314, and mouse (not shown). The computer of Figure 3 may be any conventional workstation, such as an Aptiva computer, available from IBM Corporation. Alternatively, any other form of suitable Internet access device, including the new generation of low-cost systems (effectively sub-PCs) which are currently being developed may be employed instead as the client telephone terminal.

The computer of Figure 3 is equipped with two adapter cards. The first of these is a network adapter card 330. This card, together with accompanying software such as the TCP/IP communications stack, allows messages to be transmitted onto and received from a computer network such as a Local Area Network (LAN). In this case, it is presumed that the Internet is accessed via the LAN. Alternatively (or additionally), the computer of Figure 3 may have a modem (not shown), installed either internally as another adapter card, or externally, for example via the RS-232 serial port. The modem in turn can be connected to a telephone socket, allowing dial-up access to an Internet provider. The operation of a network adapter card or modem to provide Internet access is well-known, and so again will not be described in detail.

The second card shown in Figure 3 is an audio card 362 which is connected to a headset including microphone 366 and earphones 364 for audio input and output respectively (alternatively the computer in Figure 3 may have a loudspeaker, and built-in microphone, but the use of a headset is preferred to optimise the quality of the audio signal produced and actually heard).

The audio card is shown in more detail in Figure 4. The card illustrated and used in the preferred embodiment is an M-Wave card available from IBM Corporation, although other cards are commercially available that perform an analogous function. The card 362 contains an A/D converter 442 to digitise incoming audio signals from the attached microphone 366. The A/D converter is attached to a codec 444, which samples the incoming audio signal into samples (eg 8 bit or 16 bit). Digitised samples are then passed to a digital signal processor (DSP) 446 on the card via a buffer 448 where they are compressed. The DSP is controlled by one or more programs stored in semiconductor memory 452 on the card. Data can be transferred by the DSP to and from the main PC bus.

Audio signals to be played out are received by the DSP 446 from the PC bus 328, and processed in a converse fashion to incoming audio. That is, the output audio signals are passed through the DSP 446 for decompression, and then through buffer 450 to the codec 444, from there to a D/A converter 454, and finally to a loudspeaker 364 or other appropriate output device.

There are various compression/decompression techniques that are available for audio communication over the Internet. The preferred embodiment uses a modified version of GSM, which is the standard compression technique used in Europe for cellular phones, to provide voice compression. Another possible technique to use is the DigiTalk system developed by Rockwell Communications. Such technologies as these reduce the bandwidth requirements for voice communications to less than 10 kbits per second. Note that although the embodiment shown in Figure 4 performs the compression/decompression on a DSP, in other embodiments this processing may be performed purely in software on the host computer.

In order to operate as an Internet telephone, the computer system of Figure 3 must contain appropriate application software. In the preferred embodiment, this application software is called WebTalker, and provides a user interface as shown in Figure 5. This interface includes message panel 505, which can be used to pass status messages to the user (eg "dialling", "engaged", etc), and a set of control buttons 510. The Call button 511 allows a call to be made to a named individual, whilst the User List button 512 provides access to a server directory (see below), as well as to a listing of people who have been called previously (or from whom calls have been received). Clicking on the name of a desired individual from one of these lists results in a call being made to the corresponding email address for that individual. The remaining control buttons, Set-Up 513, Help 515, Mute 514 and Exit 516 are substantially self-explanatory, and will not be described further since they are not directly relevant to an understanding of the present invention.

The window also contains slider bars, which can be used to control Volume 530, Voice Quality 531, and Microphone Sensitivity 532. It will be appreciated that it is possible to improve voice quality for example by reducing the degree of compression, or by increasing the amount of buffering at the receiving end, although these will tend to lead to increased delays. Finally, the window contains Mail Waiting light 520 and Voice Mail light 525, which will be explained in more detail below.

The operation of the Webtalker telephone is as follows (see Figure 6). When the user starts up the Webtalker software telephone client 620 on the Internet 600, the telephone logs onto a server 650 on the Internet. There may be multiple possible servers, but a default one is chosen in accordance with the set-up parameters of the Internet telephone. It will be appreciated that this requires the Web telephone to use standard Internet communications facilities, as well-known in the art.

The server maintains a list of people currently logged on to the Internet and using Webtalker. This list is accessible to the user (via the User List control button), allowing the user to select another party to call. The called party then receives notification of their incoming call in their control window 505, plus a pop-up box asking them whether or not they wish to accept the call. Assuming the call is accepted, then this establishes a direct link between the two clients (ie the two Internet telephones) independent of the server. This link provides a TCP/IP control channel and a UDP data channel. The two clients can start audio communications, using their audio subsystems as described above. Thus the audio data from each client is processed into a compressed form, and then transmitted over the UDP data channel to the other client. The audio communications can either be full duplex, or, to save bandwidth, half duplex.

In practice although the audio quality across the network is acceptable, the end to end delay Internet may be a couple of seconds or more, which means that completely natural conversation is not possible. Thus it can become confusing if one party tries to interrupt another, or does not clearly indicate when he or she finishes speaking. However, once a user is accustomed to these limitations, then an Internet telephone conversation becomes a very valuable method of communication.

As so far described, the WebTalker application has substantially the same function as known Internet telephones, and so its operation will be well understood by the skilled person. However, in accordance with the present invention, the WebTalker telephone includes the additional facility of allowing a voice message to be left when the called party is not logged onto the Internet. This is to be contrasted with the situation with known Internet telephones, which at the called end provide a voice mail message when the called party is logged onto the network, but choses not to answer an incoming call. As explained previously, most users, particularly those at home, do log-off from the Internet when not specifically using it to avoid paying unnecessary connection charges.

There can be several reasons why a call might be made to a party who is not currently logged on. One possibility is for the server list to be in error, perhaps because of a recent failue in the network, which has not yet been reflected in the list. Another possibility is that a caller exited abnormally from the Webtalker application (eg they crashed out), and so did not trigger the normal log off procedure at shut-down (normally the server would eventually time out with respect to these terminals, and eventually log them off anyway). Another possibility is that the call is made using the Call button 511, with the address of the called party directly entered, therefore by-passing the server list of users currently logged on (not all Internet telephones enable this).

Figure 6 illustrates the situation where client 1 has a first WebTalker telephone 620, connected to the Internet 600 typically via a modem and telephone line 625. Likewise client 2 has a WebTalker telephone 630, and a similar dial-up connection 635 to the Internet. However, client 2 is not currently logged onto the Internet, so this connection is shown in a dashed line.

At this point, when client 1 tries unsuccessfully to make a call to client 2, the WebTalker telephone of client 1 provides a status message indicating that the call could not be made because client 2 is not currently logged onto the network. Further, the WebTalker telephone invites client 1 to leave a voice mail message for client 2. Assuming that client 1 does opt to leave a voice mail message, then the WebTalker telephone dials up the voice mail system 610 in exactly the same manner as dialling any other WebTalker telephone on the Internet.

In the preferred embodiment, the option of sending voice mail is therefore controlled by the calling Internet telephone without reference to the server. However, as an alternative, the server may be involved. For example, in addition to presenting a list of users currently logged on, the server may present an additional list of callers for whom voice messages can be left. Selecting a caller from this list results in a call being made to the voice mail system, in the same manner that selecting a currently logged on user results in a call to that user. In such circumstances, it may in fact be desirable for the voice mail system itself to log onto the server.

In general, the voice mail system 610 will have the capability to handle many lines of incoming traffic, and have a permanent connection into the Internet. The voice mail system 610 will typically be managed by a service provider. Client 2 may pay a subscription for this service, or it might perhaps be included free with an Internet access package, or telephone line rental, in order to encourage usage. Similarly, an organisation marketing Internet telephones might offer the voice mail service to attract users to their particular offering. Alternatively, a company might provide the voice mail system 610 for all its employees.

It is assumed that client 1 was informed of the Internet address of the voice mail system 610 for client 2 at the same time that it learned of the Internet address of client 2, and that these have been stored together in a directory on client 1. It may also be possible for client 1 to interrogate the server 650 to find out the address of the voice mail system for client 2; or as suggested above, the connection may in fact be initiated through the server. Alternatively it may be that all subscribers to the WebTalker Internet telephone may use the same voice mail service. However, in the event that the WebTalker telephone is unable to determine the address of the voice mail system for client 2, then an appropriate error message is supplied to client 1.

Figure 7 illustrates the enhancements necessary to the voice mail system 610 in order to allow it to receive calls from the WebTalker telephone. As shown previously in Figure 2, the DirectTalkMail system 730 is essentially an application running on top of the DirectTalk/6000 voice processing system 720, which itself is an application running on top of the AIX operating system 710 to provide recording and playout of voice segments etc. Included within the DirectTalk voice processing system is a custom server interface 725, which allows C programs to interact directly with the DirectTalk/6000 voice database. Use of the custom server interface is required in the present instance since the voice messages are not being played out over the standard telephone interface (via the digital trunk adapter), but rather transmitted in software format over the Internet. The custom server interface is described in more detail in the manual IBM AIX DirectTalk/6000 Voice Application Development (reference SC33-1722-00).

In order for the DirectTalkMail to receive WebTalker telephone calls, appropriate software has to be provided. This software can effectively be split into two components, with an interface between them. The first component provides the WebTalker interface 750, and includes communications software 760 to allow communications over the Internet (nb some of the software necessary for this is already provided in the operating system 710, as known in the art). The WebTalker interface software 750 ensures that to a client WebTalker telephone, such as WebTalker telephone 620 (see Figure 6), the voice mail system can appear substantially similar to another WebTalker telephone. Therefore, when the voice mail system needs to play out a prompt over the Internet, the WebTalker IF component 750 is responsible for compressing the audio, packaging it correctly, and transmitting it out over the Internet to the client system. Conversely, for incoming audio from the client system, the WebTalker IF component is responsible for decompressing, buffering, and assembling the received packets into a proper audio signal. Note that this processing may be done all in software (ie there is no need for special hardware at the voice processing system such as that shown in Figure 4).

The DirectTalkMail interface component 740 is therefore passed incoming audio signals in PCM format from the WebTalker interface (A law, mu law, or any other suitable digital audio format could be used), and uses the DirectTalk/6000 custom server interface to store these as voice messages in the DirectTalk/6000 voice database. Similarly, the DirectTalkMail interface component uses the custom server interface to retrieve stored prompts and voice messages, converts to the appropriate format (eg PCM) and passes these over to the WebTalker interface component for compression, and transmission to the client. Each mailbox in the voice database may have an assigned Internet address, so that calls intended for a client at a particular Internet address (as specified by the information received from the WebTalker interface component) can be stored in the correct mailbox (alternatively incoming calls could be required to specify a mailbox number). It will be appreciated that in common with most voice mail systems, DirectTalkMail actually performs its own compression on voice messages prior to storage, to reduce storage requirements. The compression technique used by the voice mail system differs from that used by the WebTalker telephone system, hence the need to decompress and the recompress incoming calls (and similarly for outgoing calls), although it is certainly feasible for a voice mail system and Internet telephone to use the same compression scheme throughout.

The most basic embodiment of the present invention simply allows for the two-way transmission of audio, enabling the caller to hear a greeting or prompt, and then to leave a message. However, it will be appreciated that the DirectTalkMail system, in common with most voice mail systems, is normally controlled by a user pressing DTMF keys on their telephone to select between multiple commands or options. In order to provide this control to a client over an Internet, a facility is added to the WebTalker telephone, such that when it dials a voice mail system, a telephone keypad is presented to the user of the WebTalker telephone. By clicking on the desired button (ie digits 0-9, * or #), the user of the WebTalker telephone can transmit a control command to the voice mail system. The selected command is not transmitted as a DTMF signal, but rather as a simple message containing an identifier of the pressed key, since this requires far less bandwidth. This identifier is distinguished from normal audio data, so that it is properly recognised by the WebTalker interface component 650. The identifier is then passed to the DirectTalkMail interface, which interacts with the DirectMail and DirectTalk/6000 systems to ensure that the requested function is performed.

The two-way audio exchange between the voice mail system and the client Webtalker telephone, together with the facility for the client to send and the voice mail system to accept identifiers corresponding to DTMF tones, provide a full function voice mail service, thereby allowing (for example) client 1 to leave a voice mail message for client 2, when client 2 is not logged onto the Internet to personally receive a telephone call. It will be appreciated that voice mail system 610 can typically accept many incoming calls simulatneously. Indeed, unlike conventional operation of a voice mail system, which is limited by a predetermined number of telephone ports, there is not necessarily any hard limit on the number of software Internet calls which can be handled at the same time (rather, as the number of calls increase, performance will eventually start to degrade, due to the finite processing power available at the voice mail system, or the limited bandwidth of the voice mail Internet connection, or both).

Once a voice mail message has been stored within the voice mail system 610, then there are several possible mechanisms for retrieving it. Firstly, the message can be retrieved in conventional fashion over the normal telephone network. Thus the user simply dials the telephone number associated with the voice mail system, and can then access any messages they have by responding to appropriate prompts, and (generally) entering a password.

Another possibility is that the message can be retrieved over the Internet using a Web browser with audio capability. This can be implemented by having the voice mail system act as a WWW site, with universal resource locators pointing to a user mailbox, and the mail messages within. This approach is described in more detail in PCT application PCT/GB95/02009, and is also provided by the currently available release of the DirectTalk/6000 and DirectTalkMail products (see the manual IBM AIX DirectTalkMail Administration, reference SC33-1733-00). As a slight variation on the approach described therein, once a message has been selected, the Web browser may also provide the option of having this message delivered to the user's Internet telephone system (which will generally be running on the same machine as the browser). It may be advantageous to have the Internet telephone system play the message (rather than the browser itself), for example because the former allows the message to be subsequently forwarded, or will allow the caller's address to be stored in the user's directory.

The preferred embodiment provides another possibility, that of accessing the stored voice mail message from a WebTalker telephone. Thus as part of its start-up procedure, the WebTalker telephone interrogates the voice mail system associated with that client (this can be specified as part of the set-up procedure). The Webtalker telephone client therefore sends a query to the voice mail system, which is received by the WebTalker interface component and passed into the DirectTalkMail system. The DirectTalkMail system then identifies the mailbox corresponding to the specified Internet address for that client, and returns an indication of whether or not there are any new messages for that user. This indication is returned by the WebTalker interface component to the requesting WebTalker telephone, and if positive, the message waiting indicator on the WebTalker telephone client screen is activated (eg by highlighting or making a different colour).

In order to access the voice messages, the client Internet telephone then places a call over the Internet to the voice mail system. This is done by simply pressing the voice mail button, which automatically dials the default voice mail system for that client. This establishes a two-way audio call between the client Internet telephone and the voice mail system. The call can now proceed in analogous fashion to that described above for entering a voice message into the system. Again, the voice mail system plays various prompts to the user, who is provided with a simulated DTMF kepad on his or her screen. The user selects the desired button, and can navigate through the voice mail system, including typically some password protection, to obtain access to their voice mail, which can then be played out to them over the Internet. Thus the user can have immediate access to their voice mail, either via the conventional telephone network, or from any location on the Internet.

It will be appreciated that many variations on the above approach are possible. For example, instead of using a simulated DTMF kepad for inputting control commands, it may be possible to include a voice recognition facility with the voice mail system to allow a client to simply speak his or her desired choice. Another possibility is for all prompts from the voice mail system to be available for transmission in text form to a client, and displayed there textually or graphically, eg perhaps using a menu structure, thereby significantly reducing bandwidth requirements and overall delay. It is also possible for DirectTalkMail to exploit the enhanced user interface of the client (compared to a conventional telephone) to provide information in a more convenient format to the client. For example, rather than asking a user to "Press 1 to hear messages" (say), the system might simply display the command "Listen to messages", with associated button. Selecting this button would then result in the voice mail system taking the same action as if a "1" had been pressed on a conventional telephone. The enhanced capabilities of the screen interface may be further exploited to provide the user with features that are not directly available when calling from a conventional telephone. For example, the voice mail system could send a text list of stored messages, together with information such as the sender of the message, to allow a user to immediately assess all his or her outstanding mesages (nb this is the approach taken with the voice mail Web browser access described above).

It will be appreciated that although the systems described so far transmit telephone calls entirely over the Internet, for some Internet telephone systems it is in fact possible to make hybrid telephone calls. In these cases the originating or destination telephone may in fact be a conventional telephone, with one or more servers acting as interface units between the Internet and the conventional telephone network (indeed, the voice mail system itself may possibly be so connected). As used herein, the term "Internet telephone" is generally meant to cover such systems, and references to transmission over the Internet may include transmission over the conventional telephone network along some of the route.

It is possible for the voice mail service described above to be invoked not only when the called party is not logged onto the Internet, but even when they are logged on, but unable or unwilling to accept the call at the time that it is received. This would avoid the need for an Internet telephone to provide a local voice mail service. In this case, the Internet telephone, in refusing to accept a call, could transmit back to the calling party the Internet address of its preferred voice mail system.

There are also many possibilities for the action to be taken by the voice mail system on receipt of an incoming voice message. For example, it might try to page the intended recipient, or possibly place a telephone call through to them at a previously indicated location. Another possibility is to send an email notification to them, to be read next time they log on to the Internet.

It will be recognised that one of the drawbacks with current Internet telephones is that systems from different vendors are generally incompatible with one another. Although efforts are being made to provide standardisation, it will be noted that the voice mail system of the present invention may easily be adapted to support multiple formats. The simplest way of achieving this is to have software which is the equivalent of the WebTalker interface 750 and communications layer 760 for each different telephone format, with incoming/outgoing calls being identified and passed to the correct interface software.

An important aspect of the invention is that allows a subscriber to maintain a single voice mailbox, accessible either over the Internet, or the conventional telephone network. Thus a user has only a single greeting response to maintain (eg to inform callers if the user is eg in a meeting, away on vacation, or whatever). Furthermore, voice mail messages may be treated in exactly the same manner, whether received over the Internet or conventional telephone network, and whether accessed and retrieved via the Internet or the conventional telephone network. For example, when a user may notified of the number or existence of new or stored messages, without any distinction as to the origin receipt mechanism of the messages. The voice mail system of the invention therefore offers the user greater power and flexibility, without any corresponding increase in complexity.

Claims

1. A method of providing voice mail on the Internet comprising the steps of:

   selecting to make a call from a first user at a first Internet telephone system to a second user at a second Internet telephone system;
   determining that said second Internet telephone system is not currently logged onto the Internet to receive said call; and
   transmitting a voice message for said second user over the Internet to a voice mail system distinct from said second Internet telephone system for subsequent retrieval by said second user.

2. The method of claim 1, wherein responsive to a determination that the second Internet telephone system is not currently logged onto the Internet, the first user at the first Internet telephone system is offered a choice of whether or not to leave a voice message for the second user.

3. The method of claim 1 or 2, further comprising the step of establishing a communications link over the Internet between the first Internet telephone system and the voice mail system in order to transmit the voice message for said second user.

4. The method of claim 3, wherein said communications link permits two-way communications, and the voice mail system transmits to the first user one or more prompts or greetings prior to transmission of the message for the second user to the voice mail system.

5. The method of claim 4, wherein said one or more prompts or greetings include information concerning the current location and availablity of the second user.

6. The method of claim 4, wherein said information concerning the current location and availability of the second user is also heard if the voice mail system is accessed by the conventional telephone network.

7. The method of any preceding claim, wherein said voice message can be retrieved by the second user either using an Internet telephone over the Internet, or by using a conventional telephone over the conventional phone network.

8. The method of any preceding claim, wherein said second user needs to supply a password to the voice mail system prior to retrieval of said voice message.

9. The method of any preceding claim, further comprising the step of the voice mail system providing said second user of said second Internet telephone with an indication that he or she has a voice mail message waiting next time said second user logs on to the Internet using an Internet telephone.

10. The method of claim 9, wherein said indication is

provided in response to a request from the Internet telephone, which is transmitted from the Internet telephone to the voice mail system at start-up time.

11. The method of claim 9 or 10, wherein said indication is provided irrespective of whether said voice mail message was received over the Internet or over the conventional telephone network.

12. A telephone system for making a telephone call over the Internet, including:

means for allowing a first user at said system to select a second user at a second Internet telephone system to call;
means for determining that said second Internet telephone system is not currently logged onto the Internet to receive said call; and
means for transmitting a voice message for said second user over the Internet to a voice mail system distinct from said second Internet telephone system for subsequent retrieval by said second user.

13. The telephone system of claim 12, further including means responsive to a determination that the second Internet telephone system is not currently logged onto the Internet, for offering the first user the option of leaving a voice message for the second user.

14. The telephone system of claim 12 or 13, further including means for establishing a communications link over the Internet to a voice mail system associated with said second user in order to transmit a voice message for said second user.

15. The telephone system of claim 14, wherein said communications link permits two-way communications to allow said first user to receive one or more prompts or greetings prior to transmission of the message for the second user.

16. The telephone system of claim 15, further including means for displaying one or more prompts or greetings to said first user in textual or graphical form.

17. The telephone system of any of claims 12 to 15, further including means for receiving from the voice mail system information that said first user has one or more new messages in the voice mail system, and means for providing a visual indication accordingly to said first user.

18. The telephone system of claim 17, further including means responsive to the start-up of said telephone system for sending a request to the voice mail system in order to receive said information whether the first user has one or more new messages.

19. A voice mail system including:

means for receiving a voice message for over the Internet;
means for storing said voice message;
means for allowing said message to be retrieved over the Internet; and means for allowing said message to be retrieved over the conventional telephone network.

20. The voice mail system of claim 19, further comprising means for transmitting one or more prompts or greetings prior to receipt or retrieval of a voice message over the Internet.

21. The voice mail system of claim 20, wherein said one or more prompts or greetings transmitted prior to receipt of a voice message over the Internet include information concerning the current location and availablity of the second user.

22. The voice mail system of any of claims 19 to 21, further including means for receiving a voice message over the conventional telephone network.

23. The voice mail system of any of claims 19 to 22, wherein said means for receiving a voice message over the Internet includes means for receiving a voice signal in compressed form split into a plurality of packets, and means for assembling the voice signal and decompressing it into said voice message.

24. The voice mail system of any of claims 19 to 23, wherein said means for allowing a voice message to be retrieved over the Internet includes means for compressing the message and packetising it prior to transmission over the Internet.

25. The voice mail system of any of claims 19 to 24, further including means for providing an indication to a subscriber that he or she has one or more voice mail messages waiting.

26. The voice mail system of claim 25, wherein said indication includes both voice messages received over the Internet, and voice messages received over the conventional telephone network.

27. The voice mail system of claim 25 or 26, wherein said indication is provided to a subscriber using an Internet telephone system in response to a query received over the Internet from said system.

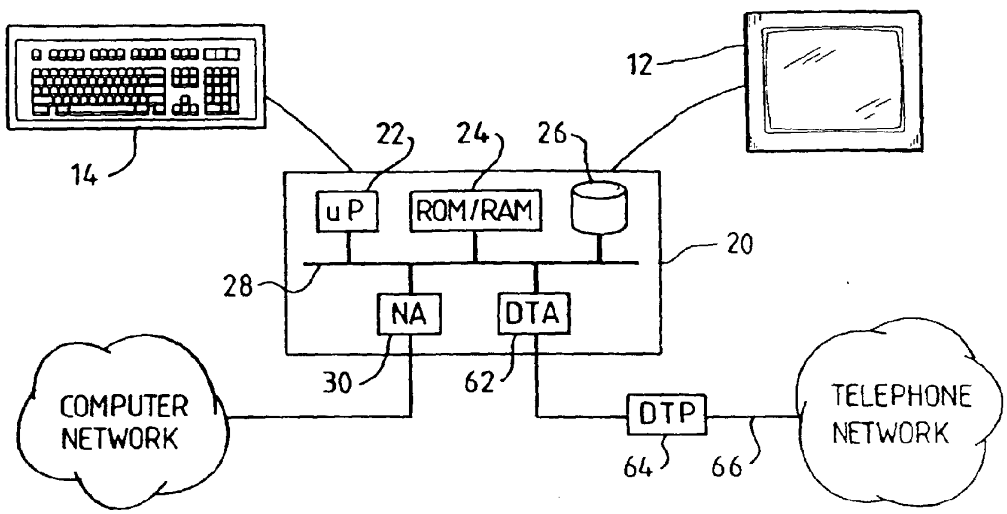
12
22
24
26
14
uP
ROM/RAM
20
28
NA
DTA
30
62
COMPUTER
NETWORK
DTP
TELEPHONE
NETWORK
64
66

FIG. 1

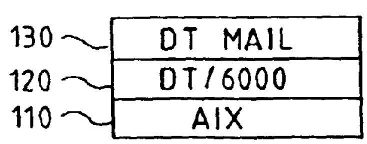
130
DT MAIL
120
DT/6000
110
AIX

FIG. 2

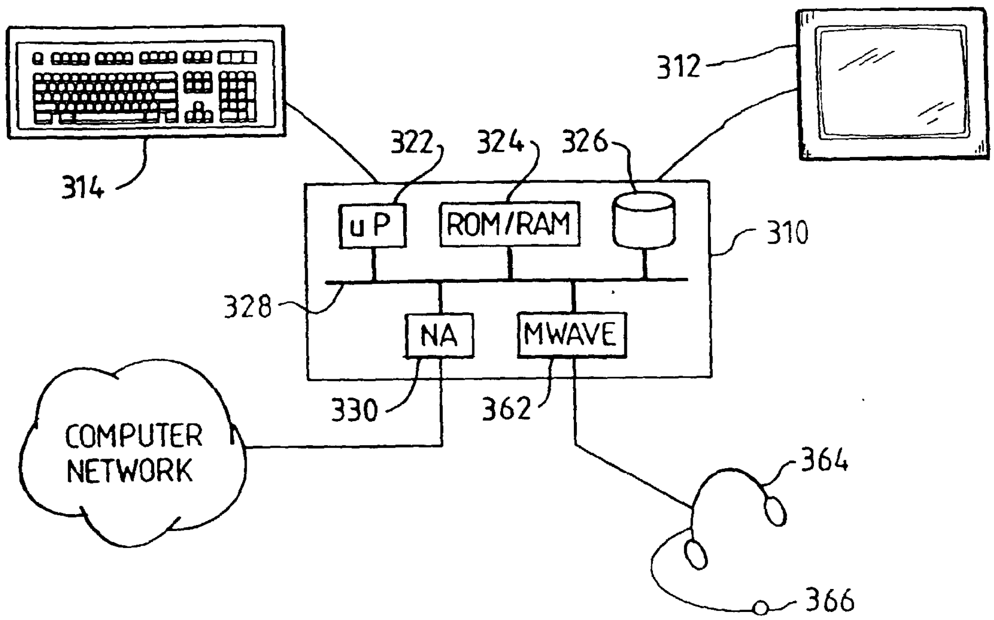
312
314
322
324
326
310
uP
ROM/RAM
328
NA
MWAVE
330
362
COMPUTER
NETWORK
364
366

FIG. 3

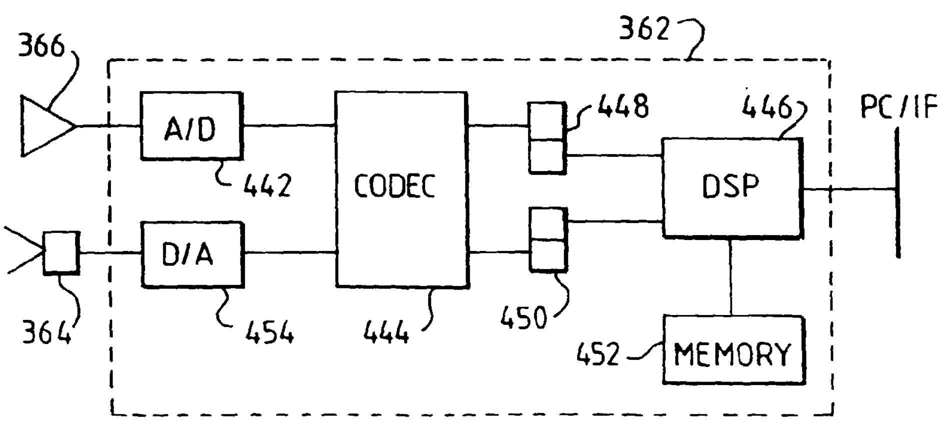
366
362
A/D
CODEC
448
446
PC/IF
442
DSP
D/A
364
454
444
450
452
MEMORY

FIG. 4

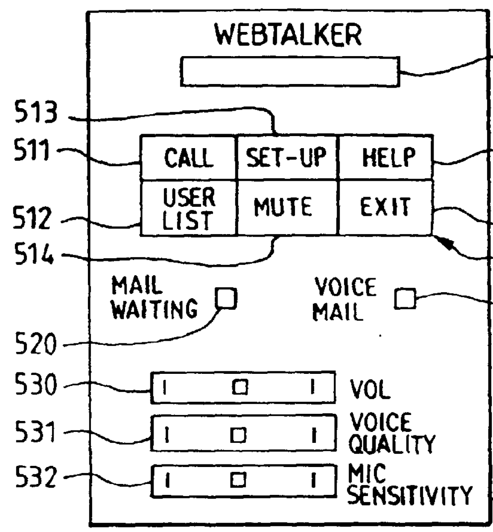

FIG. 5

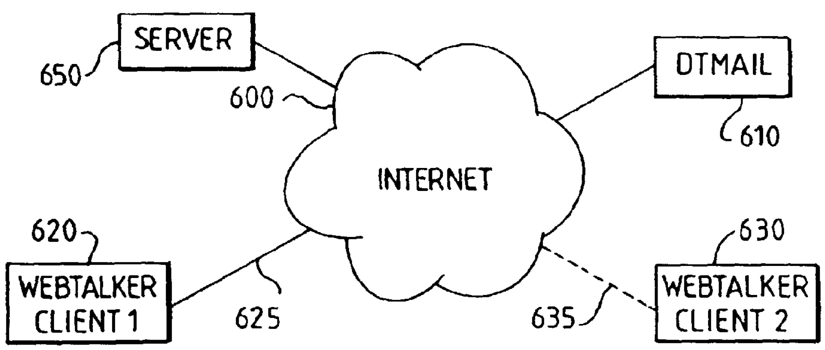

FIG. 6

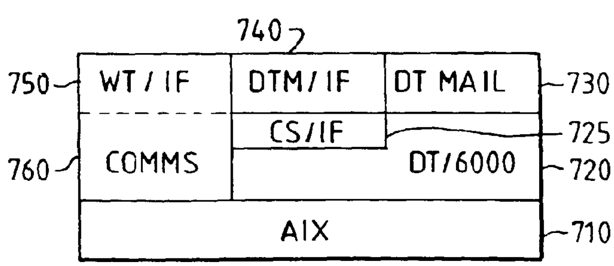

FIG. 7

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H04M 11/08

(52) UK CL (Edition J)
H4K KOD
H4F FDB FS24 FS33

(56) Documents cited
International Technology Disclosures May 88 (Published 25/5/88)

(58) Field of search
UK CL (Edition J) **H4K KOD KOF**
INT CL⁴ **H04M**
Derwent WPIL (online) INSPEC (RTM) online

(54) **Document distribution system with voice generation and facsimile**

(57) Printed or other pre-formed documents are delivered to requesters rapidly and at low cost by a microcomputer-based system which uses voice generation circuitry to invite a person calling in to the system to specify the document needed by pushing particular keys of his/her touch-tone telephone 40 or by spoken input. Upon ascertaining the identity of the document in question, the computer transmits the document to the caller in standard facsimile form, either over the telephone connection already established if the telephone call was initiated from a telephone connected with a facsimile machine, or to a different telephone number specified by the caller.

FIG. 1

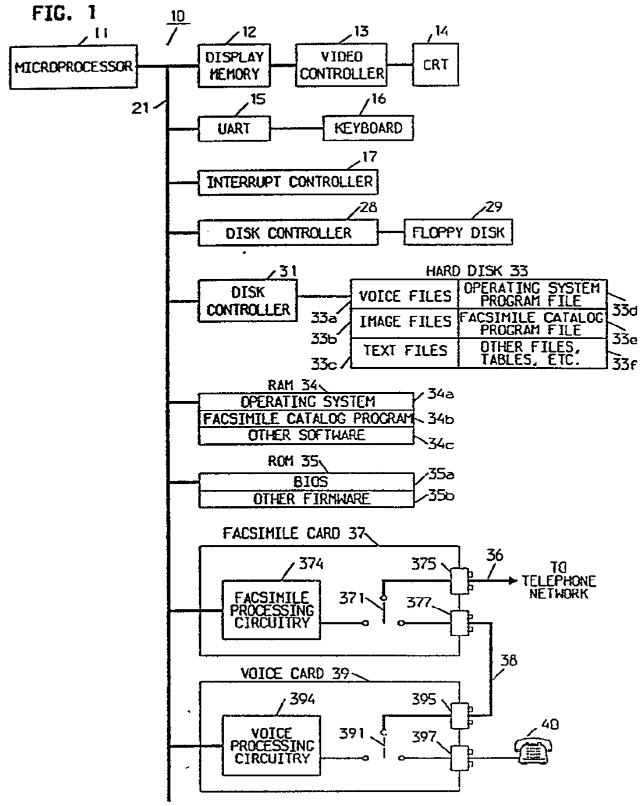

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FIG. 1

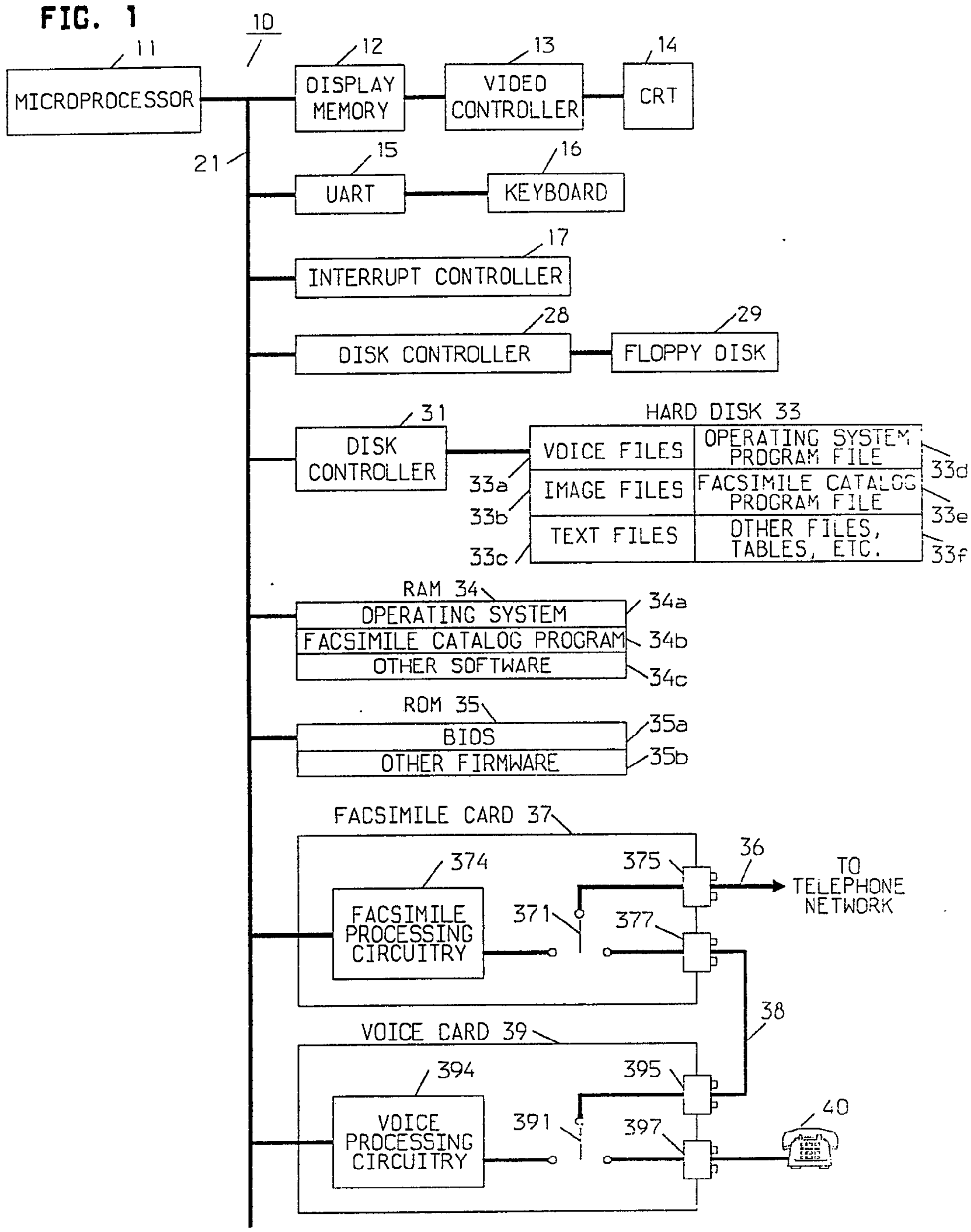


FIG. 2

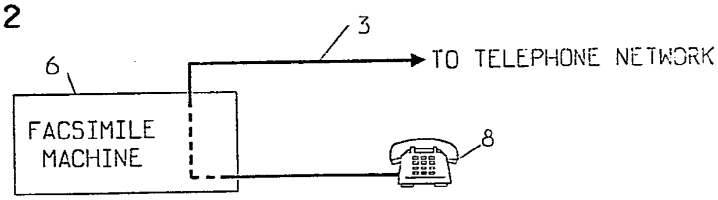
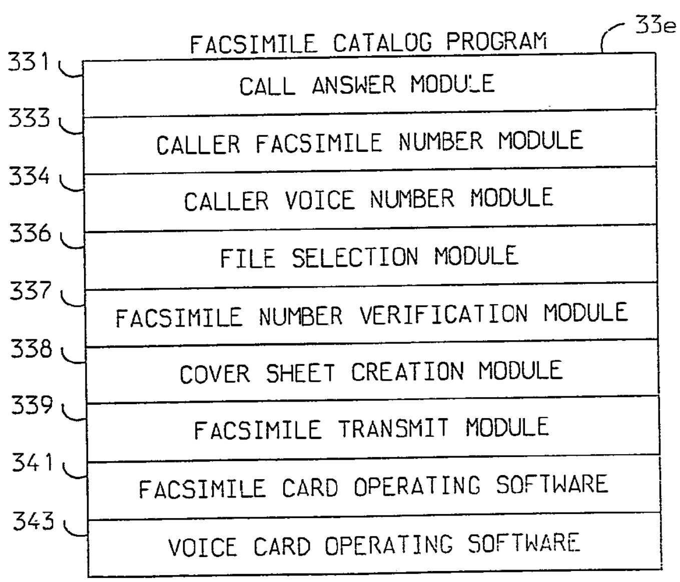

6
3
TO TELEPHONE NETWORK
FACSIMILE MACHINE
8

FIG. 4


FACSIMILE CATALOG PROGRAM
33e
331
CALL ANSWER MODULE
333
CALLER FACSIMILE NUMBER MODULE
334
CALLER VOICE NUMBER MODULE
336
FILE SELECTION MODULE
337
FACSIMILE NUMBER VERIFICATION MODULE
338
COVER SHEET CREATION MODULE
339
FACSIMILE TRANSMIT MODULE
341
FACSIMILE CARD OPERATING SOFTWARE
343
VOICE CARD OPERATING SOFTWARE

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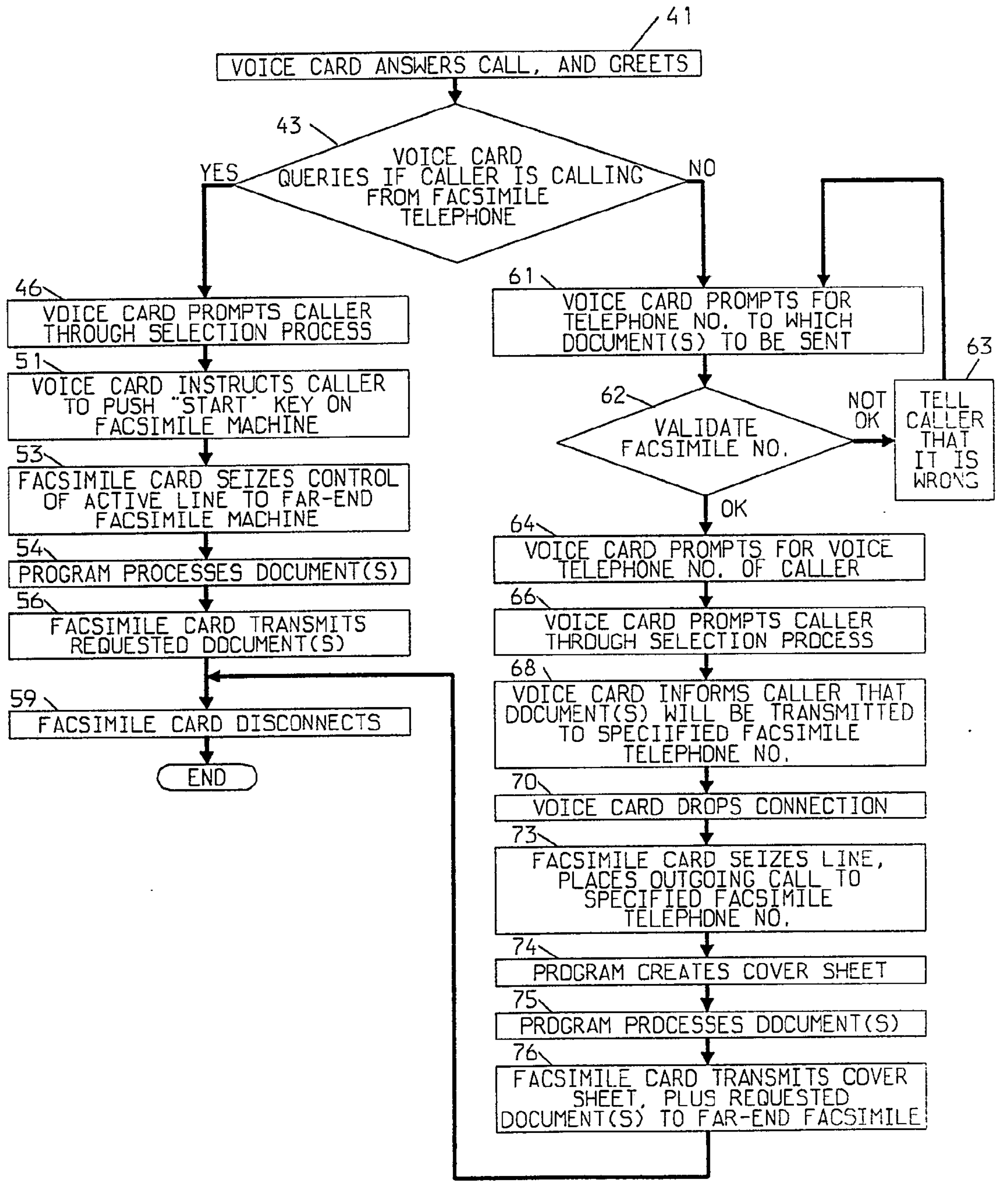
3/3

FIG. 3



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METHOD AND APPARATUS FOR TRANSMITTING DOCUMENTS

This invention relates to methods of and apparatus for transmission of documents.

