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(54) **SPEECH ANALYSIS BASED ANSWER DETECTION FOR IP BASED TELEPHONES**

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(57) **ABSTRACT**

Speaking a predetermined phrase by a user, causes a handset to transmit this audio information to a personal computer. The personal computer is responsive to the predefined phrase to determine that the user of the handset has answered an incoming call.

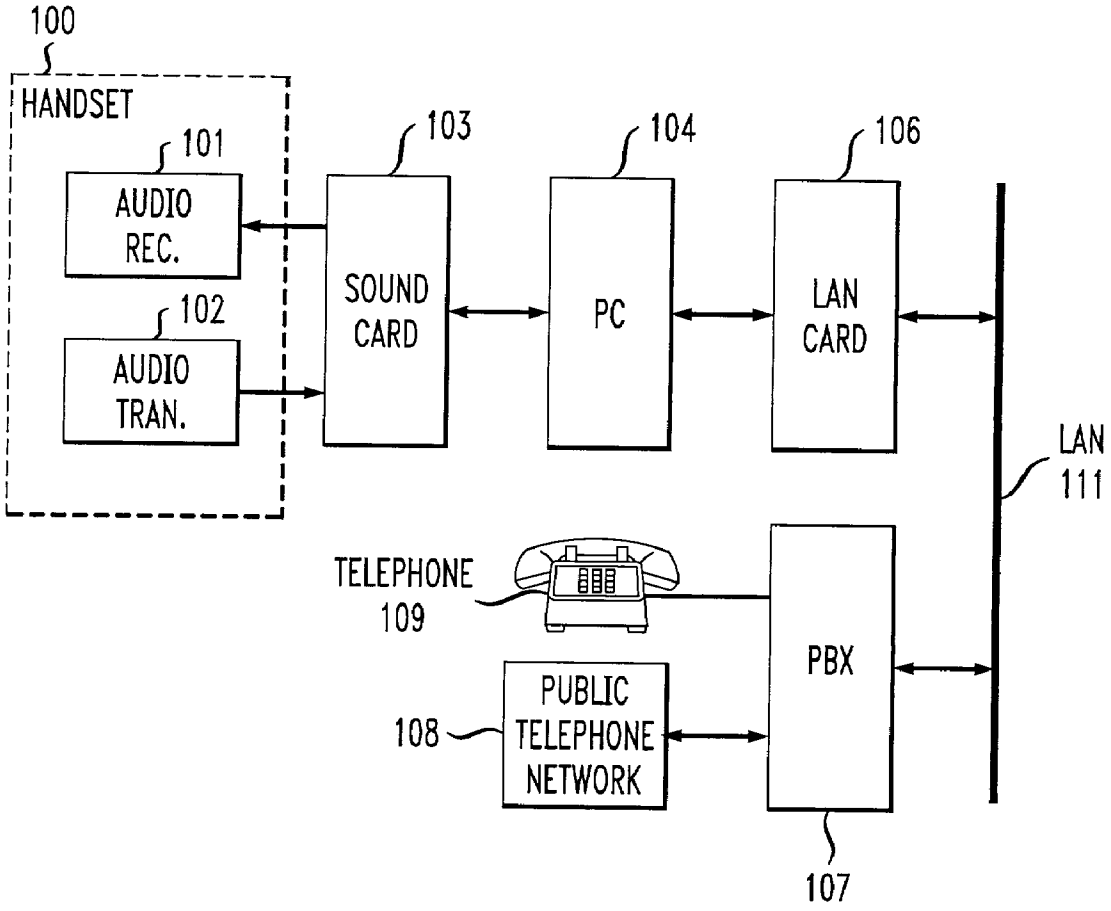


FIG. 1

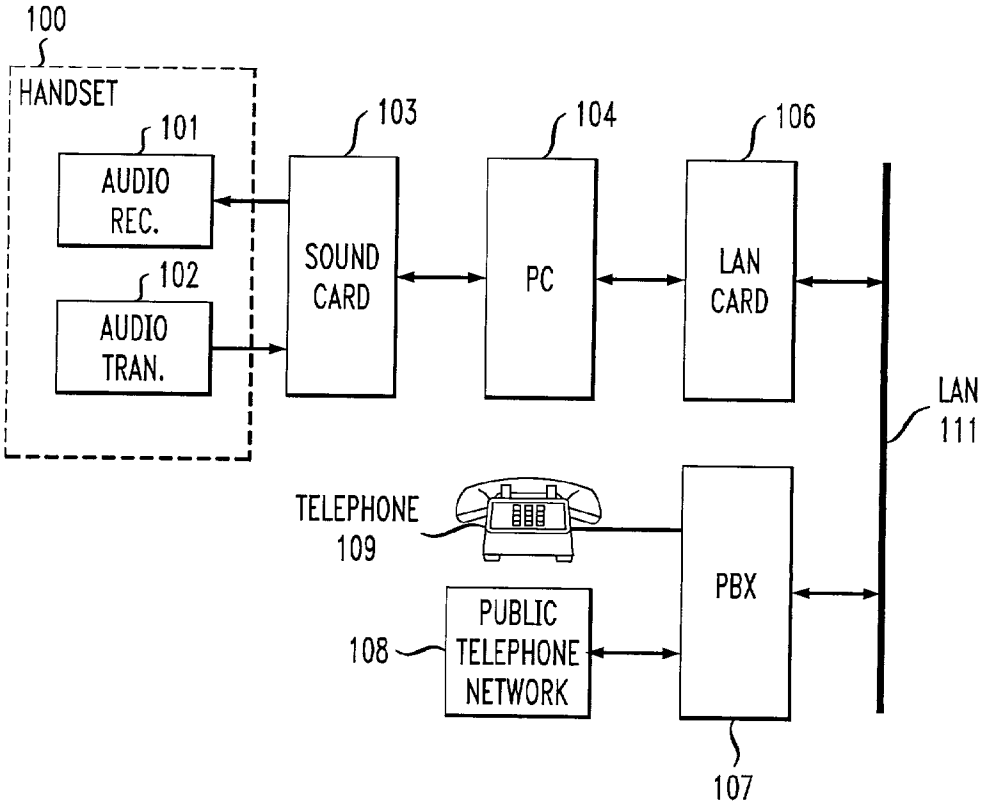

100
HANDSET
101
AUDIO REC.
102
AUDIO TRAN.
103
SOUND CARD
104
PC
106
LAN CARD
LAN 111
TELEPHONE 109
PBX
108
PUBLIC TELEPHONE NETWORK
107

