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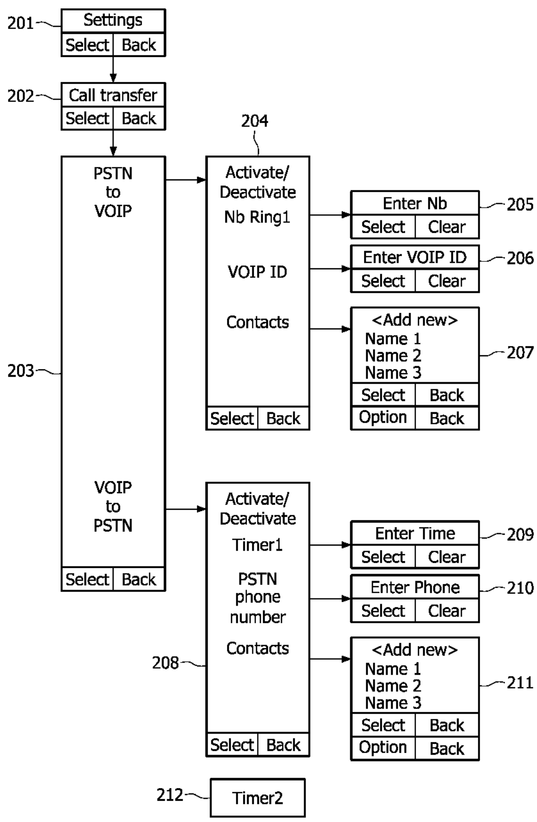

FIG. 2

(57) Abstract: The present invention relates especially to a DECT cordless dual mode phone i.e. that can establish calls between the PSTN (13) and packet networks (10) that comprises a base station (2) and a handset (1) that can perform call forwarding of VoIP calls to the PSTN and vice versa. The base station has inputs for the land line and packet network as well as means for interconnecting both inputs such that calls can be forwarded. Call forwarding features include standard functionality such as setting the time that the telephone rings before forwarding the call (call forward no answer) to another destination including voicemail in the base station or alternatively in the network.

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CALL FORWARDING PHONE CALL BETWEEN A PUBLIC SWITCHED NETWORK AND AN INTERNET PROTOCOL NETWORK

FIELD OF THE INVENTION

The invention relates to a dual mode phone comprising a first connection unit for connecting the dual mode phone to the public switched network, and a second connection unit for connecting the dual mode phone to the Internet Protocol network.

The invention also relates to a method of transferring a phone call from a dual mode phone to another phone, said dual mode phone device being adapted to be connected to the public switched network and to the Internet protocol network.

BACKGROUND OF THE INVENTION

Voice Over Internet Protocol VOIP emerging technology gives to end-users the advantage to perform phone calls at very low rate.

Conventional phones implementing VOIP technology are usually software applications running on a personal computer PC, which applications use peripheral capabilities of the PC such as a sound card and a broadband modem. These software applications are called softphones.

It is usually possible to connect a cordless phone to the PC, for example via a Universal Serial Bus USB wired connection. Such a cordless phone comprises a base station and a handset for wireless communication with the base station. The base station is connected to the Internet protocol network through the PC via the USB wired connection and may also be connected to the Public Switched Telephony Network PSTN.

A dual mode PSTN/VOIP cordless phone can usually not transfer an incoming call from a PSTN/VOIP network to the VOIP/PSTN network.

In order to solve such a problem, US patent number 6724755 discloses an automatic call transfer activation and deactivation method, which is linked with the call transfer service managed by the network. However, such an implementation is only suitable if the VOIP service provider is the same as the PSTN service provider. Moreover, this implementation is not free for the end-user (he needs to pay a subscription fee to the service provider) and such an implementation does not permit to interrupt the call transfer.

SUMMARY OF THE INVENTION

It is an object of the invention to propose a dual mode phone, which operation is more convenient to the end-user than the one of the prior art.

To this end, there is provided a dual mode phone comprising:

- a first connection unit for connecting the dual mode phone to the public switched network;
- a second connection unit for connecting the dual mode phone to the Internet Protocol network, and which is coupled to the first connection unit;
- a memory for storing a phone identifier and a Internet protocol identifier; and
- call transfer management means for transferring an incoming call received by the second connection unit from the Internet protocol network to a phone having the phone identifier through the first connection unit and the public switched network, or for transferring an incoming call received by the first connection unit from the public switched network to a phone having the Internet protocol identifier through the second connection unit and the Internet protocol network.

As a consequence, the method in accordance with the invention makes it possible to transfer an incoming call from a PSTN/VOIP network to the VOIP/PSTN network whatever the VOIP network provider and the PSTN network provider are, the call transfer being directly managed by the dual mode phone.

According to an embodiment of the invention, the dual mode phone comprises a timer for allowing the dual mode phone user to pick up the incoming call during a first predetermined time limit, the call transfer management means initiating the call transfer when said first predetermined time limit has expired.

According to another embodiment of the invention, the dual mode phone comprises means for automatically dialing the phone identifier or the Internet protocol identifier.

According to another embodiment of the invention, the dual mode phone comprises means for identifying a caller, the memory being adapted to store a list of callers allowed to be transferred, the call transfer management means being adapted to transfer the incoming call only if the caller is on the list of callers allowed to be transferred.

Still according to an embodiment of the invention, the dual mode phone comprises conversion means for converting audio data coded with a codec for the internet protocol network into audio data coded for the public switched network.

The invention also relates to a method of transferring a phone call from a dual mode phone to another phone, said dual mode phone device being adapted to be connected to the public switched network and to the Internet protocol network and comprising a memory for storing a phone identifier and an Internet protocol identifier, said method comprising the steps of transferring an incoming call received by the dual mode phone from the Internet protocol network to a phone having the phone identifier through the public switched network, or transferring an incoming call received by the dual mode phone from the public switched network to a phone having the Internet protocol identifier through the Internet protocol network.

The invention also extends to a computer program product comprising a set of instructions which, when loaded in a memory of the dual mode phone, makes the dual mode phone execute all the steps of the method in accordance with the invention.