It is a common and oft-repeated circumstance of business and consumer life that one needs to obtain a printed document from a supplier of same. Such documents range from government forms to integrated circuit "specification" sheets to airline schedules. A typical such transaction begins when the person needing the documents telephones the business establishment or agency in question and, having been put in contact with an order clerk, salesman, etc, requests the needed document. Typically, the document is delivered to the requester in the mail. When the requester is in a hurry, however, an "overnight delivery" service or, perhaps, private messenger may be used. Such expedited delivery mechanisms are relatively expensive, however.

The present invention is directed to a system, which is illustratively microcomputer-based, for delivering printed or other pre-formed documents to requesters rapidly and at low cost. In particular, the requester dials a telephone number associated with the system. The latter uses voice generation circuitry to invite the requester, hereinafter the "caller", to specify the document needed, such as by pushing particular keys of his/her touch tone telephone. Upon ascertaining the identity of the document in question, the computer transmits the document to the caller in standard facsimile form.

Advantageously, if the telephone call was initiated from a telephone connected with a facsimile machine--a fact that the system may ascertain by querying the caller during the telephone transaction--the facsimile data is communicated over the telephone connection already established. If, on the other hand, the telephone call was initiated from a telephone that is not connected with a facsimile machine, then, in preferred embodiments of the invention and in accordance with a feature thereof, the caller is prompted to enter the telephone number of a facsimile machine to which the requested document can be sent and the document is then sent there.

In accordance with a further feature of the invention, when the document is to be sent to a facsimile machine other than one connected with the callers telephone, the caller may also be prompted for caller-identifying data, such as his/her telephone number. This data is supplied by the system with the requested document--preferably on a separate cover sheet-thereby enabling an attendant at the receiving facsimile machine to identify the intended recipient.

The invention will now be described by way of example with reference to the accompanying drawings, in which:

FIG. 1 is a block diagram of a microcomputer system embodying the invention;

FIG. 2 is a block diagram of a standard facsimile machine station from which documents can be requested from the system of FIG. 1;

FIG. 3 is a flowchart of the processing performed by software within the system of FIG. 1; and

FIG. 4 is a memory map of a region of the hard disk used in the system of FIG. 1.

Referring now to FIG. 1 system 10 is based around a standard microcomputer and commercially available special-purpose circuit cards and provides individuals who call into the computer with a catalog service-specifically, in this example, the ability to request "specification" sheets for integrated circuits.

The system is illustratively based on an AT&T Model PC6300 personal computer, at the heart of which is a microprocessor 11 having address, data and control buses denoted generically as bus 21. Connected to bus 21 are a display memory 12 whose contents are used by a video controller 13 to generate video signals for a CRT 14; a universal asynchronous receiver transmitter (UART) 15, which serves as a serial interface between microprocessor 11 and a keyboard 16; an interrupt controller 17, to which hardware interrupt leads (not shown) extend, inter alia, from UART 15; a floppy disk controller 28, which serves as an interface between microprocessor 11 and a floppy disk memory 29, and a hard disk controller 31, which serves as an interface between microprocessor 11 and hard disk memory 33. The latter holds, inter alia, voice, image and text files 33a, 33b and 33c, respectively, as discussed in further detail hereinbelow; a copy of the workstation's operating system 33d--illustratively the MS-DOS® operating system; a copy of an application module, herein referred to as "facsimile catalog program" 33e, which, when executing, controls the system hardware; and a number of other files not here relevant indicated at 33f.

Also connected to bus 21 is random access memory (RAM) 34 and read-only memory (ROM) 35. When the system is in operation, RAM 34 holds the executed copies of a) the operating system, indicated as 34a, and b) facsimile catalog program, indicated as 34b, and c) other software not here relevant, indicated as 34c. ROM 35 contains the conventional Basic Input/Output System (BIOS) 35a as well as other firmware 35b.

Also connected to bus 21 are two circuit cards that play a central role in implementing the invention. One of these is voice card 39 which may be, for example, the card marketed by Natural Microsystems under the name WATSON. As called out in the drawing, voice card 39 includes input and output connectors 395 and 397, respectively, a switch 391 and voice processing circuitry 394, the latter comprising all the other circuitry on the card. When in one position, switch 391 simply interconnects connector 395 and 397, thereby providing a direct path through the card. When in its other position, switch 391 connects input connector 395 to the voice processing circuitry.

Voice card 39 has a number of capabilities. Among the capabilities relevant here are the ability to a) recognize touch tone inputs and report the identity of the digit or symbol represented thereby onto bus 21 and b) to re-create human speech from digitally stored versions thereof supplied from the bus and to output that speech to connector 395.

The other of the aforementioned circuit cards is facsimile card 37, illustratively the FAXCON-20 facsimile communication board available from AT&T. Similar to voice card 39, facsimile card 37 includes input and output connectors 375 and 377, respectively, a switch 371 and facsimile processing circuitry 374, the latter comprising all the other circuitry on the card. Among the capabilities of facsimile card 37 relevant here is the ability to accept files containing text and/or image data, convert that data into facsimile images and supply it, when switch 371 is appropriately thrown, to connector 375 using standard facsimile formats and protocols.

Circuit cards 37 and 39 are interconnected by way of a jumper cable 38 which connects facsimile card output connector 377 to voice card input connector 395. In addition, a telephone line 36 is connected to facsimile card input connector 375 and a standard telephone set 40 is connected to voice card output connector 397.

When the system is in an idle state, switch 371 within card 37 is set so as to interconnect facsimile card connectors 375 and 377 while switch 391 within card 39 is set so as to interconnect voice card connectors 395 and 397. Thus prior to the receipt of any telephone call, telephone line 36 is directly connected through both cards to telephone set 40.

Turning now to FIGS. 2 and 3, the facsimile station depicted in FIG. 2 is seen to comprise a standard facsimile machine 6 having connected thereto a standard touch-tone telephone 8 and a conventional telephone line 3 which extends into the public dialed network. When a person at telephone 8 wishes to be availed of the catalog service provided by system 10, he/she dials the telephone number associated with telephone 40. Having been instructed by facsimile catalog program 34b, which is now in control of the system, to be monitoring the incoming line, voice card 39 recognizes the ringing voltage and, as indicated at 41 of FIG. 3, answers the call. It does this, specifically, by causing its switch 391 to connect voice processing circuitry 394 to connector 395 and thence to the telephone line back through facsimile card 37. (Although not shown in the FIG., voice processing circuitry 394 has an internal connection to connector 395 that enables it to monitor the signals applied thereto, such as ringing voltage.)

Facsimile catalog program 34b at this point operates voice card 39 to deliver a sequence of messages to the caller. In each case, the message is delivered by instructing the operating system 34a to retrieve one of voice files 33a from hard disk 33 and route the file via bus 21 to voice processing circuitry 394 of voice card 39. The voice processing circuitry converts this file into audible speech which is thereupon transmitted to the caller via the telephone line.

System 10 illustratively provides to requesters "specification" sheets for integrated circuits manufactured by the (fictitious) XYZ Microchip Corporation, and, as further indicated at 41, the first message is a greeting, illustratively the greeting "You have reached the XYZ Microchip Corporation automated facsimile catalog." This is immediately followed by a second message, again delivered to the voice card via the above-outlined mechanism. As indicated at 43, this second message queries as to whether the caller is calling from a facsimile telephone, i.e., a telephone associated with a facsimile machine, the message illustratively being, "Touch '1' on your touch-tone telephone if you are calling from a facsimile telephone; otherwise touch any other key." The program now instructs the voice card to be receptive to an answering touch-tone input from the caller. The voice card, being capable of recognizing touch tone inputs, provides facsimile catalog program 34b with an indication of the caller's response.

Since in the present example, telephone 8 is, in fact, associated with a facsimile machine, the caller will touch "1" at this time. The program thereupon proceeds to step 46 at which it prompts the caller through a selection process so to elicit from the caller what specific information the caller wishes to receive. This process may involve several queries and responses, allowing the caller to be increasingly specific with respect to the desired information.

A typical such interaction might be:

FAX CATALOG:	Touch '1' for information on digital integrated circuits; touch '2' for information on analog integrated circuits.
USER:	[Enters '1'].
FAX CATALOG:	Touch '1' for 74LS series devices; touch '2' for 74LP series devices.
USER:	[Enters '1'].
FAX CATALOG:	Enter the 4-digit code of the device that you are interested in.
USER:	[Enters '7522'].

(In some applications, the caller may make selections based on information previously disseminated by the supplier, such as a printed index of available documents.)

As indicated at 51, facsimile catalog program 34b now causes the voice card to instruct the caller to now operate the "START" key on the facsimile machine with which the caller's telephone is associated. To this point, facsimile machine 6 has been providing a signal path from telephone line 3 directly through to telephone 8. However, responsive to the operation of the "START" key, facsimile machine 6 now disconnects telephone 8 from the line and connects the line to the internal facsimile circuitry of the machine. There should not be any source documents in the input tray of the caller's facsimile machine. Accordingly, the latter assumes the role of recipient in the upcoming facsimile transaction.

After a preprogrammed delay to allow the caller to comply with the instruction to operate the "START" key, facsimile catalog program 34b instructs facsimile card 37 to now operate its switch 371, causing facsimile card 37 to seize control of the active, already established telephone connection to the far-end facsimile machine, as indicated at 53. At this point, there is a direct telephone connection between facsimile circuitry 374 in facsimile card 37 and the facsimile circuitry in the caller's facsimile machine.

As also indicated at 53, facsimile card 37 now negotiates with facsimile machine 6, in the standard way, such parameters as line speed, transmission protocols and transmitter/receiver identity. Upon successful completion of these negotiations, facsimile card 37 notifies facsimile catalog program 34b of this fact. Proceeding then to step 54, the program then processes the requested document(s)--in this case the "specification" sheet for the "7522" microchip--in preparation for transmission to facsimile machine 6. In particular, it instructs the operating system to retrieve from hard disk 33 one or more of either image files 33b or text files 33c. These files contain the text and/or graphic images which comprise the document(s) that the caller has requested. In particular, each image file contains a document, or portion thereof, in bit-mapped form and, as such, is directly transmittable by facsimile card 37. A text file, by contrast, contains standard ASCII characters and must be converted by facsimile catalog program 34b to a bit-mapped form before being sent to the facsimile card. A particular document or package of documents may be stored as either an image file, a text file, or some combination of files to be assembled by the facsimile catalog program.

The various image files may be files that were, for example, created by scanning paper copies of documents through an electronic scanner (not shown) connected to the system; received from a remote location by facsimile transmission; or created on a frame creation system such as the AT&T PC Image Director® system. The text files may be files that were, for example, created from keyboard 16 or received from a remote location via a data communications link.

Having assembled the document(s) requested by the caller and, if necessary, converted them to bit-mapped form, facsimile catalog program 34b delivers them to facsimile card 37 via bus 21. Facsimile processing circuitry 374 converts these bit maps into facsimile images in accordance with the negotiated parameters. As indicated at 56, the resulting facsimile images are thereupon transmitted to the caller's facsimile machine 8 via the telephone network, using standard facsimile protocols such as CCITT Group III.

Having completed the facsimile transmissions, facsimile catalog program 34b then instructs facsimile card 37 to disconnect from the telephone line, as indicated at 59. The operation thus comes to an end, with the program now instructing facsimile and voice cards to assume their previous idle states.

Returning, now, to step 43, let us now assume that the caller is not at a telephone associated with a facsimile machine and therefore touches a key other than "1" in response to the step 43 prompt. Facsimile catalog program 34b thereupon proceeds to step 61 at which, in accordance with a feature of the invention, it prompts the caller for the telephone number of the facsimile machine to which the caller wishes to have the documents transmitted. As indicated at 62, the program validates the entered number, illustratively by verifying that it appears to be a valid telephone number; checking for the presence of an area code; stripping off the area code if it is the same as the area code of telephone line 36; and prepending an outside calling dialing code, such as "9", to the number in cases where telephone line 36 extends from a PBX. If any of these validation checks fail, the program so informs the caller, as indicated at 63, and then returns to step 61 to re-prompt the caller for the desired number.

(Similar validation checks may be provided, as desired, at various stages of the call, such as during the eliciting of the identity of the desired document at step 46 described above.)

Once the caller has supplied an apparently valid number, the program proceeds to step 64, where, for a purpose discussed hereinbelow, it prompts the caller for caller-identifying data, illustratively his/her voice telephone number, i.e., the number at which the caller receives his/her normal telephone calls.

As indicated at 66, the system now prompts the caller through a selection process similar to that carried out a step 46 in order to elicit from the caller what specific information the caller wishes to receive. The program then causes the voice card to inform the caller, at step 68, that the requested document(s) will be transmitted to the previously supplied facsimile telephone number and causes voice card 39 to disconnect from telephone line 36 by operating switch 391, as indicated at step 70.

After waiting a brief period of time, illustratively 2-3 seconds, to assure that the previously established telephone connection has been dropped, facsimile catalog program 34b, at step 73, instructs facsimile card 37 to seize telephone line 36 and place a new, outgoing telephone call to the facsimile number just supplied by the caller. This is accomplished by first having the facsimile card operate switch 371 to connect telephone line 36 to facsimile processing circuitry 374, and then having the latter dial the number. If the called facsimile number is busy or does not answer, the facsimile catalog program will wait a predetermined period of time, illustratively five minutes, and then initiate one or more retries.

Facsimile catalog program 34b now creates, at step 74, a cover sheet for the document(s) about to be transmitted. It does this by instructing the operating system to retrieve a cover sheet template, stored as one of text files 33c in hard disk 33, and modifying it to include the facsimile telephone number and caller's voice telephone number both previously supplied by the caller. As in the prior case, the program, at step 75, a) instructs the operating system to retrieve one or more of either image files 33b or text files 33c from hard disk 33, b) processes them as needed and c) and supplies them along with the cover sheet via bus 21 to facsimile card 37 for transmission. The facsimile card thereupon transmits them, as indicated at 76 and, once again at step 59, facsimile catalog program 34b instructs facsimile card 37 to disconnect from the telephone line and assume its previously idle state.

FIG. 4 is a memory map of that region of hard disk 33 that holds the "permanent" copy of the facsimile catalog program, denoted 33e, as previously discussed. As shown therein, the program includes a number of identifiable modules, copies of which comprise the executed copy of the program stored in RAM 34. The principal functions of these modules are as follows:

Module Name and Number	Principal Function and Corresponding Flowchart Steps
Call Answer 331	initial telephone call processing (steps 41,43)
Caller Facsimile Number 333	obtaining caller's facsimile number (61)
Caller Voice Number 334	obtaining caller's voice number (64)
File Selection 336	eliciting identity, and processing, of desired documents (46,51,66,68,70)
Facs. Num. Verification 337	validating facsimile number (62,63)
Cover Sheet Creation 338	creating cover sheet (74)
Facsimile Transmit 339	transmitting the requested document(s) (53,54, 56,59,73,75,76)

Two other modules within the facsimile card program are facsimile card operating software 341 and voice card operating software 343. These modules are supplied by the vendor(s) of the cards themselves and control the card hardware in response to commands from the above-listed software modules. And it will, of course, be appreciated that the executing copy of the facsimile catalog program in RAM 34 is comprised of copies of the modules shown in FIG. 4.

Variations will occur to those skilled in the art. For example, although the invention has been disclosed in the context of a particular hardware configuration, other hardware configurations providing the same functionality may be used. In addition, although the caller inputs are illustratively provided via the operation of the keys of a touch-tone telephone, it may be desired to provide the system with, for example, voice-recognition circuitry which allows the caller to provide spoken inputs instead. In addition, although the presently disclosed system maintains the image and text files in local storage, it may be desired--particularly if there is a large volume of requestable information--to store that information on a larger system and have the local system, i.e., the system interacting with the caller, request and have downloaded the information when it is needed. And in other straightforward variations, the system may be configured to handle multiple calls and document requests on a time-shared or parallel processing basis. It may also be arranged to provide the caller with the opportunity to make multiple document requests in the same transaction with the system. Additionally, it will be appreciated that the documents supplied by the system may be of the type that changes fairly frequently, such as real estate listings, as compared to documents which do not, such as government forms. Moreover, in some implementations, access to highly confidential documents may be restricted by requesting the caller to enter a security password or voice sample prior to providing the caller with the documents.

CLAIMS

1. A method of transmitting,on request, copies of documents stored in machine-readable form, including the machine-implemented steps of answering a telephone call made over a telephone line to complete a first telephone connection from a caller at the far end of said connection, providing to said caller machine-stored voice signals instructing said caller relative to the ordering of copies of one or more of said documents, receiving from said caller responses to said voice signals and determining from said responses the identity of a specific one of said documents, retrieving the machine-readable document from storage, and transmitting the retrieved machine-readable document in facsimile form.

2. A method as claimed in claim 1 including the machine-implemented steps of determining whether a facsimile machine is connected to the far end of said first telephone connection, and, if so, transmitting the retrieved document in facsimile form over said telephone line during said telephone call, and, if not, providing to said caller machine-stored instructions relative to the specification of information identifying a facsimile machine to which the retrieved document is to be transmitted, receiving from said caller responses to those instructions, and initiating a telephone call to the identified facsimile machine to establish thereto a second telephone connection and transmitting the retrieved document in facsimile form over said second telephone connection.

3. A method as claimed in claim 2 including, in the case where a facsimile machine is not connected to the far end of said first telephone connection, the machine-implemented steps of determining the identity of said caller, and transmitting information identifying said caller along with the retrieved document.

4. Apparatus for transmitting documents, including (i) memory means for storing a plurality of documents in machine-readable form, and (ii) means for answering a telephone call made over a telephone line to complete a first telephone connection from a caller at the far end of said connection and for providing to said caller machine-stored voice signals instructing said caller relative

to the ordering of copies of one or more of said documents, for receiving from said caller responses to said instructions and for identifying from said responses a specific one of said documents, and for retrieving the identified document from said memory means and for transmitting it in facsimile form.

5. Apparatus as claimed in claim 4 wherein the means in (ii) serves for determining whether a facsimile machine is connected to the far end of said first telephone connection, and, when it has been determined that a facsimile machine is connected to the far end of said first telephone connection, for transmitting the retrieved document in facsimile form over said telephone line during said telephone call.

6. Apparatus as claimed in claim 5 wherein the means in (ii) serves, when it has been determined that a facsimile machine is not connected to the far end of said first telephone connection, for providing to said caller voice instructions relative to the specification of information identifying a facsimile machine to which the requested document is to be transmitted, for receiving from said caller responses to those instructions, for thereafter initiating a telephone call to the identified facsimile machine to establish thereto a second telephone connection, and for transmitting the retrieved document in facsimile form over said second telephone connection.

7. Apparatus as claimed in claim 6 wherein the means in (ii) serves, when it has been determined that a facsimile machine is not connected to the far end of said first telephone connection, for providing to said caller machine-stored voice signals instructing said caller relative to the specifying of data identifying said caller, for receiving said caller-identifying data from said caller, and for transmitting said caller-identifying data along with the requested document.

8. A computer-based document transmission system, including first circuit means operable under program control for providing audio messages over a telephone line and for interpreting information supplied over the telephone line, second circuit means operable under program control for providing facsimile messages

over the telephone line, storage means for storing machine-readable representations of a plurality of document files, program means a) for operating said first circuit means to answer a telephone call made over a telephone line to complete a telephone connection between a caller at the far end of said connection and said system, to provide machine-stored voice signals instructing said caller relative to the ordering of copies of one or more of said documents, to receive from said caller responses to said voice signals identifying at least a specific one of said documents, and to determine whether a facsimile machine is connected to the far end of said telephone connection, b) for retrieving from said storage means at least a first one of said document files associated with said specific one of said documents, and c) for operating said second circuit means to transmit the retrieved document in facsimile form.

9. A method of transmitting documents substantially as herein described with reference to the accompanying drawings.

10. Apparatus for transmitting documents substantially as herein described with reference to the accompanying drawings.

CSTB/KW

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9004001 22.02.1990

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H04M 1/66 3/42

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H4K KFB KFD

(56) Documents cited
None

(58) Field of search
UK CL (Edition K) H4K KFB KFD
INT CL⁵ H04M
Online database: WPI

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(74) Agent and/or Address for Service
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Intellectual Property Unit, 13th Floor, 151 Gower Street,
London, WC1E 6BA, United Kingdom

(54) **Telephone call intercept system**

(57) An automatic call interception apparatus (1) provides a barrier between the subscriber and the caller. The apparatus can ask the callers identity, without the subscriber having to speak to the caller personally. The reply is recorded and then played back to the subscriber who can then decide whether or not to accept the call. The call need be intercepted only if the subscriber answers thus minimising inconvenience to the caller.

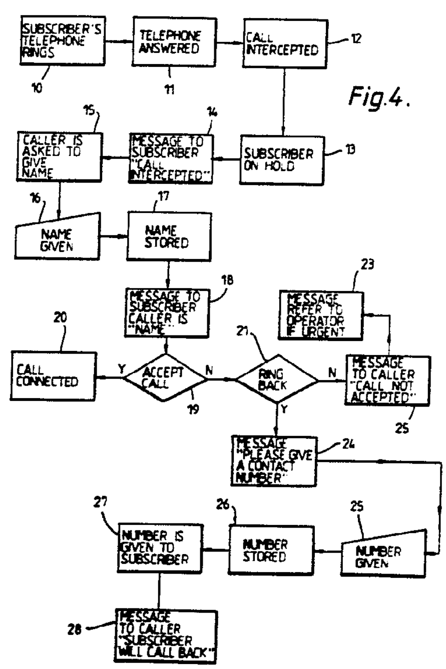

Fig.4.

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At least one drawing originally filed was informal and the print reproduced here is taken from a later filed formal copy.

Fig.1a.

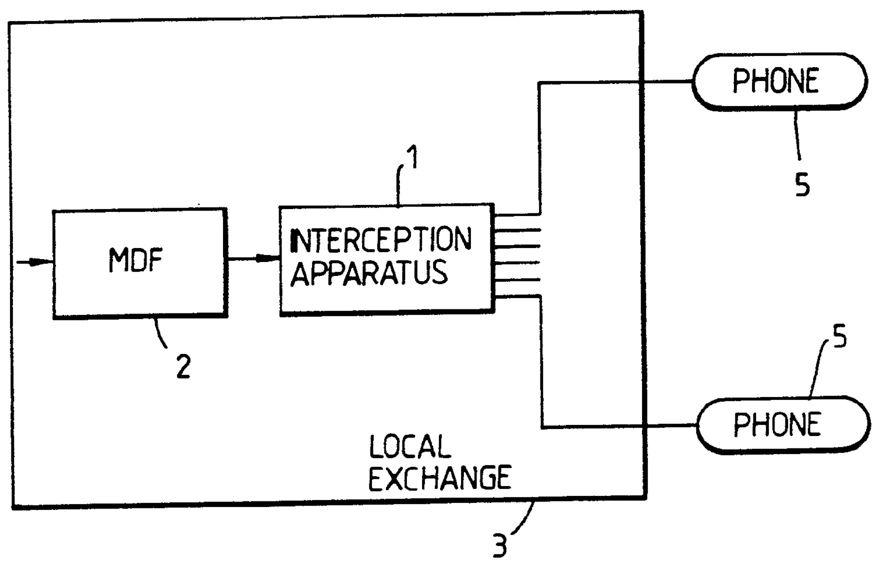
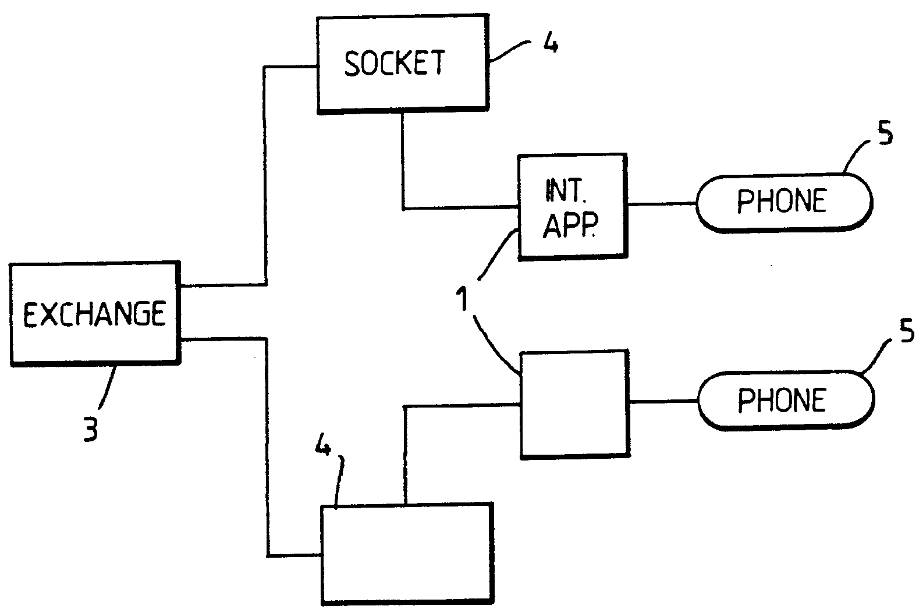
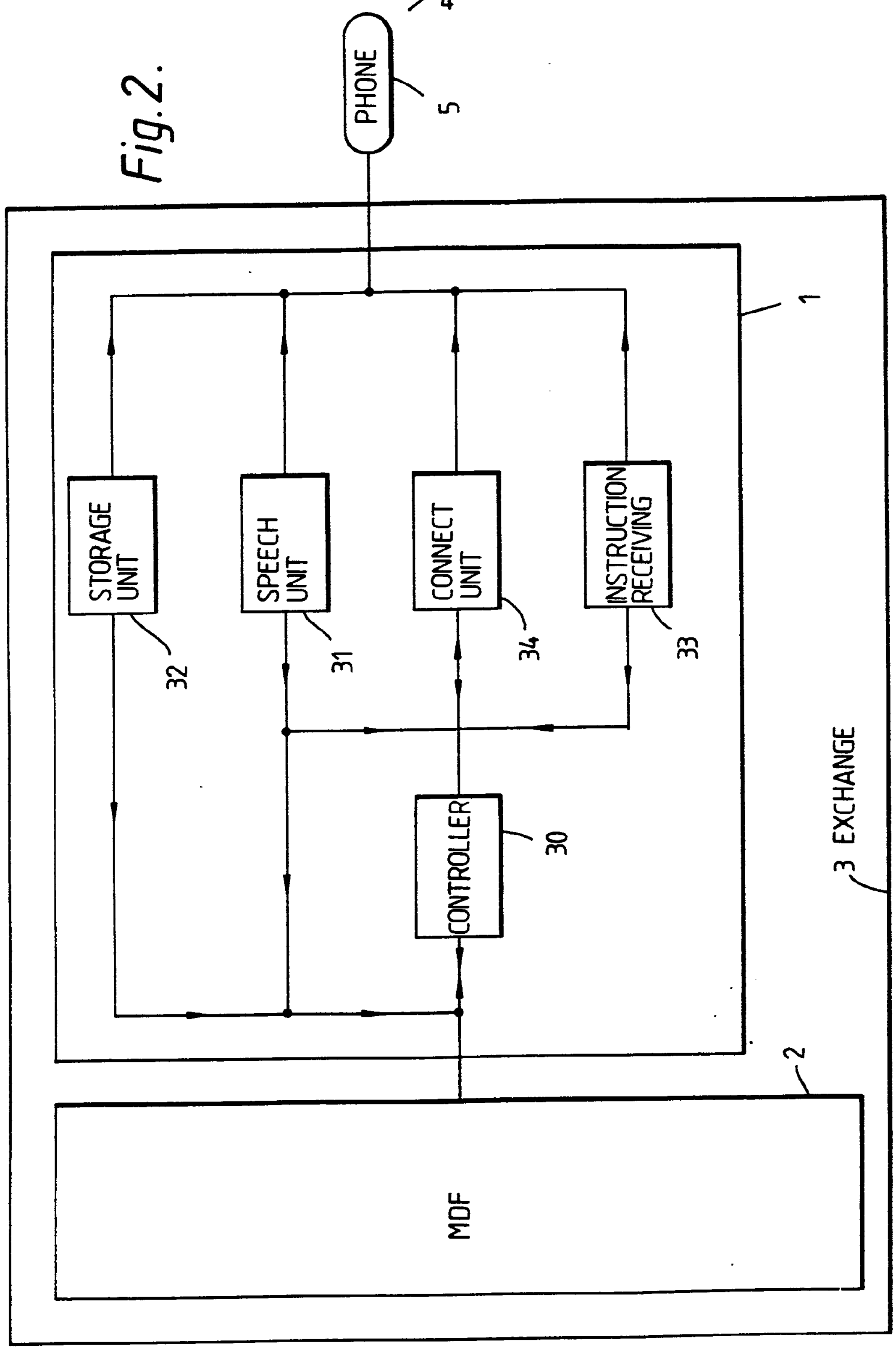

PHONE
5
1
MDF
INTERCEPTION APPARATUS
2
5
PHONE
LOCAL EXCHANGE
3

Fig.1b.


SOCKET
4
5
INT. APP.
PHONE
EXCHANGE
1
5
PHONE
3
4

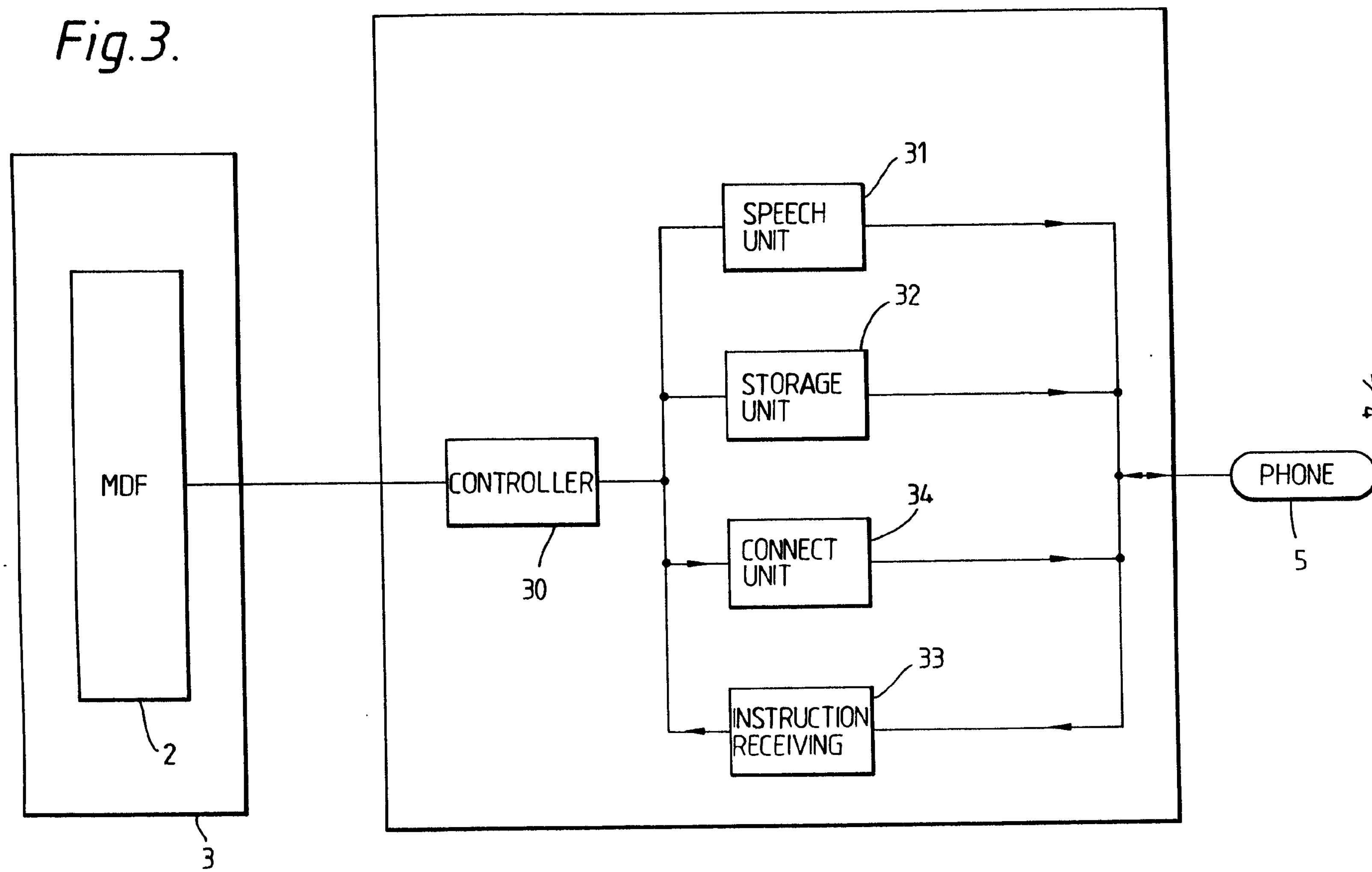
2/4

Fig. 2.



PHONE
5
1
STORAGE UNIT
32
SPEECH UNIT
31
CONNECT UNIT
34
INSTRUCTION RECEIVING
33
CONTROLLER
30
3 EXCHANGE
2
MDF

Fig.3.


MDF
2
3
CONTROLLER
30
SPEECH UNIT
31
STORAGE UNIT
32
CONNECT UNIT
34
INSTRUCTION RECEIVING
33
PHONE
5

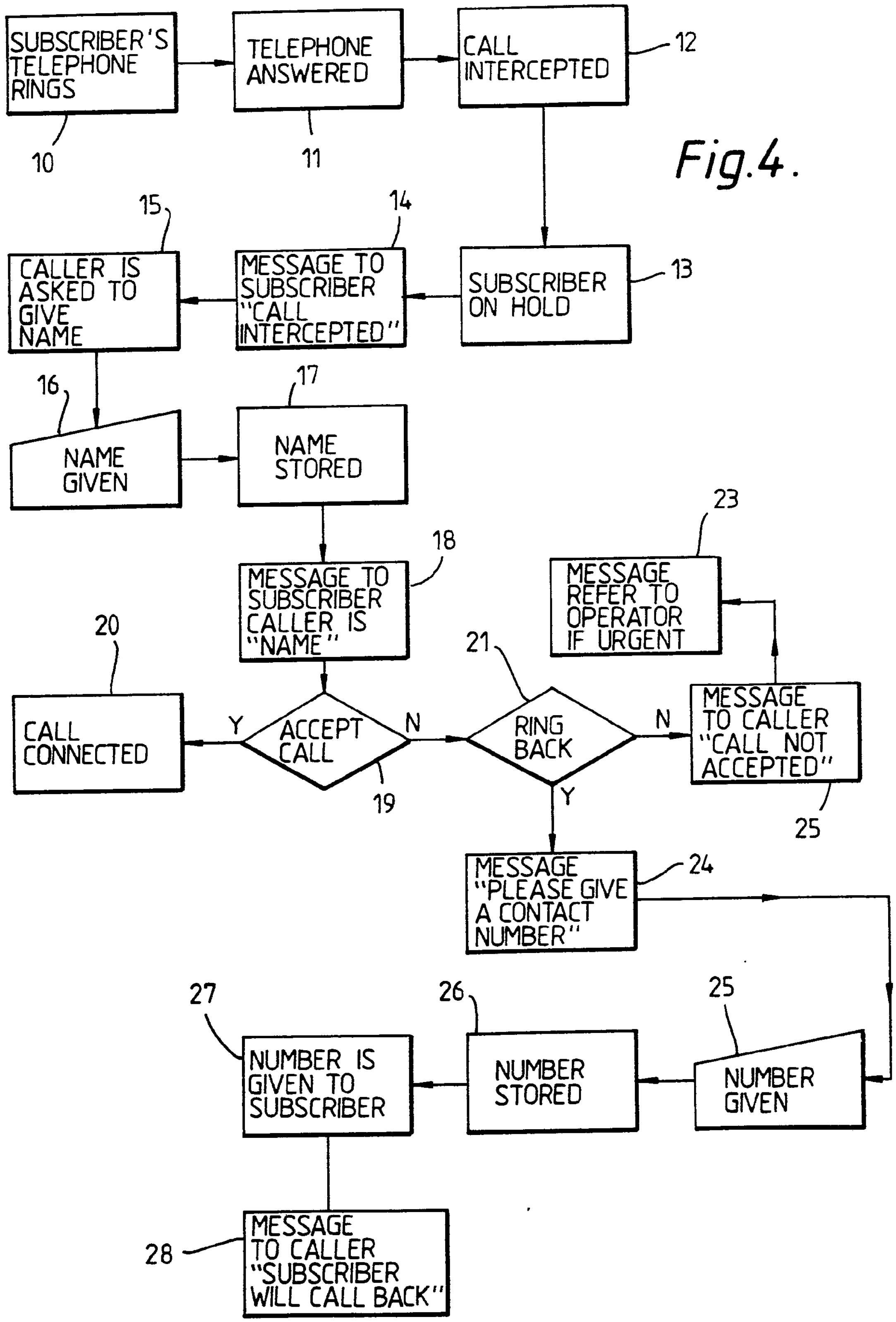

SUBSCRIBER'S TELEPHONE RINGS
10
TELEPHONE ANSWERED
11
CALL INTERCEPTED
12
SUBSCRIBER ON HOLD
13
MESSAGE TO SUBSCRIBER "CALL INTERCEPTED"
14
CALLER IS ASKED TO GIVE NAME
15
NAME GIVEN
16
NAME STORED
17
MESSAGE TO SUBSCRIBER CALLER IS "NAME"
18
ACCEPT CALL
19
Y
N
CALL CONNECTED
20
RING BACK
21
N
Y
MESSAGE REFER TO OPERATOR IF URGENT
23
MESSAGE TO CALLER "CALL NOT ACCEPTED"
25
MESSAGE "PLEASE GIVE A CONTACT NUMBER"
24
NUMBER GIVEN
25
NUMBER STORED
26
NUMBER IS GIVEN TO SUBSCRIBER
27
MESSAGE TO CALLER "SUBSCRIBER WILL CALL BACK"
28

Fig.4.

TELEPHONE CALL INTERCEPT SYSTEM

This invention relates to an automatic telephone call interception facility to allow screening of incoming calls. The invention has a particular use as a deterrent to malicious callers.

Subscribers are often bothered with unwelcome calls which may be malicious or just time consuming, eg from salespeople.

A call intercept facility is available to subscribers troubled by malicious callers. However, this facility has several disadvantages.

This facility involves all the subscriber's calls being diverted to an operator. The operator then asks:

1. the identity of the calling party,
2. the telephone number to which the calling party wishes to be connected,
3. the telephone number of the calling party,
4. who the calling party wishes to speak to.

Once the operator has asked the questions, the caller is then connected to the subscriber's line. There is no guarantee that the subscriber will answer. The operator does not check to see if the subscriber's phone will be answered before asking the questions or attempting to connect the call.

This is a very labour intensive and expensive service to provide. The calling party can get irritated at being asked a lot of seemingly pointless questions each time the subscriber is called. The operator is not able to give the subscriber the option of refusing the call. The service is uneconomical to use simply to screen out unwanted calls.

The object of this invention is to provide an inexpensive and easily implemented system of intercepting and screening incoming calls.

According to the invention there is provided a telephone call interception apparatus comprising first connection means for connecting the apparatus to a party calling a subscriber's telephone associated with the apparatus, interrogation means for interrogating the calling party, recording means for recording the responses given by the calling party to the interrogation, transfer means for transferring the recorded answers to the subscriber's telephone and second connection means for connecting the calling party to the subscriber's telephone, means for accepting signals from the called party to instruct the apparatus to connect the call and means for accepting a password from the operator allowing the apparatus to be bypassed in cases of emergency.

The apparatus preferably incorporates a speech synthesizer in the interrogation means.

The apparatus preferably also includes means for recording the responses given by the calling party to the interrogation. The recording means is preferably through a digital recording means. In a preferred embodiment the invention is a network system based at the local exchange capable of intercepting calls on a plurality of subscriber lines. Another preferred embodiment of the invention is a small portable unit based on the subscriber's premises.

Alternatively the apparatus uses pulse coded modulation (PCM) or linear predictive coding (LPC) to record the responses to the interrogation.

Conveniently the apparatus has means for disconnecting the calling party if the subscriber does not wish to accept the call.

Preferably the apparatus is such that the call is intercepted only if the called party answers.

Conveniently the apparatus incorporates a means for logging intercepted calls.

Advantageously the apparatus incorporates means such that the subscriber can switch the call intercept on or off from the telephone to be protected, or from any telephone using an identifying password or code.

The subscriber based apparatus may incorporate a plug means for plugging into a socket on the subscriber's premises.

The apparatus conveniently also incorporates an alarm means to allow the subscriber to alert the telephone company to the fact that assistance is required.

The apparatus may further comprise means to allow the subscriber to trigger a malicious call alarm.

Preferably the apparatus includes means for accepting a password to allow the apparatus to be overridden. This allows household members to get straight through and also allows an operator to bypass the system in case of emergency.

The apparatus conveniently incorporates a telephone answering apparatus obviating the need for two pieces of equipment attached to the same telephone.

The intercept system could either be at the local exchange connected to a main distribution frame (MDF) where the individual line pairs enter the local exchange, and be a chargeable service or it could take the form of a separate portable unit. The subscriber could buy or rent the unit and it could be plugged into the existing telephone socket.

The present invention will now be described by way of example only and with reference to the accompanying drawings in which:

Figure 1a) is a block diagram of an exchange based system;

Figure 1b) is a block diagram of a subscriber based system.

Figure 2 is a block diagram of a network based system constructed in accordance with the invention.

Figure 3 is a block diagram of a subscriber's premises based system constructed in accordance with the invention.