FIG. 2

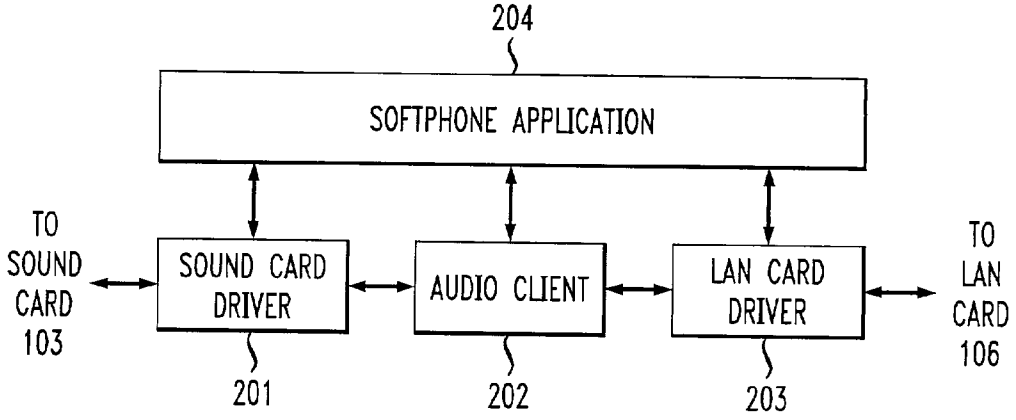

204
SOFTPHONE APPLICATION
TO SOUND CARD 103
SOUND CARD DRIVER
AUDIO CLIENT
LAN CARD DRIVER
TO LAN CARD 106
201
202
203

FIG. 3

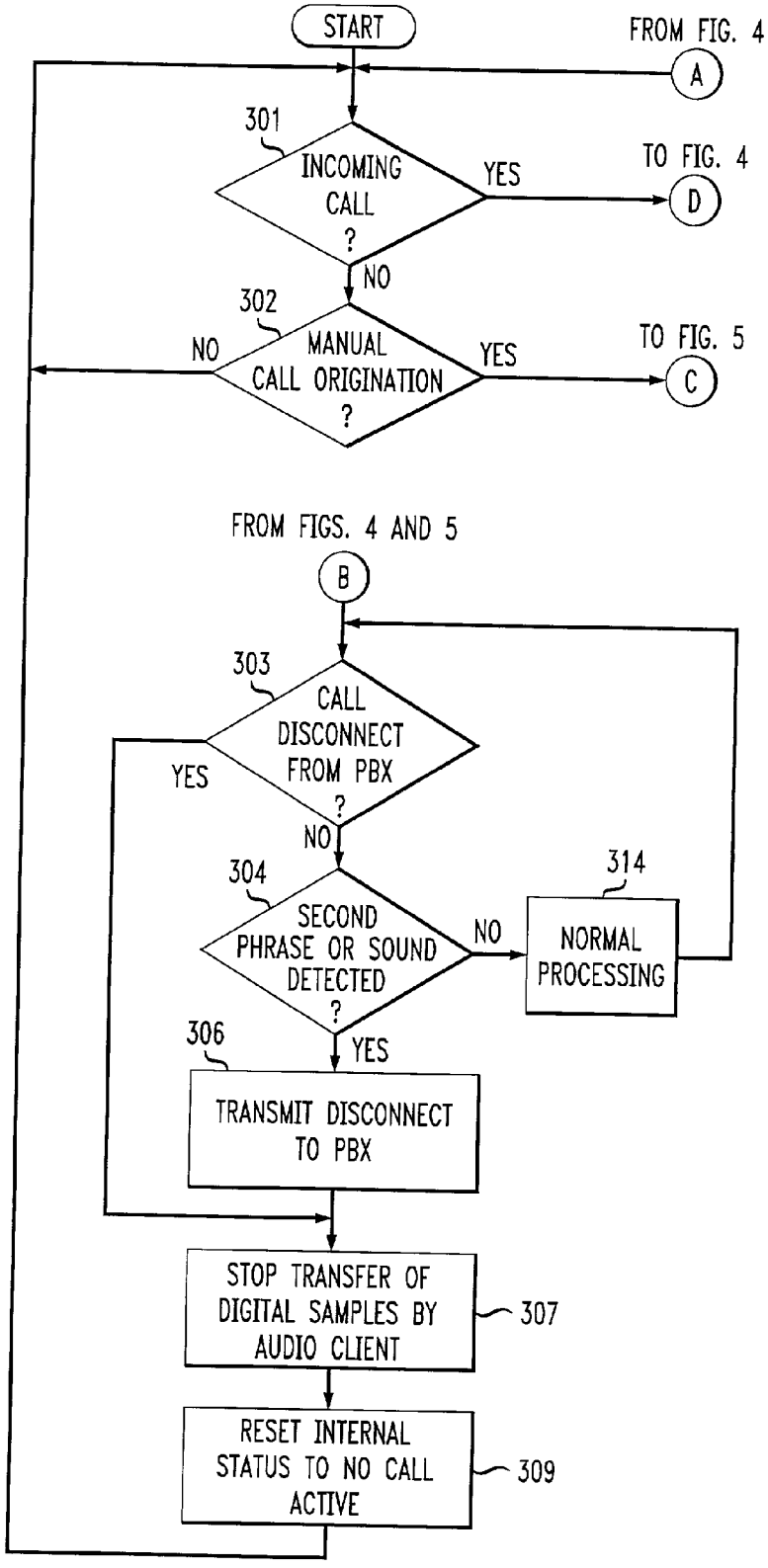

START
FROM FIG. 4
A
301
INCOMING CALL ?
YES
TO FIG. 4
D
NO
302
NO
MANUAL CALL ORIGINATION ?
YES
TO FIG. 5
C
FROM FIGS. 4 AND 5
B
303
CALL DISCONNECT FROM PBX ?
YES
NO
304
SECOND PHRASE OR SOUND DETECTED ?
NO
314
NORMAL PROCESSING
YES
306
TRANSMIT DISCONNECT TO PBX
STOP TRANSFER OF DIGITAL SAMPLES BY AUDIO CLIENT
307
RESET INTERNAL STATUS TO NO CALL ACTIVE
309