These and other aspects of the invention will be apparent from and will be elucidated with reference to the embodiments described hereinafter.

BRIEF DESCRIPTION OF THE DRAWINGS

The present invention will now be described in more detail, by way of example, with reference to the accompanying drawings, wherein:

- Fig. 1 shows a communication system in accordance with an embodiment of the invention;
- Fig. 2 shows the call transfer configuration menu structure.
- Fig. 3 shows a block diagram of the method of transferring a call from the PSTN to the VOIP network with the corresponding message flow in accordance with an embodiment of the invention;
- Fig. 4 shows a block diagram of the method of transferring a call from the PSTN to the VOIP network in accordance with an embodiment of the invention; and
- Fig. 5 shows a block diagram of the method of transferring a call from the VOIP to the PSTN network in accordance with an embodiment of the invention.

DETAILED DESCRIPTION OF THE INVENTION

The invention relates to a dual phone which can be connected to Public Switched Telephony Network PSTN, such a phone operating for example in this case according to the DECT (for Digital Enhanced Cordless Telephone) standard or a similar standard, and which can also be connected to the Internet Protocol IP network, such a phone operating in this case according to the Voice Over Internet Protocol VOIP technology. In the latter case, the user can receive an incoming IP call or transmit an outgoing IP call either from the handset or from the computer.

Said phone comprises a base station and a handset. The dual mode phone is connected either to a personal computer PC incorporating a modem or to a residential gateway (a modem or a modem router). The link between the base station and the computer or residential gateway can be wired or wireless. The phone may include a wireless handset or a wired handset connected to the base station. The base station is connected to the PSTN network.

A communication system in accordance with an embodiment of the invention is depicted in Fig. 1. Said embodiment corresponds to the case of a dual mode PSTN/VOIP PC less cordless phone. Such a communication system comprises:

- a cordless phone comprising a handset 1 and a base station 2 able to be connected to a PSTN network 13 in order to communicate with another user having a PSTN phone 14;
- a residential gateway 4 able to be connected to an IP network 10 in order to communicate with another user having a VOIP phone, for example a PC 12 connected to the IP network via another residential gateway 11;
- a wireless or wired connection 3, for example of the USB type, between the residential gateway 4 and the cordless phone;
- a collection of software applications running on the cordless phone, such software applications including a softphone such as Skype or Voipbuster software.

Figure 2 shows the call transfer configuration menu structure.

The user of the dual mode cordless phone has to perform a configuration setup via a menu using the keyboard of the cordless phone or directly on the base station, for example through a web page. This configuration setup contains information that will be used for the management of the conditional call transfer.

As a first step, the end user has to:

- connect the dual phone to the PSTN network with a RJ11 plug.
- connect the dual phone to the IP network through a modem or router with RJ45 plug.

Next step corresponds to the configuration of the conditional call transfer.

According to this configuration, the end user is able to activate or deactivate the call transfer “PSTN to VOIP” and “VOIP to PSTN” through a menu tree available in the handsets or in the base station.

To this end, the end user select the menu item “Settings” (201), then the menu item “Call transfer” (202). Subsequently, the end-user is able to set parameters for a call transfer “PSTN to VOIP” or “VOIP to PSTN” (203). Both call transfers can be activated in parallel.

If the end user selects “PSTN to VOIP” (204), then the user needs to set the “Nb Ring1” (205) parameter (e.g. via a timer duration or via a ringing counter) which defines the number of ringing needed to start the call transfer procedure in order to allow the end user of the dual mode phone to answer the call. The number of ringing must be lower than the number of ringing allowed for the launching of the PSTN voice mail service or local answering machine. If not, a warning message is displayed.

The end user also has to enter the VOIP ID identifier (206) to which the incoming PSTN call will be transferred (for example it could be a name like “Paul”, an IP address like sip:paul@192.168.1.20 or a number like “102”). The VOIP ID can also be a VOIP voice mail. Please also note that the VOIP ID identifier can be the one of the dual phone.

A first list of contacts (Name 1, Name 2, Name 3) indicates the callers allowed to be transferred. The callers not allowed to be transferred are redirected to the answering machine of the dual phone or to the voice mail service, if available. The list of contacts is stored in a table located in the memory (for example a flash memory, an Electrically-Erasable Programmable Read-Only Memory EEPROM …) of the dual mode phone. Said table is accessible to the end user by a dedicated menu item (207). It is possible to add (Add new) or remove a contact. The contact list contains at least:

- information about the PSTN caller identification.(e.g. incoming call number or name of the caller)
- a timer duration between the call transfer order and the missed call transfer. If the transfer fails (e.g. the distant caller is not available) then the call is redirected to the local answering machine.

If the end user selects “VOIP to PSTN” (208), then the end user of the dual phone needs to set the “Timer1” (209) parameter which defines the timer (in second) to reach before starting the call transfer procedure. The duration defines the time allowed for answering the call. “Timer1” has to be lower than the timer used for the VOIP voice mail. If not, a warning message is displayed.

The end user also has to enter the PSTN phone number (210), e.g. a number like 02.44.02.02.02 or a name from the phonebook, to which the incoming VOIP call will be transferred.

Finally, the end user has to select in a second list of contacts containing the callers allowed to be transferred. The list of contacts is stored in a table located in the memory of the dual mode phone, as described above. The contact list contains at least information about the PSTN caller identification, e.g. incoming call number or name of the caller.

According to an embodiment of the invention, the dual phone implements the Caller Identification Detection (CID) for both PSTN and VOIP networks in order to offer the conditional call transfer. In this case, the end user has to subscribe to the CID service of his telephone network provider. Otherwise, all incoming calls are transferred without any filtering.

The first and second lists are stored in the base station as a data structure with at least a name and at least a corresponding phone number (for PSTN to VOIP call transfer) and a VOIP ID (for VOIP to PSTN call transfer). The two lists are accessible from each handset subscribed on the base station.

When the CID is supported by the phone and an end user has subscribed to this service, the CID is sent to the final callee (for both VOIP and PSTN). That means the CID displayed on the call receiving device is the one of the caller initiator.