Figure 4 is a flow diagram showing the manner of operation of a system constructed in accordance with the invention.

Referring to the drawings, Figure 1a shows an intercept apparatus 1 connected to the main distribution frame 2 (MDF) at a local exchange 3. The intercept apparatus 1 is capable of intercepting calls on a plurality of lines (typically 100 or more lines per unit).

On compatible exchanges which allow customer controlled call diversion the subscriber would be able to switch the intercept apparatus on or off as required. Once activated all calls would be intercepted before the called party had answered.

Referring now to Figure 1b, the intercept apparatus 1 is connected to the local exchange 3 via the subscriber's telephone socket 4, the intercept apparatus being positioned between the socket and the subscriber's telephone 5.

Figure 2, shows in more detail the telephone intercept apparatus 1 situated at the local exchange 3. It has a controller 30 which controls the activities of the other parts of the apparatus. When a caller rings a subscriber and the subscriber picks up the receiver to answer the call the call is then intercepted. The controller 30 causes a speech unit 31 to tell both the caller and the subscriber that the call has been intercepted, and asks the caller to give a name. The speech unit may be a speech synthesizer, or alternatively pre-recorded messages on magnetic tape or other media could be used.

Responses from the caller to questions asked by the speech unit 31, are stored in a storage unit 32, constituted by a digital storage unit eg a RAM. Alternatively, the storage unit 32 could be a simple magnetic storage means.

The controller 30 causes the responses to be transferred to the subscriber at the relevant times in the manner described below with reference to the flow chart of Figure 4.

After information has been transferred to the subscriber the controller 30 alerts an instructions receiving unit 33, to be on standby to receive instructions from the subscriber. The instruction receiving unit 33 could be a speech recognition system with a limited vocabulary or take the form of an MF tone decoder for people with speech or language difficulties.

When the subscriber indicates that the call is accepted the controller 30 signals a call connect unit 34 to connect the call if appropriate.

Where the subscriber does not wish to accept the call, the caller is referred back to the operator at the local exchange or played a suitable message.

The controller 30 and also has means (speech recogniser or an MF tone decoder) for recognising a password or other signal sent from the exchange so that the exchange operator can, upon giving the correct password or other code, override the interception mechanism and be routed directly to the subscribers telephone 5 via the call connect unit 34. Members of the subscriber's household could also have a second password allowing the system to be bypassed. The speech recognition system could be speaker dependent for added security.

The apparatus could incorporate means to allow the subscriber to instruct the apparatus not to connect any further calls from a caller. The apparatus could use a speaker dependent recognition system to recognise unwanted repeat callers.

Where the subscriber does not wish to accept the call and the caller does not hang up, thereby blocking the subscriber's line the apparatus is able to disconnect the call.

An alternative embodiment of the invention is the subscriber based unit shown in Figure 3. This is very similar to the MDF network based system, but as the subscriber based unit only intercepts calls on the subscriber line it is connected to it may be consequently much smaller and less expensive. The controller 30 controls the operation of the unit. The other parts of the unit are the same, a speech unit 31, a storage unit 32, an instruction receiving unit 33 and a call connect unit 34. In the subscriber based unit the call connect unit 34 is not able to disconnect the call if not accepted and the caller does not hang up because calls can only be disconnected by the caller or the local exchange.

Both the network based service and the subscriber's premises based service operate in substantially the same manner, which will now be described with reference to Figure 4.

The subscriber's telephone rings (at 10), and when the subscriber answers (at 11), the call is intercepted (at 12). At this point the subscriber is put on hold (at 13) and informed that the call has been intercepted (at 14). The caller is informed, either by a light on the subscriber based apparatus or a message that the call is being intercepted and logged and asked to give a name (at 15). When the caller gives a name (at 16) this is stored (at 17) and then transferred to the subscriber (at 18). The subscriber is then given the option of accepting the call or not (at 19). If the subscriber chooses to accept the call, this is indicated to the unit by pressing a dedicated button or by MF signalling, then the call is connected (at 20). Some lower cost models might not have call logging facilities but the caller will not be able to tell whether the apparatus in question can log calls or not.

If the subscriber chooses not accept the call, the apparatus offers the subscriber the possibility of returning a call (at 21). Again the subscriber's intentions can be indicated by dedicated "yes" or "no" buttons or by MF signalling. If the subscriber does not wish to return the call then the caller is informed that the call has not been accepted (at 22) and is referred to the operator in case of emergency (at 23).

If the subscriber does choose to ring the caller back, the operator asks the caller to provide a contact telephone number (at 24). The caller can then give the number (at 25) which is stored (at 26) for transmission to the subscriber (at 27) and the caller is informed that the subscriber will ring back (at 28).

Dedicated buttons with labels and lights would overcome problems experienced by people with speech or language difficulties.

The subscriber based system could intercept internal calls on PABX exchanges which do not go via the local exchange, and so could not be intercepted by the operator or an MDF based system.

The apparatus could incorporate means for allowing subscriber's to hear the information provided by the caller eg name and telephone number, replayed via a loudspeaker instead of through the earpiece from the handset.

CLAIMS

1. Telephone call interception apparatus comprising first connection means for connecting the apparatus to a party calling a subscriber's telephone associated with the apparatus, interrogation means for interrogating the calling party, recording means for recording the responses given by the calling party to the interrogation, transfer means for transferring the recorded answers to the subscriber's telephone and second connection means for connecting the calling party to the subscriber's telephone, means for accepting signals from the called party to instruct the apparatus to connect the call and means for recognising a password from the exchange operator and means responsive to said password causing the apparatus to be by-passed in cases of emergency. A

2. An apparatus as claimed in claim 1, wherein the password recognising means is able to recognise a second password and whereby a caller having a password can cause the apparatus to be by-passed.

3. Apparatus as claimed in claim 2, comprising a multitone frequency decoder system to accept instructions from the called party.

4. Apparatus as claimed in claim 3, further comprising a speech recognition system to accept instructions from the called party.

5. Apparatus as claimed in any one of the preceding claims, wherein a digital recording means constitutes the recording means for recording the responses given by the calling party to the interrogation.

6. An apparatus as claimed in claim 2, where the password accepting means constitutes an MF tone decoder.

7. An apparatus as claimed in claim 2, wherein the password accepting means constitutes a speech recognition system.

8. An apparatus as claimed in claim 2, wherein the password accepting means constitutes a speaker dependent speech recognition system.

9. An apparatus as claimed in any one of the preceding claims, wherein the interrogation means incorporates a speech synthesizer.

10. An apparatus as claimed in any one of the preceding claims, wherein the apparatus only intercepts the call if the called party answers.

11. An apparatus as claimed in any one of the preceding claims, further comprising means for connecting the apparatus to a distribution frame at a local exchange, the apparatus being capable of intercepting calls on a plurality of subscriber lines.

12. An apparatus as claimed in claim 11, further comprising means for disconnecting the calling party if the subscriber does not wish to accept the call.

13. An apparatus as claimed in one of the preceding claims, wherein the apparatus can be switched on or off from any telephone by the subscriber.

14. An apparatus as claimed in any one of the preceding claims, further comprising programmable call rejection means to always reject calls from a certain party.

15. An apparatus as claimed in claim 14, wherein the programmable call rejection means further comprises a speaker recognition system.

16. An apparatus as claimed in any one of claims 1 to 10, further comprising a plug means for plugging the apparatus into a telephone socket on the subscriber's premises.

17. An apparatus as claimed in any one of the preceding claims, further comprising an alarm means to allow the subscriber to alert the telephone company to the fact that assistance is required.

18. An apparatus as claimed in any one of the preceding claims, further comprising means to allow the subscriber to trigger a malicious call alarm.

19. A telephone call interception apparatus substantially as described herein with reference to and as illustrated by the accompanying drawings.

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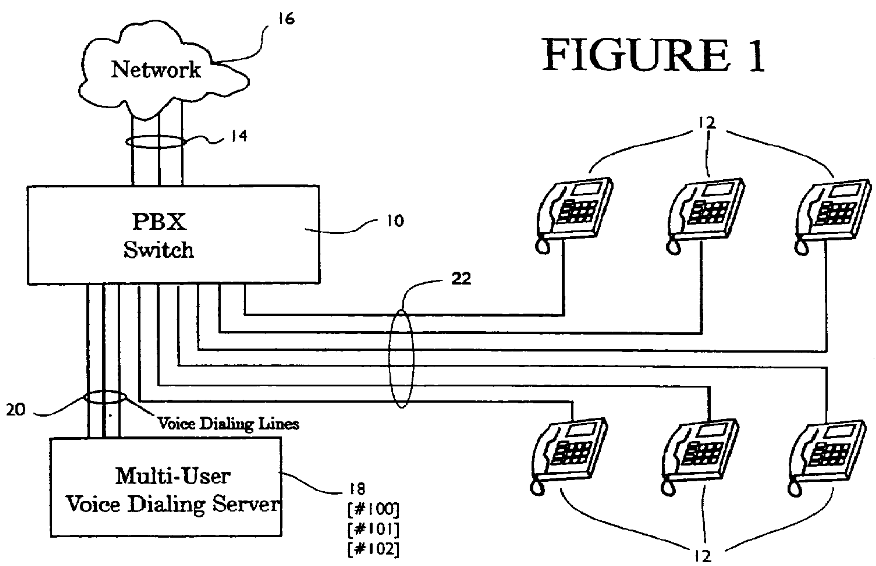
(58) Field of Search
UK CL (Edition O) **H4K KFH KF42**
INT CL[6] **H04M 3/42**
ONLINE: WPI

(54) **Voice dialling server for branch exchange telephone systems**

(57) The voice dialing server plugs into one or more unused extensions of a branch exchange system to provide each of the users on the system with voice dialing services. To use the system a user simply dials the extension to which the server is attached. The server then prompts the user to supply the name of a party to be called. The name is then looked up in a telephone number dictionary unique to that user. The system then places the telephone call by sending commands to the branch exchange system that simulate the operations a user would perform to connect to an outside line or inside extension and then place the call. The server incorporates a speech processing module having a multistage word recognizer that represents speech in terms of high phoneme similarity values. This representation is highly compact, allowing the word recognizer to perform the recognizer and fine match stages with far less processor overhead than frame-by-frame speech recognizers.

FIGURE 1



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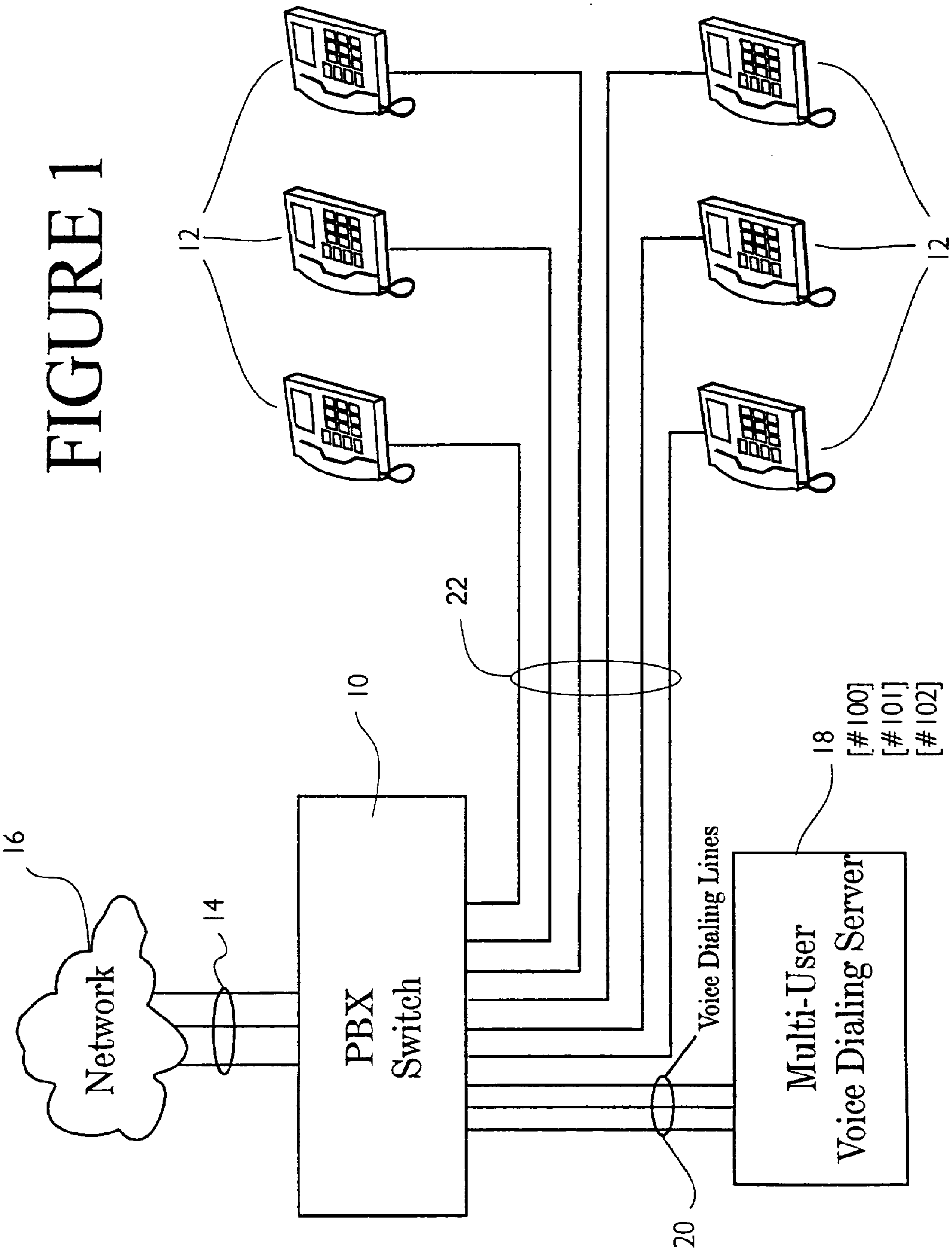

12
12
22
10
16
14
Network
PBX
Switch
Voice Dialing Lines
Multi-User
Voice Dialing Server
18
[#100]
[#101]
[#102]
20

FIGURE 1

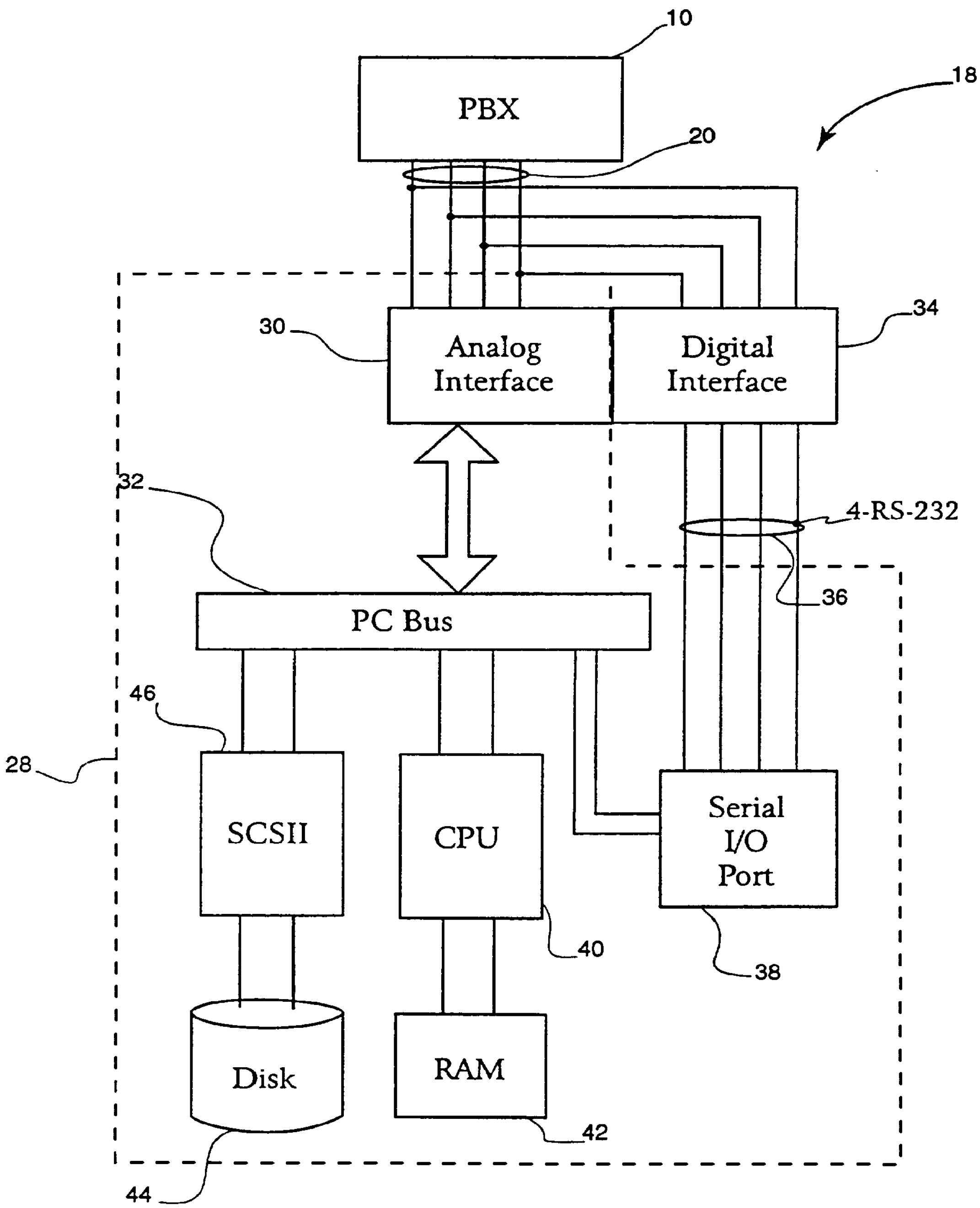

10
18
PBX
20
30
Analog
Interface
Digital
Interface
34
32
4-RS-232
36
PC Bus
46
28
SCSII
CPU
Serial
I/O
Port
40
38
Disk
RAM
42
44

Figure 2

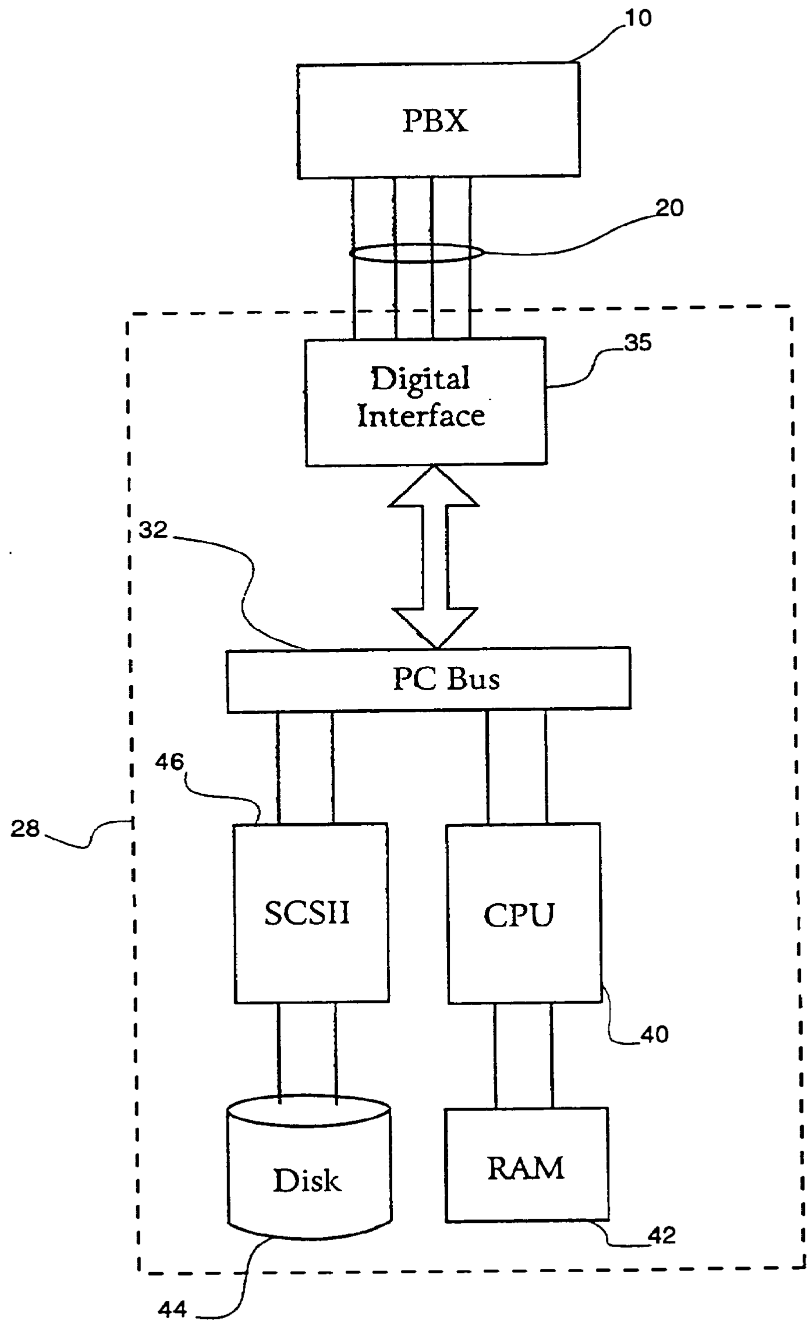

10
PBX
20
35
Digital Interface
32
PC Bus
46
28
SCSII
CPU
40
Disk
RAM
42
44

Figure 3

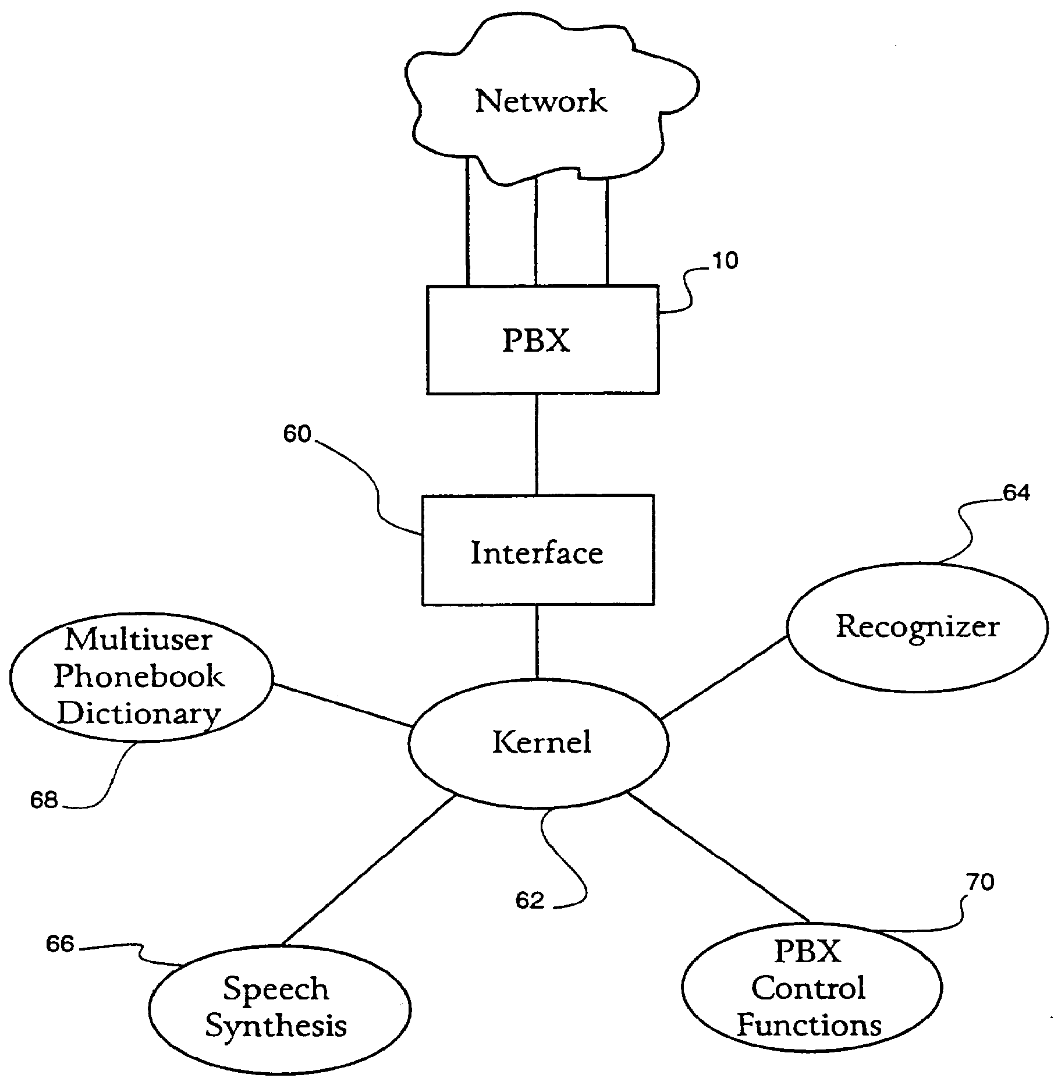
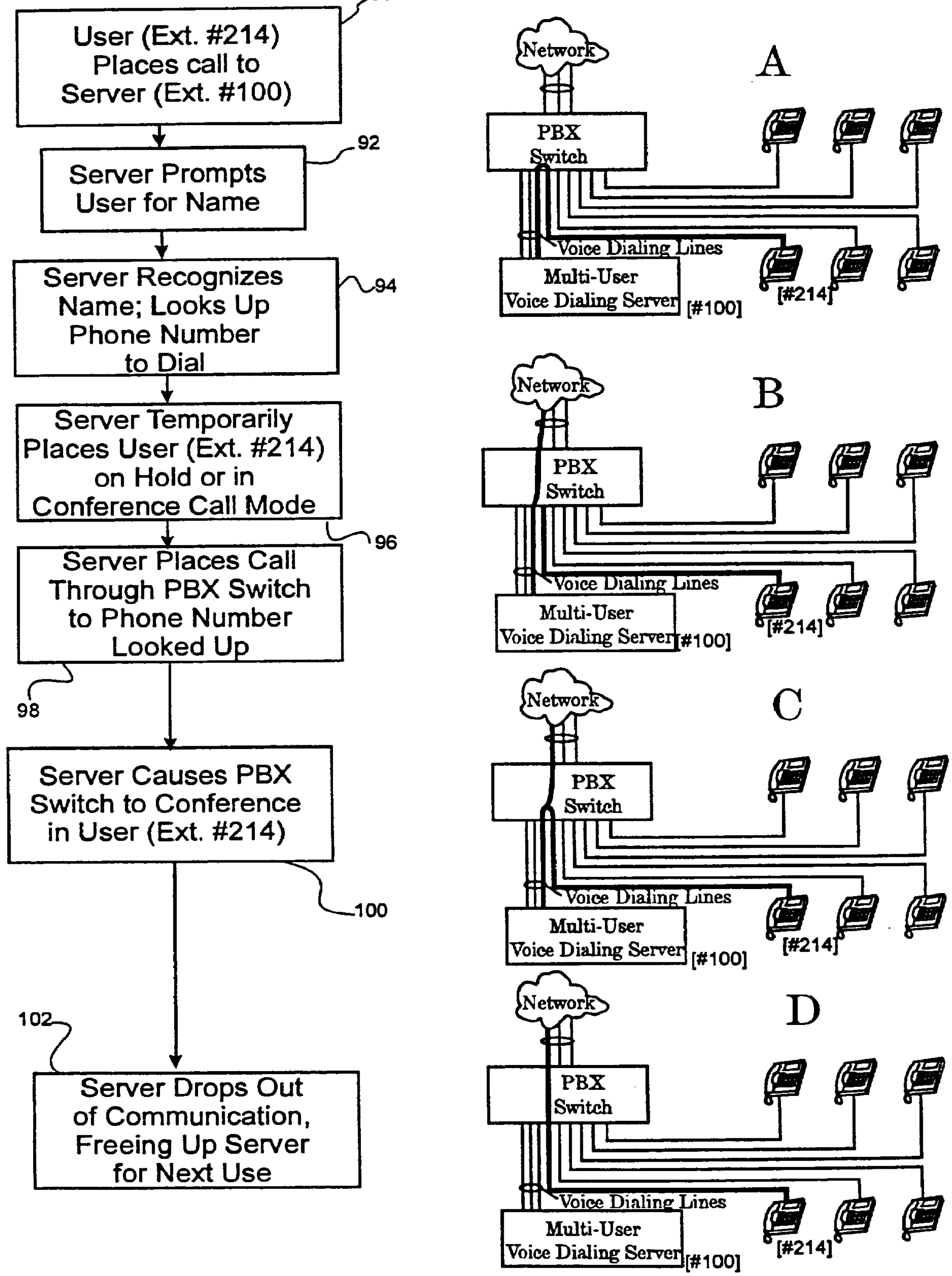

Network
10
PBX
60
Interface
64
Recognizer
Multiuser
Phonebook
Dictionary
Kernel
68
62
70
66
Speech
Synthesis
PBX
Control
Functions

Figure 4

FIGURE 5


90
User (Ext. #214) Places call to Server (Ext. #100)
92
Server Prompts User for Name
94
Server Recognizes Name; Looks Up Phone Number to Dial
Server Temporarily Places User (Ext. #214) on Hold or in Conference Call Mode
96
Server Places Call Through PBX Switch to Phone Number Looked Up
98
Server Causes PBX Switch to Conference in User (Ext. #214)
100
102
Server Drops Out of Communication, Freeing Up Server for Next Use
A
B
C
D
Network
PBX Switch
Voice Dialing Lines
Multi-User Voice Dialing Server
[#100]
[#214]

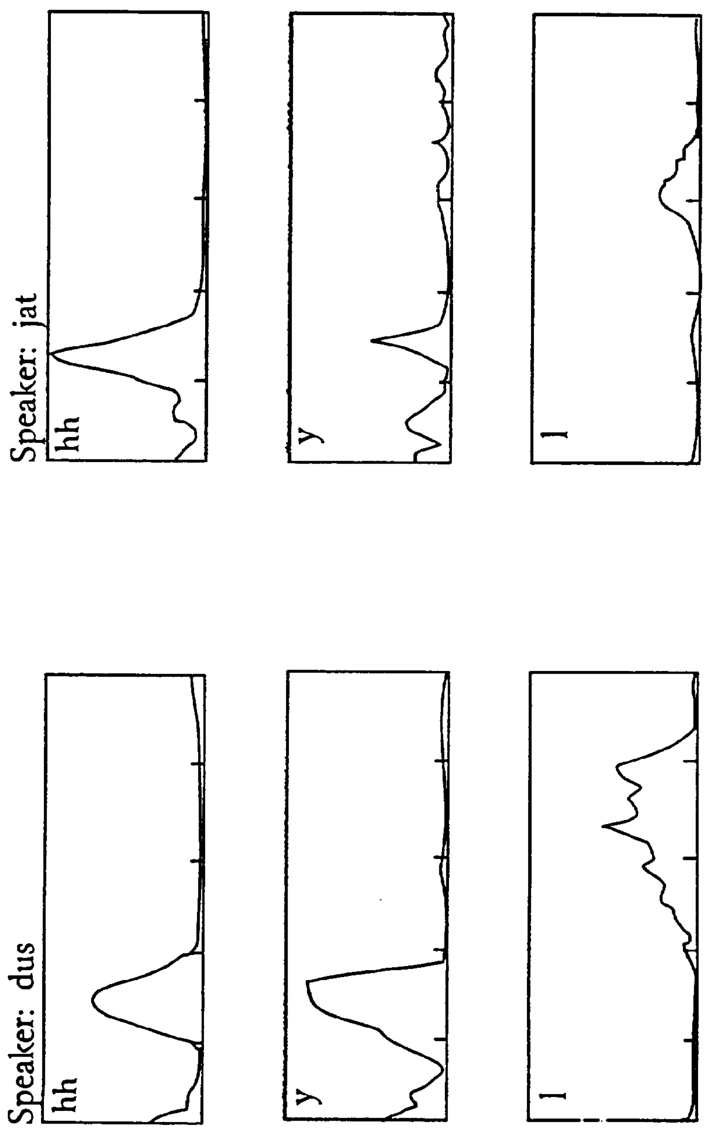


Figure 6

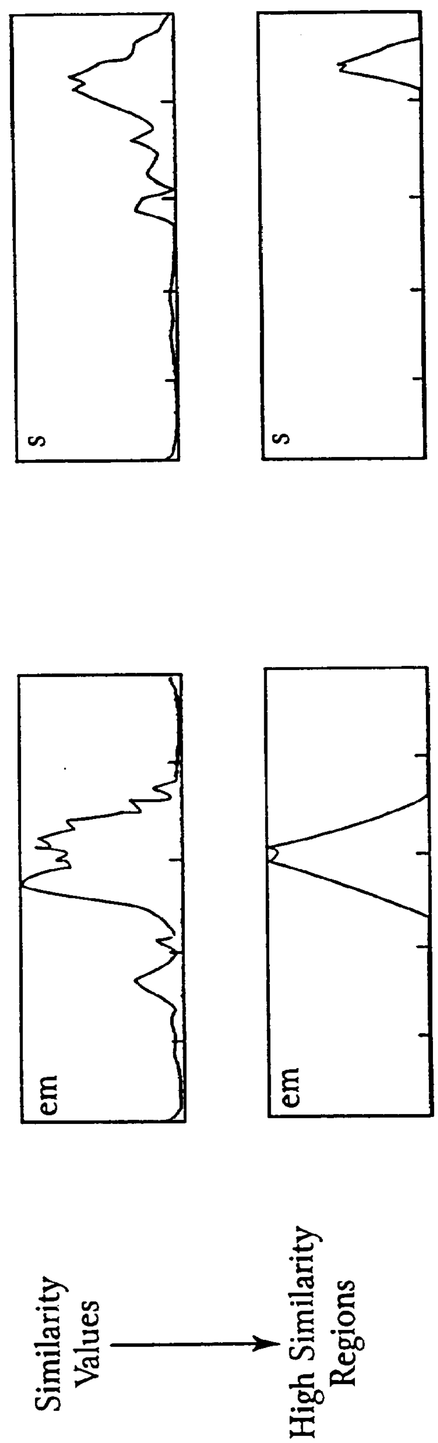
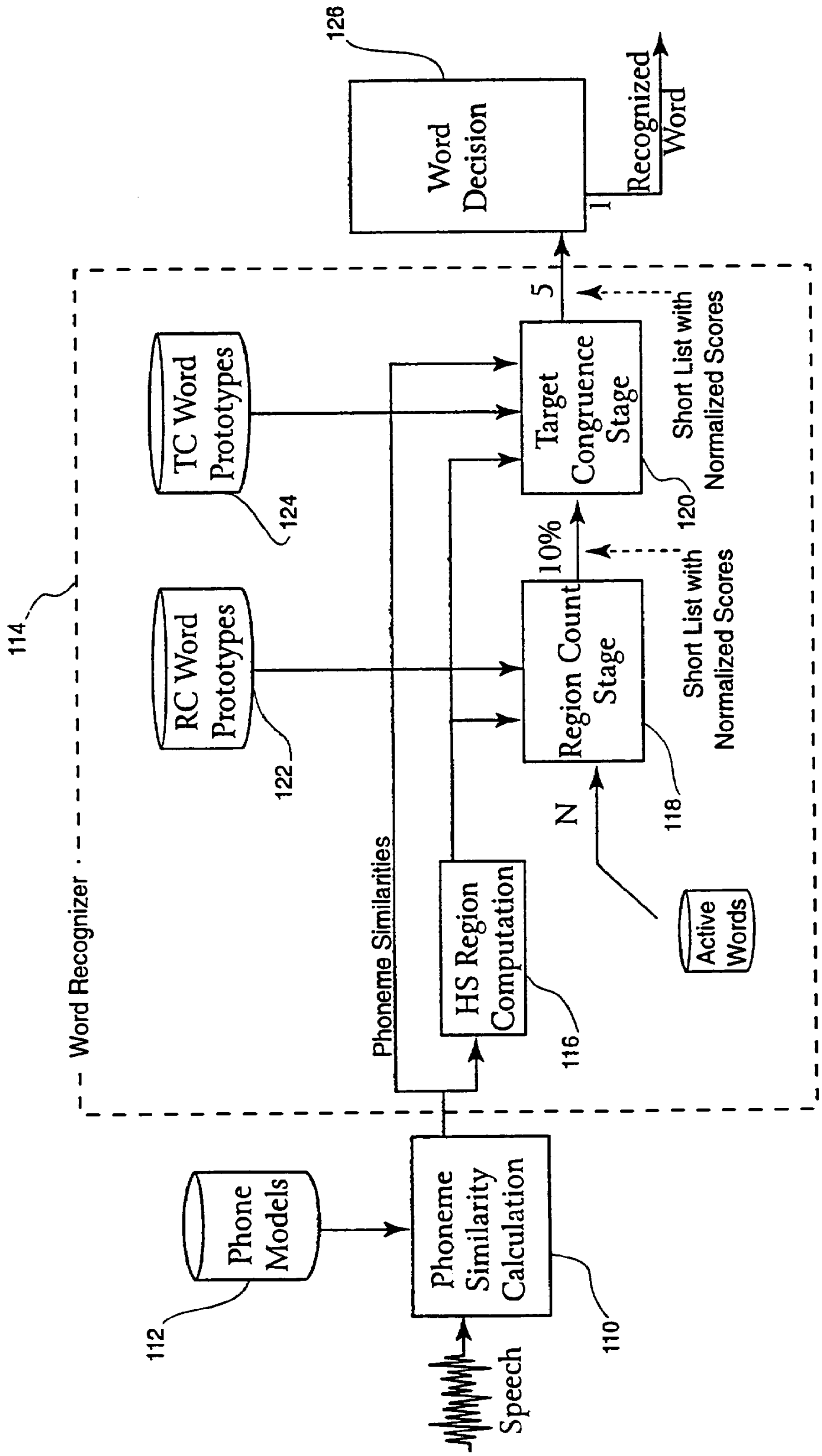


Figure 7



Speech
Phone Models
112
Phoneme Similarity Calculation
110
Word Recognizer
114
Phoneme Similarities
HS Region Computation
116
RC Word Prototypes
122
TC Word Prototypes
124
Region Count Stage
118
N
Active Words
10%
Short List with Normalized Scores
Target Congruence Stage
120
Short List with Normalized Scores
5
Word Decision
126
1
Recognized Word

Figure 8

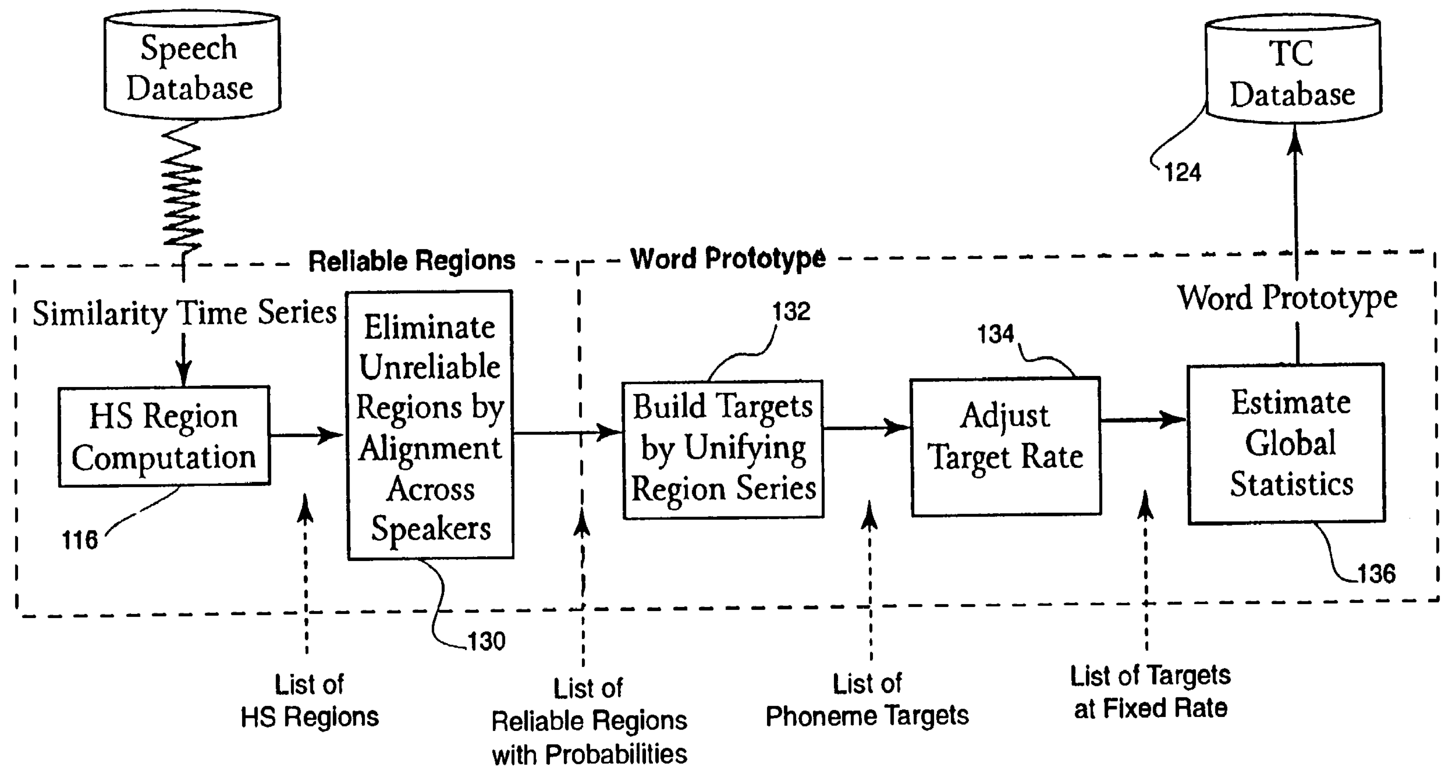

Speech Database
TC Database
124
Reliable Regions
Word Prototype
Similarity Time Series
HS Region Computation
116
Eliminate Unreliable Regions by Alignment Across Speakers
130
Build Targets by Unifying Region Series
132
Adjust Target Rate
134
Word Prototype
Estimate Global Statistics
136
List of HS Regions
List of Reliable Regions with Probabilities
List of Phoneme Targets
List of Targets at Fixed Rate

Figure 9

VOICE DIALING SERVER FOR BRANCH EXCHANGE TELEPHONE SYSTEMS

The present invention relates generally to telephone switching equipment. More particularly the invention relates to a voice dialing server that attaches to the telephone branch exchange equipment to provide voice dialing services without the need to extensively modify the branch exchange equipment. The preferred system plugs into one or more unused extensions of the branch exchange system to provide voice dialing services for multiple users of the system. Each user may have his or her own dictionary of names and phone numbers. The system integrates with the existing branch exchange network, using the existing voice and control channels to cause the existing branch exchange system to perform the necessary switching operations.