FIG. 4

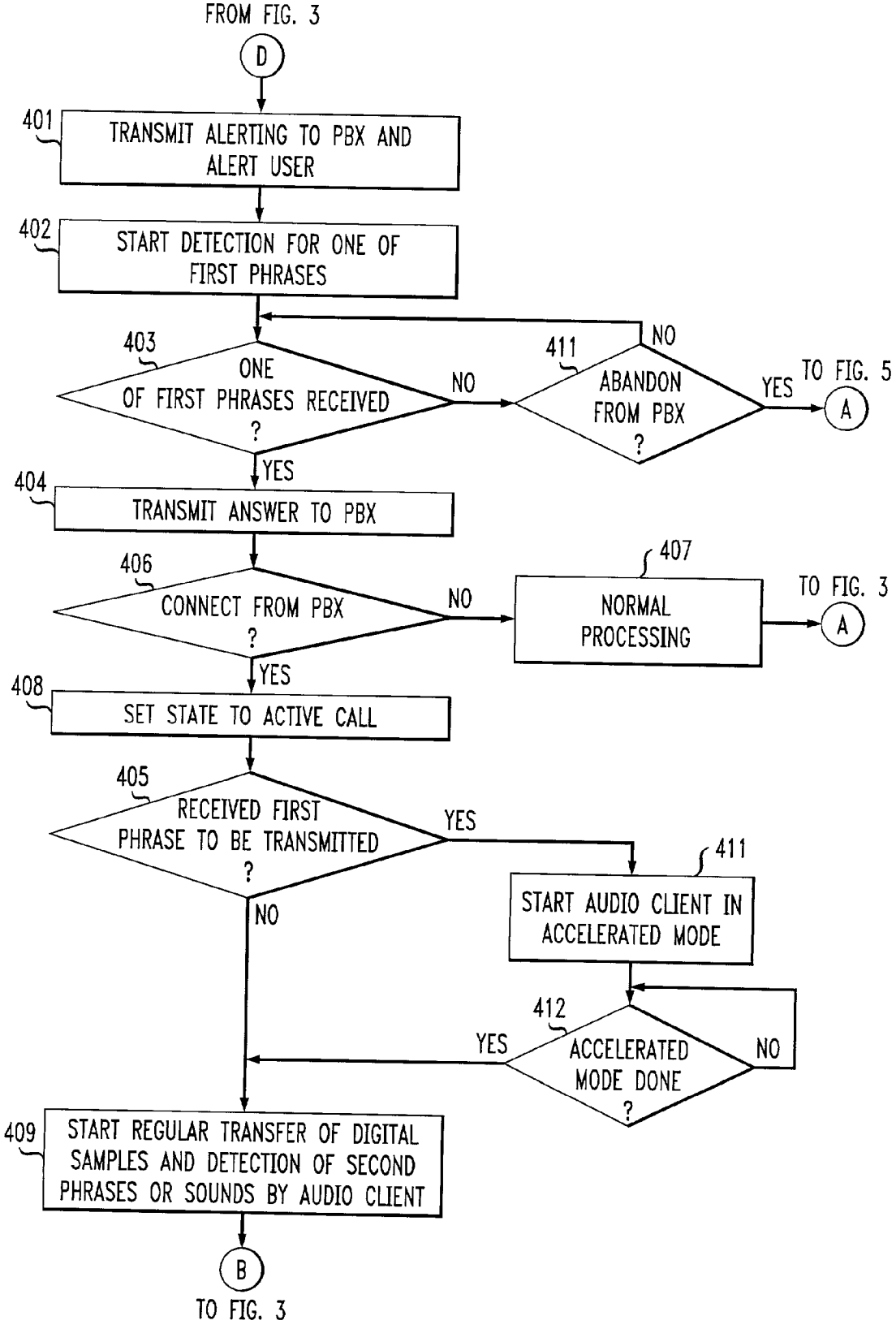

FROM FIG. 3
D
401
TRANSMIT ALERTING TO PBX AND ALERT USER
402
START DETECTION FOR ONE OF FIRST PHRASES
403
ONE OF FIRST PHRASES RECEIVED ?
NO
411
ABANDON FROM PBX ?
NO
YES
TO FIG. 5
A
YES
404
TRANSMIT ANSWER TO PBX
406
CONNECT FROM PBX ?
NO
407
NORMAL PROCESSING
TO FIG. 3
A
YES
408
SET STATE TO ACTIVE CALL
405
RECEIVED FIRST PHRASE TO BE TRANSMITTED ?
YES
411
START AUDIO CLIENT IN ACCELERATED MODE
NO
412
ACCELERATED MODE DONE ?
YES
NO
409
START REGULAR TRANSFER OF DIGITAL SAMPLES AND DETECTION OF SECOND PHRASES OR SOUNDS BY AUDIO CLIENT
B
TO FIG. 3