The configuration menu comprises a timer duration “Timer2” (212) between the call transfer order and the missed call transfer statement. If the call transfer fails (e.g. the final callee is not available) then the call is redirected to the answering machine of the dual mode phone if available or to the VOIP voice mailbox available on the network. This timer can be set by default by the manufacturer of the phone. Alternatively, the end-user can be allowed to set the value of the timer.

Figure 3 shows a block diagram of the method of transferring a call from the PSTN to the VOIP network with the corresponding message flow.

When a user A calls via the PSTN network a user B who is the owner of the dual mode Phone, the incoming call is redirected by the dual mode phone to a VOIP subscriber C according to the configuration of the dual mode phone previously set by the user B, as it will be explained hereinafter. Note that the call can be re-directed over a plurality of VOIP subscribers.

The dual phone is connected both to the PSTN network (with a RJ11 plug) and to the VOIP network (with a RJ45 plug or a Wireless fidelity Wifi link). The PSTN to VOIP call transfer is activated and the parameters “VOIP ID” and “Nb Ring1” are set.

In a step 310, the dual phone B1 receives an incoming PSTN call (the ringing signal is detected on the line interface) from a remote phone A1.

In a step 320, the dual phone B1 rings. Even if the call transfer is activated, the dual phone user might be able to pickup the dual phone before the call transfer starts or during the call transfer (in the latter case a conference call between remote phone A1, VOIP phone C1 and dual phone B1 is set up).

In a step 330, the dual phone software application checks if the CID received is allowed to be transferred. To this end, the software application compares the CID phone number with the phone numbers in the list of contacts that can be transferred.

In a step 340, the software application waits for the number of ringing “Nb Ring1” as set in the dual phone. If the number of ringing is greater than the value “Nb Ring1”, then the software application jumps to step 350.

In this step 350, the dual phone B1 answers the PSTN call automatically.

In a step 360 the dual phone B1 is in the state “call in progress”. Similarly, in a step 370, the remote PSTN phone A1 is in the state “call in progress”.

In a step 380, the dual phone B1 sends a notification to the remote phone A1 in order to indicate to the remote phone user that the call is going to be transferred.

Subsequently, in a step 390, the software application of the dual phone B1 automatically dials the identifier VOIP ID of the VOIP phone as set by the dual phone user. If the CID of the incoming call is available, the software application forwards this CID to the VOIP phone C1 and fills in the VOIP header message identified by “From” with the incoming call CID (for example, the document SIP RFC 3261 describes the following SIP message: From: Sebastien<sip:100@reg.com>).

Then, during a step 400, the software application of the dual phone B1 plays music and starts the timer “Timer2”.

In a step 410, the dual phone B1 sends through an audio stream the music to the remote phone A1.

In parallel, in a step 420, the VOIP phone C1 rings. Subsequently, in a step 430, the VOIP phone C1 answers the incoming VOIP call. Then, the dual phone B1 receives a message indicating that the VOIP call is answered in step 440 and, as a result, stops playing music in a step 450.

In a step 460, the dual phone B1 is in the state “Call PSTN and Call VOIP in progress”.

In a step 470, the software application of the dual phone B1 connects the PSTN audio channels to the VOIP audio channels, as it will be described hereinafter. As a result, the audio flows between the remote phone A1 and the VOIP phone C1 are connected to each other.

Figure 4 shows a block diagram of the method of transferring a call from the PSTN to the VOIP network as it is implemented in the dual phone in accordance with an embodiment of the invention.

In a step 4010, the dual phone is connected both to the PSTN network and to the VOIP network. The PSTN to VOIP call transfer is activated and the parameters “VOIP ID” and “Nb Ring1” are set.

Then, in a step 4020, the dual phone receives an incoming PSTN call: a ringing signal is detected on the line interface.

The software application of the dual phone checks, in a step 4030, if the CID received can be transferred or not, as described before. If the call cannot be transferred (No), the call transfer procedure is stopped in a step 4035. The incoming PSTN call is then handled as a normal call, which can be answered by the dual phone user, redirected to a voice mail service or the answering machine of the dual phone, or stored as a missed call.

Otherwise (Yes), the call transfer procedure goes on with a test 4040 in which it is checked if the number of ringing received is greater than the value “Nb Ring1”. If not (No), the software application of the dual phone waits till the value “Nb Ring1” is reached.

When the value “Nb Ring1” is reached (Yes), the software application of the dual phone answers the PSTN call automatically in a step 4050. Then, in a step 4060, the software application sends a notification to the PSTN caller in order to indicate to said PSTN caller that the call is going to be transferred.

Subsequently, in a step 4070, the software application of the dual phone B1 initiates a VOIP call with the VOIP remote phone having the VOIP ID identifier as set by the dual phone user.

In a step 4080, the software application waits for the VOIP call establishment. If the VOIP callee having the VOIP ID identifier answers the call or the VOIP callee voice message is activated (Yes), then the software application connects the VOIP audio channel to the PSTN audio channel and audio data are redirected through the dual phone in a step 4090. To this end, the software application of the dual phone has to check which audio codec is used for the VOIP call (for example G.711, G.723, G.726, G.729, G.722 or G.722.2) and comprises conversion means for converting the audio data coded with the codec of the VOIP phone into PSTN audio data (always coded with the Adaptive Difference Pulse Code Modulation ADPCM, namely G.726), if needed. If the audio codec of the VOIP phone and the audio codec of the dual phone are the same is used, then no conversion is required.

The process ends in a step 4095 when the phone call is terminated.

If the VOIP call is unanswered (No) the software application waits, in a step 4100, for the time value "Timer2" set by default or by the user in the dual phone. If the VOIP callee answers the call (No), then the software application goes to step 4090.

If a busy tone is received from the VOIP phone or if the time value "Timer2" is reached (Yes), then the software application checks, in a step 4110 if the dual phone has an answering machine (the product capabilities are stored in base station memory) and if the answering machine is activated (ON state). If the dual phone has an answering machine (Yes), then the software application stops the VOIP call in a step 4120 and starts the answering machine in step 4130. When the recording of the message of the remote caller is terminated, the process ends in a step 4135.