Voice dialing promises to make telephones easier to use, by allowing the user to simply speak a name and then have the voice dialing system look up the telephone number of the named party and automatically place the call. In the cellular telephone market, rudimentary voice dialing systems have been experimented with to provide hands-free operation. The primary technological focus in the cellular telephone market has been on how to overcome the high ambient noise level present in the cellular telephone environment, particularly in car phone applications. There has also been some work in developing voice dialing units for the home. These units typically connect between the telephone and the outside telephone line. A primary technological focus of those units has been on how to overcome the presence of the dial tone when the user lifts the handset to use the voice dialer.

While voice dialing has made some inroads, particularly in the applications discussed above, voice dialing has yet to be incorporated into more complex telephone systems such as private branch exchange switching systems (PBX systems). There are a number of reasons for this. First, voice recognition is a challenging problem and current technology does not provide suitable recognition accuracy in an economical configuration. For example, the complex Hidden Markov Model-based systems employed by state-of-the-art speech recognizers (as in dictation transcription systems) require lots of memory and computational power.

Second, in the voice dialing application, the voice recognition problem is compounded where the system must be adapted for use by a large number of users. The need to respond to the spoken commands of a large number of users makes the voice dialing problem far more difficult than it is for simple voice dialing systems designed for home use.

Third, it is not a simple matter to integrate voice dialing into a complex telephone switching network. Modern-day telephone switching networks employ an intricate labyrinth of digital control signals that effect various switching functions (e.g. placing a call on hold, transferring a call, initiating a conference call, reassigning an extension to a different location and so forth). Simple voice dialing systems of the type employed in cellular phone applications or home dialing applications will not work in this more complex environment.

Finally, office PBX equipment is expensive and difficult to replace without disrupting day-to-day office functions. Thus many businesses that would benefit from voice dialing services, were such equipment available, simply cannot afford the cost and down-time required to replace that equipment with newer equipment providing voice dialing capabilities.

Thus, while the desirability of providing voice dialing in office systems is readily appreciated, current technology does not provide the means to accomplish it.

The present invention provides a voice dialing server for coupling to a branch exchange telephone system of the type that provides call switching among a plurality of telephone extension ports. The system is designed for plug-compatible connection to the existing telephone system without the need for modifying the system extensively. The voice dialing server has an interface for connection to at least one of the telephone extension ports of the existing telephone system. The interface supports transmission of voice signals and telephone system control information.

The voice dialing server also includes a speech processing module coupled to the interface for providing the following services. The speech processing module answers calls placed to the voice dialing server by users of the system. It processes speech input from the user, corresponding to a selected party to be called; and it looks up the telephone number of the selected party.

The voice dialing system also includes a branch exchange control module that is coupled to the interface and to the speech processing module. The control module issues control information to the telephone system, causing the telephone system to connect the user's extension to an outside line while dialing the phone number of the selected party. The preferred embodiment causes the extension that has been assigned to the interface to be connected to a second telephone port on the system. The second port can be another extension or an outside line. Then the call is placed via the second port and the user's extension is then attached to the second port. In this way the user is placed in communication with the selected party.

The system integrates fully with the existing branch exchange telephone system. Thus the invention can be readily added to an existing telephone system, simply by plugging it into an unused extension port on the system. To use the system the user simply dials the extension assigned to the voice dialing server and follows the voice prompts issued by the server. The system is preferably implemented in a multitasking environment that allows multiple threads to run concurrently. Thus multiple users may use the system simultaneously. The system is capable of providing different phone directories for different users, and these may be automatically associated with the users' telephone extension. The system is able to determine the extension of the user. By determining the user's extension the voice dialing server automatically uses the phone number dictionary created by the user at that extension. Alteratively, the user can override the determined extension by supplying a different extension, thereby causing a different phone number dictionary to be used.

Although well integrated into the existing telephone system architecture, the invention can also be used by callers outside the system to reach persons inside the system or to look up numbers from the telephone book. For example, a user calling from home may connect to the voice dialing server by specifying the server's extension. Then, the user may enter his or her office telephone extension number, thereby telling the voice dialing server that the phone number dictionary assigned to the office extension should be used. Thereafter, the user calling from home can use his or her office telephone number directory just as if the user were from the office.

The voice dialing server uses very fast and yet remarkably accurate voice recognition technology based on reliably detected phoneme similarity regions. The preferred embodiment uses a multistage word recognizer that compactly represents speech in terms of high phoneme similarity values. This is a departure from conventional techniques that determine similarity based on a frame-by-frame alignment. The preferred embodiment uses a word recognizer that preserves only the interesting regions of high phoneme similarity or features. A word recognizer is used to narrow the search so that the subsequent fine match stage is able to perform its task more quickly. The word recognizer and fine match stages share the initial representation of speech as a sequence of multiple phoneme similarity values. By representing speech as features at a lower data rate in the initial stage of recognition, the complexity of the matching procedure is greatly reduced.

For a more complete understanding of the invention, its objects and advantages, reference may be had to the following specification and to the accompanying drawings.

Figure 1 is a system block diagram showing the multiuser voice dialing server connected to an existing public branch exchange (PBX) switch;

Figure 2 is a block diagram of a first embodiment of the invention;

Figure 3 is a block diagram of a second embodiment of the invention;

Figure 4 is a entity relationship diagram showing how the major software subsystems are interfaced with the existing PBX switch;

Figure 5 is a flowchart with accompanying signal flow diagrams, showing how the PBX control functions are performed;

Figure 6 is a phoneme similarity time series for the word "hill" spoken by two speakers;

Figure 7 is a series of graphs showing the output of the region picking procedure whereby similarity values are converted into high similarity regions;

Figure 8 is a block diagram of the presently preferred word recognizer system;

Figure 9 is a block diagram illustrating the target congruence word prototype training procedure.

The present voice dialing server is designed to connect to an existing telephone system of the type found in small, medium and large businesses, institutions, hotels, offices and the like. For purposes of illustrating the invention the existing telephone system will be illustrated and described as a private branch exchange system or PBX system. As will be appreciated from the following description, the invention is not limited to any particular type of telephone switching system. Hence the reference to private branch exchange or PBX systems in this written description is not intended to limit the invention.

With the foregoing in mind, Figure 1 depicts a conventional PBX switch **10** to which a plurality of telephone stations **12** are connected. PBX switch **10** is connected through a plurality of outside lines **14** to the telephone network infrastructure **16**. Each of the individual stations **12** is connected to a separate extension or port, assigned a unique extension number. When calling internally from one station to another, the extension numbers may be dialed directly and the PBX switch connects the calling station to the designated receiving station. When placing calls to the telephone network **16** the full telephone number of the intended receiving station is dialed through the PBX switch.

The multiuser voice dialing server **18** of the invention is connected to one or more extension ports of the PBX switch **10**, essentially in the same fashion as telephone stations **12** are connected. Preferably the voice dialing server is assigned an extension number different from the extension numbers assigned to the telephone stations **12**. In this example the voice dialing server is assigned extension number #100. Although it is possible to implement the invention using only one extension line, the voice dialing server will handle more traffic from users if the server is connected through a plurality of lines to the PBX switch. In Figure 1 server **18** is connected through three separate lines **20** to three separate extension ports of the PBX switch **10**. These lines may be referred to as the voice dialing lines, although it will be appreciated that these lines are physically the same as the telephone station lines **22** that connect the telephone stations **12** to the PBX switch.

When multiple voice dialing lines are used, as illustrated here, one line will be assigned the primary extension number (in this case #100). The remaining lines are assigned other extension numbers. To make the system easy to use, the PBX switch **10** is programmed so that the primary extension (#100) is used by all users. When this extension is busy (in use by an earlier user) subsequent calls to the primary extension are routed to one of the unused remaining lines. If all voice dialing lines are busy when a user attempts to employ the voice dialing server, a busy signal will be received. This does not ordinarily occur because the voice dialing server is designed to drop out of the communication path once the desired number has been dialed. The system is designed to prompt the user for a name. It then looks up the telephone number associated with that name and dials it after receiving verbal confirmation from the user. The voice recognizer of the preferred embodiment is quite fast, hence each individual use of the system does not tie up a voice dialing extension for very long.

A first embodiment of the voice dialing server is illustrated in Figure 2. In Figure 2 PBX switch **10** and the voice dialing lines **20** have been illustrated. The remaining components of the telephone system, as shown in Figure 1, have been omitted from Figure 2 to simplify the illustration. The voice dialing server can be implemented using a conventional personal computer, depicted diagrammatically at **28**, that has been equipped with the voice dialing server software described more fully below. The voice dialing server embodiment of Figure 2 uses an analog interface **30** that plugs into the PC bus **32** and has ports for connecting to voice dialing lines **20**. An optional digital interface **34** may be connected through a plurality of RS-232 lines **36** to the serial ports **38** of computer **28**. In this case there would be a digital line for each analog line. The digital interface is connected in parallel with the analog interface to the voice dialing lines **20**. Computer **28** includes a central processing unit **40** and random access memory **42**. These are coupled to PC bus **32** in conventional fashion. A disk drive **44** is used to store the multiuser phone number dictionaries, as well as the boot copy of the voice dialing server software. The voice dialing server software is loaded into

RAM 42, where it is accessed by the CPU 40 during execution. Disk drive 44 may be coupled through any suitable interface such as a SCSI interface 46 to the PC bus 32.

The analog interface of this embodiment may be a model D41E voice board available from Dialogic. Analog interface 30 includes a digital signal processor (DSP) and a general purpose microprocessor. The interface is capable of handling all telephony signal and it performs DTMF (touchtone) detection and generation as well as audio/voice signal processing tasks. The D41E voice board from Dialogic supports four independent voice channels.

The digital interface 34 is a protocol converter that converts the digital control signals from PBX switch 10 into serial signals conforming to the telephony application programming interface (TAPI) protocol established by Microsoft Corporation. The digital interface 34 is optional. Essentially, it is provided to allow the voice dialing server to determine the user's extension number automatically. The TAPI protocol is used to employ a caller ID function that will tell the voice dialing server what extension the user is calling from. Knowing this extension allows the voice dialing server to automatically use the phone number dictionary that is preassigned to that caller's extension. Without the caller ID information, the voice dialing server will need to prompt the user to enter his or her extension in order to activate the correct phone number dictionary.

An alternate embodiment of the invention is depicted in Figure 3. The embodiment of Figure 3 is similar to that of Figure 2 except that a dedicated digital interface **35** is used in place of analog interface **30** and digital interface **34**. The dedicated digital interface is designed to directly connect with a predetermined make and model of PBX switch. The availability of such a dedicated digital interface **35** depends on the make and model of the PBX system. One such system is a Norstar PBX switch using a D/42-NS voice board as digital interface **35**. The D/42-NS voice board is available from Dialogic. It functions similar to the D41E analog voice board described above, with additional digital control features built-in to interface with the Norstar PBX switch.

As noted above, the presently preferred embodiments are implemented using a suitably programmed personal computer. Figure **4** is a software entity relationship diagram showing the preferred software architecture that may be used to program the computer. Essentially, the software performs two functions: a voice interaction function and a PBX control function. From a voice and control signal standpoint, all communication with the PBX switch **10** is through an interface **60**. The interface **60** supports both bidirectional voice communication and digital control information. The software of the preferred embodiment assumes that the voice channel has been digitized, hence the voice information communicated through interface **60** is digital audio data. If analog voice signals are present in the PBX system, they may be converted into digital signals through the analog interface hardware **30** (Figure 2).

Connected to interface **60** is the kernel module **62** that oversees the operation of the server software. Attached to the kernel module **62** is the voice recognizer module **64** and speech synthesis module **66**. The voice recognizer **64** works with a multiuser phone book dictionary **68** that contains all of the multiple users' personal phone book information, that is, the names and phone numbers that the users have entered by speaking the names and entering the numbers using DTMF tones entered through the touchtone keypad of the station handset. A subset of kernel module **62** are the PBX control functions **70**. These are a stored set of digital control commands that cause the PBX **10** to execute certain control functions, in effect mimicking the control functions that a user of a telephone station handset might employ. The PBX control functions include the ability to place a call on hold and to request the PBX switch to set up a conference call. These commands are used during dialing of the selected phone number and thereafter to connect the user to the selected party. See pseudocode in the Appendix for details.

Figure 5 is a flowchart showing how a user (at extension #214) might use the voice dialing server (at extension #100) to place an outside call using the voice dialer dictionary. Alongside the numbered boxes of the flowchart several reproductions of Figure **1** have been illustrated, showing in bold lines how the switching actually occurs. The reader may wish to refer to these switching diagrams while reading the flowchart of Figure 5.

The procedure begins at Step **90**. The user at extension #214 lifts the handset of the telephone station and dials the extension of the voice dialing server (#100). The server answers the call and prompts the user for a name at Step 92. To effect this step the analog interface 30 (Figure 2) or the dedicated digital interface 35 (Figure 3) detects the ring signal and answers the incoming call. The extension number of the user's station is detected at this point for use in selecting the proper dictionary. The user may override by entering a different extension number. The incoming call event is transmitted through interface 60 (Figure 4) to the kernel module 62. In response, the kernel module 62 employs the speech synthesis module 66 to prompt the user for a name and then monitors the voice channel (through interface 60) while employing the recognizer module 64.

Returning to Figure 5, when the server recognizes the name spoken by the caller at Step 94, the server looks up the phone number to dial using the multiuser phone book dictionary 68 (Figure 4). If the voice recognizer does not identify a name in the dictionary, or if the recognized name is below a predetermined reliability threshold the kernel module 62 may employs the speech synthesis module 66 to prompt the user to try again.

After recognizing the name and looking up the phone number, the kernel module 62 of the server prompts the user by repeating the name and asking the user to verify that the name is correct. The user may then either answer yes or no. If the answer is yes, the server will proceed to place the call. If the answer is no, the server will prompt the user to try again.

During these first three steps (Steps 90-94) the user's extension is connected through the PBX switch to the voice dialing server. This is shown in the switching diagram adjacent Steps 90-94. Bold lines are used to show the connection.

After obtaining the number to call and receiving the user's verification, the server then at Step 96 temporarily places the user on hold or in conference call mode. Then in Step 98 the server places a call through the PBX switch to the phone number that was determined during the lookup procedure. As illustrated at B the user's extension (#214) is temporarily placed on hold while the server is connected to an outside line via the PBX switch. Note that this technique allows the voice dialing server to connect to an outside line without the need to employ a separate inside extension. To effect this operation the kernel module 62 uses one of the PBX control functions 70 to send a request through interface 60 to the PBX. The request causes the PBX to place the user's extension on hold or in conference call mode and then causes the PBX switch to connect the server's extension (#100) to an outside line. This is done by mimicking the control signal commands that would be sent by a user of a telephone station handset to effect these same functions.

After establishing an outside line connection and receiving a dial tone, the server places the call by dialing the number that was looked up. The kernel 62 performs this operation by using the DTMF dialing capabilities of the analog interface 30 (Figure 2) or the digital interface 35 (Figure 3).

After dialing the desired number the server causes the PBX switch to conference in the user's extension at Step 100. As shown at C, the user's extension (#214) and the voice dialing server's extension (#100) are now both connected through a conference call to the outside line. Finally, in Step 102 the server drops out of the communication as illustrated at D. This leaves the user's extension (#214) connected to the outside line and frees up the server for its next use by another user.

The present invention employs a unique compact speech representation based on regions of high phoneme similarity values. As shown in Figure 6, there is an overall consistency in the shape of the phoneme similarity time series for a given word. In Figure 6 phoneme similarity time series for the word "hill" spoken by two speakers are compared. Although the precise wave shapes differ between the two speakers, the phoneme similarity data nevertheless exhibit regions of similarity between the speakers. Similar behavior is observed in the phoneme plausibility time series that has been described by Gong and Haton in "Plausibility Functions in Continuous Speech Recognition: The VINICS System," *Speech Communication*, Vol. 13, Oct. 1993, pp. 187-196.

Conventional speech recognition systems match each input utterance to reference templates, such as templates composed on phoneme similarity vectors, as in the model speech method (MSM) of Hoshimi et al., "Speaker-Independent Speech Recognition Method Using Training Speech From a Small Number of Speakers," ICASSP, Vol. 1, pp. 469-472, 1992. In these conventional systems the reference speech representation is frame-based and requires a high data rate, typically 8 to 12 parameters every 10 to 20 milliseconds. The frame-by-frame alignment that is required with these conventional systems is computationally costly and makes this approach unsuitable for larger vocabularies, especially when using small hardware.

The present system uses a multistage word recognizer that is applied prior to a frame-by-frame alignment, in order to reduce the search space and to achieve real time performance improvements. The number of stages in the recognizer, as well as the computational complexity of each stage and the number of word candidates preserved at each stage, can be adjusted to achieve desired goals of speed, memory size and recognition accuracy for a particular application. The word recognizer uses an initial representation of speech as a sequence of multiple phoneme similarity values. However, the word recognizer further refines this speech representation to preserve only the interesting regions of high phoneme similarity. Referring to Figure 7, the interesting regions of high phoneme similarity value are represented as high similarity regions. By representing the speech as features at a lower data rate in the initial stages of recognition, the complexity of the matching procedure is greatly reduced.

The multistage word recognizer also employs a unique scoring procedure for propagating and combining the scores obtained at each stage of the word recognizer in order to produce a final word decision. By combining the quasi-independent sources of information produced at each stage, a significant gain in accuracy is obtained.

The system's architecture features three distinct components that are applied in sequence on the incoming speech to compute the best word candidate.

Referring to Figure 8, an overview of the presently preferred system will be presented. The first component of the present system is a phoneme similarity front end **110** that converts speech signals into phoneme similarity time series. Speech is digitized at 8 kilohertz and processed by 10th order linear predictive coding (LPC) analysis to produce 10 cepstral coefficients every 100th of a second. Each block of 10 successive frames of cepstral coefficients is compared to 55 phoneme reference templates (a subset of the TIMIT phoneme units) to compute a vector of multiple phoneme similarity values. The block of analysis frames is then shifted by one frame at a time to produce a vector of phoneme similarity values each centisecond (each 100th of a second). As illustrated in Figure 8, the phoneme similarity front end works in conjunction with a phone model database **112** that supplies the phoneme reference templates. The output of the phoneme similarity front end may be stored in a suitable memory for conveying the set of phoneme similarity time series so generated to the word recognizer stages.

The word recognizer stages, depicted in Figure 8 generally at **114**, comprise the second major component of the system. A peak driven procedure is first applied on the phoneme similarity time series supplied by front end **110**. The peak driven procedure extracts High Similarity Regions (HS Regions). In this process, low peaks and local peaks of phoneme similarity values are discarded, as illustrated in Figure 7. In the preferred embodiment regions are characterized by 4 parameters: phoneme symbol, height at the peak location and time locations of the left and right frames. Over our data corpus, an average of 60 regions per second of speech is observed. In Figure 8 the high similarity region extraction module **116** performs the peak driven procedure. The output of the HS region extraction module is supplied to two different word recognizer stages that operate using different recognizer techniques to provide a short list of word candidates for the fine match final recognizer stage **126**.

The first of the two stages of word recognizer **114** is the Region Count stage or RC stage **118**. This stage extracts a short list of word candidates that are then supplied to the next stage of the word recognizer **114**, the Target Congruence stage or TC stage **120**. The RC stage **118** has an RC word prototype database **122** that supplies compact word representations based on the novel compact speech representation (regions of high phoneme similarity values) of the invention. Similarly, the TC stage **120** also includes a TC word prototype database **124** that supplies a different compact word representation, also based on the compact speech representation of the invention. The TC stage provides a more selective short list of word candidates, essentially a further refinement of the list produced by the RC stage **118**.

The word decision stage **126**, the final major component of the present system, selects the word with the largest score from the short list supplied by TC stage **120**.

Region Count Modeling

The RC stage **118** of word recognizer **114** represents each reference word with statistical information on the number of HS regions over a predefined number of time intervals. The presently preferred embodiment divides words into three equal time intervals in which each phoneme interval is described by (1) the mean of the number of HS regions occurring in that interval and (2) a weight that is inversely proportional to the square of the variance, which indicates how reliable the region count is. Specifically for a score normalized between 0 and 100, the weight would be $100/(\text{variance}^2 + 2)$. These parameters are easily estimated from training data. In the currently preferred implementation, each word requires exactly 330 parameters, which corresponds to two statistics, each over three intervals each comprising 55 phoneme units (2 statistics x 3 intervals x 55 phoneme units).

Region count modeling was found to be very effective due to its fast alignment time (0.33 milliseconds per test word on a Sparc10 workstation) and its high top 10% accuracy.

The region count prototype is constructed as follows. A first utterance of a training word or phrase is represented as time-dependent phoneme similarity data. In the presently preferred embodiment each utterance is divided into N time intervals. Presently each utterance is divided into three time intervals, with each time interval being represented by data corresponding to the 55 phonemes. Thus the presently preferred implementation represents each utterance as a 3 x 55 vector. In representing the utterance as a 3 x 55 vector, each vector element in a given interval stores the number of similarity regions that are detected for each given phoneme. Thus if three occurrences of the phoneme "ah" occur in the first interval, the number 3 is stored in the vector element corresponding to the "ah" phoneme.

An inductive or iterative process is then performed for each of the successive utterances of the training word or phrase. Specifically, each successive utterance is represented as a vector like that of the first utterance. The two vectors are then combined to generate the vector sum and the vector sum of the squares. In addition, a scalar count value is maintained to keep track of the current number of utterances that have been combined.

The process proceeds inductively or iteratively in this fashion, each new utterance being combined with the previous ones such that the sum and sum of squares vectors ultimately represent the accumulated data from all of the utterances.

Once all training utterances have been processed in this fashion the vector mean and vector variance are calculated. The mean vector is calculated as the sum vector divided by the number of utterances used in the training set. The vector variance is the mean of the squares minus the square of the means. The mean and variance vectors are then stored as the region count prototype for the given word or phrase. The same procedure is followed to similarly produce a mean and variance vector for each of the remaining words or phrases in the lexicon.

When a test utterance is compared with the RC prototype, the test utterance is converted into the time dependent phoneme similarity vector, essentially in the same way as each of the training utterances were converted. The Euclidean distance between the test utterance and the prototype is computed by subtracting the test utterance RC data vector from the prototype mean vector and this difference is then squared. The Euclidean distance is then multiplied by a weighting factor, preferably the reciprocal of the prototype variance. The weighted Euclidean distance, so calculated, is then converted into a scalar number by adding each of the vector component elements. In a similar fashion the weighting factor (reciprocal of the variance) is converted into a scalar number by adding all of the vector elements. The final score is then computed by dividing the scalar distance by the scalar weight.

The above process may be repeated for each word in the prototype lexicon and the most probable word candidates are then selected based on the scalar score.

Target Congruence Modeling

The second stage of the word recognizer represents each reference word by (1) a prototype which consists of a series of phoneme targets and (2) by global statistics, namely the average word duration and the average "match rate," which represents the degree of fit of the word prototype to its training data. In the presently preferred embodiment targets are generalized HS regions described by 5 parameters:

1. phoneme symbol;
2. target weight (percentage occurrence in training data);
3. average peak height (phoneme similarity value);
4. average left frame location;
5. average right frame location.

Word prototypes are automatically created from the training data as follows. First, HS regions are extracted from the phoneme similarity time series for a number of training speakers. The training data may be generated based on speech from a plurality of different speakers or it may be based on multiple utterances of the same training words by a single speaker. Then, for each training utterance of a word, reliable HS regions are computed by aligning the given training utterance with all other utterances of the same word in the training data. This achieves region-to-region alignment.

For each training utterance the number of occurrences (or probability) of a particular region is then obtained. At that time, regions with probabilities less than a pre-established Reliability Threshold (typically 0.25) are found unreliable and are eliminated. The word prototype is constructed by merging reliably detected, high similarity regions to form *targets*. At the end of that process a target rate constraint (i.e. desired number of targets per second) is then applied to obtain a uniform word description level for all the words in the lexicon. The desired number of targets per second can be selected to meet system design constraints such as the ability of a given processor to handle data at a given rate. By controlling the target rate a reduction in the number of targets is achieved by keeping only the most reliable targets. Once the word prototype has been obtained in this fashion, the average match rate and average word duration are computed and stored as part of the word prototype data.

The number of parameters needed to represent a word depends on the average duration of the word and on the level of phonetic detail that is desired. For a typical 500 millisecond word at 50 targets per second, the speech representation used by the presently preferred embodiment employs 127 parameters, which correspond to 5 values per target x 50 targets per second x 0.5 seconds + 2 global statistics (average match rate and average word duration).

Figure 9 illustrates the word prototype training procedure by which the TC word prototype database **124** is constructed. The RC word prototype database **122** is constructed by similar, but far simpler process, in that only the presence or absence of an HS region occurring with each of the three equal time intervals must be detected.

Referring to Figure 9, the HS Region Computation Module **116** is used to convert the similarity time series from the speech database into a list of HS regions. The alignment module **130** operates on this list of HS regions to eliminate unreliable regions by alignment across speakers. Again, the process can be performed across a plurality of different speakers or across a plurality of utterances by the same speaker.

Next the list of reliable regions, together with the associated probabilities of detecting those regions is passed to the target building module **132**. This module builds targets by unifying the region series to produce a list of phoneme targets associated with each word in the database. This list of phoneme targets is then supplied to a module **134** that adjusts the target rate by applying the target rate constraint. The target rate constraint (the desired number of targets per second) may be set to a level that achieves the desired target rate. After adjusting the target rate a statistical analyzer module **136** estimates the global statistics (the average match rate and the average word duration) and these statistics along with the list of targets at the selected rate are then stored as the TC word prototype database **124**.

Word Recognition

Given an active lexicon of N words, the region count stage is first applied to produce a short list of word candidates with normalized scores. A weighted Euclidean distance is used to measure the degree of fit of a test word X to a reference word P (in RC format as supplied by the RC word prototype database). Specifically, in the current implementation the weighted Euclidean distance is defined as

$$D_{RC}(X,P)=\sum_{i=1}^{3}\sum_{j=1}^{55}(x_{ij}-p_{ij})^2 w_{ij} / \sum_{i=1}^{3}\sum_{j=1}^{55} w_{ij}$$

where x_{ij} is the number of HS regions in time interval I for phoneme j, where p_{ij} is the corresponding average number of HS regions estimated on training data, and where w_{ij} is the corresponding weight. The N/10 highest scoring word prototypes are preserved as word candidates and their scores (weighted Euclidean distances) are normalized by dividing each individual score by the highest score. This defines a normalized score S_{RC} for each word. Normalized scores range from 0 to 1 and are dimensionless, making it possible to combine scores resulting from different scoring methods.

The target congruence stage is then applied on each word candidate selected by the RC stage. A region-to-target alignment procedure is used to produce a congruence score between the test word and a given word reference (in TC format as supplied by the TC word prototype database). The congruence score of a matched target CGmatch, that is, the alignment found between target t of the prototype and region r of the test word, is defined as

$$CG_{match}(t,r) = \min(A_t / A_r, A_r / A_t)$$

where A_t and A_r respectively represent the target's area and the aligned region's area in the time similarity plane.

The congruence score of an unmatched target CGunmatch is computed in the same way, using an estimate for the area A_r of the missing HS region. The estimated area A_r is computed as the area under the similarity curve for the target's phoneme label, between the projected locations of the target's left and right frames.

The word congruence score is computed as the weighted sum of congruence scores for all the targets, divided by the sum of their weights. Normalized congruence scores S_{TC} are computed by dividing the individual congruence scores by the highest congruence score. The final score output by the word recognizer is a combination of the information obtained at each recognizer stage. In the presently preferred embodiment the final score output of the recognizer is:

$$S_{Hypo} = (S_{RC} + S_{TC})/2$$

The recognized word is the one with the highest S_{Hypo} value.

APPENDIX

Notes:

The function TransferCallDesklab(Number) does the transfer to an inside extension by calling the PBX function "feature 7 0" followed by the extension number after a hookflash. Then the line is released.

The function TransferExternalCallDeskLab(Number) does the transfer outside. In the program a message is played, then the user is put on hold (by sending "feature 7 9", then the program gets an external line, then a conference call is established, the phone number is dialed, and the line is released.

Pseudocode:

```
int TransferCallDeskLab(Number)
char *Number;
{
  int LastRet;
  int Ret;


  ghookflash((*DskLab).Desc,500);
  gdial((*DskLab).Desc,"*70",1);
  gdial((*DskLab).Desc,Number,1);
  gphone_hookswitch((*DskLab).Desc,1);
  while (Ret=gphone_status((*DskLab).Desc,&LastRet)!=G_ONHOOK)
    {
      sleep(1);
    }
}

int TransferExternalCallDeskLab (phoneNumber)
        char *phoneNumber;
```

```
{
  int LastRet;
  int Ret;
  int lastatus=-199, rtnval;
  int thereIsProblem,
      state,
      new_state,      last_state;
extern char *G_PhoneStatus[];
    char msg[] = "Calling";

    /* Play message while transfering */
    ALIPlayMessage(msg);
    esleep(1,1000);

    printf("Putting calling line on hold ...");fflush(stdout);
    ghookflash((*DskLab).Desc,500);
    gdial((*DskLab).Desc,"*79",1);
    printf("done!\n"); fflush(stdout);

    printf("Getting  external line .. "); fflush(stdout);

    gdial((*DskLab).Desc,"9",0);   /* obtain an external line */

    state = 0;
    do {
        esleep(0,1000);
        new_state = gphone_status((*DskLab).Desc,&last_state);
        if (state != new_state)  {
                state = new_state;
                printf("state = %s\n",G_PhoneStatus[state]);
        }
        thereIsProblem = 0;

        switch (state) {
           case G_ONHOOK:   /* call disconnected -- strangely */
           case G_BUSY:     /* cannot get an outside line */
           case G_REORDER:
           case G_REORDER2: thereIsProblem = 1;
           default: break;
        }
    }  while ( (state != G_DIALTONE ) && (state != G_CONNECTED)
                && !thereIsProblem );

    printf("done!\n"); fflush(stdout);
    printf("Establishing Conference Call ...");fflush(stdout);

    ghookflash((*DskLab).Desc,500);    gdial((*DskLab).Desc,"*3",1);
```

```
    ALIPlayMessage(msg);

    printf("Dialing %s... ",phoneNumber); fflush(stdout);

    gdial((*DskLab).Desc,phoneNumber,1);
    printf("done!\n"); fflush(stdout);

    state = 0;
    do  {
        esleep(0,10000);                /* 1/4 second sleep */
        new_state = gphone_status((*DskLab).Desc,&last_state);
        if (state != new_state)  {
            state = new_state;
            printf("state = %s\n",G_PhoneStatus[state]);fflush(stdout);
        }
        thereIsProblem = 0;

        switch (state) {
          case G_ONHOOK:   /* call disconnected -- strangely */
          case G_BUSY:     /* cannot get an outside line */
          case G_REORDER:
          case G_REORDER2: thereIsProblem = 1;
          default: ;
        }
    }  while ((state !=G_CONNECTED ) && (state != G_BUSY) &&
!thereIsProblem);

    printf("Putting phone ONHOOK ...");fflush(stdout);

    gphone_hookswitch((*DskLab).Desc,G_ONHOOK);
    while ((Ret=gphone_status((*DskLab).Desc,&LastRet))!=G_ONHOOK)
    {
      sleep(1);
    }
    printf("done!\n"); fflush(stdout);
}
```

CLAIMS

1. A voice dialing server for coupling to a branch exchange telephone system of the type that provides call switching among a plurality telephone extension ports, comprising:

an interface for connection to at least a first one of said telephone extension ports to support transmission of voice signals and telephone system control information;

a speech processing module coupled to said interface for (a) answering a call placed to the voice dialing server by a user (b) processing speech input from the user corresponding to a selected party to be called and (c) looking up a phone number of a selected party;

branch exchange control module coupled to said interface and to said speech processing module for issuing control information to said telephone system (a) to cause the extension assigned to said interface to connect to second port (b) to establish communication via the second port and (c) to attach the user's extension to the second port, whereby the user is placed in communication with the selected party.

2. The server of Claim 1 wherein said speech processing module supports a plurality of user phone number dictionaries.

3. The server of Claim 2 wherein said branch exchange control module includes system for communicating with said telephone system to determine the identity of the user's extension and for using this identity to select one of said plurality of phone number dictionaries for use by said speech processing module.

4. The server of Claim 2 wherein said branch exchange control module includes system responsive to keyed user input for selecting one of said plurality of phone number dictionaries for use by said speech processing module.

5. The server of claim 1, 2, 3 and 4, wherein said branch exchange control module issues control information to said telephone system to place the user's call to the voice dialling server on hold while establishing communication via said second port.

6. The server of any one of claims 1 to 5, wherein said branch exchange control module issues control information to said telephone system to transfer the user's call to the voice dialling server to another extension on said telephone system.

7. The server of any one of claims 1 to 6, wherein said branch exchange control module is implemented on a computer having a bus and said interface comprises an analog interface coupled to the bus of said computer.

8. The server of any one of claims 1 to 6, wherein said branch exchange control module is implemented on a computer having at least one

serial port and said interface comprises an analog interface coupled to the serial port of said computer.

9. The server of any one of claims 1 to 6, wherein said branch exchange control module is implemented on a computer having a bus and said interface comprises an digital interface coupled to the bus of said computer.

10. The server of any one of claims 1 to 9, wherein said speech processing module includes a speech recognizer that represents speech as high phoneme similarity values.

11. The server of any one of claims 1 to 9, wherein said speech processing module includes a speech recognizer comprising a word recognizer that employs a region count stage that extracts a list of word candidates based on regions of high phoneme similarity values.

12. The server of any one of claims 1 to 9, wherein said speech processing module includes a speech recognizer comprising a word recognizer that employs:

a region count stage that extracts a first list of word candidates based on regions of high phoneme similarity values, and

a target congruence stage that extracts a second list of word candidates from said first list based on regions of high phoneme similarity values.

13. A server constructed and arranged to operate as hereinbefore described with reference to the accompanying drawings.

The Patent Office

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Documents considered to be relevant:

Category	Identity of document and relevant passage	Relevant to claims
X	EP 0283120 A1 (BRITISH TELECOM) - see whole document	1 at least
X	EP 0166318 A2 (STANDARD ELECTRIC) - see fig.1 & US4763350	1 at least
X,P	WPI Abstract Accession No: 97-305198/199728 & JP 9116940A (MATSUSHITA) 02.05.97 (see abstract)	1 at least
X	Patent Abstracts of Japan, vol.94, No.10, & JP 6303320A (MATSUSHITA) 28.10.94 (see abstract)	1 at least
X	WPI Abstract Accession No: 94-195600/199424 & JP 6133044A (FUJITSU) 13.05.94 (see abstract)	1 at least

X	Document indicating lack of novelty or inventive step	A	Document indicating technological background and/or state of the art.
Y	Document indicating lack of inventive step if combined with one or more other documents of same category.	P	Document published on or after the declared priority date but before the filing date of this invention.
&	Member of the same patent family	E	Patent document published on or after, but with priority date earlier than, the filing date of this application.

FACSIMILE STORE AND FORWARD SWITCHING SYSTEM

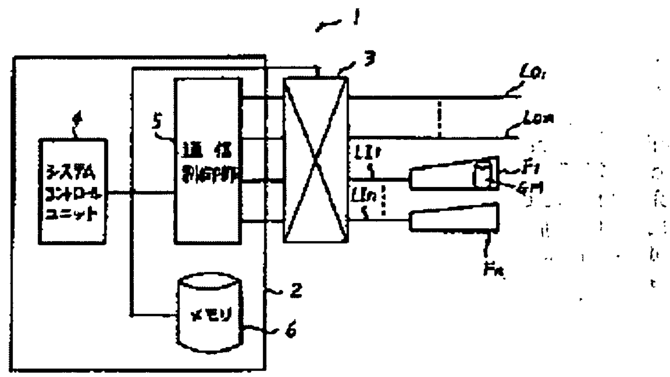
Patent number: JP1258526
Publication date: 1989-10-16

Inventor: TSUNODA NAOKI
Applicant: RICOH CO LTD

Classification:
- international: H04L11/20
- european:
Application number: JP19880086839 19880407
Priority number(s):

Abstract of **JP1258526**

PURPOSE:To execute the effective use of the memory of a Fax store and forward switching system by providing at least one terminal of the facsimile(Fax) terminal of an extension equipped with a picture memory and providing a means to accumulate prescribed picture information into the picture memory and transmit the transfer information of the effect to the related Fax terminal.
CONSTITUTION:In a Fax store and forward switching system 1, for example, when the time designation transmission is requested from extension Fax terminals F1-Fn, the system 1 accumulates temporary the picture information of an original from the extension Fax terminals F1-Fn to request the communication request to a memory 6 at the time of the designation time, the picture information is read and transmitted to a designed destination from the memory 6. When the confidential transmission service from the Fax terminal of line wires LO1-LOm to special extension Fax terminals F1-Fn is requested, the system 1 receives the picture information from a requester, accumulates the picture information into the memory 6, and thereafter, the prescribed identifier is designated from the extension Fax terminals F1-Fn, an in case of requesting reception, the system answers the reception request and transmits the picture information of the memory 6.