FIG. 5

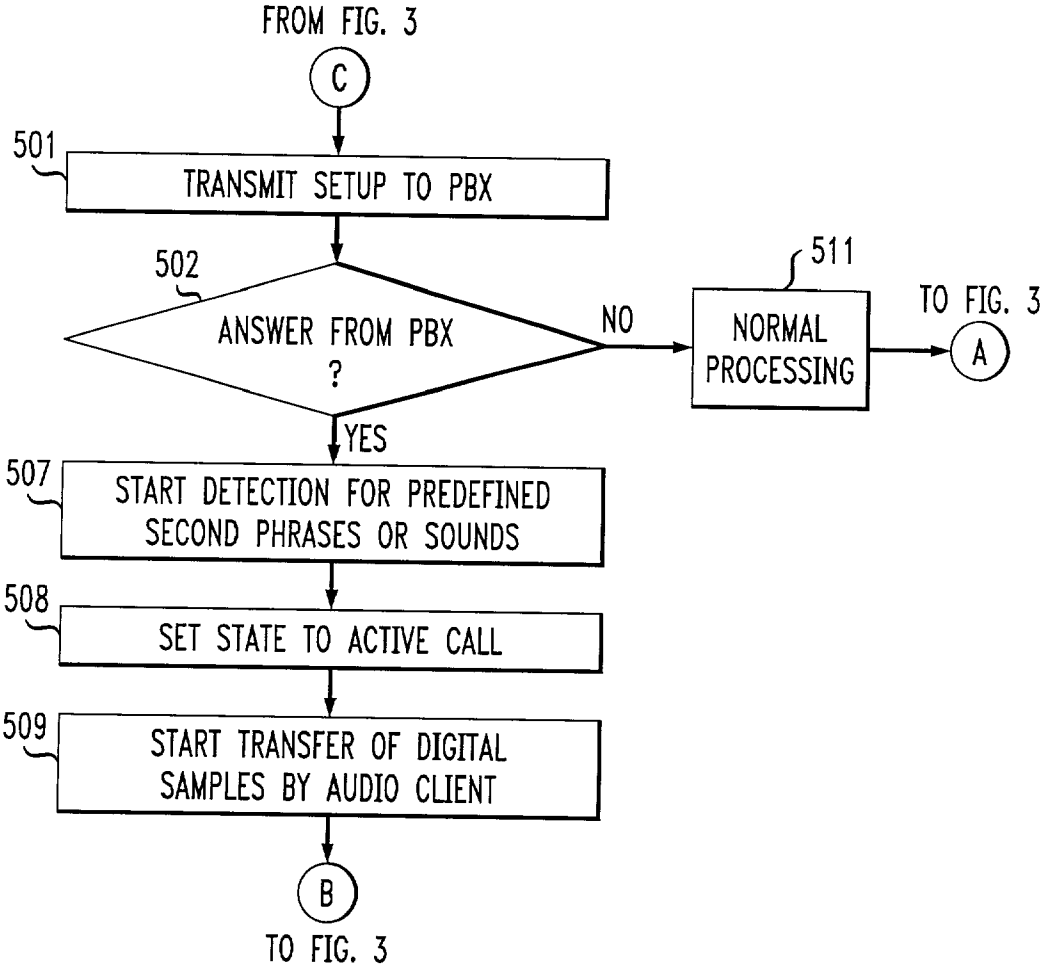

FROM FIG. 3
C
501
TRANSMIT SETUP TO PBX
502
ANSWER FROM PBX ?
NO
511
NORMAL PROCESSING
TO FIG. 3
A
YES
507
START DETECTION FOR PREDEFINED SECOND PHRASES OR SOUNDS
508
SET STATE TO ACTIVE CALL
509
START TRANSFER OF DIGITAL SAMPLES BY AUDIO CLIENT
B
TO FIG. 3

FIG. 6

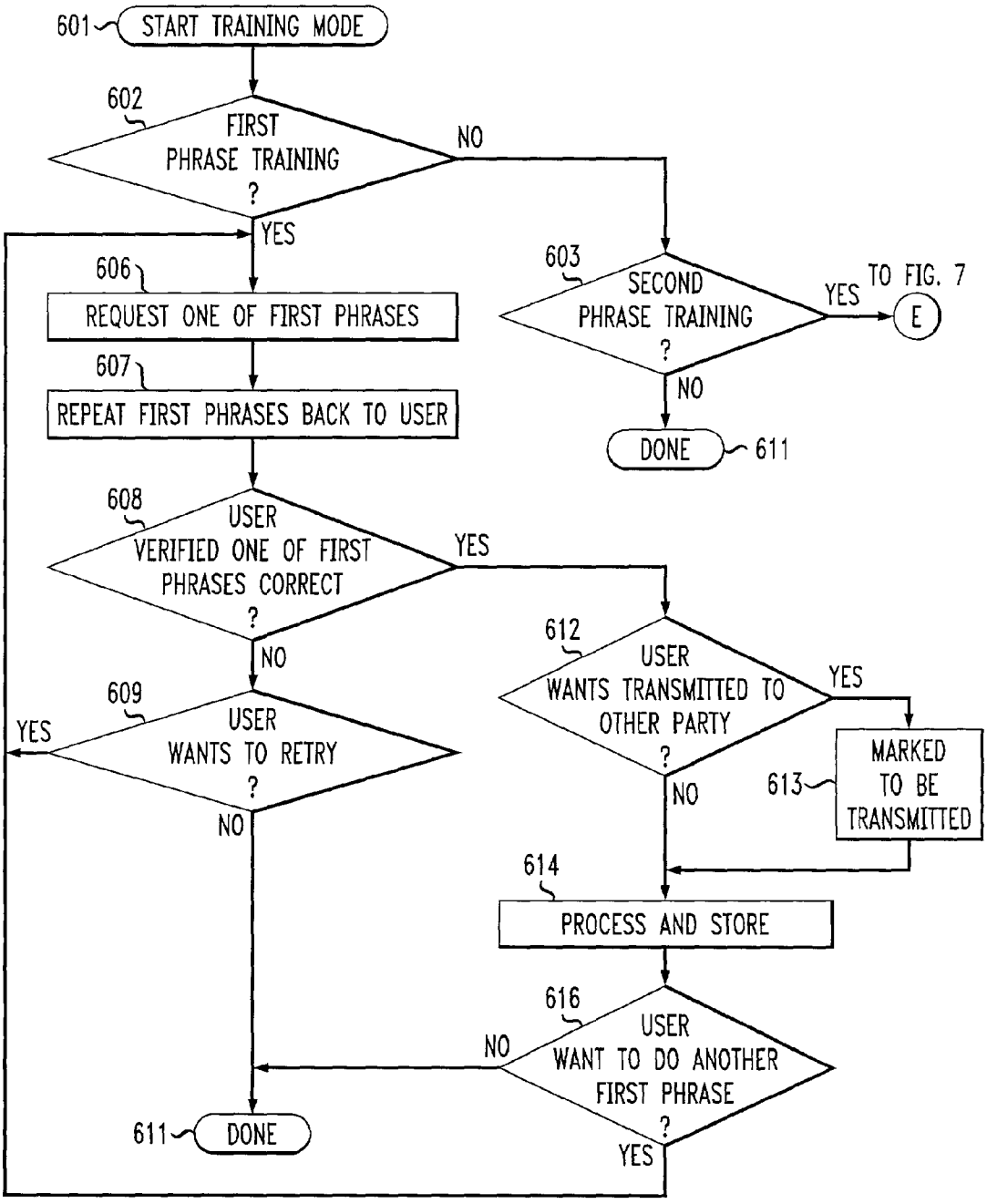
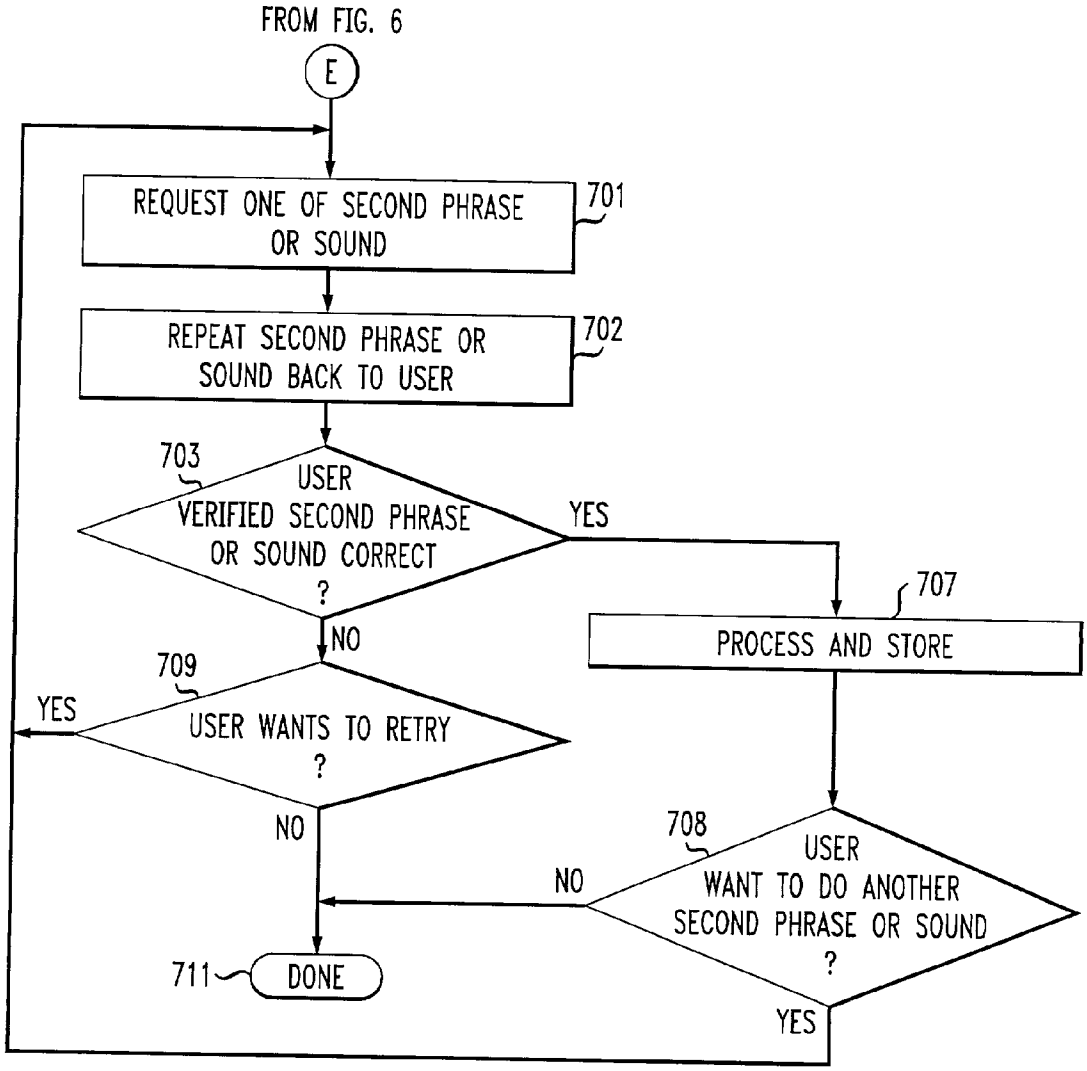