If the dual phone has no answering machine (No), then the software application stops the VOIP call in a step 4200, stops the PSTN call and store a missed call in the call history menu item in a step 4210, and the process ends in a step 4215.

Moreover, if PSTN call is released by the PSTN caller in step 4300, then the process goes on with step 4200.

Figure 5 shows a block diagram of the method of transferring a call from the VOIP to the PSTN network as it is implemented in the dual phone in accordance with an embodiment of the invention.

When a user A calls via the VOIP network the user B who is the owner of the dual mode phone, the call allowed is redirected by the dual mode phone to a PSTN phone according to the configuration of the dual mode phone previously set by the user B, as it will be explained hereinafter. Note that the call can be re-directed over a plurality of PSTN number.

The hereinafter described method is very similar to the one described above in connection with the PSTN to VOIP call transfer. Instead of redirecting the PSTN call to the VOIP network, the phone redirects the VOIP call to the PSTN network, and the setting called "Timer1" replaces "Nb Rings1" as no analog ringing signal exists on VOIP incoming call.

In a step 5010, the dual phone is connected both to the PSTN network and to the VOIP network. The PSTN to VOIP call transfer is activated and the parameters "PSTN phone number" and "Timer1" are set.

Then, in a step 5020, the dual phone receives an incoming VOIP call, i.e. a SIP "INVITE" message is received. The dual phone rings. Even if the call transfer is activated, the dual phone user is able to pickup the call before the call transfer starts or during the call transfer.

The software application of the dual phone checks, in a step 5030, if the CID received can be transferred or not. If the call cannot be transferred (No), the call transfer procedure is stopped in a step 5035. The incoming VOIP call is then handled as a normal call, which can be answered by the dual phone user, redirected to a voice mail service, or stored as a missed call.

Otherwise (Yes), the call transfer procedure goes on with a test 5040 in which it is checked if the time elapsed since the initiation of the VOIP call is greater than the value "Timer1". If not (No), the software application of the dual phone waits till the value "Timer1" is reached.

When the value "Timer1" is reached (Yes), the software application of the dual phone answers the VOIP Call automatically in a step 5050. Then, in a step 5060, the software application sends a notification to the VOIP caller in order to indicate to said VOIP caller that the call is going to be transferred.

Subsequently, in a step 5070, the software application of the dual phone B1 initiates a PSTN call with the PSTN phone having the PSTN phone number as set by the dual phone user.

In a step 5080, the software application waits for the PSTN call establishment. If the PSTN callee having the PSTN phone number answers the call or if the PSTN callee voice answering machine or voice mail service is activated (Yes), then the software application connects the PSTN audio channel to the VOIP audio channel and audio data are redirected through the dual phone in a step 5090. The process ends in a step 5095 when the phone call is terminated.

If the PSTN call is unanswered (No) the software application waits for a predetermined time value "Timer2" set by default or by the user in the dual phone in a step 5100. If the PSTN callee answers the call (No), then the software application goes to step 5090.

If a busy tone is received from the PSTN phone or if the time value “Timer2” is reached (Yes), then the software application checks, in a step 5110 if the dual phone has an answering machine. If the dual phone has an answering machine (Yes), then the software application stops the PSTN call in a step 5120 and starts the answering machine in step 5130. When the recording of the message of the remote caller is terminated, the process ends in a step 5135.

If the dual phone has no answering machine (No), then the software application stops the PSTN call in a step 5200, stops the VOIP call and store a missed call in the call history menu item in a step 5210, and the process ends in a step 5215.

Moreover, if VOIP call is released by the VOIP caller in step 5300, then the process goes on with step 5200.

The invention may be implemented by means of dedicated software. A set of instructions corresponding to this software and which is loaded into a program memory causes an integrated circuit of the dual phone to carry out the method in accordance with the embodiments of the invention described in Figure 4 and Figure 5. The set of instructions may be stored on a data carrier such as, for example, a disk. The set of instructions can be read from the data carrier so as to load it into the program memory of the integrated circuit, which will then fulfils its role. For example, the software is copied on a CD-ROM, said CD ROM being sold together with the dual phone. Alternatively, the software can also be made available through the Internet. Moreover, this dedicated software may also be integrated by default in Flash memory or a Read Only Memory ROM memory of the dual mode phone.

It should be noted that the above-mentioned embodiments illustrate rather than limit the invention, and that those skilled in the art will be capable of designing many alternative embodiments without departing from the scope of the invention as defined by the appended claims. In the claims, any reference signs placed in parentheses shall not be construed as limiting the claims. The word "comprising" and "comprises", and the like, does not exclude the presence of elements or steps other than those listed in any claim or the specification as a whole. The singular reference of an element does not exclude the plural reference of such elements and vice-versa.

CLAIMS

1. A dual mode phone comprising:

- a first connection unit for connecting the dual mode phone to the public switched network;
- a second connection unit for connecting the dual mode phone to the Internet Protocol network, and which is coupled to the first connection unit;
- a memory for storing a phone identifier (PSTN phone number) and a Internet protocol identifier (VOIP ID); and
- call transfer management means for transferring an incoming call received by the second connection unit from the Internet protocol network to a phone having the phone identifier through the first connection unit and the public switched network, or for transferring an incoming call received by the first connection unit from the public switched network to a phone having the Internet protocol identifier through the second connection unit and the Internet protocol network.

2. A dual mode phone as claimed in claim 1, comprising a timer for allowing the dual mode phone user to pick up the incoming call during a first predetermined time limit (Nb Ring1, Timer1), the call transfer management means initiating the call transfer when said first predetermined time limit has expired.

3. A dual mode phone as claimed in claim 1, comprising means for automatically dialing the phone identifier (PSTN phone number) or the Internet protocol identifier (VOIP ID).

4. A dual mode phone as claimed in claim 1, comprising means for identifying a caller, the memory being adapted to store a list of callers allowed to be transferred, the call transfer management means being adapted to transfer the incoming call only if the caller is on the list of callers allowed to be transferred.