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㉔発明の名称　ファクシミリ蓄積交換装置

㉑特　　願　昭63－86839
㉒出　　願　昭63(1988)4月7日

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明　細　書

1．発明の名称

ファクシミリ蓄積交換装置

2．特許請求の範囲

（1）複数の内線と外線を介して複数のファクシミリ端末に接続され、各ファクシミリ端末からの画情報を受信して一旦メモリに蓄積した後、所定の宛先に送信するファクシミリ蓄積交換装置において、内線のファクシミリ端末として画像メモリを備えたファクシミリ端末を少なくとも1端末設け、受信してメモリに蓄積した画情報を、所定の宛先に送信できないとき、該画情報を該画像メモリ付ファクシミリ端末に送信して該画像メモリに蓄積させ、当該宛先等にその旨の転送情報を送信することを特徴とするファクシミリ蓄積交換装置。

（2）前記画像メモリ付ファクシミリ端末への画情報の送信をあらかじめ設定した所定の時間毎あるいはメモリに画情報が所定容量まで蓄積された時に行うことを特徴とする請求項1記載のファクシミリ蓄積交換装置。

3．発明の詳細な説明

（産業上の利用分野）

本発明はファクシミリ蓄積交換装置に関し、画情報を蓄積するメモリの有効利用を図ったファクシミリ蓄積交換装置に関する。

（従来の技術）

近時、ファクシミリ装置が普及するにつれ、複数の内線および外線を介して複数のファクシミリ端末に接続され、各ファクシミリ端末からの画情報をメモリに蓄積した後、指定された宛先に送信するファクシミリ蓄積交換装置が出現している。

このようなファクシミリ蓄積交換装置においては、種々の通信サービスを行っており、例えば、内線ファクシミリ端末から発呼があると、通常のファクシミリ通信手順に従ってファクシミリ制御信号の交換を行った後、当該ファクシミリ端末の有している機能に応じた通信手順を設定し、画情報の受信を行っている。このとき、内線ファクシ

ミリ端末が時刻指定送信を要求すると、ファクシミリ蓄積交換装置は受信した画情報を一旦メモリに蓄積した後、指定の送信時刻に指定された宛先に通常のファクシミリ通信手順に従って送信する。

　また、外線ファクシミリ端末から発呼があると、送信相手の内線ファクシミリ端末に接続したり、ファクシミリ蓄積交換装置が当該外線ファクシミリ端末からの画情報を受信してメモリに蓄積した後、内線ファクシミリ端末に送信している。また、外線ファクシミリ端末が親展サービスを要求すると、ファクシミリ蓄積交換装置は、受信した画情報をメモリに蓄積し、その後、内線ファクシミリ端末からの受取要求があると、メモリ内の画情報を送信する。

（発明が解決しようとする課題）

　しかしながら、このような従来のファクシミリ蓄積交換装置にあっては、メモリに蓄積した画情報を相手ファクシミリ端末がビジー状態であるとか、紙切れである等の理由により送信できない場合、画情報が送信されずに不達原稿等としてメモリにいつまでも蓄積される。したがって、メモリを有効に利用することができなかった。

　そこで、従来、蓄積した画情報の送信先にあらかじめ設定した所定回数だけ再発呼処理を行っても送信できない場合や、親展サービスで所定の許容時間内に受取要求がない場合には、ファクシミリ蓄積交換装置のメモリに蓄積した不達原稿や親展原稿の画情報を消去するファクシミリ蓄積交換装置もある。

　しかしながら、ファクシミリ蓄積交換装置のメモリに蓄積した不達原稿や親展原稿の画情報を消去してしまうと、ファクシミリ蓄積交換装置に通信サービスを依頼したファクシミリ端末のオペレータは同様の通信サービスを繰り返し要求する必要があり、ファクシミリ蓄積交換装置の利用性が低下し、また、不達原稿の画情報を有効に利用することができない。

（発明の目的）

　そこで、本発明は、内線ファクシミリ端末として画像メモリを備えたファクシミリ端末を少なくとも１端末設け、不達原稿となった画情報や親展原稿の画情報を該ファクシミリ端末に送信してその画像メモリに蓄積し、その旨転送情報を該当するファクシミリ端末に送信することにより、ファクシミリ蓄積交換装置のメモリが不達原稿や親展原稿の画情報により占領されることを防止するとともに、不達原稿や親展原稿の画情報を内線ファクシミリ端末の画像メモリに利用可能な状態で蓄積して、ファクシミリ蓄積交換装置のメモリの有効利用を図るとともに、ファクシミリ蓄積交換装置を利用した通信サービスの便利性をより一層向上させることを目的としている。

（発明の構成）

　本発明は、上記目的を達成するため、

（１）複数の内線と外線を介して複数のファクシミリ端末に接続され、各ファクシミリ端末からの画情報を受信して一旦メモリに蓄積した後、所定の宛先に送信するファクシミリ蓄積交換装置において、内線のファクシミリ端末として画像メモリを備えたファクシミリ端末を少なくとも１端末設け、受信してメモリに蓄積した画情報を、所定の宛先に送信できないとき、該画情報を該画像メモリ付ファクシミリ端末に送信して該画像メモリに蓄積させ、当該宛先等にその旨の転送情報を送信することを特徴とするもの。

および、

（２）前記画像メモリ付ファクシミリ端末への画情報の送信をあらかじめ設定した所定の時間毎あるいはメモリに画情報が所定容量まで蓄積された時に行うことを特徴とするものである。

　以下、本発明の実施例に基づいて具体的に説明する。

　第１、２図は本発明の一実施例を示す図である。

　第１図において、１はファクシミリ蓄積交換装置であり、ファクシミリ蓄積交換装置１は本体２と構内交換機（Private Branch Exchange：ＰＢＸ）３を備えている。

　本体２は、システムコントロールユニット４、通信制御部５およびメモリ６を備えており、構内交換機３には内線$L1_1$～$L1_n$を介して内線フ

ファクシミリ端末F_1～F_nが接続されるとともに、外線LO_1～LO_mが接続されている。構内交換機3はシステムコントロールユニット4からの指示に従い、発呼動作を行うとともに、内線LI_1～LI_n相互および内線LI_1～LI_nと外線LO_1～LO_mとの接続・交換を行う。

内線ファクシミリ端末F_1～F_nのうち内線ファクシミリ端末F_1は画像メモリＧＭを備えており、画像メモリＧＭは原稿複数ページ分の画情報を蓄積する容量を有している。

通信制御部５は変・復調器、圧縮・再生器、ダブルバッファメモリ等を備えている。通信制御部５はシステムコントロールユニット４からの指示に従い、相手ファクシミリ端末（内線ファクシミリ端末F_1～F_nまたは外線に接続されたファクシミリ端末）とファクシミリ制御信号の交換を行うとともに、相手ファクシミリ端末より送信されてきた画情報の復調、冗長度抑圧符号化を行いながらダブルバッファメモリの片方に貯めこみ、一杯になると、メモリ６に転送するという入力動作、およびメモリ６から転送されてきた画情報をダブルバッファメモリの片方に貯めた後、変調して相手ファクシミリ端末に送出するという出力動作を行う。

システムコントロールユニット４は内部ＲＯＭに格納するプログラムに従ってファクシミリ蓄積交換装置１の各部を制御してファクシミリ蓄積交換装置１としてのシーケンスを実行するとともに、本発明の不達原稿の画情報転送処理を実行する。

メモリ６は各ファクシミリ端末から送信されてきた画情報を蓄積し、メモリ６に蓄積された画情報は指定宛先毎にシステムコントロールユニット４によりファイル管理されている。

次に、作用を第２図に示すフローチャートに基づいて説明する。

ファクシミリ蓄積交換装置１は内線ファクシミリ端末F_1～F_nあるいは外線LO_1～LO_mに接続されたファクシミリ端末からの通信サービス要求があると、その通信サービス要求に応じて種々の通信サービスを提供する。例えば内線ファクシミリ端末F_1～F_nから時刻指定送信の要求があると、ファクシミリ蓄積交換装置１は通信要求を依頼してきた内線ファクシミリ端末F_1～F_nからの原稿の画情報を受信して一旦メモリ６に蓄積し、指定時刻になると、メモリ６から当該画情報を読み出して指定された宛先に送信する。また、外線LO_1～LO_mのファクシミリ端末から特定の内線ファクシミリ端末F_1～F_n宛への親展送信サービスの要求があると、ファクシミリ蓄積交換装置１は依頼元からの画情報を受信してメモリ６に蓄積し、その後、内線ファクシミリ端末F_1～F_nから所定のＩＤ（識別子）を指定して受取要求があると、その受取要求に答えてメモリ６の画情報を送信する。

このように、ファクシミリ蓄積交換装置１では通信サービスの内容によってはメモリ６に画情報を蓄積し、通信サービスが完了するまで、メモリ６に蓄積される。しかし、送信先がビジー状態や紙切れ状態等で、メモリ６内の画情報を送信できないときや、親展送信サービスで受取要求がない場合には不達原稿としていつまでもメモリ６に蓄積され、メモリ６の有効利用が図れない。

そこで、本実施例では、内線ファクシミリ端末として画像メモリＧＭを備えたファクシミリ端末F_1を設置し、所定の転送時刻、あるいはメモリ６の蓄積容量等に基づいてメモリ６の不達原稿の画情報を内線ファクシミリ端末F_1に転送してその画像メモリＧＭに蓄積させる。

すなわち、システムコントロールユニット４は、まず、不達原稿がメモリ６に有るか、すなわち、時刻指定送信要求に応じて受信した原稿の画情報で指定時刻に発呼したが受信できず、所定回数再発呼して送信できない画情報がメモリ６に有るかどうか、をチェックし（ステップP_1）、不達原稿が無いときには、受取要求の無い親展原稿の画情報がメモリ６に有るかどうかチェックする（ステップP_2）。親展原稿の画情報も無いときには、転送レポート要求が有るかチェックし（ステップP_3）、転送レポート要求が無いときにはステップP_1に戻る。

ステップP_2あるいはステップP_3で不達原稿あるいは受取要求の無い親展原稿が有るときには、あらかじめ設定された転送時刻かどうかチェックし（ステップP4）、転送時刻になっていないときには、メモリ6への蓄積量があらかじめ設定した蓄積量に達しているかどうかチェックする（ステップP_5）。転送時刻としては、例えば、1日1回あるいは数回適当な時刻を転送時刻として設定してもよいし、また不達原稿では送信指定時刻から何時間か後を転送時刻とし、親展原稿では原稿受信時刻から何時間か後を転送時刻としてもよい。また、メモリ6の蓄積量としてはメモリ6の全メモリ容量から以降の通信サービスとの関係から適宜原稿のページ数等で設定する。

転送時刻になっておらず、メモリ6の蓄積量も所定量に達していないときには（ステップP_4、P_5）、ステップP_1に移行して同様の処理を繰り返す。転送時刻になるか、メモリ6への蓄積量が所定量に達すると、画像メモリＧＭを備えた内線ファクシミリ端末F_1を発呼し（ステップP_6）、不達原稿や親展原稿を内線ファクシミリ端末F_1に転送する（ステップP_7）。このとき、ファクシミリ蓄積交換装置1は、不達原稿についてはメモリ送信して内線ファクシミリ端末F_1の画像メモリＧＭに蓄積させ、親展原稿については親展送信して内線ファクシミリ端末F_1の画像メモリＧＭに蓄積させる。原稿の送信が終了すると、回線を切断する（ステップP_8）。

次いで、ファクシミリ蓄積交換装置1は該当する内線ファクシミリ端末F_1～Ｆｎを発呼して転送レポートを出力する（ステップP_9、P_{10}、P_{11}）。この転送レポートは不達原稿や親展原稿を内線ファクシミリ端末F_1の画像メモリＧＭに転送した旨およびその原稿のリストを示すレポートであり、ファクシミリ装置のレポート機能を利用している。すなわち、ファクシミリ蓄積交換装置1は、まず、不達原稿や親展原稿を転送した内線ファクシミリ端末F_1に発呼し、転送レポートを出力する。したがって、内線ファクシミリ端末F_1のオペレータは内線ファクシミリ端末F_1の画像メモリＧＭに不達原稿や親展原稿の画情報が蓄積されていることを知ることができる。次いで、ファクシミリ蓄積交換装置1は転送原稿が不達原稿のときは、時刻指定送信を依頼してきた内線ファクシミリ端末F_1～Ｆｎを発呼し、転送レポートを出力する。したがって、時刻指定送信を依頼した内線ファクシミリ端末F_1～Ｆｎのオペレータは原稿が消去をされないで、内線ファクシミリ端末F_1の画像メモリＧＭに蓄積されていることを知り、内線ファクシミリ端末F_1の画像メモリＧＭに蓄積されている原稿の画情報を有効に利用することができる。

また、前記ステップP_1で、既に内線ファクシミリ端末F_1に転送した不達原稿や親展原稿に対して送信要求や受取要求があると、ファクシミリ蓄積交換装置1は転送レポート要求と判断して、ステップP_9に移行し、当該内線ファクシミリ端末F_1～Ｆｎに転送レポートを出力する（ステップP_9～P_{11}）。

このように、本実施例のファクシミリ蓄積交換装置1においては、不達原稿や親展原稿を内線ファクシミリ端末F_1の画像メモリＧＭに転送し、画像メモリＧＭに蓄積させることができるので、ファクシミリ蓄積交換装置1のメモリ6を有効に利用することができるとともに、内線ファクシミリ端末F_1の画像メモリＧＭを有効に利用することができる。また、内線ファクシミリ端末F_1の画像メモリＧＭに蓄積した不達原稿や親展原稿を有効に利用することができる。

なお、上記実施例においては、ファクシミリ蓄積交換装置1のメモリ6に蓄積した原稿を転送する条件として、転送時刻とメモリの蓄積量を取り上げたが、これに限るものではなく、また、その条件の設定方法も上記実施例のものに限るものではない。

（効果）

本考案によれば、ファクシミリ蓄積交換装置のメモリが不達原稿や親展原稿の画情報により占領されることを防止することができるとともに、不達原稿や親展原稿の画情報を内線ファクシミリ端末

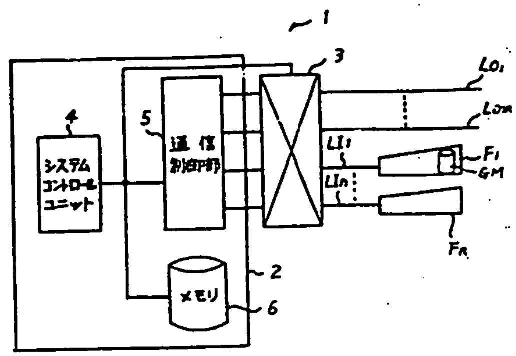
の画像メモリに利用可能な状態で蓄積することができ、ファクシミリ蓄積交換装置のメモリの有効に利用することができるとともに、ファクシミリ蓄積交換装置を利用した通信サービスの便利性をより一層向上させることができる。

4．図面の簡単な説明

第1、2図は本発明のファクシミリ蓄積交換装置の一実施例を示す図であり、第1図はそのファクシミリ蓄積交換装置のブロック図、第2図はその原稿転送処理を示すフローチャートである。

1……ファクシミリ蓄積交換装置、
2……本体、
3……構内交換機、
4……システムコントロールユニット、
5……通信制御部、
6……メモリ、
LI₁～LIn……内線、
LO₁～LOm……外線、
F₁～Fn……内線ファクシミリ端末、
GM……画像メモリ。

第 1 図



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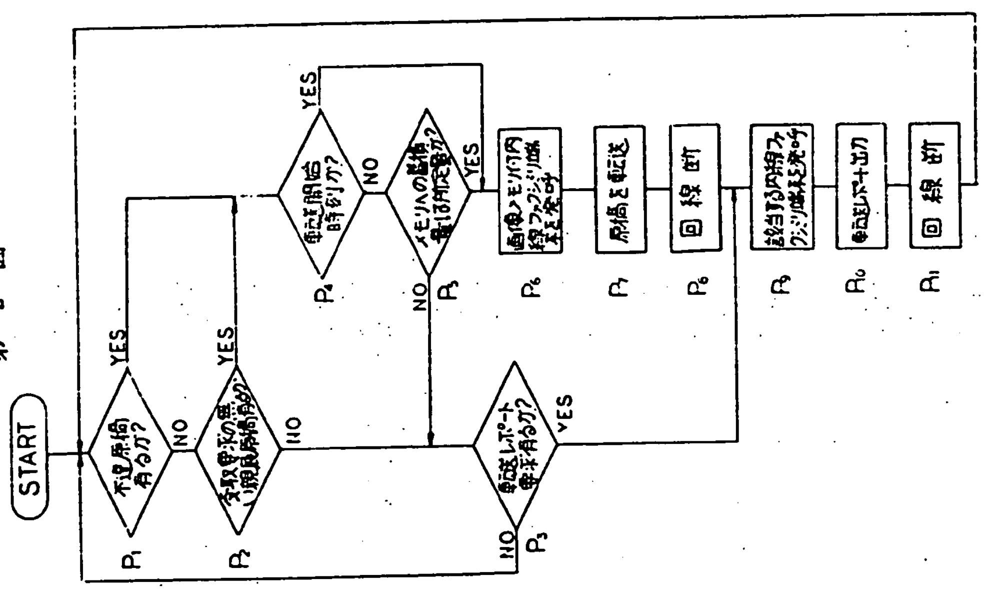

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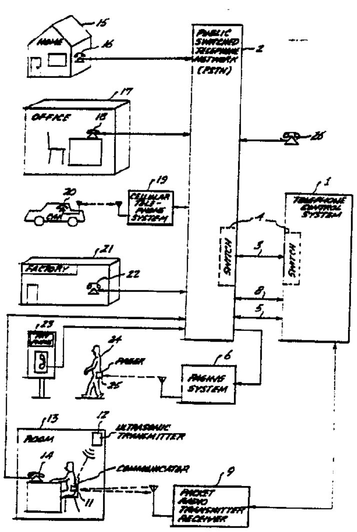

(57) Abstract

A telephone control system (1) which enhances the accessability of system subscribers by providing a variety of call-handling modes, and various ways of programming by which those subscribers can tailor the system. In operation, subscribers are each assigned a telephone number (ACCESS NUMBER) which can be dialed from any location via the Public Switched Telephone Network (PSTN) (2). The control system determines which subscriber a call is intended for, and by referring to a data base determines the method of call handling which has been preselected by that subscriber (e.g. switching a call to another telephone number). The switching function (4) may be located in the control system itself, or located in the PSTN but under the control of the control system. Hence, the ACCESS NUMBER may be used as the sole telephone number for a subscriber. Callers need not know the subscribers specific whereabouts nor the subscriber's various location-specific telephone numbers such as home, office, car phone, and so forth.

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REMOTE ACCESS TELEPHONE CONTROL SYSTEM

TECHNICAL FIELD

The present invention pertains to the telephone equipment art and, more particularly, to a telephone control system which allows subscribers to remotely control a plurality of call handling utilities to predeterminedly direct incoming calls.

INCORPORATION BY REFERENCE

The subject matter disclosed and claimed in U.S. Application Patent No. 4,893,335, issued January 9, 1990, entitled "Remote Access Telephone Control System", invented by the same inventors and assigned to the applicant of the instant application, is hereby incorporated by reference.

BACKGROUND OF THE INVENTION

Despite the availability of numerous telephone central exchange provided functions, such as "call-forwarding", "three-way calling", "call-waiting" and "speed-dialing", as well as the advent and availability of paging and mobile telephone systems, the completion of a call to a system subscriber can often be a complicated, time consuming and tedious task. Unless the telephone subscriber is located at the telephone which receives calls to his assigned phone number, completion of a call from a calling party, despite the aforementioned services, generally involves the calling party leaving a message and awaiting a call back by the subscriber. Even if the subscriber is accessible via mobile telephone or a paging system, human intervention is commonly required to look up and dial specific mobile telephone or paging numbers to attempt to contact the subscriber. Thus, additional delays and costs are incurred.

In addition, even if the subscriber is on a paging system, the successful direction of a message to the subscriber requires that the subscriber manually inform the system of his whereabouts.

Finally, existing telephone control systems offer very limited control to either the subscriber or the calling party with respect to the processing of calls.

SUMMARY OF THE INVENTION

In summary, the present invention is directed to a control system which is connected to a telephone exchange and wherein each user of the system is assigned a unique telephone number with the control system routing calls to the user via a user controlled number. The control system comprises an input/output device which is adapted for connection to the telephone exchange trunks or lines to input and output telephone calls. A switching control means controllably connects a call on one line to a different line. Provided memory controllably stores and recalls electronic signals. An electronic processing means is provided for accessing the memory, switching control means and input/output device to direct the flow of input and output calls. The electronic processing means includes means for: a) identifying a call to the system from a caller directed to a specific user, b) recalling from the memory a forwarding number for the user, c) implementing a call to the forwarding number, d) switching the caller's call to the forwarding number, e) identifying a call to the system from a specific user, f) changing the specific user's memory stored forwarding number responsive to a command from the specific user, g) processing calls to the system directed to a specific user in a selected one of a plurality of modes and, h) changing a users call processing mode responsive to a command from the user.

In accordance with one feature of the invention, the control system may forward incoming calls for a subscriber to a telephone number that has been preselected by the subscriber. The call may be directly forwarded, or may be preceded by a brief announcement informing the caller that the call is being forwarded.

In accordance with another feature of the invention, the control system may first screen incoming calls before causing them to be forwarded. The call may be screened by any of several methods, including a 'priority-screen' which allows the caller to determine whether or not the call should be forwarded to the subscriber, a 'VIP code-screen' which will only forward calls if the caller enters a preselected code, and a 'voice-screen' which records the caller's name and business, places the caller on hold while contacting the subscriber, plays the recorded message, and allows the subscriber to determine if he wishes to be connected to the caller. With any of these screening methods, should the call not ultimately be forwarded to the subscriber, then the call may be delivered to a 'message center', where a message may be left for later pickup by the subscriber.

In accordance with another feature of the invention, the control system may cause the subscriber's pager to be activated in conjunction with forwarding of a call. In this way, the subscriber is given notice that the call is being forwarded to the pre-selected telephone number.

In accordance with another feature of the invention, the control system may, prior to forwarding a call, provide a courtesy message to the caller such as "..when the call is answered, please ask for your party by name..", or alternately inform the caller "..when the call is answered, please ask for extension number 1234."

In accordance with another feature of the invention, the control system may send all calls to a 'message center' where all messages may be left for later pickup by the subscriber. In this way the call is handled without disturbing the subscriber.

In accordance with another feature of the invention, the control system may handle calls via a 'branch-routing' mode. In this mode, callers may choose from a directory of options. As an example, the caller may be told: "You have reached ABC Real Estate. Please touch 1 to speak to Mr. Smith, 2 to speak to Mr. Jones, or hold the line to speak to the receptionist." Depending on the programming of the system, each of these selections may cause the call to be forwarded to a specific telephone number, or to another access number. No command may cause the system to follow a predetermined default method for disposing of the call.

In accordance with another feature of the invention, the control system may handle calls via a 'meet-me' mode. In this mode, the caller is placed on hold after being told that the user is being paged to a phone. The control system then pages the user. The user then may call the access system, and enter a code to be connected to the caller. In addition, the caller may be given the opportunity to touch 9 to leave a message if he does not wish to wait. If the caller chooses to leave a message, then when the subscriber calls in, he will be informed that the caller did not wait and instead left a message. If the caller hangs up without leaving a message, then the subscriber will be so informed.

In accordance with another feature of the invention, the call-handling modes and other features of the control system may be programmed by the subscriber by accessing a command mode by dialing the access number from a touch-tone phone, entering a Personal Identification Number (PIN) code, and then following a series of menu items to select the desired feature, call-handling mode, or forwarding number. This process is further simplified by providing the subscriber with 'mode memories' which contain the most often used call handling modes, and 'number memories' which contain often used telephone numbers such as home phone, office phone, car phone, and so on.

In accordance with another feature of the invention, the subscriber may invoke a 'feature timer' which will cause a new call handling mode to take effect at the expiration of a selected time interval.

In accordance with another feature of the invention, the subscriber may invoke a 'weekly schedule' which will cause the current call-handling mode to be automatically selected from a pre-programmed list of call-handling modes, based upon the time-of-day and day-of-week.

In accordance with another feature of the invention, the subscriber may cause his calls to be forwarded to him at his current location, without the subscriber needing to enter the telephone number of the current location. Instead, the telephone number of the location is automatically received by the control system as an Automatic Number Identification (ANI) via ANI-capable facilities from the Public Switched Telephone Network (PSTN). The subscriber need only call the control system, enter his PIN code, and select a forwarding mode.

In accordance with another feature of the invention, the subscriber may program the operation of the control system by picking up a preregistered phone such as his office or home telephone and touching a 2 or 3 digit speed-dial code. In combination with the ANI capability described above this makes programming very simple. By way of example, if a subscriber is at home, he may pick up the phone and dial 10#. This causes the telephone company central office to memory dial an 800 number with an NNX that points to ANI-capable trunks connected to the control system. The control system recognizes the ANI as belonging to the home phone of one of it's subscribers. The control system then causes all calls to that subscriber to be priority-screen forwarded to the home. If the subscriber had dialed 11#, an 800 number with the same NNX but different last four digits would be dialed, which would cause the control system to select voice-screened forwarding to the home, and so forth.

In accordance with another feature of the invention, the subscriber may elect to make an 'outside call' while in the control system command mode, by touching 9 and dialing the desired telephone number.

The control system is further enhanced by the addition of a communicator feature. This feature allows a subscriber to move from place to place and have his calls follow him without the need for him to call into the control system. The communicator is a portable device carried on the subscriber's person. The device contains an RF transmitter, an RF receiver, an ultrasonic receiver, a keypad, a 'beeper', and control circuitry. The communicator receives ultrasonic messages from small wall mounted ultrasonic transmitters. These transmitters contain the phone number, and optionally the extension number, of the nearest telephone or a mode appropriate for the location such as do not disturb in a hospital operating room. The communicator also receives radio frequency messages from the control system indicating, or paging, an incoming call for the user. The communicator device can send various radio frequency messages back to the control system, including a message containing the phone number received from the ultrasonic transmitter, a message acknowledging receipt of the page, and messages in response to keypad selections by the subscriber indicating a desire to select a new mode of call handling.

A feature of the present invention provides a telephone control system which allows subscribers to access the system, even remotely, and implement any of a number of utilities for the handling of subscriber directed calls.

Another feature of the invention provides a telephone control system wherein each subscriber is assigned one telephone number and calls to the system on that number are automatically directed to the subscriber regardless of his location.

Another feature of the invention provides the above described telephone control system wherein the system interfaces with callers and subscribers via courtesy messages which minimize or eliminate the need for users to recall complicated control commands.

Another feature of the invention provides the above described control system wherein subscribers are provided with portable pager-like communicators which include RF transmitters for transmission directly to the paging system.

Another feature of the present invention provides the above described control system wherein the communicator devices automatically respond to remote locater transmitters to transmit back to the control system the phone number of a telephone proximate the user.

BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 is a block diagram illustrating the various modes of operation and interfacing equipment for the preferred embodiment of the telephone control system;

Figure 2 is a block diagram illustrating the principle components of the telephone control system;

Figure 3 is a block diagram of the Communicator;

Figure 4 is a block diagram of the Ultrasonic Transmitter;

Figure 5 is a block diagram of the Call Processing facility;

Figure 6 is a flowchart illustrating operation of the E & M Control Circuit;

Figure 7 is a diagram illustrating the Subscriber Master Record;

Figure 8 is a diagram illustrating the Mode Memory;

Figure 9 is a flowchart of the Main Task for the Call Processing facility;

Figure 10 is flowchart of the Code Processing Facility-Network Message Task;

Figure 11 is a flowchart of the Code Processing Facility-Call Termination

Task;

Figure 12 is a flowchart of the Call Processing Facility-Call Handler Task;

Figure 13 is a flowchart of the Call Processing Facility-Dynamic Mode Assignment;

Figure 14 is a flowchart of the Call Processing Facility-Direct Forwarding Function;

Figure 15 is a flowchart of the Call Processing Facility-Announced Forwarding Function;

Figure 16 is a flowchart of the Call Processing Facility-Priority/Urgent Screen Function;

Figure 17 is a flowchart of the Call Processing Facility-VIP Code Screen Function;

Figure 18 is a flowchart of the Call Processing Facility-Branch Routing Function;

Figure 19 is a flowchart of the Call Processing Facility-Caller Message Center Function;

Figure 20 is a flowchart of the Call Processing Facility-Voice-Screen Function;

Figure 21 is a flowchart of the Call Processing Facility-Meet Me Caller Function;

Figure 22 is a flowchart of the Call Processing Facility-'Send Page' Subroutine;

Figure 23 is a flowchart of the Call Processing Facility-Command Mode Function;

Figure 24 is a flowchart of the Call Processing Facility-Command Message Center Function;

Figure 25 is a flowchart of the Call Processing Facility-Command Forwarding Number Function;

Figure 26 is a flowchart of the Call Processing Facility-Command Feature Timer Function;

Figure 27 is a flowchart of the Call Processing Facility-Command Memory Function;

Figure 28 is a flowchart of the Call Processing Facility-Command Outside Call Function;

Figure 29 is a flowchart of the Call Processing Facility-Command Help Function;

Figure 30 is a flowchart of the Call Processing Facility-Command Meet Me Function;

Figure 31 is a flowchart of the Call Processing Facility-Command Branch Route Function;

Figure 32 is a flowchart of the Call Processing Facility-Command Advanced Features Function;

Figure 33 is a block diagram illustrating the principle components of the Meet Me Facility;

Figure 34 is a flowchart of the Meet Me Facility Main Task;

Figure 35 is a block diagram illustrating the principle components of the Subscriber Access Facility;

Figure 36 is a flowchart illustrating operation of the E & M Control Circuit for the Subscriber Access Facility;

Figure 37 is a flowchart of the Subscriber Access Facility Main Task;

Figure 38 is a block diagram illustrating the principle components of the Communicator Access Facility;

Figure 39 is a flowchart of the Communicator Access Facility Main Task;

Figure 40 is a flowchart of the Communicator Main Task;

Figure 41 is a block diagram illustrating the principle components of the Pager Dialing Facility;

Figure 42 is a flowchart of the Pager Dialing Facility Main Task;

Figure 43 is a block diagram illustrating the principle components of the Client Services Facility; and;

Figure 44 is a flowchart of the Client Services Facility Main Task.

OVERVIEW OF THE INVENTION

FIG. 1 illustrates in block diagram form, the manner in which the Telephone Control System may be used to enhance the accessibility of it's subscribers. As is shown, the Telephone Control System 1 connects with the PSTN 2 via facilities 3. The Telephone Control System 1 may control switch 4, causing it to connect incoming and outgoing trunks.

As is shown, alternate preferred embodiments exist with respect to switch 4.

In the first preferred embodiment, the switch 4 is actually part of the PSTN 1. In this embodiment, the facilities 3 must be capable of transmitting switch control signals from the Telephone Control System 1 to the switch 4. An example of this type of facility is a CENTREX line, which allows the transmission of switch control signals in the form of 'hookswitch flashes' and touch tones to initiate call-conferencing and call-transfer. A recently available variation of the CENTREX facility is a CENTREX DID trunk, which not only has the 'hookflash' capability, but also provides the called number in the form of Direct-Inward-Dialing digits. This is the form of facility 3 which is referred to in the detailed description of the preferred embodiment. Another variation of the CENTREX facility provides the called number via a separate data-link known as Simplified Message Desk Interface (SMDI).

U.S. Patent 4,893,335, which issued January 9, 1990, incorporated by reference herein, describes in detail a system for controlling the PSTN switch.

In an alternate preferred embodiment, the switch 4 is part of the Telephone Control System 1. In this embodiment, the facilities 3 need only include standard DID trunks for the incoming calls, and standard outgoing trunks. The access control system 1 controls switch 4 directly, causing it to connect paths between various incoming and outgoing trunks as required.

Again referring to FIG. 1, the Telephone Control System 1 also connects to the PSTN 2 via standard tip-ring phone lines 5, for purposes of communicating with Paging System 6. The Paging System may be any of the commonly known paging systems such as those comprised of transmitters such as Motorola's PACE or Quintron model QT250B and paging terminals such as Glenayre model GL3000XL or BBL System 3, which send encoded messages via radio frequency to cause a unique pager, or beeper, worn by a paging system subscriber, to sound an alert, produce a message in a display, activate a light, vibrate, or produce any of a variety of other alerting mechanisms. Typically, these paging systems will cause a pager to be alerted in response to another individual dialing a phone number which corresponds to that individual's pager. This phone number is routed via the PSTN 2 to a paging terminal via facilities 7, which in turn determines, typically via DID digits, who the call is intended for, and then sends a radio frequency message to alert that individual's pager. To cause a subscriber's pager to be activated, the Telephone Control System 1 then need only come off hook on one of the lines 5, and dial the phone number that corresponds to the subscriber's pager. Although not described in this preferred embodiment, it is anticipated that the Telephone Control System 1 could also interface to a paging system directly via a dedicated data link.

An additional facility 5 connects the Telephone Control System 1 to the PSTN 2. This facility is a trunk which provides the Automatic Number Identification (ANI) of the calling party. An example of such a trunk is the Feature Group D (FGD) trunk which is commonly used by interexchange carriers. The interexchange carriers use the ANI information to properly bill the calling party. The Telephone Control System 1 uses this ANI information in a new and different manner. As will be described in further detail herein, subscribers of the Telephone Control System 1 may program the Telephone Control System 1 by calling it through trunking facilities 5. The access control system 1 automatically acquires the ANI, or phone number of the calling party. This allows the access control system 1 to program the forwarding number for the subscriber without the subscriber needing to manually enter it. Although not described in the preferred embodiment, it is anticipated that other types of facilities which provide ANI information may also be used for this purpose. An example of another type of facility which provides ANI is a CENTREX line with an SMDI data link, which is now available from several types of central offices. The

SMDI data link is capable of passing both the called party number and the calling party number (ANI).

Still referring to FIG. 1, The Telephone Control System 1 is also connected to a Packet Radio Transmitter/Receiver 9 via data-link 10. The Packet Radio Transmitter/Receiver 9 may consist of any of the commonly known radio transceivers such as YAESU FT-470 and ICOM IC-u 4AT, equipped with a packet radio interface such as HEATHKIT HK-21. As will be described in further detail herein, the Packet Radio Transmitter/Receiver 9 is used by the Telephone Control System 1 to interface with the portable Communicator device 11, carried by an Telephone Control System subscriber. The Communicator 11 may both send and receive DATA messages via radio frequency. The Communicator 11 may also receive ultrasonic messages from fixed ultrasonic transmitter 12, shown located in room 13. Ultrasonic transmitter 12 continuously transmits the phone number, and, if appropriate, the extension, of the phone 14 located in the same room or a signal indicating an appropriate call control mode for a given situation such as do not disturb in a hospital operating room. It should be noted that, although the preferred embodiment disclosed herein describes transmitter 12 as ultrasonic, it is anticipated that an infrared transmitter may also be used. The ultrasonic transmitter has the advantage that it will pass signals through a layer of clothes, which would be important for example if the subscriber were carrying the Communicator 11 in a shirt pocket.

To aide in the discussion of the illustrative examples which follow, FIG. 1 also shows a subscriber's home 15, with a home phone 16; a subscriber's office 17, with an office phone 18; a cellular telephone system 19, which interfaces to a subscriber's car-phone 20; a factory 21, with a factory phone 22; a pay telephone 23; a subscriber 24 with pager 25; and a caller's telephone 26.

The illustrative examples which follow are intended only to clarify some of the concepts, features, and objects of the invention, and do not define the scope of the invention.

Methods of Call-Handling

Following are several illustrative examples of the various call-handling modes of the Telephone Control System 1.

Direct Forwarding

For the sake of this example, assume that a caller at phone 26 wishes to speak to a subscriber to the access control system 1, and further assume that the subscriber is at home 15, and that he has preprogrammed the system to 'direct forward' his calls to him at his home phone 16. The caller dials the access number for the subscriber, and the PSTN delivers the call to the Telephone Control System 1 via facilities 3. The facilities 3 provide the access control system 1 with the called party information (DID) digits. The Telephone Control System then refers to it's internal database to determine how to handle the call. The access control system determines that calls for this subscriber are to be handled via 'direct forwarding' mode, and that the call is to be forwarded to the subscriber's home. The access control system then dials the subscriber's home on an outgoing facility 3, and instructs the switch 4 to connect the incoming facility to the outgoing facility to complete the call. The PSTN 2 then delivers the call to home phone 16, causing it to ring, and the subscriber may pick up the phone and connect to the caller. Note that this mode of call-handling is referred to as 'direct forwarding' because the call is forwarded without any announcement or prompting from the Telephone Control System 1.

Announced Forwarding

Assume again that a caller at phone 26 wishes to speak to a subscriber to the Telephone Control System 1. Also assume that the subscriber is at home 15, and that this time he has preprogrammed the system to 'Announce-forward' his calls to him at his home phone 16. Again, the caller dials the access number for the subscriber, and the PSTN delivers the call to the Telephone Control System 1 via facilities 3. Upon receiving the DID digits for this subscriber, the Telephone Control System 1 determines that calls for this subscriber are to be handled via 'announced forwarding' mode, and that the call is to be forwarded to the subscriber's home. The access control system then plays a brief greeting to the caller: "Hello, you have reached the **ACCESSLINE** for Mr. Jones. We're Connecting your call." The Telephone Control System then dials the phone number for phone 16 on an outgoing facility 3, and instructs the switch 4 to connect the incoming facility to the outgoing facility to complete the call. The PSTN 2 then delivers the call to home phone 16, causing it to ring, and the subscriber may pick up the phone and connect to the caller.

Forwarding with Page

Assume again that a caller at phone 26 wishes to speak to a subscriber to the Telephone Control System 1. Also assume again that the subscriber is at home 15, and that this time he has preprogrammed the system to 'Forward with page' his calls to him at his home phone 16. Again, the caller dials the access number for the subscriber, and the PSTN delivers the call to the Telephone Control System 1 via facilities 3.

Upon receiving the DID digits for this subscriber, the access control system 1 determines that calls for this subscriber are to be handled via 'forward with page' mode, and that the call is to be forwarded to the subscriber's home. The access control system then plays a brief greeting to the caller: "Hello, you have reached the **ACCESSLINE** for Mr. Jones. We are sending a page to inform your party of the call. Please stay on the line." The Telephone Control System 1 then dials the phone number for the pager corresponding to this subscriber and informs the caller "We have sent a page to alert your party and will connect you momentarily." The access control system then dials the phone number for phone 16 on an outgoing facility 3, and instructs the switch 4 to connect the incoming facility to the outgoing facility to complete the call. The PSTN 2 then delivers the call to home phone 16, causing it to ring, and the subscriber may pick up the phone and connect to the caller. The subscriber, having been alerted to the incoming call by his pager, was ready to receive it.

Message Center

In some cases the subscriber may not be able to take calls and may wish that his callers simply leave a message. In these cases, the subscriber may program the access control system 1 to connect calls to the subscriber's preselected 'message center'. The Telephone Control System 1 may connect calls to either an 'internal message center' or an 'external message center'. The 'external message center' is simply a phone number that the Telephone Control System 1 will forward calls to if in this mode. This may be the phone number for an answering service, a receptionist, a voice mail system, or any other appropriate location as desired by the subscriber. If the subscriber elects to use the 'internal message center', then an example of a typical call may be as follows.

Assume that a caller at phone 26 wishes to speak to a subscriber to the Telephone Control System 1. Also assume that the subscriber does not wish to be disturbed and that he has preprogrammed the system to 'internal message center' mode. The caller dials the access number for the subscriber, and the PSTN delivers the call to the Telephone Control System 1 via facilities 3. Upon receiving the DID digits for this subscriber, the Telephone Control System 1 determines that calls for this subscriber are to be handled via 'internal message center' mode. The Telephone Control System then plays a brief greeting to the caller: "Hello, you have reached the **ACCESSLINE** for Mr. Jones. Your party is not readily available at the moment, however we will connect you to your party's message center where you may leave a detailed message.....Please leave your message at the tone." The Telephone Control System 1 then records the callers message and saves it for later playback by the subscriber. In addition, should the subscriber have so elected, the access control system 1 may dial the phone number corresponding to the subscriber's pager, to alert the subscriber to the message.

Priority-Call Screening

Assume again that a caller at phone 26 wishes to speak to a subscriber to the

Telephone Control System 1. This time assume that the subscriber is at his office 17, and that he has preprogrammed the system to send his calls to him at his office via 'priority call-screening', with a message asking the caller to ask for extension 123, which in this example is the extension number of the phone 18 on his desk. Again, the caller at phone 26 dials the access number for the subscriber, and the PSTN 2 delivers the call to the access control system 1 via facilities 3. Upon receiving the DID digits for this subscriber, the Telephone Control System 1 determines that calls for this subscriber are to be handled via 'priority call-screening' mode, and that the call is to be forwarded to the subscriber's office. The access control system 1 then plays a brief greeting to the caller: "Hello, you have reached the **ACCESSLINE** for Mr. Jones. Your party is not readily available at the moment. If this call is urgent then please touch 0 now and we will attempt to connect you to your party. Otherwise, please hold the line and we will connect you to your party's message center where you may leave a detailed message." If the caller does not touch 0, then the call is delivered to the 'message center' as described above. If the caller does touch 0, then the Telephone Control System 1 may inform the caller: "Please standby while we connect your call. When the call is answered please ask for extension number 123." The access control system then dials the preprogrammed lead phone number for the subscriber's office 17 on an outgoing facility 3, and instructs the switch 4 to connect the incoming facility to the outgoing facility to complete the call. The PSTN 2 then delivers the call to the receptionist at office 17, causing it to ring. When the receptionist answers, the caller will ask for extension 123, as he was instructed by access control system 1, and the receptionist may connect the call to the subscriber's desk phone 18.

VIP Code Screening

Assume again that a caller at phone 26 wishes to speak to a subscriber to the Telephone Control System 1. This time assume that the subscriber is in his car, and that he has preprogrammed the system to send his calls to him at his car-phone 20 via 'VIP code-screening'. In this mode, only those callers who have been told a special

VIP code will be able to reach the subscriber. All other callers will be sent to the message center. Again, the caller at phone 26 dials the access number for the subscriber, and the PSTN 2 delivers the call to the Telephone Control System 1 via facilities 3.

Upon receiving the DID digits for this subscriber, the access control system 1 determines that calls for this subscriber are to be handled via 'VIP code-screening' mode, and that the call is to be forwarded to the subscriber's car phone 20. The Telephone Control System 1 then plays a brief greeting to the caller: "Hello, you have reached the **ACCESSLINE** for Mr. Jones. Your party is not readily available at the moment. Please enter your VIP code now, or hold the line and we will connect you to your party's message center where you may leave a detailed message." If the caller does not enter the correct VIP code, then the call is delivered to the 'message center' as described above. If the caller does enter the VIP code, then the Telephone Control System 1 may inform the caller: "Please standby while we connect your call." The Telephone Control System then dials the telephone number for car-phone 20 on an outgoing facility 3, and instructs the switch 4 to connect the incoming facility to the outgoing facility to complete the call. The PSTN 2 then delivers the call to the cellular telephone system 19, which in turn delivers the call to the car-phone 20.

Voice-Screening

Assume again that a caller at phone 26 wishes to speak to a subscriber of the Telephone Control System 1. This time assume that the subscriber, Mr. Jones, is visiting his client's factory 21, and that he has preprogrammed the system to send his calls to him at this location via 'voice-screening'. Again, the caller at phone 26 dials the access number for the subscriber, and the PSTN 2 delivers the call to the access control system 1 via facilities 3. Upon receiving the DID digits for this subscriber, the Telephone Control System 1 determines that calls for this subscriber are to be handled via 'voice-screening' mode, and that the call is to be forwarded to his client's factory 21. The access control system 1 then plays a brief greeting to the caller:

"Hello, you have reached the **ACCESSLINE** for Mr. Jones. Please state your name and the purpose of your call at the tone. After the tone, please stay on the line while we attempt to locate your party and connect your call." The access control system 1 then records the caller's name and business, and then responds: "Thank you, please standby." The access control system then dials the telephone number for factory 21 on an outgoing facility 3, leaving the incoming call on hold. The PSTN 2 then delivers the outgoing call to the lead telephone number of factory 21, which is answered by the factory's receptionist. The Telephone Control System tells the receptionist "We have a call holding for Mr. Jones. Please locate the party." The receptionist pages Mr. Jones via the factory's speaker system, informing him of the call. Mr. Jones then answers the call at phone 22, and enters his Personal Identification Number (PIN) code. The access control system 1 then plays back the callers name and business. The Telephone Control System 1 then asks Mr. Jones: "Please touch 1 to connect the call, 2 to send the caller away, or 3 to send the caller to your message center." In this example, Mr. Jones wishes to speak to the caller, so he touches 1. The Telephone Control System 1 instructs the switch 4 to connect the incoming facility to the outgoing facility to complete the call.

Branch-Routing

Assume again that a caller at phone 26 wishes to speak to a subscriber of the Telephone Control System 1. This time assume that the subscriber is not a person, but a business; the ABC Factory Company 21, and that the Telephone Control System 1 has been preprogrammed to handle their calls via 'branch-routing' mode. Again, the caller at phone 26 dials the access number for the subscriber, and the PSTN 2 delivers the call to the Telephone Control System 1 via facilities 3. Upon receiving the DID digits for this subscriber, the Telephone Control System 1 determines that calls for this subscriber are to be handled via 'branch-routing' mode. The Telephone Control System 1 then refers to it's memory and plays a pre-recorded 'branch-routing' greeting to the caller: "Hello, you have reached the ABC Factory Company. Please touch 1 to speak to the manufacturing manager, 2 to speak to accounts receivable, 3 to speak to accounts payable, 4 to speak to purchasing, or hold the line to speak to the receptionist." Should the caller need help, he will hold the line for a moment, and the Telephone Control System 1 responds: "Please standby." The Telephone Control System then dials the telephone number for the factory's reception phone 22 on an outgoing facility 3, and instructs the switch 4 to connect the incoming facility to the outgoing facility to complete the call.

Meet-Me

Assume again that a caller at phone 26 wishes to speak to a subscriber to the Telephone Control System 1. This time assume that the subscriber, Mr. Jones, is away from the office today, and that he has preprogrammed the system to handle his calls via 'meet-me' mode. Again, the caller at phone 26 dials the access number for the subscriber, and the PSTN 2 delivers the call to the Telephone Control System 1 via facilities 3. Upon receiving the DID digits for this subscriber, the Telephone Control System 1 determines that calls for this subscriber are to be handled via 'meet-me' mode. The Telephone Control System 1 then responds by producing audible ringback to the caller, while dialing the phone number for the pager corresponding to this subscriber on facilities 5. The Telephone Control System 1 then plays a brief message to the caller: "Hello, you have reached the **ACCESSLINE** for Mr. Jones. We are paging your party to a phone, please standby. If you are unable to wait you may touch 9 to leave a message. Otherwise please hold the line." The Telephone Control System 1 then places the caller on hold and waits for the subscriber to call in. Meanwhile the subscriber 24 has received the page via his pager 25, and is proceeding to pay phone 23 to answer the call. The subscriber dials his own access number and the PSTN 2 delivers the call to the Telephone Control System 1 via facilities 3. The subscriber then enters his own PIN code and is informed "You have a caller holding for you on your meet-me service. Please touch 4 to be connected to the caller." It is also anticipated that if the caller had hung up or left a message in the meantime, that the subscriber would be so informed. Assuming that the caller is still holding, and that the subscriber touches 4, the access control system 1 instructs the switch 4 to connect the incoming facility to the outgoing facility to complete the call.