601
START TRAINING MODE
602
FIRST PHRASE TRAINING ?
NO
YES
603
SECOND PHRASE TRAINING ?
YES
TO FIG. 7
E
NO
DONE
611
606
REQUEST ONE OF FIRST PHRASES
607
REPEAT FIRST PHRASES BACK TO USER
608
USER VERIFIED ONE OF FIRST PHRASES CORRECT ?
YES
NO
609
USER WANTS TO RETRY ?
YES
NO
612
USER WANTS TRANSMITTED TO OTHER PARTY ?
YES
NO
613
MARKED TO BE TRANSMITTED
614
PROCESS AND STORE
616
USER WANT TO DO ANOTHER FIRST PHRASE ?
NO
YES
611
DONE

FIG. 7


FROM FIG. 6
E
REQUEST ONE OF SECOND PHRASE OR SOUND
701
REPEAT SECOND PHRASE OR SOUND BACK TO USER
702
703
USER VERIFIED SECOND PHRASE OR SOUND CORRECT ?
YES
NO
707
PROCESS AND STORE
709
USER WANTS TO RETRY ?
YES
NO
708
USER WANT TO DO ANOTHER SECOND PHRASE OR SOUND ?
NO
YES
711
DONE

SPEECH ANALYSIS BASED ANSWER DETECTION FOR IP BASED TELEPHONES

TECHNICAL FIELD

[0001] This invention relates to telephones in general and, in particular, two or more telephones communicating using an IP protocol.

BACKGROUND OF THE INVENTION

[0002] An IP softphone consists of a software package running on a personal computer (PC). An example of such an IP softphone is the Avaya IP Softphone manufactured by Avaya Inc. The software emulates a telephone and communicates signaling and voice information via a IP-network connection between the PC and a telecommunication switching system (also referred to as a PBX). This network connection is often the Internet or a LAN. Audio information being received from the network is communicated with the user via a headset or handset via a sound card that interfaces the headset or handset to the PC. When a user speaks, the sound card converts the analog information into digital information that the software then transmits to a destination via the network. When a call is received via the network for the IP softphone, the software alerts the user via an internal or external speaker attached to the PC. The user then answers the call by selecting the softphone software application and selecting a soft button labeled "answer". The user then conducts a conversation with the caller via either the handset or the headset. To place a call, the user once again selects the software and selects a soft button labeled "Number" and proceeds to dial the destination number using a keypad or a pre-stored list of telephone numbers. The keypad can by either a soft keypad or the PC's keyboard.

[0003] A common problem is that while a user is walking into their office, they then hear the IP softphone ringing. To answer the call, the user must unlock the screen blanker to access the PC, select the software application, and actuate the soft button labeled "answer". Then, the user can pick up the handset or the headset and talk. The problem is that a great amount of time is involved in performing these steps. Frequently, before the user has completed the steps, the call is transferred to coverage such as a voice mail system.

[0004] An apparently obvious solution would be to have a switch in the handset that was activated when the handset was picked up to answer the call. However, the vast majority of sound cards do not have a physical input that can sense a switch. One solution that has been utilized is in the Microsoft Netmeeting application. Microsoft Netmeeting attempts to solve this problem by having an option referred to as "call, automatically accept calls". Utilizing this option, a user need only start listening and talking in order to participate in the call. The disadvantage of this option is that a call is always answered whether or not the user is available or not. This leads to a great deal of frustration for callers if the user is not there. Further, at times, the called party may simply not want to answer the call at a particular time.