5. A dual mode phone as claimed in claim 1, comprising conversion means for converting audio data coded with a codec for the internet protocol network into audio data coded for the public switched network.

6. A dual mode phone as claimed in claim 1, comprising a timer for allowing a phone user to which the incoming call is to be transferred to pick up the incoming call during a second predetermined time limit (Timer2).

7. A dual mode phone as claimed in claim 6, wherein the call transfer management means are adapted to transfer the incoming call to an answering machine of the dual phone or to a voice mail service, or to store the missed call in the memory after the second predetermined time limit has expired.

8. A dual mode phone as claimed in claim 1, comprising means for notifying a phone user to which the incoming call is to be transferred that the incoming call is going to be transferred.

9. A dual mode phone as claimed in claim 1, wherein the Internet protocol identifier is the one of the dual mode phone.

10. A dual mode phone as claimed in claim 1, comprising means for identifying an incoming call identifier and for forwarding the incoming call identifier to the phone to which the incoming call is transferred.

11. A method of transferring a phone call from a dual mode phone to another phone, said dual mode phone device being adapted to be connected to the public switched network and to the Internet protocol network and comprising a memory for storing a phone identifier and an Internet protocol identifier, said method comprising the steps of:

- transferring an incoming call received by the dual mode phone from the Internet protocol network to a phone having the phone identifier through the public switched network, or
- transferring an incoming call received by the dual mode phone from the public switched network to a phone having the Internet protocol identifier through the Internet protocol network.

12. A computer program product comprising a set of instructions which, when loaded in a memory of a dual mode phone, makes the dual mode phone execute all the steps of the method as claimed in claim 10.

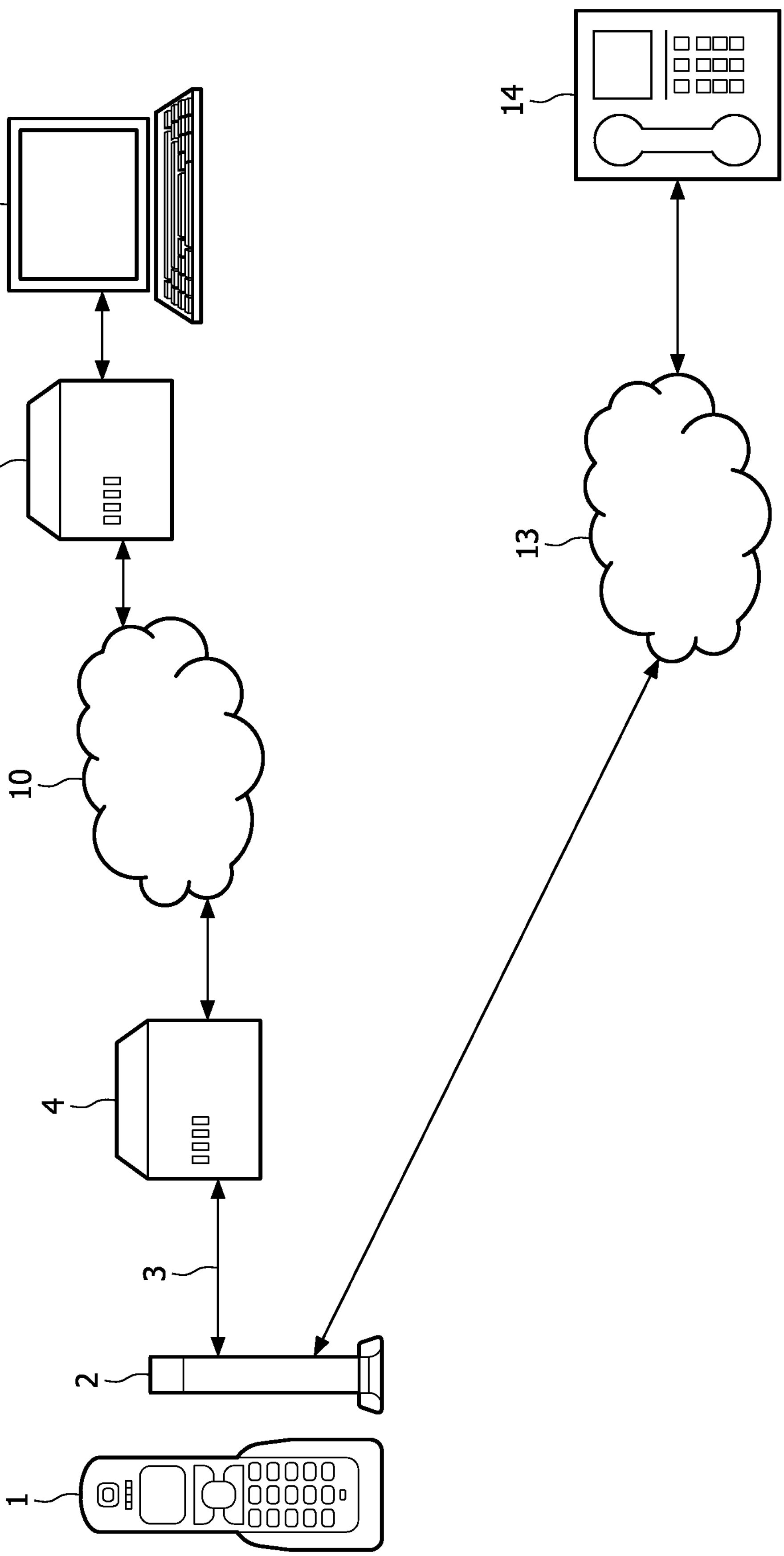
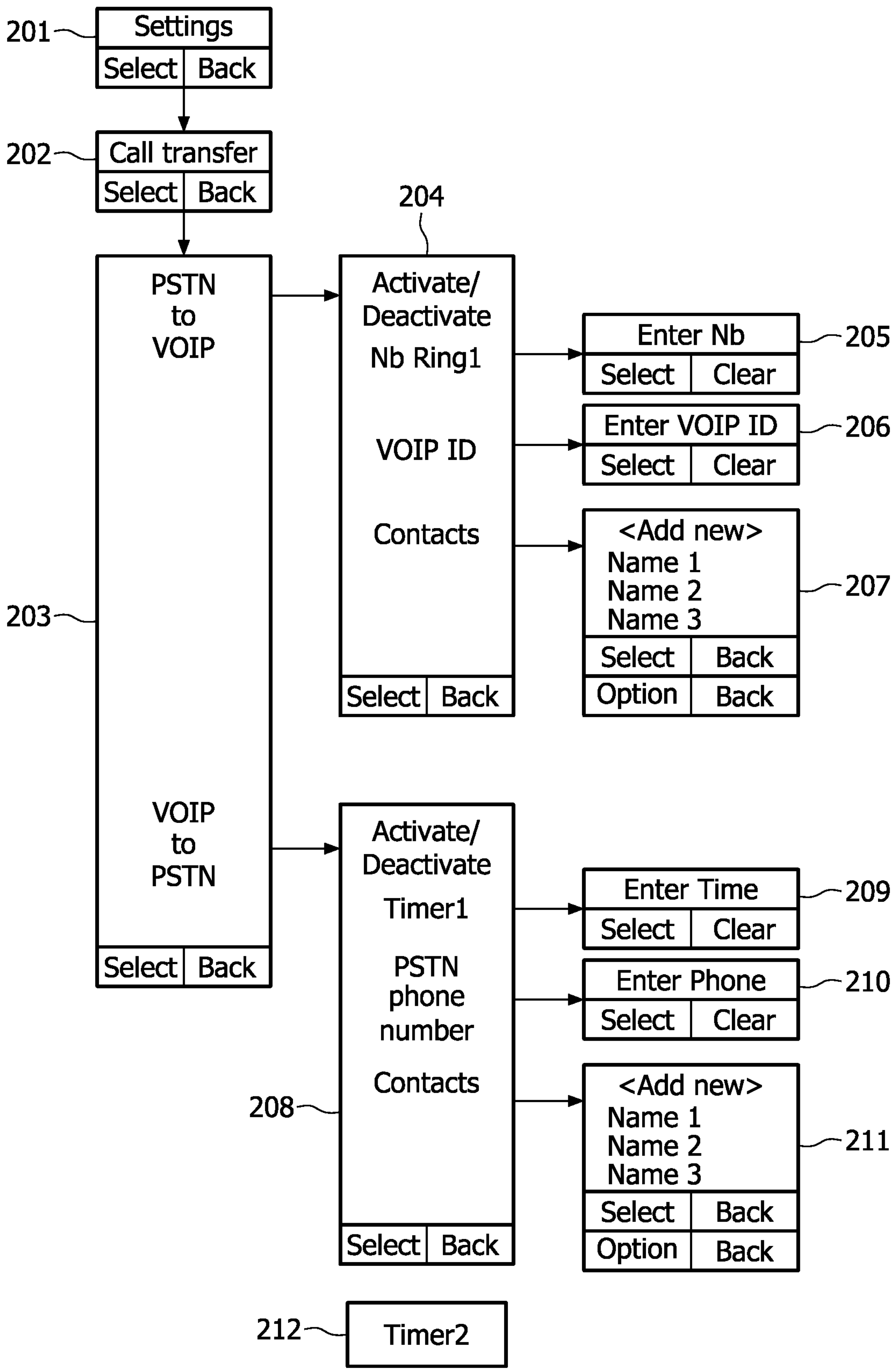