Methods of Programming

Although various methods of 'remotely programmable call forwarding' have been attempted in the prior art, these have proven to have limited widespread appeal due to the complicated and inflexible methods by which the users were required to program the systems. By contrast, the access control system employs a variety of different methods to allow the subscriber to easily and effectively program the operation of the Telephone Control System. Following are several illustrative examples of the various methods a subscriber may employ to program the Telephone Control System 1.

Command Mode

To use the Command Mode method of programming, the subscriber simply dials his own access number from any telephone, and enters his PIN code. The PSTN 2 delivers the call to the Telephone Control System 1 via facilities 3. The Telephone Control System 1 then informs the caller of the current feature which is selected, and then provides the subscriber with a simple menu of options by which he can choose a new feature. For example, in response to the entry of the PIN code, the subscriber may be prompted: "Hello Mr. Jones. Your calls are currently being VIP code screened to your office. Touch 1 to check messages, 2 to change your forwarding number, 3 to select a memory, 8 for advanced features, 9 to make a call, or touch 0 for help." The subscriber may then make his desired selection. Please note that although the preferred embodiment herein discusses the use of touch tone as the signalling method by which the subscriber controls the Command Mode of the access control system, the inventors anticipate that other signalling methods may also be employed, including, but not limited to, rotary dial pulse detection and voice recognition.

Another feature of the Telephone Control System Command Mode allows the subscriber to program 'mode memories' which contain the most often used call handling methods for that subscriber. For example, a subscriber may program memory 10 to be announced forwarding to his office, memory 20 to be voice screened forwarding to his home, memory 30 to be message center mode, and so forth.

Weekly Schedule

Subscribers who have some routine to their weekly activities may soon grow weary of constantly calling the Telephone Control System and selecting the same call handling methods at the same time, day after day. For this reason, the weekly schedule feature provides a very real benefit to the subscriber. As an example, let us assume that a subscriber, Mr. Jones, starts work at 8:00 AM every morning, and leaves for home at 5:00 PM in the evening. Let us further assume that he takes an hour for lunch from 12:00 to 1:00PM each day. The weekly schedule for this subscriber might be pre-programmed into the Telephone Control System's database to be:

DAY	TIME	MODE MEMORY	FEATURE
Mon-Fri	7:30am	30	Message Center
Mon-Fri	8:00am	10	Announced forwarding to office
Mon-Fri	12:00noon	30	Message center
Mon-Fri	1:00pm	10	Announced forwarding to office
Mon-Fri	5:00pm	20	Voice screen forwarding to home

As calls are received by the Telephone Control System for this subscriber, the Telephone Control System 1 refers to this database and determines the correct call handling method for the day of week and time of day, and delivers the call accordingly.

Another feature of the weekly schedule allows the subscriber to override the weekly schedule should his schedule deviate from the preprogrammed sequence. In this way the subscriber can move freely about his routine activities, and only he needs to program the access control system should his schedule change from the normal.

Feature Timer

To illustrate the use of the Feature Timer capability of the Telephone Control System, assume that a subscriber is visiting a client's office for a one hour appointment, and wishes to have his calls forwarded to him at this location during that time. He may use the Command Mode as described above to select announced forwarding mode and change the forwarding number to be the telephone number of his client's phone. When he leaves he intends to change the call handling mode back to his message center. However, if the subscriber forgets to call the Telephone Control System as he is leaving the client's office, then the client may still continue to receive calls intended for the subscriber. To overcome this problem, the subscriber may have instead used the Feature Timer function when he first called the access control system when he got to the client's office. In this example, the subscriber could have called the access control system, and used Command Mode to select announced forwarding to his client's office. However, instead of hanging up at that point, he could have activated the Feature Timer, programming it to maintain the current mode for one hour and then automatically change the call handling mode to message center mode. In this way, the subscriber would not have to remember to call the Telephone Control System as he leaves the client's office, and the client would not be bothered with the subscriber's calls after he left.

Programming a Forwarding Number Using ANI

One difficulty in prior art implementations of remotely programmable call forwarding devices, is that it takes quite a few digits for the user to call the system, enter an access code, and then enter the forwarding number. One means by which the invention described herein overcomes this difficulty is by employing special trunks which provide the called party number, also referred to as ANI. To see how this improves the ease of programming, consider the following example. Assume that the subscriber is visiting factory 21, and that this is a location that he does not visit regularly, and therefore does not have it's telephone number preprogrammed into the Telephone Control System 1. Further assume, as was discussed earlier, that the access control system 1 is connected to the PSTN with Feature Group D trunks 8 which provide ANI, and which can be reached by dialing an 800 number. To cause his calls to get to him at the factory 21, the subscriber in this example would pick up telephone 22 and dial the 800 number which corresponds to the Feature Group D trunk. The PSTN 2 would deliver the call to the Telephone Control System 1 and the Telephone Control System 1 would receive the ANI information digits containing the telephone number of telephone 22. The subscriber then need only enter his PIN code and the call handling feature memory he wishes to use, which might be memory 40, announced forwarding. The subscriber could then hang up and the Telephone Control System 1 would program the database to send all calls for that subscriber to telephone 22 via announced forwarding. As should be obvious, the sequence of digits entered by the subscriber was shorter than if he had to actually enter the phone number. It should also be pointed out that another advantage of this method of programming is that the same sequence of digits is used to program the system each time. In other words, if the subscriber went to another location and wanted his calls to be sent to him via announced forwarding, he could pick up a phone and dial the exact same sequence of digits as was described above. This makes the programming of the Telephone Control System less demanding on the subscriber since he only has to memorize one sequence to accomplish this function. It is also anticipated by the inventors that a subscriber to this service may employ a 'pocket dialer' preprogrammed with this fixed digit sequence, thereby even further simplifying the ease of programming.

Programming the Telephone Control System using Speed Calling and ANI

A well known service offered by many telephone companies is 'Speed Calling'. This service allows users to preprogram often used telephone numbers into memories which can be recalled by dialing a one, two, or three digit code. To see how subscribers can use this service to improve the ease of programming the Telephone Control System, consider the following example. Assume that an Telephone Control System subscriber, who lives at home 15, has preprogrammed the access control system with his home phone number 16. Let us further assume, as was discussed earlier, that the Telephone Control System 1 is connected to the PSTN 2 with Feature Group D trunks 8 which provide ANI, and which can be reached by dialing an 800 number, and assume further that an entire 800-NNX has been dedicated to this trunk group, in this example 800-999-XXXX. By this invention, the last four digits of the 800 number will be used to signify the mode memory which is being selected. In this example, also assume that the subscriber has preprogrammed his telco speed dialing feature so that the sequence 2# causes the telephone number 1-800-999-0010 to be dialed, and that the sequence 3# causes the telephone number 1-800-999-0011 to be dialed. In this example, when the subscriber picks up telephone 16 and dials 2#, the speed dialing feature will cause the number 1-800-999-0010 to be dialed. The PSTN 2 will deliver the call to the access control system 1 via Feature Group D trunks 8. The access control system 1 will receive the ANI digits, and referring to it's database recognize the call as originating at the home telephone of one of it's subscribers. It then will invoke the preprogrammed mode memory 10 for that subscriber, which in this example might be voice-screened forwarding to his home phone. As a further example, if the subscriber had dialed 3#, the Telephone Control System 1 would have invoked memory 11 for that subscriber.

Communicator

As described earlier, the Communicator is a portable device carried on the subscriber's person. This example demonstrates some of the ways by which the Communicator can simplify the call handling and programming operations for the Telephone Control System subscriber. Still referring to FIG. 1, assume that the subscriber is carrying a communicator 11 on his belt, and that he has just entered room 13. Also assume that he has selected the 'automatic phone number' mode of operation for the Communicator 11. When he enters the room, the Communicator 11 detects a signal from the fixed ultrasonic transmitter 12 located near the ceiling. This signal is decoded by the Communicator 11 and is determined to contain a phone number, which in this example happens to correspond to the phone instrument 14 located in the same room 13. Upon receipt of the ultrasonic signal, the Communicator 11 transmits a brief packet message via radio frequency. This message contains the subscriber's access number and the phone number just received form the ultrasonic transmitter 12. This radio frequency message is detected by packet radio transceiver 9 and passed on to the access control system 1 via data link 10. The Telephone Control System 1 then changes the forwarding number for this subscriber to be the new number.

Assume now that a caller at phone 26 wishes to speak to this subscriber. The caller dials the access number for the subscriber, and the PSTN delivers the call to the access control system 1 via facilities 3. Upon receiving the DID digits for this subscriber, the Telephone Control System 1 determines that calls for this subscriber are to be handled via 'direct forwarding' mode, and that the call is to be forwarded to the subscriber at telephone 14. The access control system 1 then sends a page message to the packet radio transceiver 9 via data-link 10. The packet radio transceiver 9 in turn transmits a radio frequency packet message to Communicator 11, causing the beeper in the Communicator 11 to alert the subscriber to the incoming call. The Communicator 11 may also then send an acknowledgment message back via radio frequency to the packet radio transceiver 9. Meanwhile, the Telephone Control System 1 has begun to dial the phone number for phone 14 on an outgoing facility 3, and instructs the switch 4 to connect the incoming facility to the outgoing facility to complete the call. The PSTN 2 then delivers the call to phone 14, causing it to ring, and the subscriber may pick up the phone and connect to the caller. Continuing the illustrative example, assume that the subscriber completes the call and leaves the room 13. Communicator 11 detects the loss of signal from ultrasonic transmitter 12, and realizes therefore that the subscriber has left the room and is no longer able to receive calls at this location. The Communicator 11 then transmits a brief packet message via radio frequency. This message contains the subscriber's access number and a special message indicating that no phone number is available and that a default mode memory is to be used for call handling. This radio frequency message is detected by packet radio transceiver 9 and passed on to the Telephone Control System 1 via data link 10. The Telephone Control System 1 then changes the method of call handling for this subscriber to the default mode, which may typically be message center mode. It should be obvious now that if the subscriber were to reenter room 13, or to enter another room with a similar ultrasonic transmitter, that a similar sequence of events would occur causing the calls for this subscriber to be routed to the appropriate room. In this way, without any specific action or effort on the part of the subscriber, his calls will follow him from location to location and be handled automatically and properly.

Another feature of the Communicator allows the subscriber to enter a 'manual phone number' mode whereby the Communicator will ignore the messages received from the ultrasonic transmitters, maintaining the last used mode or forwarding number.

Another feature of the Communicator allows the subscriber to select a new method of call-handling by touching keys on the Communicator's keypad. This will cause the Communicator 11 to transmit a 'new mode memory request' packet message via radio frequency to the packet radio transceiver, which in turn will send the message to Telephone Control System 1 via data-link 10, causing the Telephone Control System 1 to change the call handling method for that subscriber.

DETAILED DESCRIPTION

FIG. 2 is a block diagram of the telephone control system 1. The preferred embodiment of the telephone control system 1 consists of a variety of subsystems, or facilities. A Call Processing Facility (CPF) 100 is shown connected to trunks 3. A Pager Dialing Facility (PDF) 105 is shown connected to telephone lines 5. A Subscriber Access Facility (SAF) 110 is shown connected to trunks 8. A Meet-Me Facility (MMF) 115 is shown connected to lines 120. A Communicator Access

Facility (CAF) 125 is shown connected to datalink 10. A Client Services Facility (CSF) 130 is also shown. Each of these facilities is connected to high speed data network 150.

A block diagram of the Communicator 11 is shown in FIG. 3. The operation of the Communicator 11 is controlled by microprocessor 200, which in the preferred embodiment is a single chip microprocessor containing it's own Read-Only-Memory (ROM) and Random-Access-Memory (RAM). A keypad 205, and display 210 are shown connected to microprocessor 10. A transmit data output port 215 is provided by the microprocessor 200. This transmit data is passed to packet data encoder 220 which formats the data into packets before sending the packets to antenna 230 via radio frequency transmitter 225. Radio frequency receiver 235 receives data packets from antenna 230 and passes them to packet data decoder 240, which removes the packet format and passes the raw received data to the receive data input port 245 on microprocessor 200. Output port 250 may be used to apply a tone signal to beeper driver 255 which supplies sufficient current to drive beeper 260 to produce an alerting sound. A transducer 265, capable of receiving signals in the ultrasonic frequency range, passes these signals to a 40 Kilohertz filter and amplifier 270. The resulting signal is applied to detector 275 which removes the 40 Kilohertz component from the signal and passes the resulting serial data to input port 280 of microprocessor 200. Also shown is a crystal oscillator 285 which controls the timing of all operations of the microprocessor 200. Power for the Communicator 11 is supplied by battery 290. Improvements that currently exist in the ART may be made to the ultrasonic transmitter and detector to enhance its ability to communicate the ultrasonic data in the presence of multi-path echoes. These improvements include, but are not limited to: frequency shift keying; the transmission of synchronized data and the use of a clock recovery logic to extract the clock timing over a long integration period resulting in a clock move resistant to echoes; the use of error detecting and correcting codes; and the use of sampling and voting techniques to determine the correct bit value after multiple samples during a bit period, the bit period being determined by the clock recovery logic above. In addition, multiple transmitters may be used as a method of obtaining increased coverage and area transmission diversity.

A block diagram of the Ultrasonic Transmitter 12 is illustrated in FIG. 4. Oscillator 300 provides a 40 Kilohertz signal to one input of AND gate 305. The output of crystal oscillator 310 is applied to divider 311 which produces a 75 Hertz signal to the clock input of 6-bit counter 315. The 6 outputs of counter 315 are applied to the address inputs of Read-Only-Memory 320. Memory 320 is pre-programmed with data representing the phone number of the nearest telephone. This data may contain the area code plus seven digits and the extension number if appropriate. With each digit represented by 4 bits, 14 digits and 8 bits of checksum may be transmitted. Only the least significant bits in ROM 320 need be programmed, as the LSB output 325 is used to apply this data to a second input of AND gate 305. As can be seen, the serial data at 325 will continuously output the phone number at a rate of 75 bits per second. AND gate 305 combines the serial data 325 and the 40 Kilohertz signal from oscillator 300 producing a resultant signal which is applied to transducer driver 330. The output of driver 330 is then applied to transducer 335. As should be obvious, the transducer will be generating a 40 Kilohertz signal while the serial data output 325 is high, and will be generating no signal while the serial data output 325 is low. The data is therefore modulated on the 40 Kilohertz carrier at a rate of 75 baud. The 64 bits from the ROM 320 are thus transmitted in a period of 0.853 seconds. This is adequate for the transmission of a phone number and extension. Although this baud rate is relatively low, it has the advantage of reducing the effect of multipath (reflections of the ultrasonic signal arriving at the receiver at different times and phases), and thus improves the reliability of transmission as compared with higher baud rates.

A block diagram of the Call Processing Facility (CPF) 100 is shown in FIG. 5.

Referring to FIG. 5, CPF trunk interface 400 interfaces the CPF 100 with trunk 3. As was discussed earlier, the preferred embodiment of the telephone control system 1 employs a CENTREX DID trunk. In this embodiment trunk 3 is provided via a 4-wire E&M trunk provisioned with TYPE I signalling, which is well known in the art. This type of trunk provides a 2-wire balanced transmit audio connection,shown terminated by line termination 405. This type of trunk also provides a 2-wire balanced receive audio connection, shown terminated by line termination 410. The E-Lead of trunk 3 is shown connected to the current limiting and over voltage protection at reference 415. In a similar fashion, the M-Lead of trunk 3 is shown connected to the current limiting and over voltage protection at reference 420. 2-to-4 wire convertor 425 takes the separate balanced transmit and receive signals from line terminators 405 and 410 and combines them into one single-ended signal at reference 430, which is applied to call processor 435. The E-Lead signal from limiter/protector 415 is passed to current detector 440. As is shown, current detector 440 provides a path for the E-Lead signal through to the negative battery reference of -48 volts at 445. Current Detector 440 also provides an "E-Lead Detect" logic signal indicating the presence of current via the E-lead. This signal is applied to E&M lead control circuit 450. The M-Lead signal for limiter/protector 420 is passed to M-Lead relay 455. This relay is controlled by a "M-Lead Control" signal from control circuit 450. By activating or deactivating relay 455, control circuit 450 is able to take the M-Lead on-hook or off-hook, as the M-Lead will be connected either to -48 volts or to ground. The control circuit 450 outputs an "Incoming Call" signal 470 to buffer 460, and outputs a "Loop Status" signal 475 to call processor 435. The control circuit 450 has as additional inputs a "DID Received" signal 480 from latch 465, and an "On/Off-Hook Control" signal 485 from call processor 435. The output of buffer 460, and the input of latch 465 is applied to CPF internal data bus 490.

Still referring to FIG. 5, the CPF internal data bus 490 connects CPF trunk interface 400, call processor 435, precision busy/ring detector 437, Central Processing Unit (CPU) 495, Random Access Memory (RAM) 500, Disk memory 505, and data network interface 510. Call processor 435 performs the functions of voice record and playback, dual-tone-multi-frequency (DTMF) detection and generation, and call control. The functions of call processor 435 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. A voice recognition module 436 is shown connected to call processor 435. Voice recognition module 436 allows call processor 435 to detect, on a speaker-independent basis, a simple set of spoken commands from callers. The simple set consists of 16 words including the digits '0' through '9'. This capability, in combination with the DTMF detection capability of the call processor 435, allows caller to either speak their commands, or enter them from a DTMF phone. A commercially available product which performs this voice recognition function is the Model VR/10 manufactured by Dialogic Corporation.

A precision busy/ring detector 437 is shown connected to the audio signal 430 from CPF trunk interface 400. This detector may be used to perform a rapid detection of precise busy and precise ringing signals, even in the presence of voice. Unlike the busy and ring detection functions of Call Processor 435 which require a cadence match as well as a frequency match, precision busy/ring detector 437 does not require a cadence match and is therefore capable of rapidly detecting precise busy and ringing signals even if a party is conferenced in and speaking on the line. This capability is useful in providing the 'busy/no-answer option' as will be described later. The functions of precise busy/ring detector 437 are well known in the art, and may be typically implemented as follows. The incoming audio signal is applied to the inputs of several identical circuits, each with center frequencies set to detect a different component of the expected busy or ringing signal. Each of these circuits contains a low Q bandpass filter. This filter prevents out-of-band signals such as voice from interfering with the detection function. The output of the bandpass filter is fed to a zero crossing detector. The output of the zero crossing detector is then fed to a high Q bandpass filter again set at the frequency to be detected. The output of the bandpass filter is then sent to a level detector which provides a positive indication if the incoming signal is above a preset amplitude. The output of the busy/ring detector 437 may be read by CPU 495 via the CPF internal data bus 490. CPU 495 controls all functions of CPF 100. The software program which the CPU 100 uses is loaded into RAM 500, from disk memory 505. The disk 505 also is used to store a variety of other data including the subscriber database for the telephone control system 1. A data network interface 510 is used to connect the CPF 100 to the other subsystems of the telephone control system 1. Data network interface 510 passes data messages between the CPU 495 and these other subsystems. The functions of data network interface 510 are well known in the art, and many products, such as the Model COM4i from Digiboard Corporation, exist commercially which can accomplish these functions. Although only one trunk interface 400, one trunk 3, and one call processor 435 are shown in FIG. 5, it should be readily evident to one skilled in the art that additional trunk interfaces and call processors may be added to support additional trunks.

A flowchart of the operation of E&M control circuit 450 is shown in FIG. 6. At reference 600, the control circuit 450 idles waiting for an indication from current detector 440 that the E-Lead has gone off-hook. When the E-Lead does go off-hook, as shown at reference 605, an "Incoming Call" signal is sent to CPU 495 via buffer 460. The control circuit 450 then idles at 610, waiting for an off-hook signal from call processor 435, signifying that the CPU 495 is ready to accept the call. When the off-hook signal is received, the "Loop Status" 475 is set active as shown at reference 615. The M-Lead is then winked by taking M-Lead relay 455 momentarily off-hook, as shown at reference 620. The control circuit 450 then idles again at 625, waiting for the receipt of the "DID Received" signal 480 from CPU 495 via latch 465. Call processor 435 detects the incoming DID digits on it's audio line 430 and decodes the digits passing the digit sequence on to CPU 495. This DID digit sequence represents the 'called number' or ACCESS NUMBER of a subscriber to the telephone control system 1. When the CPU 495 receives the DID digits from call processor 435, the CPU 495 sets the "DID Received" signal 480 active, and the control circuit 450 takes the M-Lead off-hook as shown at reference 630. At this point a call has been established, and the control circuit 450 must now wait until either CPU 495 terminates the call, as signified by on/off-hook control signal 485 going on-hook, or by the calling party terminating the call as signified by the E-Lead detect from current detector 440 going on-hook. These functions are accomplished by control circuit 450 as illustrated at references 635 through 670. At 635 a check is made to determine if on/off-hook control signal 485 is on-hook. If it is still off-hook then a check is made at 640 to determine if the E-lead detect signal is on-hook. If it is on-hook, then the calling party has hung up, and the control circuit 450 sets the "Loop Status" 475 inactive at 645, signalling the call processor 435 that the call is terminated. The call processor 435 may in turn signal CPU 495 that the call is terminated. Control circuit 450 then waits as shown at reference 650 for the "DID Received" signal 480 to become inactive, signifying that the CPU 495 is completed with the call and is ready to accept a new call. When the "DID Received" signal 480 goes inactive, control passes to step 675 where the M-Lead is placed on hook. Control then returns to step 600 where the control circuit 450 restarts it's sequence at reference 600. Returning again to reference 635, if the On/off-hook control signal 485 were found to be on-hook, then a 1.5 second timer is started as shown at reference 655. This timer is used to distinguish between a 'hookflash' signal, which is typically less than 1.5 seconds, and an on hook command which should be at least 1.5 seconds. At 660, the control circuit 450 causes the M-Lead relay 455 to place the M-Lead on-hook. At 665 a check is made to determine if the 1.5 second timer has expired. If it has expired, then the signal was a true on-hook command, and control passes to reference 645 for the termination of the call. If at 665 it is determined that the 1.5 second timer has not expired, then a check is made at 670 to determine if the on/off-hook control 485 is still on-hook. If it is still on-hook then the timer is tested again at 665. If the on/off-hook control 485 is off-hook again, then the command was a flash, and control returns to reference 630, where the M-Lead is taken off-hook once more.

A diagram of the Subscriber Master Record, reference 700, is illustrated in FIG 7. The subscriber master record contains information regarding a given subscriber's chosen method of call handling. This information is used by the Telephone Control System 1 to determine how to process thecall. One unique subscriber master record exists for each subscriber to the Telephone Control System 1. All subscriber master records are stored in disk 505 and, upon initialization of the Telephone Control System 1, are copied to a common database in RAM memory 500 by CPU 495. This facilitates fast retrieval of this information, which is necessary for the real-time processing of calls to the access control system 1.

Still referring to FIG. 7, the subscriber master record 700 contains as its first element an access (DID) number, shown at reference 701. As described earlier, this access number 701 is the unique phone number which is used to reach a given subscriber via the Public Switched Telephone Network 2. A PIN code, which is used by the subscriber to identify himself to the Telephone Control System 1, is shown at 702. At 703, a call handling mode is shown. The call handling mode defines the basic method of call handling which has been chosen by the subscriber. Possible call handling modes include direct forwarding, announced forwarding, message center, voice screen forwarding, urgent screen forwarding, VIP code screen forwarding, and branch-routing. At 704, a standard greeting type is shown. The standard greeting type, 704, defines the courtesy greeting announcement which the subscriber has selected for the Telephone Control System 1 to use when first answering a call. At 705, an options field is shown. This options field is used to contain several miscellaneous option flags which may be used to modify the operation of the basic call handling mode. Options 705 include 'page option' which causes a page to be sent when a call is forwarded, an 'emergency/urgent' option which modifies the operation of the priority or urgent screen forwarding mode, a 'busy/no-answer' option which causes calls to be sent to the message center if a busy or no-answer condition is detected, and a voice screen hold off option which modifies the operation of the voice screen forwarding mode. A transfer message type is shown at reference 706. The transfer message type, 706, defines the courtesy announcement which the subscriber has selected for the access control system 1 to use as a call is being transferred. A transfer number is shown at 707. This is the number which the access control system 1 will use when forwarding, or transferring calls for this subscriber. At 708 an extension number is shown which may be used by the Telephone Control System 1 to announce to a caller the extension number of the phone at which the subscriber is located. The subscriber's message center number is shown stored at 709. The subscriber's pager number is shown stored at 710. The subscriber's office number is shown stored at 711. The subscriber's home number is shown stored at 712. The subscriber's mobile phone number is shown stored at 713. At 714 a VIP screen code is shown. The VIP screen code is a code which may be used by a caller to cause his call to be forwarded to the subscriber, when the subscriber has selected VIP code screened forwarding mode. The number of the current feature memory which is being used is shown stored at 715. At 716 a pager message display number is shown. This is the number which the subscriber wishes to appear in the display of his pager when the access control system 1 has taken a message from a caller. At 717 a pager forwarding display number is shown. This is the number which the subscriber wishes to appear in the display of his pager when the Telephone Control System 1 is in the process of forwarding a call to the subscriber. At 718 a communicator dynamic mode display number is shown. This is the number which the subscriber wishes to appear in the display of his communicator 11 when the Telephone Control System 1 has a call holding, and requires that the subscriber select a method of call handling to dispose of the call. The time and date that the last caller abandoned a meet-me call by hanging up without leaving a message is shown stored at 719. The time and date that the last caller abandoned a meet-me call by leaving a message is shown stored at 720. A count of the number of calls which have been transferred to an external message center by the Telephone Control System 1 is shown stored at 721. Shown generally at 722 are the branch routing numbers 0 through 9. These fields contain the phone numbers to which a call should be transferred if a caller touches one of the digits 0 to 9 when the access control system 1 is processing a call using branch-routing mode. The branch-routing default transfer number is shown stored at 723. This is the number to which the call is transferred should a caller not enter one of the digits 0 to 9. Shown stored at 724 is the feature timer duration. This value determines when the feature timer expires. Shown stored at 725 is the feature timer termination mode. This field contains the mode memory which the subscriber wishes to use upon expiration of the feature timer. The fields necessary to implement the weekly schedule function are shown generally at reference 726. In the preferred embodiment the weekly schedule may contain up to 32 events (steps). For each step, the subscriber master record 700 stores a time and date, and a mode memory number to be used at that time and date. The current step number (1 to 32) is shown stored at 727. A flag which indicates to Telephone Control System 1 that the weekly schedule is on, is shown stored at 728. A flag which informs the Telephone Control System 1 as to whether the subscriber is allowed to make multiple outside calls is shown stored at 729. A flag indicating that the subscriber has selected to use his communicator 11 in the 'dynamic mode assignment' mode is shown stored at 730. A count of the number of calls made to this ACCESS NUMBER is shown stored at 731.

A diagram of a Mode Memory, reference 800, is illustrated in FIG 8. As described previously, the mode memories are used by subscribers of the Telephone Control System 1 to store their commonly used call handling modes and options. As with the subscriber master records 700, the mode memories 800 are stored in disk 505 and, upon initialization of the Telephone Control System 1, are copied to a common database in RAM memory 500 by CPU 495. Each mode memory is unique to an individual subscriber, and is identified by storing the subscriber's ACCESS NUMBER as part of the mode memory, as shown at reference 801. In the preferred embodiment, the subscriber may have up to 90 mode memories. Each mode memory is identified by a unique mode memory number, 10 through 99. This mode memory number is shown stored as part of the mode memory 800 at reference 802. Shown generally at reference 803, are the various fields which the subscriber has selected to store in the mode memory 800. As can be seen, these are a subset of the fields which are stored in the subscriber master record 700. To invoke a mode memory, the Telephone Control System 1 need only copy the fields from the mode memory 800 to the corresponding fields in the subscriber master record 700. The access control system 1 also copies the mode memory number 802 to the current feature memory field 715 of the subscriber master record 700.

A flowchart of the Main Task for the Call Processing Facility (CPF) 100 is shown in FIG. 9. This flowchart represents the instructions, or steps, followed by CPU 495, as it controls functions of CPF 100. The starting point for the CPF -

MAIN TASK is shown at reference 900. At step 901 the CPU 495 performs initialization processes. These processes are well known in the art and include such activities as using a boot PROM to load the operational program from disk, checking for memory errors, performing hardware diagnostics, etc. The subscriber master records 700 are read from disk 505 and copied to a database in memory 500, as shown at step 902. In a similar manner, the mode memories 800 are read from disk and copied to a database in memory, as shown at step 903. At step 904, the multitasking processes are established. The multitasking allows the software to perform more than one process simultaneously. Multitasking techniques are well known in the art. As an example, UNIX is a widely used multitasking operating system. Other well known techniques allow a pseudo-multitasking function to be accomplished on a non-UNIX based operating system by constructing a 'round-robin' scheduler, by which a main process allocates 'time slices' to each of a number of sub-processes. At step 905 a process for the Network Message Task is initialized, and at step 906 control is passed to the CPF - Network Message Task. At step 907 a process for the Call Handler Task for the first trunk 3 is initialized, and at step 908 control is passed to the CPF - Call Handler Task. In a similar fashion, at step 909 a process for the Call Handler Task for the last trunk 3 is initialized, and at step 910 control is passed to the CPF - Call Handler Task. At step 911 a process for the Call Termination Task is initialized, and at step 912 control is passed to the CPF - Call Termination Task.

A flowchart of the CPF - Network Message Task is illustrated in FIG. 10. The function of this task is to receive and process messages received by Data Network Interface 510 from other facilities of the access control system 1. The CPF - Network Message Task is entered at step 1000. At step 1001 a determination is made as to whether a "request master record" message has been received. If this message has been received, then at step 1002 the subscriber master record 700 which corresponds the DID number, ANI number, or PIN code provided in the message is searched for in the database of memory 500. Upon finding this subscriber master record, at step 1003 a message containing a copy of this subscriber master record is sent back to the requesting facility via data network interface 510. Control then returns to step 1000. Should it be determined at step 1001, that no "request master record" message has been received, then at step 1004 a check is made to determine if an "update master record" message has been received. If such a message has been received, then at step 1005 a master record for a subscriber is recovered from the message and copied to the subscriber's master record 700, at which point control returns to step 1000. If an "update master record" message is not detected at step 1004, then at step 1006 a check is made to determine if a "clear dynamic mode assignment flag" message has been received. If such a message has been received, and the message identifies a specific subscriber DID number, then the dynamic mode assignment flag 730 in the subscriber master record 700 for this subscriber is cleared at step 1007, and control returns to step 1000. If a "clear dynamic mode assignment flag" message is not detected at step 1006, then at step 1008 a check is made to determine if a "set dynamic mode assignment flag" message has been received. If such a message has been received, and the message identifies a specific subscriber DID number, then the dynamic mode assignment flag 730 in the subscriber master record 700 for this subscriber is set at step 1009, and control passes to step 1000. If a "set dynamic mode assignment flag" message is not detected at step 1008, then at step 1010 a check is made to determine if "change to new mode memory" message has been received. If such a message has not been received, then control passes to step 1014. If such a message has been received, and the message identifies a specific subscriber DID number, then at step 1011 a check is made to determine if the message was sent by the Communicator Access Facility (CAF) 125. If the message was not from the CAF 125, then control passes to step 1013. If the message was from the CAF 125, then at step 1012 an indication is sent to the CPF - Dynamic Mode Assignment routine (shown in FIG. 13), that this message was received, and control passes to step 1013. At step 1013, the mode memory number and the subscriber's DID number are removed from the message, and the corresponding mode memory 800 is copied to the corresponding subscriber's subscriber master record 700. Control then returns to step 1000. If at step 1010 it is determined that a 'change to new mode memory message' is not received, then control passes to step 1014, where a determination is made as to whether a 'mode memory inquiry' message is received, and if this is the case then control passes to step 1015 where the mode memory 800 identified in the message for the subscriber identified in the message is retrieved from the database of memory 500 and a message is constructed and sent back to the requesting facility via data network interface 510. Control then returns to step 1001. If at step 1014 it is determined that a 'mode memory inquiry message is not received, then control passes to step 1016, where a determination is made as to whether a 'mode memory update' message is received, and if this is the case, then control passes to step 1017 where the new mode memory contents are retrieved from the message, and the mode memory is copied to the appropriate mode memory 800 in the database of memory 500. Control then returns to step 1001. If at step 1016, it is determined that a 'mode memory update' message is not received, then control passes to step 1018 where a determination is made as to whether a 'create new subscriber' message is received, and if this is the case then control passes to step 1019 where the DID number is retrieved from the message, a subscriber master record 700 is created for this DID number 701, and a set of mode memories 800 are created for this DID number 701. Control then returns to step 1001. If at step 1018 it is determined that a 'create master record' message is not received, then control returns to step 1001.

A flowchart of the CPF - Call Termination Task is illustrated in FIG. 11. The purpose of this task is to monitor the loop status signal 475 for each trunk interface 400, and to terminate any call in progress should the loop status become inactive. In this way the system detects if the calling party has hung up. The CPF - Call Termination Task is entered at step 1100. At step 1101 a trunk pointer is set to a value of 1. At step 1102 the loop status signal 475 for the trunk pointed to by the trunk pointer is read via call processor 435. At step 1103, a determination is made as to whether the loop status signal 475 is active. If the signal is active, then control passes to step 1106. If the signal is not active, then the caller must have hung up, and at step 1104 the trunk is placed on hook by call processor 435 via on/off hook control signal 485. Then at step 1108 a determination is made as to whether this trunk was processing the "CPF-Meet Me Caller" function, and if this is the case then control passes to step 1109 where the current time and date is stored in the "last meet-me abandon" field 719 of the subscriber master record 700. Control then passes to step 1105. Control also passes to step 1105 if at step 1108 it is determined that the trunk was not processing the "CPF-Meet Me Caller" function. At step 1105 the CPU 495 signals the multitasking process 907 controlling the call handler task for this trunk to return to it's entry point 908, thereby terminating any activity on that trunk. Control then passes to step 1106, where a check is made to determine if the trunk pointer is pointing to the last trunk. If the trunk pointer is pointing to the last trunk, then control returns to step 1101. If the trunk pointer is not pointing to the last trunk, then at step 1107 the trunk pointer is incremented and control returns to step 1102.

A flowchart of the CPF - Call Handler Task is illustrated in FIG. 12. The function of this task is to respond to an incoming call on a trunk 3, receive the DID digits identifying the subscriber's ACCESS NUMBER being dialed, determine the method of call handling as specified in the database of memory 500 by subscriber master record 700 which corresponds to that ACCESS NUMBER, and then cause the call to be processed accordingly. The CPF - Call Handler Task is entered at step 1200, a connection point labelled "CPF IDLE" is passed at reference 1201, and at step 1202 the DID received signal 480 is cleared, allowing trunk interface 400 to receive a new call. Control then remains at step 1203 until an incoming call signal 470 is received from the trunk interface 400, at which point control passes to step 1204 causing the call processor to issue an off hook signal via it's on/off hook control line 485. Then at step 1205, incoming DID digits are decoded and accepted by the DTMF detector of call processor 435. Then at step 1206,after the DID digits have been received, the DID received signal 480 is set, causing E & M control circuit 450 to force the M-Lead active, thereby seizing the trunk. Control then passes through a connection point labelled "CPF VIRTUAL TRANSFER" at reference 1207. At step 1208 the subscriber master record 700 which corresponds to the received DID number is retrieved from the database of memory 500. At step 1209 the call count 731 is incremented in the subscriber master record 700. At step 1210 a check is made to determine if the feature timer is active. This is accomplished by checking the feature timer duration 724. The feature timer is active if the feature timer duration 724 is non zero. If the feature timer is not active, then control passes to step 1213. If the feature is active, then another check is made at step 1211 to determine if the feature timer has expired since the last call. This is determined by comparing the feature timer duration 724 with the current time and date maintained by CPU 495. The feature timer has expired if the feature timer duration 724 does not extend beyond the current time and date. If the feature timer has not expired, then control passes to step 1213. If the feature timer has expired, then at step 1212 the mode memory 800 specified by the feature timer termination mode 725 is copied to the subscriber master record 800, and the feature timer duration 724 is cleared to zero. Control then passes to step 1213. A determination is made at step 1213 as to whether the weekly schedule is active by checking the status of the weekly schedule active flag 728. If the weekly schedule is not active, then control passes to step 1217. If the weekly schedule is active, then a determination is made a step 1214 as to whether the current step of the weekly schedule is correct. This is accomplished by finding the current step of weekly schedule 726 as pointed to by the weekly schedule current step 727, and comparing the time and date of the next step with the current time and date. If the current time and date fall between the current step and the next step, then the current step is correct. If, by this process, it is determined that the current step is correct, then control passes to step 1217. Otherwise, at step 1215, the weekly schedule current step 727 is incremented to point to the next step. Then at step 1216, the mode memory number corresponding to the new step of the weekly schedule 726 is found, the corresponding mode memory 800 is copied to the subscriber master record 700, and then control passes to step 1217. At step 1217 the status of the dynamic mode assignment flag 730 is checked, and if it is found to be active, control is passed at step 1218 to the CPF - Dynamic Mode Assignment. Otherwise, control passes through a connector labelled " CPF MODE" at reference 1219, and then passes to step 1220. At step 1220 a determination is made as to whether the current call handling mode 703 is set for 'direct forwarding', and if so control is passed via step 1221 to the CPF - Direct Forwarding. Otherwise a determination is made at step 1222 as to whether the current call handling mode 703 is set for 'announced forwarding' and if not control passes to step 1230. If the call handling mode 703 is set for 'announced forwarding', then at step 1223 a further check is made to determine if the transfer number 707 is set for meet-me, and if not control passes to step 1230. If the transfer number 707 is set for meet-me, then at step 1224 one audible ring is played to the caller by call processor 435. Then at step 1225, the 'pager display digits' are set to be equal to the DID number, prior to the "send page" subroutine being called at step 1226. Upon receiving a page with his own DID number in the display, the subscriber may recognize this as a meet-me call. Then a 4 second delay is initiated at step 1227, another ring is generated at step 1228, and another 4 second delay is initiated at step 1229, before passing control to step 1230. As can be seen, the effect of steps 1224 to 1229 is to simulate the typical ringing cadence expected by a caller, and in addition allow some time for the page sent at step 1225 to reach the subscriber's pager. Because the caller must wait for the subscriber to get to a phone when the subscriber is using meet-me, the sequence of steps 1224 to 1229 has the effect of reducing the delay perceived by the caller. At step 1230, an audible ring is generated towards the caller by call processor 435. Then at step 1231 a determination is made as to whether the caller has entered the PIN code 702. If the PIN code has been entered, then the caller must be the subscriber, and therefore control passes to the CPF - Command Mode via step 1233. If the PIN code has not been entered, then the control passes to step 1270 where a determination is made as to whether the 'message center access code' has been entered by the caller. In the preferred embodiment the 'message center access code' consists of the digits "*9" and is the same for all subscribers. This code may be used by callers who simply wish to leave a message and do not need to speak with the subscriber. If the 'message center access code' has been entered by the caller, then control passes to the "CPF - Caller Message Center" function as shown at step 1271.

If the 'message center access code' has not been entered, then control passes to connector label "CPF GREETING" as shown at reference 1232.

Still referring to FIG. 12, control passes through the connector labelled "CPF GREETING" at reference 1233 to step 1234, where a determination is made as to whether the call handling mode 703 is set for 'branch-routing', and if so control passes through a connector labelled "CPF PIN" at reference 1235. Otherwise, at step 1236 the standard greeting type 704 is retrieved from the subscriber master record 700. If the standard greeting type is 'stock' as determined at step 1237, then control passes to step 1238 where a stock generic greeting is played to the caller: "Hello, you have reached your party's telephone control system." Control then passes through a connector labelled "CPF PIN" at reference 1239. If the standard greeting type is not determined to be 'stock' at step 1237, then a further check is made at step 1240 to determine if the standard greeting type is 'drop-in', and if not the control is passed to step 1245. If the standard greeting type is 'drop-in' then at step 1241 the subscriber's prerecorded drop-in name is retrieved from disk 505. Then at step 1242 the call processor 435 plays the greeting: "Hello, you have reach the telephone control system for....", and then at step 1243 completes the greeting by playing back the pre-recorded name of the subscriber retrieved form disk in step 1241. Control then passes through a connector labelled "CPF PIN" at 1244. As was described earlier, if the standard greeting type was found not to be 'drop-in' at step 1240 then control passed to step 1245. At step 1245, a check is made to determine if the standard greeting type is 'personalized', and if not control passes to step 1238, described earlier. Otherwise, control passes to step 1246 where the subscriber's pre-recorded personalized greeting is retrieved from disk 505, and is then played back to the caller at step 1247 by call processor 435. Control then passes through connector labelled "CPF PIN" at reference 1248 and a check is made at step 1249 to determine if the caller has entered a PIN code. If the PIN code has been entered, then the caller must be the subscriber, and therefore control passes to the CPF - Command Mode via step 1250. If the PIN code has not been entered, then the control passes to step 1272 where a determination is made as to whether the 'message center access code' has been entered by the caller. As was described earlier, in the preferred embodiment the 'message center access code' consists of the digits "*9" and is the same for all subscribers. This code may be used by callers who simply wish to leave a message and do not need to speak with the subscriber. If the 'message center access code' has been entered by the caller, then control passes to the "CPF - Caller Message Center" function as shown at step 1273. If the 'message center access code' has not been entered, then control passes to step 1251, where the call handling mode 703 is retrieved from the subscriber master record 700. Then at step 1252 a check is made to determine if the call handling mode 703 is set for 'announced forwarding', and if so a further check is made at step 1253 to determine if the transfer number 707 is set for meet-me. If the transfer number is not set for meet-me then control passes to CPF - Announced Forwarding via step 1254. If the transfer number is set for meet-me, then control passes to CPF - Meet-Me Caller via step 1255. If at step 1252 it was determined that the call handling mode was not set for 'announced forwarding', then control passes to step 1256. At step 1256 a check is made to determine if the call handling mode 703 is set for 'urgent screen forwarding', and if so control passes to CPF - Urgent Screen via step 1257. Otherwise, at step 1258 a check is made to determine if the call handling mode 703 is set for 'VIP code screen forwarding', and if so control passes to CPF - VIP Code Screen via step 1259. Otherwise, at step 1260 a check is made to determine if the call handling mode 703 is set for 'voice screen forwarding', and if so control passes to CPF - Voice Screen via step 1261. Otherwise, at step 1262 a check is made to determine if the call handling mode 703 is set for 'branch-routing', and if so control passes to CPF - Branch-Routing via step 1263. Otherwise control passes to CPF - Message Center via step 1264.

