

Title**A Portable Communications Device****Technical Field**

- 5 The invention concerns a portable communications device to be coupled to a computer for enabling a user to call at least another user via a network.

Background of the Invention

- 10 Voice over IP (VoIP) technology is a very economical choice for people who wish to make international calls. VoIP calls are much cheaper in comparison with traditional IDD calls and thus savings as much as 50% are possible. The VoIP call quality is very similar to traditional IDD calls.

- 15 Presently, SIP hard phones may be purchased. However, these phones are bulky and are typically the same size as conventional house or office phones. SIP hard phones limit a user to a fixed location (home or office) due to its bulky size. Some places only have one Internet connection available and thus carrying a SIP hard phone is not ideal. In public places such as Internet cafes or airport lounges, it may not be possible to find a free RJ45 outlet to connect the SIP hard phone to, or is not permissible to disconnect a computer to free up a RJ45 outlet.

- 20 Recently, there has been increased attention on VoIP software applications such as Skype™ and GoogleTalk™. These applications must be downloaded from a web site and installed on a computer to operate. They are typically used on personal computers or office computers, where there is regular use by the same user. However, for frequent travelers, students and businesspeople, it is inconvenient to have to download software onto every computer they use (if they are permitted to and the software is not already installed). They must wait for the download to complete, and then configure the application each time to their personal needs. This is very time consuming and not convenient.

Accordingly, there is a desire to provide a more convenient way of making low-cost calls to other parties using the Internet.

- 35 **Summary of the Invention**

In a first preferred aspect, there is provided a portable communications device to be coupled to a computer for enabling a user to call at least another user via a network, the portable communications device comprising:

5 a non-volatile memory device to store connection software for communicating via the network;

an audio recorder and reproducer to record and reproduce audio signals;

an audio signal connector for communicating the audio signals between the audio recorder and reproducer and a speaker and microphone;

10 wherein the connection software is executed by the computer and comprises a communication client to establish the call using a communication protocol, and to route the call via a communication server on the network.

The connection software may be preloaded onto the non-volatile memory device.

15 The connection software may be copied to the computer and the copy of the connection software on the computer may be executed by the computer.

The communication client may be a Session Initiation Protocol (SIP) client and the communication protocol is Session Initiation Protocol (SIP).

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The connection software may be automatically executed when the portable communications device is coupled to the computer and the connection software connects to a web server, the web server having a web site for the user to verify their identity before calls from the user are permitted.

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The web site may store a phonebook for the user.

The user may enter a phone number on the web site or select a user stored in the phonebook to make the call, the web server forwarding the number to be called to the communication server.

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The communication server may determine whether the call is to a virtual number or to a PSTN number.

If a PSTN number is called, the call may be routed to a PSTN switch.

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If call is to a virtual number, the communication server may route the call to the another user over a communication protocol from any one from the group consisting of: SIP, H.323 protocol and Media Gateway Control Protocol (MGCP).

- 5 The communication server may be a SIP (Session Initiation Protocol) server and have a predetermined IP (Internet Protocol) address.

The audio recorder and reproducer may be a sound card or a software audio codec.

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The audio signal connector may be an external device connector or a wireless device connector for wireless connection with external audio devices.

The speaker may be an earpiece connected to the microphone via a cable.

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A coupling device may be provided for coupling the portable communications device to a socket on the computer.

The coupling device may be a USB plug and the socket may be a USB socket, and
20 the USB plug is directly coupled to the USB socket.

The network may be the Internet.

The call may be a telephone call or a video call.

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The connection software may be upgradeable via the network.

The user may be billed according to a predetermined billing scheme if a PSTN number is called.

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The user may have a prepaid balance for making calls to a PSTN number, the prepaid balance being deducted by a predetermined amount for each call to a PSTN number.

The prepaid balance may be increased via credit card payment, direct debit
35 payment or a top-up card.

The top-up card may be scratched to reveal a secret code to be entered which causes the prepaid balance to be increased by a predetermined amount.

The portable communications device may be assigned a Direct Inward Dialing (DID) number.

- 5 The connection software and the call may be encrypted.

In a second aspect, there is provided a communication server for enabling a user to call at least another user via a network, the communication server comprising:

10 a receiver module to receive a telephone number for the call from a web server;

a communications module to determine whether the call is to a virtual number or a PSTN number;

15 wherein if the call is to a virtual number, the communication server routes the call to the at least another user over a communication protocol to establish the call over the network; and

wherein if the call is to a PSTN number, the communication server routes the call to the at least another user via a tandem routing switch to establish the call over the network and a PSTN network.