FIG. 1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Back
Activate/
Deactivate
Nb Ring1
VOIP ID
Contacts
Select
Back
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Select
Clear
205
Enter VOIP ID
Select
Clear
206
<Add new>
Name 1
Name 2
Name 3
Select
Back
Option
Back
207
208
Activate/
Deactivate
Timer1
PSTN
phone
number
Contacts
Select
Back
Enter Time
Select
Clear
209
Enter Phone
Select
Clear
210
<Add new>
Name 1
Name 2
Name 3
Select
Back
Option
Back
211
212
Timer2

FIG. 2

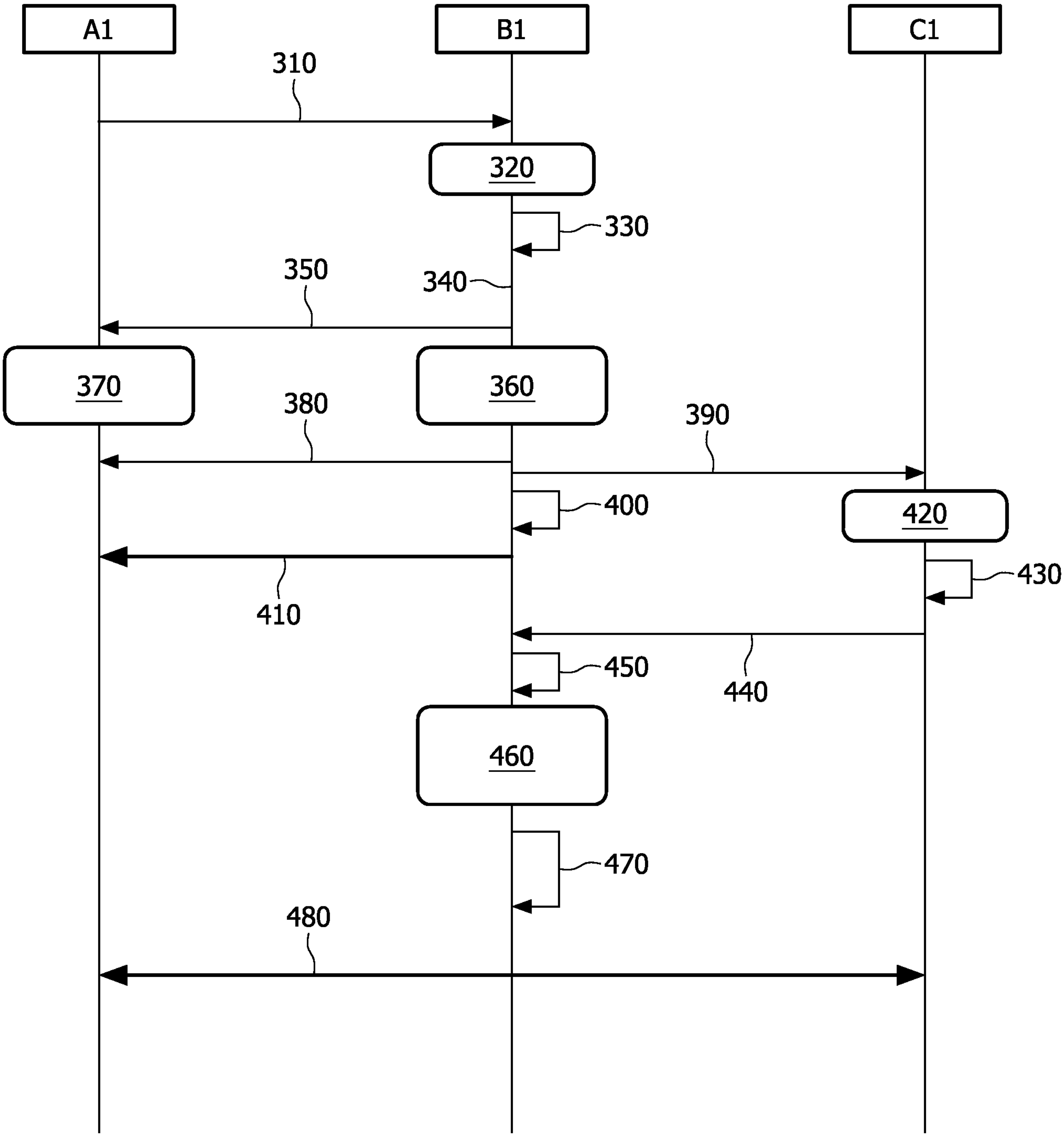
A1
B1
C1
310
320
330
350
340
370
360
380
390
400
420
410
430
440
450
460
470
480

FIG. 3

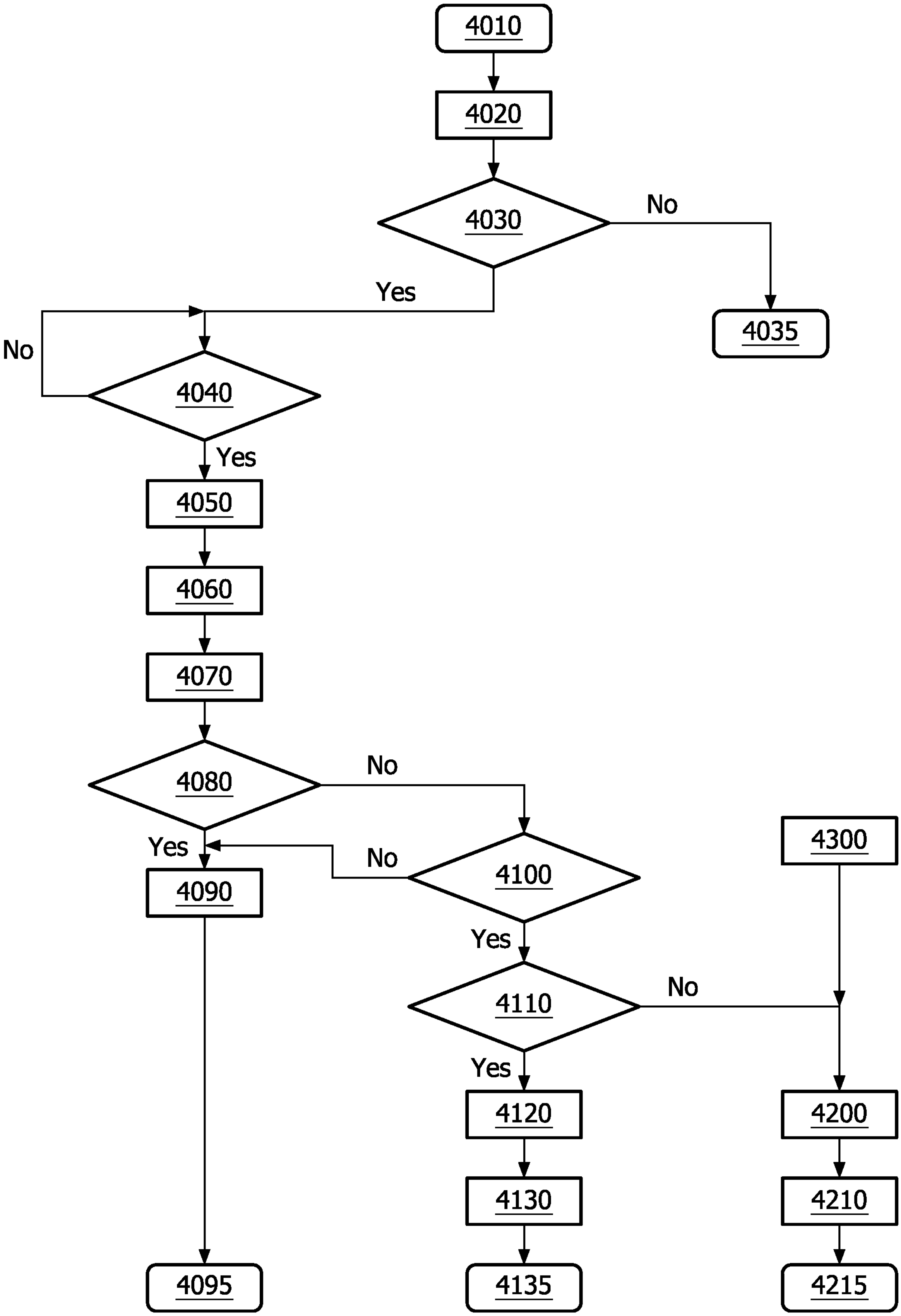
4010
4020
4030
No
Yes
4035
No
4040
Yes
4050
4060
4070
4080
No
Yes
No
4090
4100
4300
Yes
4110
No
Yes
4120
4200
4130
4210
4095
4135
4215