SUMMARY OF THE INVENTION

[0005] This invention is directed to solving these and other problems and disadvantages of the prior art. According to an embodiment of the invention, when a user speaks a predetermined phrase, a handset transmits this audio information to a personal computer. The personal computer is responsive to the predefined phrase to determine that the user of the handset has answered an incoming call. The predefined phrase is not transmitted to the calling party. The user of the handset trains the personal computer to recognize not only the predefined phrase but also the user's voice. Advantageously, this training prevents unauthorized use of the IP softphone. In addition, the personal computer advantageously is capable of utilizing user-independent predefined phrases such that any individual can use the IP softphone. The use of user-independent predefined phrases does allow unauthorized use of the IP softphone.

[0006] Advantageously, the first predefined phrase spoken by the user to answer an incoming call is subsequently transmitted to the calling party. The first phrase is initially delayed by the personal computer before it is transmitted to the calling party; however, the transmission of the first phrase to the calling party is accelerated so as not to delay the conversation. Advantageously, the user can utilize one of predefined audio segments to end a call. Predefined audio segments are second predefined phrases and predefined sounds. The personal computer is trained to recognize these second phrases or sounds. These second phrases can be user-independent predefined phrases such that any individual can use the IP softphone. Advantageously, a predefined sound can be that made when a headset or handset is returned to its stationary position.

[0007] These and other advantages and features of the present invention will become apparent from the following description of an illustrative embodiment of the invention taken together with the drawing.

BRIEF DESCRIPTION OF THE DRAWING

[0008] **FIG. 1** illustrates, in block diagram form, a system for utilizing the present invention;

[0009] **FIG. 2** illustrates, in block diagram form, a software structure for utilizing the present invention;

[0010] FIGS. **3-5** illustrate, in flowchart form, the steps performed by the software applications in implementing an embodiment of the invention; and

[0011] **FIGS. 6 and 7** illustrate, in flowchart form, the steps performed by the software applications during the training for the first and second predefined phrases.

DETAILED DESCRIPTION

[0012] **FIG. 1** illustrates a system for implementing the invention. In **FIG. 1**, PBX **107** is supplying telephone service for a user that is utilizing handset **100** via sound card **103**, personal computer (PC) **104**, LAN card **106** and LAN **111**. PBX **107** supplies access to the general public by being interconnected to public telephone network **108**. One skilled in the art would readily see that LAN card **106** and sound card **103** could be internal to PC **104**. PBX **107** can advantageously be an Avaya Definity® Business Communication Switching System. Whereas a PBX is illustrated in **FIG. 1**, one skilled in the art would readily envision that PBX **107** could be any of a variety of switching equipment. For example, PBX **107** could be a public switching telephone network (PSTN) gateway box in an H.323 calling system. PBX **107** supplies service for other users utilizing conventional telephones such as telephone **109**. In addition,

other handsets similar to handset **100** can be interconnected to LAN **111** by having individual sound cards, PCs and LAN cards. PC **104** receives control signaling and voice information from PBX **107**, and PC **104** in turn transmits voice information and control signaling information to PBX **107** via LAN card **106** and LAN **111**. Sound card **103** performs the function of converting audio information received from PC **104** to analog signals and transmitting these to audio receiver **101** that advantageously can be a simple speaker arranged in handset **100**. Sound card **103** similarly converts audio information from audio transmitter **102** (that advantageously can be a simple microphone) to digital information, which is transmitted to PC **104**. A software application in PC **104** (audio client **202**) provides the facilities for analyzing the speech of the user of the IP softphone to determine call acceptance and termination. The user of the IP softphone trains audio client **202** for these phrases. In addition, the user of the IP softphone can choose to use standard, user-independent predefined phrases. In utilizing the user-independent predefined phrases, the user does lose the ability to prevent unauthorized use of the IP softphone. In the second embodiment, audio client **202** performs the necessary acceleration of the transmission of the speech for the predefined phrase that accepts an incoming call. Other software applications in PC **104** emulate a telephone such as telephone **109** with respect to PBX **107** with the exception that PBX **107** is transmitting and receiving information via LAN **111** rather than a telephone link. Sound card **103**, PC **104**, and LAN card **106** are of a conventional design well-known to those skilled in the art.

[**0013**] Consider the following example of an embodiment of the invention. If telephone **109** places a call to handset **100**, PBX **107** transmits control signaling information to PC **104**. Softphone application **204** of **FIG. 2** of PC **104** then provides an audio alerting sound via an internal or external speaker of PC **104** or other apparatus well-known to those skilled in the art. In addition, softphone application **204** transmits control information to audio client **202** that causes audio client **202** to perform the speech analysis operation for first predefined phrases that will be received via audio transmitter **102** via sound card **103** from the user of handset **100**. Advantageously, a first predefined phrase could be "off hook". When the user of handset **100** speaks the first predefined phrase, audio client **202** detects the presence of the first predefined phrase and alerts softphone application **204** to this fact. In response, softphone application **204** transmits control information back to PBX **107** indicating that the call has been answered by handset **100**. The telephone conversation then takes place in a normal manner by the user of handset **100** hearing speech from the user of telephone **109** via audio receiver **101** and speaking to the user of telephone **109** via audio transmitter **102**. When softphone application **204** transmits control information back to PBX **107** indicating that the call has been disconnected, a manual disconnect operation by the user has been detected or a predefined second phrase or sound. This fact is transmitted as control information to PBX **107** by softphone application **204** indicating that the call has been disconnected, and PBX **107** responds in a normal manner. The user performs a manual disconnect operation by selecting a soft button labeled "disconnect".