A flowchart of the CPF - Dynamic Mode Assignment is illustrated in FIG. 13. The purpose of this function is to process calls for a subscriber who is using a Communicator 11, and who has selected the dynamic mode assignment mode of operation. The dynamic mode assignment mode of operation allows a subscriber to chose dynamically, with each incoming call, the call handling mode to be used for the call. The subscriber is alerted via his Communicator 11 that an incoming call is present, and the subscriber may then transmit a 'new mode memory' message from his Communicator 11 thereby informing the Telephone Control System 1 as to how thecall should be handled. The CPF - Dynamic Mode Assignment function is entered at step 1300, and at step 1301 the 'pager display digits' are set equal to the communicator dynamic mode display number 718. At step 1302, the 'send page' subroutine is called, causing a page to be sent to the subscriber's Communicator 11. At step 1303, a ring count is set to a value of 4. At step 1304 an audible ring is played to the caller by call processor 435, and at step 1313 a determination is made as to whether the caller has entered the PIN code 702. If the PIN code has been entered, then the caller must be the subscriber, and therefore control passes to the CPF - Command Mode via step 1314. If the PIN code has not been entered, then the control passes to step 1315 where a determination is made as to whether the 'message center access code' has been entered by the caller. As was described earlier, in the preferred embodiment the 'message center access code' consists of the digits "*9" and is the same for all subscribers. This code may be used by callers who simply wish to leave a message and do not need to speak with the subscriber. If the 'message center access code' has been entered by the caller, then control passes to the "CPF - Caller Message Center" function as shown at step 1316. If the 'message center access code' has not been entered, then at step 1305 a 4 second delay is initiated, thus creating a typical ring cadence. At step 1306 a check is made to determine if the Network Message Task (Fig. 10) has received a 'new mode memory' message from the Communicator 11 belonging to this subscriber. If such a message has been received, then the control passes to connector labelled "CPF MODE" as indicated at reference 1307. If message was not received, then the ring count is decremented at step 1308, and at step 1309 a check is made to determine if the ring count is 0. If the ring count is not '0, then control returns to step 1304 and the ring cycle is repeated. If the ring count is 0, indicating four rings cycles have been generated without the subscriber responding, then control passes to step 1310, labelled "CPF SORRY" by the connector at reference 1311, and the caller is informed, via call processor 435: "I'm sorry, your party is not available at the moment. We will connect you to your party's message center where you may leave a detailed message." Control then passes to CPF - Caller Message Center via step 1312.

A flowchart of the CPF - Direct Forwarding function is illustrated in FIG. 14. The purpose of this function is to process calls for a subscriber who has selected the 'direct forwarding' call handling mode. In this mode, calls are transferred without any announcement. Low amplitude 'confidence tones' are generated just prior to the transfer so that the subscriber may have an opportunity to enter his PIN Code. The CPF - Direct Forwarding function is entered at step 1400, and 'confidence tones' are generated at step 1401 by call processor 435. The 'confidence tones' are a prerecorded sequence of tones which are designed to sound similar to the interoffice multifrequency signalling tones that callers are familiar with. In this way the caller has no clear indication that the call is being answered and transferred, and yet at the same time the subscriber is given an indication as to when he may enter his PIN Code. At step 1402, a determination is made as to whether the subscriber has entered his PIN code. If the PIN code has been entered, then control passes to CPF - Command Mode via step 1403. Otherwise control passes to step 1416 where a determination is made as to whether the 'message center access code' has been entered by the caller. As was described earlier, in the preferred embodiment the 'message center access code' consists of the digits "*9" and is the same for all subscribers. This code may be used by callers who simply wish to leave a message and do not need to speak with the subscriber. If the 'message center access code' has been entered by the caller, then control passes to the "CPF - Caller Message Center" function as shown at step 1417. If the 'message center access code' has not been entered, then control passes to a connector labelled "CPF DIAL TRANSFER" at reference 1404 to step 1405, where a check is made to determine if the transfer number 707 contains a reference to a reserved phone number (a tag) or a reference to a mode memory. Tags may be used as follows:

TAG DIGIT RESERVED PHONE NUMBER

1	message center number 709
2	pager number 710
3	office number 711
4	home number 712
5	mobile phone number 713
6	meet-me tag
7	'externally entered' number

As will be described in more detail later in this discussion, if a mode memory 800 has a transfer number 707 that is an 'externally entered' number tag, then when that mode memory is invoked, the transfer number is not changed from the previous value. Also, if a mode memory which contains a transfer number 707 that is an 'externally entered' number tag can be invoked remotely by the subscriber via a Feature Group D trunk, causing the ANI number received by the trunk to be used as the transfer number 707. The reference to the mode memory may be in the form of the two digit mode memory number 10 - 99. Therefore, at step 1405, if the transfer number 707 contains the digits 1 through 6, or the digits 10 through 99, then control will pass to step 1406. Otherwise, control will pass to step 1409. At step 1406 a determination is made as to whether the transfer number 707 contains the meet-me tag (ie. digit 6). If the meet-me tag is found, then control passes to the CPF - Meet-Me Caller Function via step 1407. If the meet-me tag is not found at step 1406, then control passes to step 1408, where the tag or mode memory is expanded to a real phone number which can be dialed. If the transfer number 707 contains a tag, then the corresponding reserved phone number per the table above is used as the expanded number to be dialed. If the transfer number 707 contained a mode memory number, then the transfer number 707 from the corresponding mode memory 800 is used as the expanded number to be dialed. Control then passes to step 1409, where a determination is made as to whether the transfer number to be dialed can be found as the ACCESS NUMBER 801 in any of the subscriber master records 700. If so, then it is not necessary to do a physical transfer, and the call can be continued on the same trunk by passing control through the connector labelled "CPF VIRTUAL TRANSFER" at reference 1410. Otherwise, at step 1411 a flash is generated by call processor 435 by producing a 700 millisecond on hook signal on the on/off hook control line 485. This flash places the calling party on hold and causes a second dial tone to be returned on trunk 3 by the serving central office of the PSTN 2. At step 1412 a brief pause is introduced to allow time for the dial tone to appear on the trunk, and then at step 1413 the transfer number is dialed via the DTMF generator of call processor 435. Then at step 1418 the 'busy/no-answer' option flag of options 705 of subscriber master record 700 is checked. The function of this option is to handle calls which are being sent to a subscriber even if the subscriber's line is busy or does not answer. If this option is active then at step 1419 a flash is generated by call processor 435 causing the calling party to be taken off hold and connected to the call being placed to the transfer number. The calling party will thus be able to hear the progress of the call and will therefore hear the subscriber answer if the subscriber does indeed answer. At step 1420 a determination is made as to whether the call was local or long distance. If the transfer number was longer than 7 digits, or if the 7 digit number contained a prefix which is long distance in this area, then the call was long distance and a 40 second timer is started at step 1422. If the transfer number was less than or equal to 7 digits, then the call was local and a 25 second timer is started at step 1421. Then at step 1423 a determination is made as to whether precision busy/ring detector 437 is detecting busy signal, and if not control passes to step 1424. If a busy signal is detected at step 1423, indicating that the subscriber's line is busy, then control passes to step 1429 where a flash is generated by call processor 435 causing the call attempt to be dropped but leaving the calling party connected to the telephone control system 1. Control then passes to a connector labelled "CPF - Sorry" as shown at step 1430, which causes the caller to be sent to the subscriber's message center function. If at step 1423 a busy signal was not detected, then control passes to step 1424 where a determination is made as to whether the timer has expired. If the timer has expired, indicating that neither busy or ringing where detected, then control passes to 1414. If at step 1424 it is determined that the timer has not expired, then control passes to step 1425 where a determination is made as to whether precision busy/ring detector 437 is detecting a first ringing signal, and if not control returns to step 1423. If the first ringing signal is detected at step 1426, then control passes to step 1426, where a determination is made as to whether this is the fourth ring signal, and if so, indicating that the subscriber is not answering the call, then control passes to step 1429 causing the caller to be ultimately routed to the subscriber's message center function as was described earlier. If at step 1426 it is determined that this is not the fourth ring, then control passes to step 1427 where control idles until an end-of-ring is detected by precision busy/ring detector 437. Control then passes to step 1428 where a 6 second 'inter-ring timer' is started. Control then passes to step 1431 where a determination is made as to whether precision busy/ring detector 437 is detecting ringing signal and if so control returns to step 1426. If however at step 1431 it is determined that ringing signal is not being detected, then control passes to step 1432 where the 'inter-ring timer' is checked. If the 'inter-ring timer' has not expired then control returns to step 1431. If the 'inter-ring timer' has expired, indicating the subscriber has answered the call, then control passes to step 1414. At step 1414 an on hook signal is generated on the on/off hook control line 485, causing the call to be transferred to the dialed number, and freeing up the trunk 3 to handle another incoming call. Control then passes to the connector labelled "CPF IDLE" at reference 1415.

A flowchart of the CPF - Announced Forwarding function is illustrated in FIG. 15. The purpose of this function is to process calls for a subscriber who has selected the 'announced forwarding' call handling mode. In this mode, callers are greeted with a brief courtesy announcement prior to being transferred. In addition, if a 'page option' has been selected, then a page is sent to the subscriber's pager prior to transferring the call. The CPF - Announced Forwarding function is entered at step 1500 and at step 1501, a determination is made as to whether the page flag of options 705 is set, and if it is not set, then control passes to the connector labelled "CPF AF2" at reference 1502. If the page flag is set, the control passes to step 1503 where the display digits are set equal to the pager forwarding display number 717. At step 1504 the 'send page' subroutine is called causing a page to be sent to the subscriber's pager. Then at step 1505, the call processor 435 plays to the caller the message: "We are sending a page to inform your party of your call. Please stay on the line." At step 1506 a delay is initiated to allow the pager sufficient time to receive the page. Then at step 1507, another message is played to the subscriber: "We have sent a page to your party and we will connect your call momentarily. Please stay on the line." At step 1508 an additional delay is initiated to allow the subscriber the opportunity to get to a phone. Control then passes to the connector labelled "CPF AF2" at reference 1509. The connector labelled "CPF AF2" at reference 1510 passes control to step 1511, where the transfer message type 706 is retrieved from subscriber master record 700. Then at step 1512, a check is made as to whether the transfer message type is '0'. If the transfer message type is '0', indicating no transfer message is to be played, then control passes to the connector labelled "CPF DIAL TRANSFER" at reference 1520. If the transfer message type is not '0', then control passes to step 1513 where a check is made to determine if the transfer message type is '1'. If the transfer message type is '1', then at step 1514 the call processor 435 plays to the caller the message: "We're connecting your call", and then control passes to the connector labelled "CPF DIAL TRANSFER" at reference 1520. If the transfer message type is not '1', then control passes to step 1515 where a check is made to determine if the transfer message type is '2'. If the transfer message type is '2', then at step 1516 the call processor 435 plays to the caller the message: "We're connecting your call. When thecall is answered, please ask for your party by name", and then control passes to the connector labelled "CPF DIAL TRANSFER" at reference 1520. If the transfer message type is not '2', then control passes to step 1517 where a check is made to determine if the transfer message type is '3'. If the transfer message type is '3', then at step 1518 the call processor 435 plays to the caller the message: "We're connecting your call. When the call is answered, please ask for extension number...." Then at step 1519, the extension number 708 is retrieved from the subscriber master record 700 and is voiced to the caller by call processor 435. Control then passes to the connector labelled "CPF DIAL TRANSFER" at reference 1520.

A flowchart of the CPF - Urgent Screen function is illustrated in FIG. 16. The purpose of this function is to process calls for a subscriber who has selected the "priority screen" or 'urgent screen' call handling mode. The CPF - Priority/Urgent Screen function is entered at step 1600, and control passes to step 1601, where call processor 435 plays to the caller the message: "Your party is not readily available at the moment. If this call is..." Control then passes to step 1602 where the urgent/emergency flag of the options 705 is checked. If the flag is set for 'urgent', then the call processor 435 plays to the caller "...urgent...", and if the flag is set for emergency then call processor 435 plays to the caller "...an emergency..." Control then passes to step 1603 where the call processor completes the sentence by playing the message "...then touch 0 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." Then at step 1604, a 5 second timer is started. At step 1605 a determination is made as to whether the caller has touched 0. If the caller has touched 0, then control passes to the CPF - Announced Forwarding function via step 1606. If the caller has not touched 0, then at step 1607 a determination is made as to whether the 5 second timer has expired. If the 5 second timer has not expired then control returns to step 1605. If the 5 second timer has expired, then at step 1608 the caller is informed: "Please standby". Control then passes to the CPF - Caller Message Center function via step 1609.

A flowchart of the CPF - VIP Code Screen function is illustrated in FIG. 17. The purpose of this function is to process calls for a subscriber who has selected the 'VIP code screen' call handling mode. The CPF - VIP Code Screen function is entered at step 1700 and control is passed to step 1701 where the call processor 435 plays to the caller the message: "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. "Control then passes to step 1702 where a 5 second timer is started. At step 1703 a determination is made as to whether the caller has entered the VIP screen code 714 as stored in the subscriber master record 700. If the caller has entered the correct VIP screen code, then control passes to the CPF - Announced Forwarding function via step 1704. If the caller has not entered the VIP screen code 714, then at step 1705 a determination is made as to whether the 5 second timer has expired. If the 5 second timer has not expired then control returns to step 1703. If the 5 second timer has expired, then at step 1706 the caller is informed: "Please standby". Control then passes to the CPF - Caller Message Center function via step 1707.

A flowchart of the CPF - Branch Routing function is illustrated in FIG. 18. The purpose of this function is to process calls for a subscriber who has selected the 'branch-routing' call handling mode. The CPF - Branch Routing function is entered at step 1800, and control passes to step 1801 where the prerecorded branch-routing greeting is retrieved from disk 505. Then at step 1802, the playback to the caller of the branch-routing greeting is begun by call processor 435. At step 1803 a determination is made as to whether the caller has entered a digit. If the caller has not entered a digit then at step 1804 a determination is made as to whether the call processor 435 has completed the playback of the branch-routing greeting, and if an additional 5 second shave expired. If this is the case then control passes to step 1805. If this is not the case, then control returns to step 1803. If at step 1803 it is determined that the caller has entered a digit, then control passes to step 1806 where a determination is made as to whether there exists a branch-routing number 722 in the subscriber master record 700 which corresponds to the digit entered by the caller. For example, if the subscriber entered digit 4, then a determination is made as to whether the subscriber master record holds a phone number entry in the branch routing number4 position at 722. If an entry is found in such a manner, then control passes to step 1807. Otherwise, control passes to step 1805 where a the branch routing default number 723 is retrieved for the subscriber master record 700, and is set up to be used as the transfer number for this call. Control then passes to the connector labelled "CPF DIAL TRANSFER" at reference 1808. Should the caller have entered a digit which corresponded to a branch routing number 722, then at step

1807 the corresponding branch routing number 722 is retrieved form the subscriber master record 700, and is setup to be used as the transfer number for this call. Control then passes to the connector labelled "CPF DIAL TRANSFER" at reference 1808.

A flowchart of the CPF - Caller Message Center function is illustrated in FIG. 19. The purpose of this function is to process calls for a subscriber who has selected the 'message center' call handling mode. The CPF - Caller Message Center function is entered at step 1900 and control passes to step 1901 where a determination is made as to whether an external message center has been selected. This determination is made by examining the message center number 709 in the subscriber master record 700. If the message center number 709 contains a phone number, then external has been selected. If the message center number 709 does not contain a phone number, then internal message center has been selected. The subscriber may choose an external voice mail system, an answering service, his secretary, or any other appropriate phone number for the external message center number 709. If, at step 1901, it is determined that the subscriber has selected the internal message center, then control passes to step 1902, where the caller is prompted: "Please leave your message at the tone ...BEEP." Then at step 1903 the callers message is recorded by call processor 435 and stored on disk 505. At step 1904 a determination is made as to whether the caller has completed leaving the message. This is accomplished by call processor 435 determining if there has been at least 3 seconds of silence on the line since the last sound. If the caller has not completed leaving a message then the recording continues at step 1903. If the caller has completed leaving a message, then control passes to step 1905, where the caller is prompted: "Thank you for calling. Good-bye." Then at step 1906 the trunk is placed on hook by call processor 435 via on/off hook control signal 485. Control then passes to step 1907, where a decision is made as to whether a message was actually left. A message is determined to be left if at least 3 seconds of non-silence has been recorded, as determined by call processor 435. If a message was not left, then control passes to connector labelled "CPF IDLE" at reference 1908. If a message was left by the caller, then control passes to step 1909, where the 'display digits' are set equal to the pager message center display number. The 'send page' subroutine is then called at step 1910, and control passes to a connector labelled "CPF IDLE" at reference 1911. Returning now to step 1901, if a determination is made, in the manner described above, that an external message center is selected, then control passes to step 1912 where the party is informed by call processor 435: "Please standby," then control passes to step 193 where a flash is generated by call processor 435 by producing a 700 millisecond on hook signal on the on/off hook control line 485. This flash places the calling party on hold and causes a second dial tone to be returned on trunk 3 by the serving central office of the PSTN 2. At step 1914 a brief pause is introduced to allow time for the dial tone to appear on the trunk, and then at step 1915 the message center number 709 is dialed via the DTMF generator of call processor 435. The message center number may contain special dialing characters, including characters for pausing, waiting for tones, and waiting for answer. Thus a sequence of dialing characters may be constructed to allow the Telephone Control System 1 to transfer calls to a voice mail system requiring the entry of a subscriber ID. For example, an external message center dialing sequence for a typical voice mail system may be: 7 digit phone number of voice mail system + Wait for answer + 4 digit voice mail subscriber ID. Continuing now, at step 1916 an on hook signal is generated on the on/off hook control line 485, causing the call to be transferred to the dialed number, and freeing up the trunk 3 to handle another incoming call. Control then passes to step 1917 where the 'external message center count' 721 in the subscriber master record 700 is incremented. Control then passes to steps 1909 and 1910 where a page is generated as described above, before returning control to the connector labelled "CPF IDLE" at reference 1911.

A flowchart of the CPF - Voice Screen function is illustrated in FIG. 20. The purpose of this function is to process calls for a subscriber who has selected the 'voice screen' call handling mode. The CPF - Voice Screen function is entered at step 2000 and control passes to step 2001 where the caller is prompted by call processor 435: "Please state your name and business at the tone. After the tone please stay on the line while we attempt to locate your party and connect your call....BEEP." Control then passes to step 2090 where a determination is made as to whether the caller has entered the 'VIP screen code' 714, and if so control passes to the "CPF - Announced Forwarding" function as shown at step 2091. In this way, a caller who knows the 'VIP screen code' is able to be forwarded directly to the subscriber without being voice-screened. If, however, at step 2090 it is determined that the caller has not entered the 'VIP screen code' 714 then control passes to step 2002 where the caller's message is recorded by call processor 435 and stored temporarily on disk 505. At step 2003, the call processor 435 determines that the caller has completed stating his name and business, by detecting the sound of the voice followed by approximately 3 seconds of silence, at which point the call processor 435 prompts the caller: "Thank you, please standby." A flash is generated at step 2004, causing the caller to be placed on hold by the switch 4, and at step 2005 a pause is initiated to allow time for the switch 4 to provide a dial tone, at which point the transfer number 707 is dialed by the DTMF generator of call processor 435. At step 2006, an 'answer timer' is started. At step 2007 a determination is made by call processor 435 as to whether the call has been answered. If the call has not been answered, then at step 2008 a decision is made as to whether a time-out or non-answer signal such as a busy, reorder, or operator intercept has been detected by call processor 435. If so, then control proceeds through a connector labelled "CPF VSCRN FLASH" at reference 2009, to step 2010. Otherwise control returns to step 2207. At step 2010, a flash is generated, causing switch 4 to temporarily conference the caller through to the non-answer signal, and at step 2011 a 2 second pause is invoked. Then at step 2012 another flash is generated causing switch 4 to drop the conference and restore a simple 2-way connection between the caller and the trunk 3. Control then passes to a connector labelled "CPF SORRY" at reference 2013, resulting in the caller being connected to the subscriber's message center as described earlier in FIG. 13. Returning the discussion now to step 2007, if a determination is made that the call is answered, then control passes to step 2014 where the 'voice screen PIN code hold-off flag' of options 705 of the master record 700 is checked.

If this flag has been set, it means that the subscriber wishes to require that a PIN code be entered by the answering party before the called party's message is played. This is very useful if the subscriber is having his calls voice-screen forwarded to his office, for example, where the receptionist may answer the call. In this case the receptionist would connect the call to the subscriber and the subscriber would enter his PIN code to hear the calling party's message before determining whether he wishes to be connected to the calling party. If this flag is set, then control passes to step 2015, where call processor 435 prompts the answering party: "We are trying to reach...". Then at step 2016, the subscriber's prerecorded 'drop-in' name is retrieved from disk 505 and played back to the answering party. At step 2017, the answering party is informed: "Please locate the party or enter your PIN code." Then at step 2018, a 5 second delay is introduced, and at step 2019, a determination is made as to whether the answering party has entered the PIN code 702. If the PIN code is entered, then control passes to step 2021. Otherwise, control passes to step 2020, where a determination is made as to whether the sequence of steps 2015 through 2020 has been repeated ten times. If not, then control returns to step 2015, and the sequence is repeated again. However, if this is the tenth repeat, then control passes to the connector labelled "CPF VSCRN FLASH" at reference 2009, and the calling party is sent to the subscriber's message center as described earlier. If, at step 2014, it is determined that the 'voice screen PIN code hold-off flag' is not set, or if it is set and the PIN code has been entered as determined at step 2019, then control passes to step 2021, where the answering party is informed by call processor 435: "We have a call holding for...", and then to step 2022 where the subscriber's 'drop-in' name is retrieved from disk and played. Then at step 2023, which is identified by the connector labelled "CPF VSCRN LISTEN" at reference 2024, the caller's message which was originally recorded at step 2002 is retrieved from disk 505 and played by call processor 435 to the subscriber. Then at step 2025, which is identified by the connector labelled "CPF VSCRN MENU" at reference 2026, the subscriber is prompted: "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." Control then passes through a connector labelled "CPF VSCRN LOOP" at reference 2027. At reference 2028, the connector labelled "CPF VSCRN LOOP" passes control to step 2029 where a 10 second timer is started. Then at step 2030, a determination is made as to whether a digit has been entered by the subscriber, and if so control passes to step 2033. Otherwise, control passes to step 2031 where the 10 second timer is checked, and if it has not expired control returns to step 2030. If the timer has expired, then control is passes to the connector labelled "CPF VSCRN FLASH" at reference 2032, and the caller is connected to the message center as described earlier. If a digit has been entered by the subscriber, then at step 2033, the digit is checked and if it is not '1', control is passes to step 2043. If the digit is '1', then control passes to step 2034, where a flash is generated causing the calling party and the subscriber to be conferenced by switch 4. Then at step 2035 a determination is made as to whether dial tone is present on the line. If so this would indicate that the conference failed, most likely because the calling party had hung up. If this is the case, then control passes to step 2039. Otherwise, if dial tone is not detected, then at step 2036, both the calling party and the subscriber hear call processor 435 play the prompt: "Go ahead please.", and at step 2037, the trunk 3 is placed on hook causing the switch 4 to transfer the call allowing the calling party and the subscriber to continue their conversation, while at the same time freeing up trunk 3 to handle another incoming call by passing control back to the CPF - Call Handler Task via the connector labelled "CPF IDLE" at reference 2038. If the caller had hung up as determined by the detection of dial tone at step 2035, then at step 2039 another flash is generated to cause switch 4 to take the subscriber off of hold. Then at step 2040, the subscriber is informed: "I'm sorry, your party has hung up", and at step 2041 trunk 3 is placed on hook and control is returned to the CPF - Call Handler Task via the connector labelled "CPF IDLE" at reference 2042. If, as described earlier, at step 2033 it is determined that the digit is not '1', then control is passes to step 2043 where it is determined whether the digit is a '2', and if so control is passes to the connector labelled "CPF VSCRN FLASH" at reference 2044

causing the calling party to be connected to the message center as was described earlier. If the digit is not '2', then control passes to step 2045, where it is determined if the digit is a '3', and if not control passes to step 2052. If the digit is a '3', then control passes to step 2046, where a flash is generated causing switch 4 to temporarily conference the calling party and the subscriber. Then. after a 2 second pause at step 2047, another flash is generated at step 2048 causing switch 4 to terminate the conference by dropping the subscriber, leaving just the calling party connected to trunk 3. Then at step 2049, the calling party is informed by call processor 435: "I'm sorry, your party is unable to take your call at this time. Thank you for calling. Good-bye." Then at step 2050, the trunk 3 is placed on hook thereby disconnecting the calling party and control is returned to the CPF - Call Handler Task via the connector labelled "CPF IDLE" at reference 2051. If, as described earlier, at step 2045 it is determined that the digit is not '3', then control is passes to step 2052 where it is determined whether the digit is a '4', and if so control is passes to the connector labelled "CPF VSCRN LISTEN" at reference 2053, allowing the subscriber to again hear the calling party's message as described earlier. If at step 2052 it is determined that the digit is not a '4', then control passes to step 2054 where the digit is checked to determine if it is a '5', and if it is not control passes to a connector labelled "CPF VSCRN DIGIT" at reference 2060. If the digit is a '5, then control passes to step 2055 where a one minute timer in started. Then at step 2056 a determination is made whether any further digits have been entered by the subscriber, and if not control passes to step 2058 where the one minute timer is checked. If the timer is found to have expired, then control passes to a connector labelled "CPF VSCRN MENU" at reference 2059, causing the menu of step 2025 to be replayed to the subscriber. If the timer is found to have not expired, then control returns to step 2056 where a determination is again made as to whether any digits have been entered by the subscriber, and if so control passes to a connector labelled "CPF VSCRN LOOP" at reference 2057, thereby allowing the digit to be processed. The connector labelled "CPF VSCRN DIGIT6" at reference 2061 causes control to be passed to step 2062 where a determination is made as to whether the digit pressed by the subscriber is a 6, and if not control passes to step 2075. If the digit is a '6', then control passes to step 2063 where the subscriber is prompted by call processor 435: "Please enter the telephone number you wish to have this call transferred to." Then at step 2064, the control idles, waiting for a telephone number to be entered by the subscriber. If a phone number is entered, then control passes to step 2065, where the subscriber is prompted: "Number accepted. Please hang up now." Then at step 2066 a flash is generated causing switch 4 to temporarily conference the calling party and the subscriber. Then, after a 2 second pause at step 2067, another flash is generated at step 2068 causing switch 4 to terminate the conference by dropping the subscriber, leaving just the calling party connected to trunk 3. Then at step 2069 the calling party is prompted by call processor 435: " Please standby." Then at step 2070 a flash is generated causing switch 4 to place the calling party on hold and providing a dial tone to the trunk 3. Then, after a pause for dial tone at step 2071, the phone number detected in step 2064 is dialed at step 2072, and at step 2073 trunk 3 is placed on hook causing switch 4 to transfer the calling party to the phone number dialed, and control is returned to the CPF - Call Handler Task via the connector labelled "CPF IDLE" at reference 2074. If, as described earlier, at step 2062 it is determined that the digit entered by the subscriber is not a '6', then control passes to step 2075. At step 2075, a determination is made as to whether the digit is a '7', and if not then control passes to a connector labelled "CPF VSCRN LOOP" at reference 2076 thereby allowing the subscriber to enter another digit. If the digit entered is a '7', as determined at step 2075, then control passes to step 2077 where a flash is generated causing switch 4 to temporarily conference the calling party and the subscriber. Then, after a 2 second pause at step 2078, another flash is generated at step 2079 causing switch 4 to terminate the conference by dropping the subscriber, leaving just the calling party connected to trunk 3. Then at step 2080 the calling party is prompted by call processor 435: "Your party is not interested in your call. Please remove this party from your list and do not call again. Good-bye." Then at step 2081 trunk 3 is placed on hook causing switch 4 to disconnect the calling party, and control is returned to the CPF - Call Handler Task via the connector labelled

"CPF IDLE" at reference 2082.

A flowchart of the CPF - Meet Me Caller function is illustrated in FIG. 21. The purpose of this function is to process calls for a subscriber who has chosen to have his calls handled by the meet-me function. In the preferred embodiment these calls are handled by conferencing the caller on a trunk 3 of the CPF 100 with a line 120 on the Meet-Me Facility (MMF) 115. Then when the subscriber calls in to be connected, he is also conference from a trunk 3 of the CPF 100 with a line 120 on the MMF 115. The CPF may then hang up on both the caller's trunk 3 and then subscriber's trunk 3, thereby transferring the caller and subscriber to the MMF lines 120. The lines 120 are provisioned with the CENTREX features of 'call transfer' and 'barge-in', so that the caller and subscriber may then be connected as follows: The line 120 which is connected to the caller dials the CENTREX barge-in command (*77) followed by the CENTREX 'intercom code' for the line 120 which is connected to the subscriber. The caller and the subscriber are thus connected, and the line 120 which connects to the subscriber may then go on hook, transferring the subscriber to the caller's line 120. The conversation may then take place and only one line of line 120 is used. To fully understand the explanation of the CPF - Meet Me Caller function which follows, it is necessary to also review the explanations which are associated with FIG. 30 (CPF - Command Meet Me), FIG. 33 (MMF block diagram), and FIG. 34 (Meet Me Facility Main Task). Referring now to FIG. 21, the CPF - Meet Me Caller function is entered at step 2100 and control passes to step 2101 where the calling party is informed by call processor 435: "We are paging your party to a phone. Please stay on the line." Then at step 2102 a flash is generated causing switch 4 to place the calling party on hold and generate a dial tone on trunk 3. Then at step 2103 the call processor 435 dials the phone number which is associated with one of the lines 120 which are connected to the Meet-Me Facility (MMF) 115, and at step 2104 a 15 second timer is initiated. Then at step 2105, a determination is made as to whether DTMF '*' tone has been detected by call processor 435, indicating that the MMF 115 has answered. If the '*' tone is not detected, then control passes to step 2106 where the 15 second timer is checked, and if found to have not expired then control returns to step 2105. If the 15 second timer is found to have expired, then control passes to step 2107, where a flash is generated causing switch 4 to temporarily conference the calling party to the number dialed above. Then after a 2 second pause at step 2108, another flash is generated at step 2019, causing switch 4 to drop the dialed number from the conference, leaving just the calling party connected to trunk 3. Then at step 2110, a check is made to determine if dial tone is present on trunk 3. If dial tone is found to be present, indicating that the sequence of steps 2107 through 2109 had failed to restore the calling party possibly because the conference thought to be created at step 2107 was not allowed by switch 4, then another flash is generated at step 2111, which is identified by a connector labelled "CPF MM FLASH" at reference 2134. This flash causes switch 4 to reconnect the calling party to trunk 3, and then control passes via a connector labelled "CPF MM MSSG" at reference 2112 to step 2113. If dial tone is not detected at step 2110, then control passes directly to step 2113. At step 2113, the current time and date is stored in the 'last meet-me message left' field 720 of the subscriber master record 700, and then control passes to the connector labelled "CPF SORRY" at reference 2114, causing the caller to be connected to the message center as was described earlier. If at step 2105, the '*' tone is detected, indicating that the MMF 115 has answered, then control passes to step 2115 where call processor 435 dials the digit '00' signifying that this is a caller, not a subscriber. Then at step 2116, the call processor 435 dials the subscriber's DID number 701, to identify to the MMF who the calling party is waiting for. Then at step 2117, a flash is generated, causing switch 4 to conference the calling party through to this line 120 of the MMF 115, and at step 2118 a 2 second timer is initiated. Then at step 2119, a determination is made as to whether another '*' tone is detected by call processor 435, indicating the conference was successful. If the '*' tone is not detected, this implies that the conference was not successful, most likely because the calling party has hung up. In this case the 2 second timer is checked at step 2120, and if it is found not to have expired control returns to step 2119. If the 2 second timer has expired, then control passes via a connector labelled "CPF MM ABANDON" at reference 2121 to step 2122 where the current time and date are stored in the 'last meet-me abandon' field 719 of the subscriber master record. Then control passes to step 2123 where the trunk 3 is placed on hook and control is returned to the CPF - Call Handler Task via the connector labelled "CPF IDLE" at reference 2124. If at step 2119 the '*' tone is detected, then control passes to step 2125 where a repeat count is set to a value of 3. Then at step 2126, the calling party is informed by call processor 435: "We have sent a page to alert your party of the call. Your party is currently proceeding to a phone and we will connect you momentarily. If you are unable to wait, you may touch 9 at any time to leave a detailed message which we will relay to your party as soon as they pick up the line." Then at step 2127, the call processor 435 plays a 'music on-hold' selection of duration 40 seconds. Then at step 2128 a determination is made as to whether the subscriber has called in, by checking the CPF - Call Handler Tasks controlling the other trunks 3 connected to the CPF 100. If the subscriber is found, and if he has entered his PIN code 702 and touched 4 indicating he wishes to be connected to the caller, then step 2128 will return a positive indication causing control to pass to the connector labelled "CPF MM SUBCALL" at reference 2129. If a negative indication is returned at step 2128, then control passes to step 2130 where a determination is made as to whether the caller has entered '9'. If the caller does enter '9', then control passes to step 2111, causing the caller to be connected to the message center as describer earlier. If the caller has not entered '9', then control passes to step 2131, where a determination is made as to whether the 40 second music-on-hold selection is complete. If it is not complete, then control returns to step 2128. If the selection is complete, then control passes to step 2132 where the repeat count is decremented. Then at step 2133 a determination is made as to whether the value of the repeat count is now zero. If the value is zero, then control passes to step 2111, causing the caller to be connected to the message center as describer earlier. If the value of the repeat count is not yet zero, then control returns to step 2127, where the sequence of steps 2127 through 2133 is repeated once more. The connector labelled "CPF MM SUBCALL" at reference 2135 causes control to be passed to step 2136 where the DTMF fourth column tone digit 'd' is dialed by call processor 435 to inform the MMF 115 that the subscriber has arrived. Then at step 2137, the call processor 435 prompts the calling party: "Your party has picked up the line. One moment and we will connect you." Then at step 2138, a 20 second timer is initiated, and at step 2139 the call processor 435 begins to play audible ringing, with a cadence of 2 seconds on, 4 seconds off, to the caller. Then at step 2140, the 20 second timer is checked, and if it is found to have expired, indicating that the subscriber did not connect to the MMF 115, then control passes to the connector labelled "CPF MM FLASH" at reference 2141, causing the caller to be connected to the message center as was described earlier. If the 20 second timer has not expired, then control passes to step 2142 where a determination is made as to whether the subscriber has connected to the MMF 115, as determined by checking with the CPF - Call Handler Task that was found to be controlling the trunk 3 connected to the subscriber. If the subscriber has not connected to the MMF 115, then control returns to step 2140. If the subscriber has connected to the MMF 115, then the trunk 3 is placed on hook causing switch 4 to transfer the calling party to the line 120 of the MMF 115, and control is returned to the CPF - Call Handler Task via the connector labelled "CPF IDLE" at reference 2144.

A flowchart of the CPF - Send Page Subroutine is illustrated in FIG. 22. The purpose of this subroutine is to send page messages via the high speed data network 150 from the CPF 100 to either the PDF 105, or the CAF 125. These messages contain the pager number and any digits which are to be transmitted to the display of the pager. The CPF - Send Page Subroutine is entered at step 2200 and control passes to step 2201 where the pager number 710 for this subscriber is retrieved from the subscriber master record 700. Then at step 2202, a message is constructed consisting of the pager number 710, and the display digits which were identified as this subroutine was called. Then at step 2203 a determination is made as to whether the pager number 710 represents a communicator. In the preferred embodiment, each communicator 11 is identified by a pager number 710 which starts with the digits '000'. If the pager number 710 is found to be that of a communicator 11, then control passes to step 2204, and the data network interface 510 is instructed to send the message via network 150 to the Communicator Access Facility (CAF) 125, and then control is returned from this subroutine to the calling program, as represented at step 2206. If the pager number 710 is not found to be that of a communicator 11, then control passes to step 2205, where the data network interface 510 is instructed to send the message via network 150 to the Pager Dialing Facility (PDF) 105, and then control is returned from this subroutine to the calling program, as represented at step 2206.

A flowchart of the CPF - Command Mode function is illustrated in FIG. 23. The CPF - Command Mode function is entered at step 2300 and control passes to step 2301 where the subscriber's pre-recorded 'drop-in' name is retrieved from disk 505. Then at step 2302, the call processor 435 prompts the subscriber: "Hello..", and then plays the back the 'drop-in' name. Then at step 2303, a determination is made as to whether a meet-me call is currently holding for this subscriber. This is determined by checking the CPF - Call Handler Tasks which control the other trunks 3 connected to CPF 100. If a meet-me call is found to be holding for this subscriber, then at step 2304, the subscriber is prompted: "A call is holding on your meet-me service. Touch 4 to be connected to the caller." Control then passes to step 2317. If a meet-me call is not holding for this subscriber, then control passes to step 2305 where a determination is made as to whether a meet-me caller was recently holding, but hung up without leaving a message. If the time and date stored in the 'last meet me abandon' field 719 of the subscriber master record 700 is not more than 20 minutes older than the current date and time, then it is determined that a meet-me caller recently abandoned a call, and control passes to step 2306 where the subscriber is prompted: "A call was recently holding on your meet-me service, however the caller chose not to wait and hung up without leaving a message." Control then passes to step 2307 where the 'last meet-me abandon' field 719 of the subscriber master record 700 is cleared. Control then passes to step 2308. Control also passes to step 2308 if, at step 2305, it is determined that a meet-me caller did not recently abandon a call. At step 2308, a determination is made as to whether a meet-me caller is currently leaving a message for the subscriber. This is determined by checking the CPF - Call

Handler Tasks which control the other trunks 3 connected to CPF 100. If it is determined that a meet-me caller is currently leaving a message for this subscriber, then control passes to step 2309, where the subscriber is prompted: "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." Control then passes to step 2310 where 'music-on-hold' is played to the subscriber by call processor 435. Control then passes to step 2311 where a determination is made as to whether the meet-me caller has finished leaving the message. If the message is complete, then control passes to the CPF - Command Message Center function as shown at reference 2313. If the caller is still leaving the message, then control passes to step 2312 where a determination is made as to whether the subscriber has entered the '*' digit. If the '*' digit is not entered, then control returns to step 2311. If the '*' digit is entered, then control passes to step 2317. If at step 2308, a determination is made that a meet-me caller is not currently leaving a message for this subscriber, then control passes to step 2314, where a determination is made as to whether a meet-me caller recently left a message for this subscriber. If the time and date stored in the 'last meet me message left' field 720 of the subscriber master record 700 is not more than 20 minutes older than the current date and time, then it is determined that a meet-me caller recently left a message, and control passes to step 2315 where the subscriber is prompted: "A call was recently holding on your meet-me service, however the caller chose not to wait and instead left you a message." Control then passes to step 2316 where the 'last meet-me message left' field 720 of the subscriber master record 700 is cleared. Control then passes to step 2317. Control also passes to step 2317 if, at step 2314, it is determined that a meet-me caller did not recently leave a message. At step 2317, a determination is made as to whether the subscriber has selected an external message center. As described earlier, this determination is made by examining the message center number 709 in the subscriber master record 700. If the message center number 709 contains a phone number, then external has been selected. If the message center number 709 does not contain a phone number, then internal message center has been selected. If, at step 2317, it is determined that the subscriber has selected an external message center, then control passes to step 2318, where the caller is prompted: "We have transferred...". Control then passes to step 2319 where the 'external message center transfer count' 721 is retrieved from the subscriber master record 700, and is voiced to the subscriber by call processor 435. Control then passes to step 2320, where the prompt is completed by playing: "...since you last checked messages." Control then passes to the connector labelled "CPF MODE DESCR" at reference 2323. If, at step 2317, it is determined that the subscriber had selected the internal message center, then control passes to step 2321 where the number of messages currently stored for this subscriber on disk 505 is determined. At step 2322 then, the call processor 435 prompts the subscriber: "You have X messages.", where X is the number determined above. Control then passes to the connector labelled "CPF MODE DESCR" at reference 2323. The connector labelled "CPF MODE DESCR" at reference 2324, causes control to be passed to step 2325, where a description is played of the current call handling mode. This description includes the current mode memory number 715, the current call handling mode 703, and the current transfer number 707, if appropriate. For example, the subscriber may hear: "Your calls are currently being handled by mode memory 10, urgent-screened forwarding to 555-1111." If the transfer number 707 is the tag for the subscriber's home, office, pager, mobile-phone, or message center, then this would be voiced in words, ie: "..to your home." After playing a description of the current call handling mode, then control passes to step 2326, where a determination is made as to whether the feature timer is currently active. As was described earlier, this determination is made by checking the feature timer duration 724. If the feature timer is found to be active, then it's status is voiced to the subscriber at step 2327. For example, the subscriber may be prompted: "The feature timer is currently active and will cause mode memory 10 to be invoked at 5:30 today." Control then passes to step 2328. Control also passes to step 2308 if the feature timer was found to be inactive at step 2326. At step 2328, a determination is made as to whether the weekly schedule is active. This determination is made by checking the weekly schedule active flag 728 of the subscriber master record 700. If the weekly schedule is found to be active, then the status of the weekly schedule is voiced to the subscriber at step 2329. For example, the subscriber may be prompted: "The weekly schedule is on, and the next step will cause memory 20 to be invoked at 7:30 PM on Tuesday." Control then passes via connector labelled "CPF MAIN DIRECTORY" at reference 2330 to step 2331. Control also passes via connector 2330 to step 2331 if the weekly schedule is found to be inactive at step 2328. At step 2331 the call processor 435 prompts the subscriber: "Main Directory. Enter 1 to check messages, 2 to change your forwarding number, 3 to select a memory, 9 to make a call, or 0 for help." Control then passes to a loop consisting of steps 2332 through 2339. At each of these steps a determination is made as to whether a particular digit has been entered by the subscriber. If the result is positive on any of these steps, then control is passed to another function. If the digit '1' is found at step 2332, then control passes to the CPF - Command Message Center function, as shown at reference 2340. If the digit '2' is found at step 2333, then control passes to the CPF - Command Forwarding Number function, as shown at reference 2341. If the digit '3' is found at step 2334, then control passes to the CPF - Command Memory function, as shown at reference 2342. If the digit '9' is found at step 2335, then control passes to the CPF - Command Outside Call function, as shown at reference 2343. If the digit '0' is found at step 2336, then control passes to the CPF - Command Help function, as shown at reference 2344. If the digit '4' is found at step 2337, then control passes to the CPF - Command Meet Me function, as shown at reference 2345. If the digit '5' is found at step 2338, then control passes to the CPF - Command Branch Route function, as shown at reference 2346. If the digit '8' is found at step 2339, then control passes to the CPF - Command Advanced features function, as shown at reference 2347.