FIG. 4

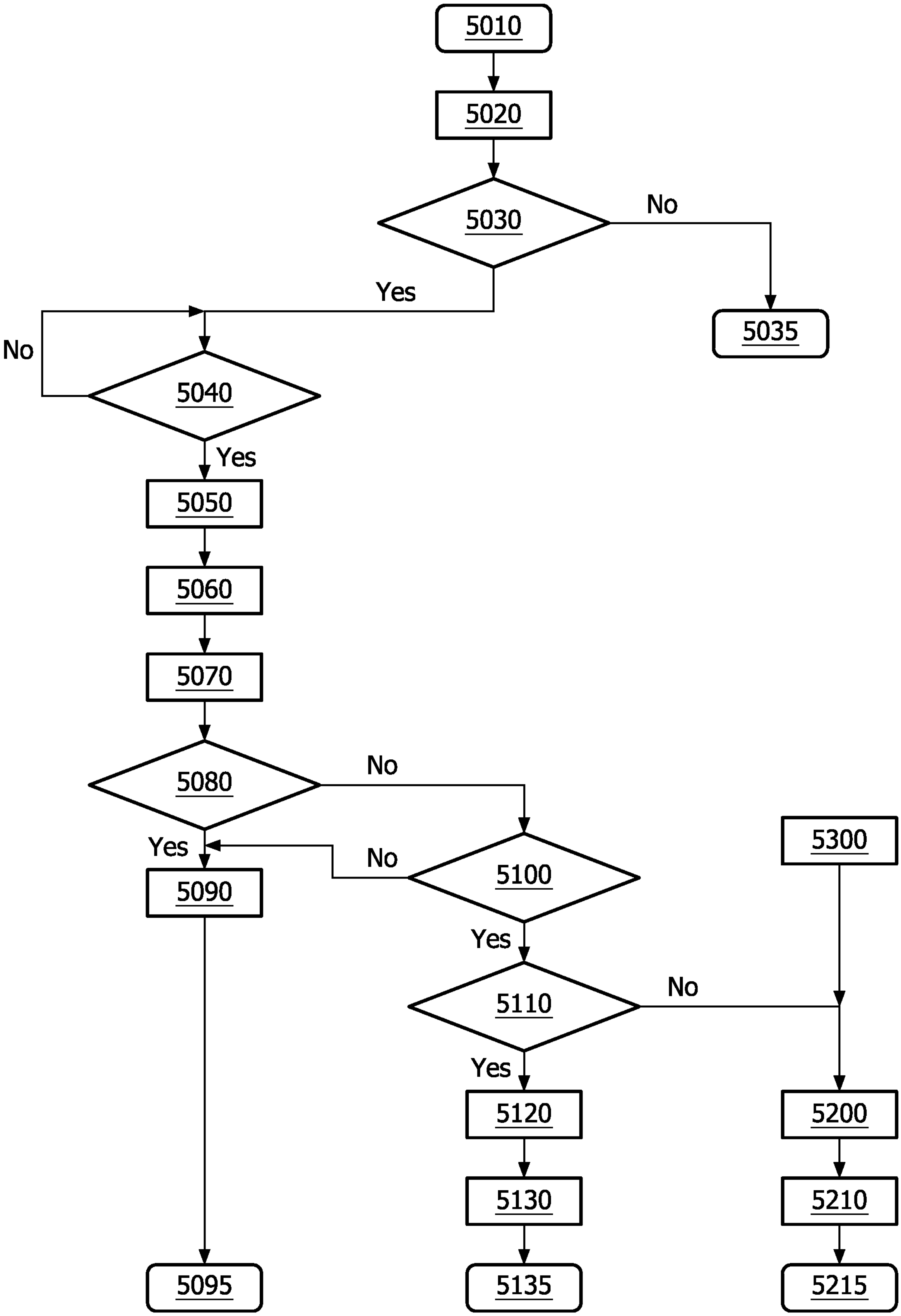
5010
5020
5030
No
Yes
5035
No
5040
Yes
5050
5060
5070
5080
No
Yes
No
5100
5300
5090
Yes
5110
No
Yes
5120
5200
5130
5210
5095
5135
5215

FIG. 5

INTERNATIONAL SEARCH REPORT

International application No
PCT/IB2008/055379

A. CLASSIFICATION OF SUBJECT MATTER
INV. H04M3/54 H04M7/00

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

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
H04M

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

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

EPO-Internal, WPI Data

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	ANONYMOUS: "CORDLESS INTERNET PHONE WITH SKYPE MODEL SPH200D"[Online] November 2006 (2006-11), XP002525974 Santa Clara, CA, USA Retrieved from the Internet: URL:ftp://downloads.netgear.com/files/SPH200D_UM_3Nov06.pdf> [retrieved on 2009-04-28] the whole document -----	1-12
X	US 2005/069121 A1 (FARUQUE MOHAMMAD A [US] ET AL) 31 March 2005 (2005-03-31) paragraph [0008] - paragraph [0056]; figures 1,4 -----	1-12
A	US 2001/046237 A1 (CHAN SHUN-SHING [US] ET AL) 29 November 2001 (2001-11-29) paragraph [0006] - paragraph [0056]; figures 1,7 -----	1-12

☐ Further documents are listed in the continuation of Box C.

☒ See patent family annex.

* Special categories of cited documents :

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"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

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

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
29 April 2009	13/05/2009

Name and mailing address of the ISA/	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL – 2280 HV Rijswijk Tel. (+31-70) 340-2040, Fax: (+31-70) 340-3016	Nash, Michael

INTERNATIONAL SEARCH REPORT

International application No
PCT/IB2008/055379

Patent document cited in search report		Publication date	Patent family member(s)		Publication date
US 2005069121	A1	31-03-2005	NONE		
US 2001046237	A1	29-11-2001	JP 3202003	B2	27-08-2001
			JP 11355474	A	24-12-1999