[**0014**] **FIG. 2** illustrates, in block diagram form, the structure of the software within PC **104** utilized to provide the functions for handset **100**. The messages transmitted among the software elements **201-204** are conveyed via a standard operating system not illustrated in **FIG. 2**. Drivers **201** and **203** provide the standard software interface to sound card **103** and LAN card **106**, respectively. Such drivers are well-known to those skilled in the art. Softphone application **204** provides the overall control not only of the communication of calls with PBX **107** but also control over the functions of audio client **202**. When the control information is received from PBX **107** indicating that there is an incoming call, this information is relayed from LAN card driver **203** to softphone application **204**. Softphone application **204** then initiates sound card driver **201** to produce a ringing sound on the internal or external speaker of PC **104**. In addition, softphone application **204** requests that sound card driver **201** start converting audio information from handset **100** to digital information and that audio client **202** start the detection for the first predefined phrase. When audio client **202** detects the first predefined phrase from sound card **103**, it transmits control information to softphone application **204** indicating this fact. Softphone application **204** is responsive to the notification from audio client **202** that it has detected the predefined phrase to transmit a control message to PBX **107** via LAN card driver **203** indicating that the call has been answered. Softphone application **204** then interacts with PBX **107** to exchange the necessary control information so that the call can be completed. Softphone application **204** controls audio client **202** so that audio client **202** is responsive to digital samples received from LAN card driver **203** to relay these to sound card driver **201** and to relay digital samples received from sound card driver **201** to LAN card driver **203**. When the predefined second phrase or sound from sound card **103** is detected by audio client **202**, it signals this fact to softphone application **204**. In response, softphone application **204** transmits a disconnect message to PBX **107** to cause the call to be terminated. If the user of telephone **109** hangs up on the call first, PBX **107** sends a disconnect message to softphone application **204** that resets the call state of the soft phone with respect to the software elements **201-204**.

[**0015**] FIGS. **3-5** illustrate the operations performed by softphone application **204** in implementing an embodiment of the invention. Upon being started, decision block **301** checks to see if an incoming call is being received from PBX **107**. If the answer is yes, control is transferred to block **401** of **FIG. 4**. The latter block transmits an alerting message back to PBX **107** and alerts the user of the softphone. Block **402** instructs audio client **202** to start the speech analysis to detect one of a plurality of first phrases that can be utilized by the user to answer the incoming call. Decision block **403** determines if one of the first phrases has been received by audio client **202**. If the answer is no, decision block **411** determines if the PBX has abandon the call. If the answer is no, control is transferred back to decision block **403**. If the answer in decision block **411** is yes, control is transferred back to decision block **301** of **FIG. 3**. If the answer is yes in decision block **403**, block **404** transmits an answer message to PBX **107**.

[**0016**] After execution of block **404**, control is transferred to decision block **406** that determines whether or not a connect message has been received back from the PBX. If the answer is no, block **407** performs normal processing before returning control back to decision block **301** of **FIG. 3**. If a connect message has been received from PBX **107**, block **408** sets the state of softphone application **204** and

audio client **202** to the active call state before transferring control to decision block **405**. Decision block **405** determines if the received first phrase that answered the incoming call is one that is to be transmitted to the calling party. If the answer is yes, block **411** instructs audio client **202** to enter the accelerated mode to transmit the received first phrase and any buffered subsequent speech to the calling party. In response, audio client application **202** starts to transmit the first phrase and subsequent speech from the user faster than the speech is being received by audio client application **202**. This is done by using well known techniques such as eliminating a portion of silence interval between words or time domain harmonic scaling or other techniques known to those skilled in the art.

[**0017**] When all of the buffered speech has been transmitted, audio client application **202** signals softphone application **204**. Decision block **412** determines when all of the received first phrase and buffered subsequent speech has been transmitted to the calling party. If the answer is no, decision block **412** is executed again. If the answer is yes, control is transferred to block **409**. Block **409** instructs audio client **202** to start the regular transfer of digital samples to the calling party and to begin the speech analysis for the detection of the predefined second phrases or sounds so as to detect when the user wishes to end the incoming call before control is transferred back to decision block **303** of **FIG. 3**.

[**0018**] Returning to decision block **301**, if the answer in decision block **301** is no indicating that an incoming call is not being received from PBX **107**, decision block **302** determines if the user has indicated a manual call origination utilizing a softkey. If the answer is yes in decision block **302**, this indicates that the user wishes originate a call and control is transferred to block **501** of **FIG. 5**. The latter block transmits a setup message to the PBX **107**, and decision block **502** determines if an answer message is received back from PBX **107**. If the answer is no in decision block **502**, block **511** performs normal processing before transferring control back to decision block **301** of **FIG. 3**. If the answer in decision block **502** indicates that a call has been established, block **507** starts the detection by audio client **202** for detecting predefined second phrases or sounds, and block **508** sets the call state to active call. Finally, block **509** starts the transfer of digital samples by audio client **202** directly from audio transmitter **102** to PBX **107** before transferring control to decision block **301** of **FIG. 3**.

[**0019**] Returning to decision block **303** of **FIG. 3**, decision blocks **303** and **304** determine if a disconnect has been received from PBX **107** or if the user has disconnected, respectively. If a disconnect has been received from PBX **107**, decision block **303** transfers control to block **307**. If the answer is no in decision block **303**, decision block **304** determines if the user has spoken one of the second phrases. If the answer is yes, control is transferred to block **306** that transmits a disconnect to PBX **107** before transferring control to block **307**. Block **307** stops the transfer of digital samples by audio client **202**. Finally, block **309** resets the internal status to no-call-active before transferring control back to decision block **301**. If the answer in decision block **304** is no, control is transferred to block **314** that performs normal processing before returning control back to decision block **303**.

[**0020**] **FIGS. 6 and 7** illustrate, in flowchart form, the steps performed to train audio client application **202** in the second embodiment for a plurality of first and second phrases. Once being started in block **601**, decision block **602** determines if the training should be for first phrases. If the answer is no, control is transferred to decision block **603** which determines if training is to be done for second phrases. If the answer is yes, control is transferred to decision block **701** of **FIG. 7**. If the answer in block **603** is no, control is transferred to block **604**, and the operations are terminated. Returning to decision block **602**, if the answer is yes, control is transferred to block **606** which requested that the user speak one of the first phrases. Block **607** then repeats that one of the first phrases back to the user, and decision block **608** verifies that this is what the user meant to say. If it is the phrase that the user meant to say, control is transferred to decision block **612** that determines if the user wants this phrase transmitted to the other party. If the answer is yes, control is transferred to block **613** that marks the phrase as one to be transmitted to the other party before control is transferred to block **614**. If the answer is no in block **612**, control is transferred to block **614** that processes and stores the phrase. Decision block **616** then determines if the user wants to enter another first phrase. If the answer is no, control is transferred to block **611**, and the operations are done. If the answer is yes in decision block **616**, control is transferred back to decision block **606**.

[**0021**] Returning to decision block **608**, if the user indicates that the repeated phrase is not correct, control is transferred to decision block **609** that determines if the user wants to retry. If the answer is yes, control is transferred back to decision block **606**. If the answer is no, control is transferred to block **611**.