A flowchart of the CPF - Command Message Center function is illustrated in FIG. 24. The CPF - Command Message Center function is entered at step 2400 and control passes to step 2401 where a determination is made as to whether the subscriber has selected an external message center. As described earlier, this determination is made by examining the message center number 709 in the subscriber master record 700. If the message center number 709 contains a phone number, then external has been selected. If the message center number 709 does not contain a phone number, then internal message center has been selected. If, at step 2401, it is determined that the subscriber has selected an external message center, then control passes to step 2402, where the caller is prompted: "Please standby." Control then passes to step 2403 where a flash is generated causing switch 4 to place the subscriber on hold a apply a dial tone to trunk 3. Then, after pausing for dial tone at step 2404, the message center number 709 is dialed by call processor 435 at step 2405. Then at step 2406, the trunk 3 is placed on hook, causing switch 4 to transfer the subscriber to the message center number. Then at step 2407, the 'external message center transfer count' 721 of the subscriber master record 700 is cleared. Control is then returned to the CPF - Call Handler Task via the connector labelled "CPF IDLE" at reference 2408. If, at step 2401, it is determined that the subscriber has selected internal message center, then control passes to step 2409, where a determination is made as to whether the subscriber has any messages stored on disk 505. If there are no messages stored for this subscriber, then control passes to step 2410, where the subscriber is informed: "You have no messages." Then at step 2411, the subscriber is prompted: "Enter 8 to return to the main directory." Control then passes to step 2412, where a determination is made as to whether the digit '8' has been entered, and if it has not been entered, then control returns to step 2410. If the digit '8' has been entered, then control passes to the connector labelled "CPF MAIN DIRECTORY" at reference 2413, allowing the subscriber to make additional selections from the main directory of the command mode. If at step 2409, the determination is made that the subscriber does have messages stored on disk 505, then control passes to step 2414, where one of the stored messages is played back to the subscriber, and the subscriber is given the opportunity to save or delete the message by entering the digits '1' or '2' respectively. Control then passes to step 2415, where a determination is made as to whether the digit '9' has been entered. If the digit '9' is entered, then control passes to the CPF - Command Outside Call function as shown at reference 2416, where the subscriber is given the opportunity to make a call, perhaps returning a call to the person who left the message. If the digit '9' was not entered, then control passes to step 2417, where a determination is made as to whether any more messages are stored on the disk 505. If more messages exist, then control is returned to step 2414. If no more messages exist, then control passes to step 2418, where the subscriber is prompted: "You have no more messages. Enter 8 to return to the main directory." Control then passes to step 2419, where a determination is made as to whether the digit '8' has been entered, and if it has not been entered, then control returns to step 2418. If the digit '8' has been entered, then control passes to the connector labelled "CPF MAIN DIRECTORY" at reference 2413, allowing the subscriber to make additional selections from the main directory of the command mode.

A flowchart of the CPF - Command Forwarding Number function is illustrated in FIG. 25. The CPF - Command Forwarding Number function is entered at step 2500 and control passes to step 2501 where a determination is made as to whether the current call handling mode 703 for this subscriber is 'message center' or branch routing'. Since it is not logical to change a forwarding number in a mode that does not require a forwarding number, if it is determined that either of these modes are active, then control will pass to step 2502, where the subscriber will be prompted: "Invalid command." Control then passes to the connector labelled "CPF MAIN DIRECTORY" at reference 2503, allowing the subscriber to make additional selections from the main directory of the command mode. If, at step 2501, it is determined that the current call handling mode 703 is not 'message center' or 'branch routing', then control passes to step 2504, where a determination is made as to whether the transfer number 707 is set equal to the 'meet-me tag'. If so, then control passes to step 2505, where the subscriber is prompted: "Your calls are currently being forwarded to you via your meet me service." Control then passes to step 2508. If, at step 2504, it is determined that the forwarding number is not equal to the 'meet-me tag' then control passes to step 2506, where the subscriber is prompted: "Your calls are currently being forwarded to..." Control then passes to step 2507, where the transfer number 707 is retrieved from the subscriber master record 700, and voiced to the subscriber. Control then passes to step 2508, where the subscriber is prompted: "Please enter your new forwarding number, or enter * to skip this, or enter 6 to use your meet-me service." Control then passes to step 2509, where a determination is made as to whether the subscriber has entered a valid phone number, or a tag for home, office, or mobile phone. If so, then control passes to step 2510, where the phone number is stored as the new transfer number 707 in the subscriber master record 700, and the subscriber is prompted: "Accepted, your calls are currently being forwarded to..." , and the transfer number 707 is voiced. Control then passes to the CPF - Command Feature Timer function as shown at reference 2511. If it is determined at step 2509 that the subscriber has not entered a phone number, then control passes to step 2512, where a determination is made as to whether the subscriber has entered the digit '6', and if so control passes to step 2513 where the 'meet-me tag' is stored as the transfer number 707 in the subscriber master record 700, and the subscriber is prompted: "Accepted, your calls are currently being forwarded to you via your meet-me service." Control then passes to the CPF - Command Feature Timer function as shown at reference 2511. If it is determined at step 2512 that the digit '6' has not been entered, then control passes to step 2514, where a determination is made as to whether the digit '*' has been entered, and if not control returns to step 2509. If the digit '*' has been entered, then the transfer number 707 remains unchanged, and control passes to the CPF - Command Feature Timer function as shown at reference 2511.

A flowchart of the CPF - Command Feature Timer function is illustrated in FIG. 26. The CPF - Command Feature Timer function is entered at step 2600 and control passes to step 2601 where the subscriber is prompted: "Please enter the length of time you wold like your current feature to be in effect, or enter * to skip the feature timer." Control then passes to step 2602, where a determination is made as to whether the digit '*' has been entered by the subscriber. If the digit '*' has been entered, then control passes to step 2603, where the subscriber is prompted: "Accepted, your feature will be in effect until further notice." Control then passes to the connector labelled "CPF MAIN DIRECTORY" at reference 2613, allowing the subscriber to make additional selections from the main directory of the command mode. If at step 2602, it is determined that the digit '*' has not been entered, then control passes to step 2604 where a determination is made as to whether a valid duration has been entered. This duration must be in the form of hours first followed by minutes. For example, to enter 1 hour and 20 minutes, the subscriber must enter '1 2 0'. If a valid duration is not found to be entered at step 2604 then control returns to step 2602. If a valid duration is entered, then control passes to step 2605, where the duration is stored as the 'feature timer duration' 724 of the subscriber master record 700, and the subscriber is prompted: "Accepted,..." and the duration time is voiced. Control then passes to step 2606 where the subscriber is prompted: "Please enter the memory number you wish to invoke upon termination of the feature timer, or enter * to use the previous mode, or touch 9 to use the schedule." Control then passes to step 2607, where a determination is made as to whether the subscriber has entered the digit '*', and if so then at step 2608 the previous mode memory is saved in temporary mode memory 0, mode memory 0 is stored as the feature timer termination mode 725 of the subscriber master record, and the subscriber is prompted: "Accepted, your feature will be in effect until (time) at which time the previous mode will be invoked.", where the value of (time) corresponds to the current time plus the feature timer duration 724. Control then passes to the connector labelled "CPF MAIN DIRECTORY" at reference 2613, allowing the subscriber to make additional selections from the main directory of the command mode. If at step 2607 it is determined that the digit '*' has not been entered, then control passes to step 2609, where a determination is made as to whether the digit '9' is entered, and if so control passes to step 2610 where the weekly schedule is saved in temporary mode memory 0, mode memory 0 is stored as the feature timer termination mode 725 of the subscriber master record, and the subscriber is prompted: "Accepted, your feature will be in effect until (time) at which time the weekly schedule will be invoked.", where the value of (time) corresponds to the current time plus the feature timer duration 724. Control passes to the connector labelled "CPF MAIN DIRECTORY" at reference 2613, allowing the subscriber to make additional selections from the main directory of the command mode. If at step 2609 it is determined that the digit '9' has not been entered, then control passes to step 2611, where a determination is made as to whether a valid mode memory 10 through 99 is entered, and if so control passes to step 2612 where the mode memory is stored as the feature timer termination mode 725 of the subscriber master record, and the subscriber is prompted: "Accepted, your feature will be in effect until (time) at which time mode memory XX will be invoked.", where the value of (time) corresponds to the current time plus the feature timer duration 724. Control passes to the connector labelled "CPF MAIN DIRECTORY" at reference 2613, allowing the subscriber to make additional selections from the main directory of the command mode.

A flowchart of the CPF - Command Memory function is illustrated in FIG. 27. The CPF - Command Memory function is entered at step 2700 and control passes to step 2701 where the subscriber is prompted: "Your calls are currently being handled via mode memory XX.", where XX is the current mode memory number 715 of the subscriber master record. Control then passes to step 2702, where a description of the mode is voiced to the subscriber. This description includes the current call handling mode 703, and the current transfer number 707, if appropriate. For example, the subscriber may hear: "Your calls are currently being urgent-screened forwarded to 555-1111." If the transfer number 707 is the tag for the subscriber's home, office, pager, mobile-phone, or message center, then this would be voiced in words, ie: "..to your home." Control then passes to step 2703 where the subscriber is prompted: "Please enter a new mode memory number or enter * to skip this." Control then passes to step 2704 where a determination is made as to whether the digit '*' has been entered. If the digit '*' is entered, then control passes to the CPF - Command Feature Timer function as shown at reference 2722. If at step 2704, it is determined that the '*' digit has not been entered, then control passes to step 2705, where a determination is made as to whether a valid 2 digit mode memory number has been entered, and if such a mode memory number has not been entered, then control returns to step 2703. If a valid mode memory number is entered, then control passes to step 2706, where the mode memory number is stored as the current mode memory number 715 of the subscriber master record 700, the corresponding mode memory 800 is copied to the subscriber master record, the subscriber is prompted: "Accepted, you have selected mode memory XX, which causes your calls to be...", and then control passes to step 2707. At step 2707, the prompt is completed by playing a brief description of the selected mode memory. The description includes the call handling mode 703, and the transfer number 707 if appropriate. For example, at step 2707 the remainder of the prompt may be: "...handled by your message center." At step 2708 a determination is made as to whether the selected mode memory requires an 'externally entered number'. This is determined by checking the transfer number field 707 of the selected mode memory, to determine if it contains an 'externally entered number' tag. If this is the case, then the subscriber must enter a transfer number after selecting this memory. If the externally entered number is not required, then control passes to the CPF - Command Feature Timer function as shown at reference 2722. If it is determined at step 2708 that an externally entered number is required, then control passes to step 2710, where the subscriber is prompted: "Please enter your new forwarding number or enter * to skip this and use....". Control then passes to step 2711 where the previous transfer number is voiced to the subscriber. Control then proceeds to step 2712 where the subscriber is further prompted with: "..or enter 6 to use your meet-me service." Control then passes to step 2713 where a determination is made as to whether the digit '*' has been entered. If the digit '*' has been entered then control passes to step 2716. Otherwise, control passes to step 2714 where a determination is made as to whether a phone number or tag has been entered by the subscriber. If not, then control returns to step 2710. If a phone number or tag has been entered, then control passes to step 2715 where the phone number is stored as the transfer number 707 of the subscriber master record 700, the subscriber is prompted: "Accepted...", and the phone number or tag is voiced. Control then passes to step 2716, where a determination is made as to whether the memory has an extension number in the extension number field 708 of the subscriber master record 700. If an extension number does not exist in this field, then control passes to the CPF - Command

Feature Timer function as shown at reference 2722. If at step 2716 it is determined that an extension number does exist, then control passes to step 2717 where the subscriber is given an opportunity to modify the extension number. At step 2717 the subscriber is prompted: "Please enter your new extension number or enter * to skip this and use ....". Control then passes to step 2718 where the phrase is completed by voicing the current extension number 708. Control then passes to step 2719 where a determination is made as to whether a new extension number has been entered, and if so, then control passes to step 2720 where the extension number is stored in the extension number field 708, the subscriber is prompted: "Accepted,..", the new extension number is voiced, and control passes to the CPF - Command Feature Timer function as shown at reference 2722. If at step 2719 it is determined that an extension number has not been entered then control passes to step 2721, where a determination is made as to whether the digit "*" is entered, and if the "*" digit is not entered then control returns to step 2717. If at step 2721 it is determined that the "*" digit is entered then control passes to the CPF - Command Feature Timer function as shown at reference 2722.

A flowchart of the CPF - Command Outside Call function is illustrated in FIG. 28. The CPF - Command Outside Call function is entered at step 2800 and control passes to step 2801 where the call processor 435 plays a 'stutter dial tone' to the subscriber. Control then passes to step 2802 where a determination is made as to whether the subscriber has entered the '#' digit. If the '#' digit has been entered, then control passes to the connector labelled "CPF MAIN DIRECTORY" at reference 2803, allowing the subscriber to make additional selections from the main directory of the command mode. If at step 2802 it is determined that the subscriber has not entered the '#' digit, then control passes to step 2804 where a determination is made as to whether the subscriber has entered a phone number that he wishes to be connected to. If a phone number has not been entered, then control returns to step 2802. If at step 2804 it is determined that a phone number has been entered, then control passes to step 2805, where a flash is generated, causing switch 4 to place the subscriber on hold, and applying a dial tone to the trunk 3. Then, after a pause for dial tone at step 2806, the call processor 435 dials the phone number at step 2807, which had been entered by the subscriber. Control then passes to step 2808, where the 'multiple outside calls allowed' flag 729 of the subscriber master record 700 is checked. If this flag is not active, then control passes to step 2809 where the trunk 3 is placed on hook, causing switch 4 to transfer the subscriber to the dialed number, and control is returned to the CPF - Call Handler Task via the connector labelled "CPF IDLE" at reference 2810. If at step 2808 it is determined that the 'multiple outside calls allowed' flag 729 is set, then control passes to step 2811, where a flash is generated which causes the subscriber to be conferenced to the dialed number while maintaining trunk 3 in the connection. In this way the subscriber may make additional calls, or later return to the main directory without hanging up and calling back. After the flash is generated at step 2811, control passes to step 2812 where a loop is entered waiting for the subscriber to enter the digit sequence '##'. When trying to detect DTMF digits while conversation may be present, as is the case in this situation, requiring the two digits in sequence reduces the likelihood of falsing on voice. If the subscriber enters '##', then control passes to step 2813 where a flash is generated causing switch 4 to disconnect the third party from the conference, leaving only the subscriber connected to trunk 3. Control then returns to step 2801, where the subscriber may make another call, or enter '#' to return to the main directory.

A flowchart of the CPF - Command Help function is illustrated in FIG. 29. The purpose of this function is to provide help to the subscriber who is using the Command Mode features of the Telephone Control System. This is accomplished by allowing the subscriber to enter the digit '0' for help from any of the command mode functions. Once the digit '0' is entered, the subscriber is provided with an explanation of the function which was being accessed at that moment. The subscriber may also touch another digit '0' to reach a live client services representative. The CPF - Command Help function is entered at step 2900 and control passes to step 2901 where the step number of the function from which the CPF - Command Help was requested is saved for later use. Then at step 2902, the subscriber is prompted:

"You have selected the help function. You may enter 0 to be connected to a client services operator, 8 to return to the main directory, or # to return to the point where you were when you entered the help function." Control then passes to step 2903, where a context sensitive help prompt is played, based on the step number saved in by step 2901. For example, if the saved step number indicated that the help function was accessed while in the CPF - Command Forwarding Number function, then the call processor 435 would play the prestored help prompt associated with that function: "When you selected the help function you were in the process of changing your forwarding number." Control then passes to step 2904 where a determination is made as to whether the digit '0' has been entered, and if so, control then passes to step 2905 where a flash is generated, causing switch 4 to place the subscriber on hold, and applying a dial tone to the trunk 3. Then, after a pause for dial tone at step 2906, the call processor 435 dials the phone number at block 2907, which is associated with a client services representative. Control then passes to step 2908 where the trunk 3 is placed on hook, causing switch 4 to transfer the subscriber to the dialed number, and control is returned to the CPF - Call Handler Task via the connector labelled "CPF IDLE" at reference 2909. If at step 2904 it is determined that the digit '0' is not entered, then control passes to step 2910 where a determination is made as to whether the digit '8' has been entered, and if so, control passes to the connector labelled "CPF MAIN DIRECTORY" at reference 2911, allowing the subscriber to make additional selections from the main directory of the command mode. If at step 2910 it is determined that the digit '8' is not entered, then control passes to step 2912 where a determination is made as to whether the digit '#' is entered, and if not control returns to step 2902. If the digit '#' is entered, then the step number that was saved at step 2901 is retrieved and the control returns to that step.

A flowchart of the CPF - Command Meet Me function is illustrated in FIG. 30. The purpose of this function is to process the call for a subscriber who has called in, entered his PIN code 702, and wishes to be connected to a meet-me caller who is holding for him. The CPF - Command Meet Me function is entered at step 3000 and control passes to step 3001 where a flash is generated, causing switch 4 to place the subscriber on hold and apply a dial tone to the trunk 3. Then, after pausing for dial tone at step 3002, the call processor 435 dials the phone number of the lines 120 which are connected to the Meet Me Facility (MMF) 115 at step 3003. Control then passes to step 3004 where a 15 second timer is started, and then to step 3005 where the 15 second timer is checked. If the 15 second timer has not expired, then control passes to step 3006 where a determination is made as to whether a '*' digit is detected, indicating that the MMF 115 has answered the call. If the '*' digit is not detected then control returns to step 3005. If at step 3006, the '*' digit is detected, then control passes to step 3007 where the call processor 435 dials '01' indicating this call is from a subscriber, and then dials the subscriber's DID number 701, thereby fully identifying the call to the MMF 115. Control then passes to step 3008 where a flash is generated causing switch 4 to take the subscriber off of hold and create a conference between the subscriber, trunk 3, and the line 120 of the MMF 115. Control then passes to step 3009 where a 4 second pause is initiated to allow the MMF 115 time to connect the caller into the call. Control then passes to step 3010 where the subscriber and caller are prompted by call processor 435: "Go ahead please." Control then passes to step 3011 where the trunk 3 is placed on hook, causing the switch 4 to transfer the subscriber to the line 120 of the MMF 115. Control is then returned to the CPF - Call Handler Task via the connector labelled "CPF IDLE" at reference 3012. If at step 3005 it is determined that the 15 second timer had expired, indicating that the MMF 115 had not answered the call, then control passes to step 3013 where a flash is generated causing switch 4 to create a temporary conference between the subscriber and the dialed number, and then after a 2 second pause at step 3014 another flash is generated at step3015 causing switch 4 to drop the dialed number from the conference and leave just the subscriber connected to the trunk 3. Control then passes to step 3016 where the subscriber is prompted: "I'm sorry, we are unable to connect your call at this time." Control then passes to the connector labelled "CPF MAIN DIRECTORY" at reference 3017, allowing the subscriber to make additional selections from the main directory of the command mode.

A flowchart of the CPF - Command Branch Route function is illustrated in FIG. 31. The CPF - Command Branch Route function is entered at step 3100 and control passes to step 3101 where a determination is made as to whether the current call handling mode 703 is 'branch-routing', and if it is not control is passes to step 3102 where the subscriber is prompted: "Invalid command.", and control passes to the connector labelled "CPF MAIN DIRECTORY" at reference 3103, allowing the subscriber to make additional selections from the main directory of the command mode. If at step 3101 it is determined that the call handling mode 703 is 'branch routing' then control passes to step 3104 where the subscriber is prompted: "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." Control then passes to step 3105 where a determination is made as to whether the digit '1' is entered, and if so, control passes to step 3106 where the subscriber is prompted: "Your branch routing greeting is...". Then at step 3107, the branch routing greeting for this subscriber is retrieved from disk 505 and played back to the subscriber. Control then passes to step 3108 where the subscriber is prompted: "Please record your new branch routing greeting at the tone, or enter * to skip this and use the existing branch routing greeting...BEEP." Then at step 3109, the new greeting is recorded and control passes to step 3110 where a determination is made as to whether the greeting is done, and if it is control passes to step 3111 where the new branch routing greeting is saved to disk 505 and then control returns to step 3104. If at step 3110 it is determined that the greeting is not done, then control passes to step 3112 where a determination is made as to whether the digit '*' is entered, and if not control returns to step 3109. If at step 3112 it is determined that the digit '*' is entered, then control passes to step 3113 where the old branch routing greeting on disk 505 is left unchanged, and control then passes to step 3104. If at step 3105 it is determined that the digit '1' is not pressed, then control passes to step 3114 where a determination is made as to whether the digit '2' is entered, and if it is entered then control passes to step 3115 where the subscriber is prompted: "Please enter the branch routing directory digit 1 to 9 for the phone number your wish to change, or touch * to skip this." Control then passes to step 3116 where a determination is made as to whether the digit '*' is entered, and if it is entered control returns to step 3104. If at step 3116 it is determined that the digit '*' is not entered, then control passes to step 3117 where a determination is made as to whether one of the digits '1' to '9' are entered, and if one of those digits is entered control then passes to step 3118. Otherwise control returns to step 3116. At step 3118 the subscriber is prompted: "The branch routing transfer number for digit X is....". Control then passes to step 3119 where the branch routing number 722 that corresponds to the digit entered in step 3117 is retrieved from the subscriber master record 700 and voiced to the subscriber. Then at step 3120 the subscriber is prompted: "Please enter the new branch routing transfer number or touch * to skip this without making a change." Control then passes to step 3121 where a determination is made as to whether the digit '*' is entered, and if it is entered control returns to step 3104. If at step 3121 it is determined that the digit '*' is not entered, then control passes to step 3122 where a determination is made as to whether a phone number has been entered, and if a phone number has not been entered, control returns to step 3121. If at step 3122 a phone number is entered, then control passes to step 3123 where the subscriber is prompted: "Accepted." Then at step 3124 the new phone number is voiced to the subscriber. Control then passes to step 3125 where the new phone number is saved as the branch routing number 722 which corresponds to the digit entered in step 3117. Control then returns to step 3104. If at step 3114 it is determined that the digit '2' is not entered, then control passes to step 3126 where a determination is made as to whether the digit '3' is entered, and if it is entered, then control passes to step 3127 where the subscriber is prompted: "The branch routing default transfer number is ..." Control then passes to step 3128 where the branch routing default number 723 is retrieved from the subscriber master record 700 and voiced to the subscriber. Then at step 3129 the subscriber is prompted: "Please enter the new branch routing default transfer number, or touch * to skip this without making a change." Control then passes to step 3130 where a determination is made as to whether the digit '*' is entered, and if the digit '*' is entered control returns to step 3104. If at step 3130 it is determined that the digit '*' is not entered, then control passes to step 3131 where a determination is made as to whether a phone number has been entered, and if a phone number has not been entered control returns to step 3130. If it is determined at step 3131 that a phone number has been entered, then control passes to step 3132 where the subscriber is prompted: "Accepted." Control then passes to step 3133 where the new phone number is voiced to the subscriber. Control then passes to step 3134 where the new phone number is saved as the branch routing default number 723 in the subscriber master record. Control then returns to step 3104. If at step 3126 it is determined that the digit '3' is not entered, then control passes to step 3135 where a determination is made as to whether the digit '#' is entered, and if it is not entered control then returns to step 3104. If at step 3135 it is determined that the digit '#' is entered then control passes to the connector labelled "CPF MAIN DIRECTORY" at reference 3136, allowing the subscriber to make additional selections from the main directory of the command mode.

A flowchart of the CPF - Command Advanced Features function is illustrated in FIG. 32. The purpose of this function is provide the subscriber with the opportunity to modify those features of the Telephone Control System 1 which do not need to be modified on a regular basis. These features include allowing the subscriber to program mode memories, allowing the subscriber to record his 'drop-in' name, or his personalized greeting, allowing the subscriber to program his reserved numbers, and allowing the subscriber to activate or deactivate the weekly schedule. The CPF - Command Advanced Features function is entered at step 3200 and control passes to step 3201 where the subscriber is prompted: "Advanced Features Directory. Enter 1 to program mode memories, 2 to record greetings, 3 to program reserved numbers, 4 to activate the weekly schedule, 5 to deactivate the weekly schedule, or # to return to the Main Directory." Control then passes to step 3202 where a determination is made as to whether the digit '1' is entered. If the digit '1' is entered, then control passes to step 3203 where the subscriber is allowed to specify a mode memory number 802 of value '10 to '99'. If the mode memory number 802 specified already exists the call processor voices the status of that memory. The subscriber is then given an opportunity to modify the parameters 803 contained in the mode memory. When the subscriber is finished modifying the contents, the changes are saved in mode memory 800. Control then returns to step 3201. If at step 3202 it is determined that the digit '1' is not entered, then control passes to step 3204 where a determination is made as to whether the digit 2 is entered. If the digit '2' is entered, then control passes to step 3205 where the subscriber is allowed to listen to and re-record the 'drop-in' name and the 'personalized greeting'. If the subscriber does re-record either of these, then the changed name or greeting is saved on disk 505. Control then returns to step 3201. If at step 3204 it is determined that the digit '2' is not entered, then control passes to step 3206 where a determination is made as to whether the digit '3' is entered. If the digit '3' is entered, then control passes to step 3207 where the subscriber is allowed to modify the 'message center number' 709, the 'pager number' 710, the 'office number' 711, the 'home number' 712, or the 'mobile phone number' 713. If the subscriber changes any of these numbers then the new number is saved in the corresponding field of the subscriber master record 700. Control then returns to step 3201. If at step 3206 it is determined that the digit '3' is not entered, then control passes to step 3208 where a determination is made as to whether the digit '4' is entered. If the digit '4' is entered, then control passes to step 3209 where the subscriber is allowed to activate the weekly schedule. If the subscriber chooses to activate the weekly schedule, then the 'weekly schedule active' flag 728 of the subscriber master record 700 is set. Control then returns to step 3201. If at step 3208 it is determined that the digit '4' is not entered, then control passes to step 3210 where a determination is made as to whether the digit '5' is entered. If the digit '5' is entered, then control passes to step 3211 where the subscriber is allowed to deactivate the weekly schedule. If the subscriber chooses to deactivate the weekly schedule, then the 'weekly schedule active' flag 728 of the subscriber master record 700 is cleared. Control then returns to step 3201. If at step 3210 it is determined that the digit '5' is not entered, then control passes to step 3212 where a determination is made as to whether the digit '#' is entered. If the digit '#'

is not entered, then control returns to step 3201. If at step 3212 it is determined that the digit '#' is entered then control passes to the connector labelled "CPF MAIN DIRECTORY" at reference 3213, allowing the subscriber to make additional selections from the main directory of the command mode.

A block diagram of the Meet-Me Facility (MMF) 115 is illustrated in FIG. 33. As was discussed earlier, the MMF 115 interfaces to tip-ring lines 120. These lines are provisioned by switch 4 with the CENTREX feature of 'call transfer', which allows a caller to be transferred to another number by flashing, dialing the number, and then going on hook. These lines are also provisioned by switch 4 with the CENTREX feature of 'barge-in', which allows a party on one of the lines 120 to barge into a conversation in progress on another of the lines 120. This is accomplished by flashing, dialing a barge-in code (*77), and then dialing the intercom code associated with the line 120 of the conversation to be barged-in on. As was mentioned earlier, to fully understand the operation of the meet-me feature it is necessary to also review the explanations which are associated with FIG. 21 (CPF - Meet Me Caller), FIG. 30 (CPF - Command Meet Me), and FIG. 34 (Meet Me Facility Main Task). Referring now to FIG. 33, the lines 120 are shown connected to call processors 3300, which contain a tip-ring interface and DTMF generators and detectors. The functions of call processor 330 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 MMF 115 also contains a CPU 3301 which contains among other things a microprocessor, a boot ROM, a RAM, and a disk. The MMF 115 also contains a data network interface module 3302 which connects to the high speed data network 150. The functions of data network interface 150 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 processors 3300, the CPU 3301, and the data network interface 3302 are all shown connected to an internal data bus 3303. The CPU 3301 initializes itself at power-up using the boot ROM and then loads a control program into memory which it then executes. The control program allows for the control of simultaneous activities on the lines 120.

A flowchart of the Meet Me Facility Main Task is illustrated in FIG. 34. The Meet Me Facility Main Task is the part of the MMF 115 control program which controls the activities on one of the lines 120. The Meet Me Facility Main Task is entered at step 3400 and control passes to step 3401 where a determination is made as to whether the call processor 3300 has detected a ring signal on the line 120, and if a ring signal is not detected, then control remains at step 3401. If a ring signal is detected, then control passes to step 3402, where the line 120 is taken off hook by call processor 3300, thereby answering the incoming call. At step 3403, a 1 second pause is initiated to allow for the line 120 to settle, and then at step 3404 the call processor 3300 dials the DTMF digit '*' as an answer indication to the CPF 100 which is calling. Then at step 3405 a 5 second timer is started, and control then passes to step 3406 where the 5 second timer is checked. If the 5 second timer has expired, then control passes to step 3407 where the line 120 is placed on-hook by call processor 3300, and then control returns to step 3401. If at step 3406 it is determined that the 5 second timer has not expired, then control passes to step 3408 where a determination is made as to whether the DTMF digit sequence '00' is detected by call processor 3300, indicating the call is a meet-me caller from CPF 100. If the digit sequence '00' is not detected, then control passes to step 3409 where a determination is made as to whether the DTMF digit sequence '01' is detected, indicating the call is a meet-me subscriber from CPF 100. If the digit sequence '01' is not detected, then control returns to step 3406 where the 5 second timer is again checked. If at step 3408 it is determined that the digit sequence '00' is detected, then control passes via a connector labelled "MMF CALLER" at reference 3410 to step 3411, where a determination is made as to whether a DTMF digit sequence representing the Access Number 701 of the subscriber being called is detected by call processor 3300. If a valid phone number is not detected, then control remains at step 3411. If a valid phone number is detected, then control passes to step 3412 where a 2 second pause is initiated. Then at step 3413, the call processor 3300 dials the DTMF digit '*' to inform the CPF 100 that the connection has been successful so far.

Control then passes to step 3414 where a determination is made as to whether the DTMF digit 'd' is detected, indicating that the subscriber has called into the CPF 100, and the CPF 100 is about to conference him to the MMF 115. If the digit 'd' is not detected then control remains at step 3414. If the digit 'd' is detected, then control passes to step 3415 where a flash is generated on lines 120 causing the CENTREX system serving lines 120 to place the calling party (in this case the CPF 100) on hold, and a dial tone to be applied to line 120. Control then passes to step 3416 where a determination is made as to whether the subscriber has yet been connected to one of the other of lines 120 on the MMF 115. If the subscriber has not yet been connected, then control remains at step 3416. If it is determined that the subscriber has connected to one of the other of lines 120, then control passes to step 3417, where the CENTREX 'barge-in code' (*77) is dialed by the DTMF generator of call processor 3300. Then at step 3418, the call processor 3300 dials the intercom code for the line 120 which is currently connected to the subscriber. Control then passes to step 3419 where a 2 second pause is generated, and then to step 3420 where a flash is generated. This causes the line 120 which is connected to the subscriber to be connected via the CENTREX system to the line 120 which is connected to the caller. Control then passes to step 3421 where control remains while the subscriber and caller converse, until a loop interruption signal is detected on line 120 indicating at least one of the two parties has disconnected. Control then passes to step 3422 where the line 120 is placed on hook, and control the returns to step 3401. If at step 3409 it is determined that the digit sequence '01' is detected, then control passes via a connector labelled "MMF SUBSCRIBER" at reference 3423 to step 3424, where a determination is made as to whether a DTMF digit sequence representing the Access Number 701 of the subscriber is detected by call processor 3300. If a valid phone number is not detected, then control remains at step 3424. If a valid phone number is detected, then control passes to step 3425 where an indication is made available that a subscriber is connected to the MMF 115 on this line 120. Control then passes to step 3426 where a 3 second pause is initiated, allowing time for the line 120 connected to the subscriber to perform the barge-in sequence. Control then passes to step 3427 where the line 120 is placed on hook causing the CENTREX system to call transfer the subscriber to the line 120 which has just barged-in. Control then returns to step 3401.

A block diagram of the Subscriber Access Facility (SAF) 110 is illustrated in FIG. 35. The SAF 110 provides a means by which subscribers can access the Telephone Control System 1 via trunks which provide automatic number identification (ANI). SAF trunk interface 3500 interfaces the SAF 110 with trunks 8. The trunk interface 3500 is the same trunk interface as was described earlier at reference 400 used in the CPF 100, however the E & M Lead Control Circuit operates under a slightly different set of instructions, as will be described below in the explanation which accompanies FIG. 36. Still referring to FIG. 35, as was discussed earlier, the preferred embodiment of the TELEPHONE CONTROL SYSTEM 1 employs a Feature Group D (FGD) facility for trunks 8. This is provided via a 4-wire E&M trunk provisioned with TYPE I signaling, which is well known in the art. These type of trunks provide a 2-wire balanced transmit audio connection, a 2-wire balanced receive audio connection, an E-Lead, and an M-Lead. Although only one trunk interface 3500, one trunk 8, and one call processor 3504 are shown in FIG. 35, it should be readily evident to one skilled in the art that additional trunk interfaces and call processors may be added to support additional trunks. The trunk interface 3500 provides a two-way audio path shown at reference 3501, a loop status output shown at reference 3502, and a on/off hook control input shown at reference 3503. These lines are shown connected to call processor 3504 which performs the functions of voice storage and playback, DTMF generation and detection, and call control. Devices which perform the functions of call processor 3504 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. Also shown is a multi frequency detector module 3505 which is shown connected to the call processor 3504. A commercially available multi-frequency module capable of performing this function is the Model MF/40 manufactured by Dialogic Corporation. A data network interface 3507 is used to connect the SAF 110 to the other subsystems of the Telephone

Control System 1. Data network interface 3507 passes data messages between the SAF 110 and these other subsystems. The functions of data network interface 3507 are well known in the art, and many products, such as the Model COM4i from Digiboard Corporation, exist commercially which can accomplish these functions. CPU 3506, which contains a microprocessor, a boot ROM, a RAM, and a disk, controls all functions of the SAF 110. The trunk interface 3500, the CPU 3506, the call processor 3504, and the data network interface 3507 are all shown connected to an internal data bus 3508. The CPU 3506 initializes itself at power-up using the boot ROM and then loads a control program into memory which it then executes. The control program allows for the control of simultaneous activities on the trunks 8. An explanation of the control program for the SAF 110 accompanies FIG. 37.

A flowchart of the operation of E&M control circuit for the trunk interface 3500 of the SAF 115 is shown in FIG. 36. As the construction of the trunk interface 3500 of FIG. 35 is identical to that of the trunk interface 400 of FIG. 5, the explanation which follows will refer to the reference numbers of FIG 5 when discussing the internal components of the trunk interface 3500. Referring now to FIG. 36, the E&M Lead Control Circuit Operation for the SAF Trunk Interface function is entered at step 3600 and control passes to step 3601 where the control circuit 450 idles waiting for an indication from current detector 440 that the E-Lead has gone off-hook. When the E-Lead does go off-hook, control passes to step 3602 where an "Incoming Call" signal is sent to CPU 3506 via buffer 460. Control then passes to step 3603 where a determination is made as to whether an off-hook signal is received from call processor 3504 signifying that the CPU 3506 is ready to accept the call. If the off hook signal is detected, then control passes to step 3604 where the 'loop status' 3502 is set active. Control then passes to step 3605 where the M-Lead is winked by taking the M-Lead relay 455 momentarily off-hook. Control then passes to step 3606 where a determination is made as to whether a 'DID received' signal 480 is detected. If the signal is not detected then control remains at step 3606. If the signal is received, indicating that the multifrequency detector 3505 has detected the 'ANI' and 'called number' digits from the FGD trunk 8, then control passes to step

3607 where the M-lead relay is once more winked momentarily off hook to acknowledge receipt of the multifrequency data. A 300 millisecond pause is then initiated at step 3608, prior to taking the M-Lead off hook at 3609 to 'answer' the trunk. At this point audio is established by the switch which is providing the FGD service, and the calling party is connected to the SAF 110. Control then passes to a loop consisting of steps 3610 and 3611. This loop persists until either at step 3610 the on/off hook signal 3503 is taken on hook by the call processor 3504, or at 3611 the E-Lead is determined to be on-hook. In either case the call is ended, and control passes to step 3612 where the 'loop status' signal 3502 is set inactive. Then at step 3613 a determination is made as to whether the 'DID received' signal 480 still remains active indicating the CPU 3506 is not yet ready to receive a new call. If this signal is still active, then control remains at step 3613. If it is determined at step 3613 that the 'DID received' signal 480 is now inactive, then control passes to step 3614 where the M-Lead is placed on hook, terminating the call, and control passes to step 3601.

A flowchart of the Subscriber Access Facility Main Task is illustrated in FIG. 37. This program is loaded into memory and executed by SAF CPU 3506. The Subscriber Access Facility Main Task is entered at step 3700 and control passes to step 3701 where a 'clear DID received' signal is sent to the trunk interface 3500. Control then passes to step 3702 where the 'incoming call' signal of the trunk interface 3500 is checked. If this signal is not active then control remains at step 3702. If this signal is active, then control passes to step 3703, where the output of the multifrequency detector 3505 is checked via call processor 3504. The incoming multifrequency digit sequence 'KP' + '00' + ANI + 'ST' is decoded where 'KP' is the start digit, 'ST' is the stop digit, and the ANI is the phone number of the phone from which the subscriber is calling. Control then passes to step 3704 where, in a similar manner, the incoming multifrequency digit sequence 'KP' + 800 + NXX + XXXX + 'ST' is decoded. Again the 'KP' is the start digit, the 'ST' is the stop digit, and the sequence 800 NXX XXXX is the phone number dialed by the subscriber to reach the trunk 8, the NXX being the prefix which identifies trunk group 8 to the PSTN 2. This phone number represents the programming function which the subscriber wishes to accomplish. Control then passes to step 3705 where a 'set DID received' signal is sent to the trunk interface 3500, indicating that the multifrequency data has been received. Control then passes to step 3706 where a determination is made as to whether the dialed number was of the form 800 - NXX - 00ab, and if the dialed number was not of this form then control passes to a connector labelled "SAF EXT" at reference 3707. If at step 3706 it is determined that the dialed number is of the form 800 - NXX - 00ab, then control passes to step 3710 where a 'request master record' message is constructed using the ANI received in step 3703, and the message is sent via data network interface 3507 to the CPF 100. Control then passes to step 3711 where a determination is made as to whether a response has been received to the 'request master record' message, and if such a response is not received control remains at step 3711. If the response message is received by network interface 3507, then control passes to step 3712 where a determination is made as to whether a valid subscriber master record 700 is included in the returned message. If a valid subscriber master record is not included, then control passes to the connector labelled "SAF REORDER" shown at reference 3713. If at step 3712 a valid subscriber master record 700 is found, then control passes to step 3714 where the DID number 701 is removed from the subscriber master record 700. Control then passes to step 3715 where a 'mode memory inquiry' message is constructed using the DID number 701 and the digits 'ab' as detected in step 3706, and the message is sent via data network interface 3507 to the CPF 100. Control then passes to step 3716 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 3716. If the response message is received by network interface 3507, then control passes to step 3717 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 3718. If at step 3717 a valid mode memory 800 is found, then control passes to step 3720, which is identified by a connector labelled "SAF UPDATE" at reference 3719. At step 3720 a 'change to new mode memory' message is constructed, again using DID number 701 and the digits 'ab' as detected in step 3706, and the message is sent via data network interface 3507 to the CPF 100. Control then passes to step 3721 where the subscriber is prompted by call processor 3504: "Accepted, you have selected mode memory...'ab'", where 'ab' are the digits detected in step 3706. Control then passes to step 3722 where the call processor 3504 causes trunk 8 to be placed on hook, and then control returns to step 3701. If at step 3706 it is determined that the dialed number was not of the form '800 - NXX - 00ab', then control passes to a connector labelled "SAF EXT" as shown at reference 3707. A connector labelled "SAF EXT" is shown at reference 3724, which causes control to be passed to step 3725 where a determination is made as to whether the dialed number was of the form 800 - NXX - cdef, where the digits 'cd' do not equal '00'. If the dialed number is not of this form, then control passes to a connector labelled "SAF REORDER" shown at reference 3745. If at step 3725 it is determined that the dialed number is of this form, then control passes to step 3726 where call processor 3504 prompts the subscriber by playing a 'bong tone'. Control then passes to step 3727 where a determination is made as to whether the subscriber has entered a PIN code, and if the PIN code is not entered control remains at step 3727. If a PIN code is entered, then control passes to step 3728 where a 'request master record' message is constructed using then PIN code entered in step 3727 and the digits 'cdef' detected in step 3706. This message is then sent via data network interface 3507 to the CPF 100. Control then passes to step 3729 where a determination is made as to whether a response has been received to the 'request master record' message, and if such a response is not received control remains at step 3729. If the response message is received by network interface 3507, then control passes to step 3750 where a determination is made as to whether a valid subscriber master record 700 is included in the returned message. If a valid subscriber master record is not included, then control passes to the connector labelled "SAF REORDER" shown at reference 3751. If at step 3750 a valid subscriber master record 700 is found, then this indicates that CPF 100 found