[**0022**] Returning to decision block **602**, if the answer is no, control is transferred to block **701** of **FIG. 7** which requested that the user speak one of the second phrases or create one of the predefined sounds. Block **702** then repeats that one of the second phrases back to the user, and decision block **703** verifies that this is what the user meant to say. If it is the phrase that the user meant to say, block **707** processes and stores the phrase or sound. Decision block **708** then determines if the user wants to enter another second phrase or predefined sound. If the answer is no, control is transferred to block **711**, and the operations are done. If the answer is yes in decision block **708**, control is transferred back to decision block **701**.

[**0023**] Returning to decision block **703**, if the user indicates that the repeated phrase or sound is not correct, control is transferred to decision block **709** that determines if the user wants to retry. If the answer is yes, control is transferred back to decision block **701**. If the answer is no, control is transferred to block **711**.

[**0024**] Of course, various changes and modifications to the illustrated embodiments described above will be apparent to those skilled in the art. Such changes and modifications can be made without departing from the spirit and scope of the invention and without diminishing its intended advantages. It is therefore, intended that such changes and modifications be covered by the following claims except insofar as limited by the prior art.

What is claimed is:

1. A method of determining call answer by an IP softphone that is implemented by a personal computer and including an audio transmitter, comprising the steps of:

receiving an incoming call from a telecommunication switching system;

detecting one of a plurality of first predefined speech phrases from the audio transmitter and spoken by a user of the IP softphone; and

accepting the incoming call from the telecommunication switching system in response to the detection of the one of the plurality of first predefined speech phrases.

2. The method of claim 1 further comprises the step of transmitting the one of the plurality of first predefined speech phrases to a calling party of the incoming call.

3. The method of claim 2 wherein the step of transmitting comprises the steps of buffering the one of the plurality of first predefined speech phrases and a portion of subsequent speech as buffered information; and

accelerating transmission of the buffered information.

4. The method of claim 3 wherein the step of accelerating comprises eliminating a portion of silence intervals between words.

5. The method of claim 3 wherein the step of accelerating comprises using time domain harmonic scaling.

6. The method of claim 1 wherein the one of the plurality of first predefined speech phrases is user-independent.

7. The method of claim 1 wherein the one of the plurality of first predefined speech phrases is user-dependent.

8. The method of claim 1 further comprises the steps of detecting one of a plurality of predefined audio information segments from the audio transmitter; and

disconnecting the incoming call from the telecommunication switching system in response to the detection of the one of the plurality of predefined audio information segments.

9. The method of claim 8 wherein the one of the plurality of predefined audio information segments is an user-independent predefined speech phrase.

10. The method of claim 8 wherein the one of the plurality of predefined audio information segments is an user-dependent second predefined speech phrase.

11. The method of claim 8 wherein the one of the plurality predefined audio information segments is a sound of a handset being placed in a stationary position.

12. The method of claim 8 wherein the one of the plurality predefined audio information segments is a sound of a headset being placed in a stationary position.

13. The method of claim 2 wherein the one of the plurality of first predefined speech phrases is user-independent.

14. The method of claim 2 wherein the one of the plurality of first predefined speech phrases is user-dependent.

15. The method of claim 2 further comprises the steps of detecting one of a plurality of predefined audio information segments from the audio transmitter; and

disconnecting the incoming call from the telecommunication switching system in response to the detection of the one of the plurality of predefined audio information segments.

16. The method of claim 15 wherein the one of the plurality of predefined audio information segments is an user-independent predefined speech phrase.

17. The method of claim 15 wherein the one of the plurality of predefined audio information segments is an user-dependent second predefined speech phrase.

18. The method of claim 15 wherein the one of the plurality predefined audio information segments is a sound of a handset being placed in a stationary position.

19. The method of claim 15 wherein the one of the plurality predefined audio information segments is a sound of a headset being placed in a stationary position.

20. A method of determining call answer by an IP softphone that is implemented by a softphone application and an audio client application executing on a personal computer and including an audio transmitter, comprising the steps of:

receiving an incoming call from a telecommunication switching system;

detecting a first predefined speech phrase from the audio transmitter by the audio client application and spoken by a user of the IP softphone;

accepting the incoming call from the telecommunication switching system by the softphone application in response to the detection of the first predefined speech phrase;

buffering the first predefined speech phrase and a portion of subsequent speech as buffered information;

transmitting the buffered information to a calling party by accelerating transmission of the buffered information;

detecting a predefined audio information segment from the audio transmitter by the audio client application; and

disconnecting the incoming call from the telecommunication switching system by the softphone application in response to the detection of the predefined audio information segment.

21. The method of claim 20 wherein the step of accelerating comprises eliminating a portion of silence intervals between words.

22. The method of claim 20 wherein the step of accelerating comprises using time domain harmonic scaling.

23. The method of claim 20 wherein the first predefined speech phrase is user-independent.

24. The method of claim 20 wherein the first predefined speech phrase is user-dependent.

25. The method of claim 20 wherein the predefined audio information segment is an user-independent predefined speech phrase.

26. The method of claim 20 wherein the predefined audio information segment is an user-dependent second predefined speech phrase.

27. The method of claim 20 wherein the predefined audio information segment is a sound of a handset being placed in a stationary position.

28. The method of claim 20 wherein the predefined audio information segment is a sound of a headset being placed in a stationary position.

29. A method of determining call answer by an IP softphone that is implemented by a softphone application and an audio client application executing on a personal computer and including an audio transmitter, comprising the steps of:

receiving an incoming call from a telecommunication switching system;

detecting a first predefined speech phrase from the audio transmitter by the audio client application and spoken by a user of the IP softphone;

accepting the incoming call from the telecommunication switching system by the softphone application in response to the detection of the first predefined speech phrase;

detecting a predefined audio information segment from the audio transmitter by the audio client application; and

disconnecting the incoming call from the telecommunication switching system by the softphone application in response to the detection of the predefined audio information segment.

30. The method of claim 29 wherein the first predefined speech phrase is user-independent.

31. The method of claim 29 wherein the first predefined speech phrase is user-dependent.

32. The method of claim 29 wherein the predefined audio information segment is an user-independent predefined speech phrase.

33. The method of claim 29 wherein the predefined audio information segment is an user-dependent second predefined speech phrase.

34. The method of claim 29 wherein the predefined audio information segment is a sound of a handset being placed in a stationary position.

35. The method of claim 29 wherein the predefined audio information segment is a sound of a headset being placed in a stationary position.

36. Apparatus for implementing the steps of claim 1.

37. Apparatus for implementing the steps of claim 2.

38. Apparatus for implementing the steps of claim 8.

39. Apparatus for implementing the steps of claim **15**.

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