- 20 The communication protocol may be Session Initiation Protocol (SIP), H.323 protocol or Media Gateway Control Protocol (MGCP).

The call may be originated from or received by a portable communications device to be coupled to a computer, the portable communications device comprising:

25 a non-volatile memory device to store connection software for communicating via the network;

an audio recorder and reproducer to record and reproduce audio signals;

an audio signal connector for communicating the audio signals between the audio recorder and reproducer and a speaker and microphone;

30 wherein the connection software is executed by the computer and comprises a communication client to establish the call using a communication protocol, and to route the call via the communication server.

- 35 In a third aspect, there is provided a communication system for enabling a user to call at least another user via a network, the system comprising:

a communication client to establish the call using a communication protocol, and to route the call via a communication server on the network;

a communication server to route calls initiated by the communication client according to whether the call is to a virtual number or to a PSTN number; and
a payment gateway to bill the user for calls to PSTN numbers.

- 5 The communication client may be stored in a non-volatile memory device of a portable communications device to be coupled to a computer, wherein the portable communications device further comprises:

an audio recorder and reproducer to record and reproduce audio signals;
and

- 10 an audio signal connector for communicating the audio signals between the audio recorder and reproducer and a speaker and microphone.

The communication server may comprise:

- a receiver module to receive a telephone number for the call from a web
15 server;

a communications module to determine whether the call is to a virtual number or to a PSTN number;

- wherein if the call is to a virtual number, the communication server routes the call to the at least another user over a communication protocol to establish the
20 call over the network; and

wherein if the call is to a PSTN number, the communication server routes the call to the at least another user via a tandem routing switch to establish the call over the network and a PSTN network.

- 25 The communication protocol may be any one from the group consisting of: Session Initiation Protocol (SIP), H.323 protocol and Media Gateway Control Protocol (MGCP).

- In a fourth aspect, there is provided a package for enabling a user to call at least
30 another user via a network, the package comprising:

a portable communications device according to claim 1;

an earpiece connected to a microphone via a cable, to be operatively connected to the audio signal connector;

- a card device to provide authentication information to the user for
35 authentication with a web site before the user is permitted to make a call.

The card device may provide the user with a predetermined balance to make calls to PSTN numbers.

The balance may be credited via online payment or funds transfer, or using a top-up card.

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Brief Description of the Drawings

An example of the invention will now be described with reference to the accompanying drawings, in which:

Figure 1 is a pictorial representation of a portable communications device in accordance with a preferred embodiment of the present invention;

Figure 2 is a block diagram of the portable communications device of Figure 1;

Figure 3 is a system diagram of the portable communications device of Figure 1 in use and a server-side system in accordance with a preferred embodiment of the present invention; and

Figure 4 is a process flow diagram of using the portable communications device of Figure 1.

Detailed Description of the Drawings

Figure 1 and the following discussion are intended to provide a brief, general description of a suitable computing environment in which the present invention may be implemented. Although not required, the invention will be described in the general context of computer-executable instructions, such as program modules, being executed by a computer such as a personal computer, laptop computer, notebook computer, tablet computer, PDA and the like. Generally, program modules include routines, programs, characters, components, data structures, that perform particular tasks or implement particular abstract data types. As those skilled in the art will appreciate, the invention may be practiced with other computer system configurations, including hand-held devices, multiprocessor systems, microprocessor-based or programmable consumer electronics, network PCs, minicomputers, mainframe computers, and the like. The invention may also be practiced in distributed computing environments where tasks are performed by remote processing devices that are linked through a communications network. In a distributed computing environment, program modules may be located in both local and remote memory storage devices.

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Referring to Figures 1 and 2, a portable communications device 10 to be coupled to a computer 50 for enabling a user to call at least another user via the Internet is

provided. The device 10 comprises: a non-volatile memory device 20, an audio recorder and reproducer 21 and an audio signal connector 22. The non-volatile memory device 20 stores connection software 23 for communicating audio and/or video data via the Internet. The audio recorder and reproducer 21 records and reproduces audio signals. The audio signal connector 23 communicates the audio signals between the audio recorder and reproducer 21 and a speaker 30 and microphone 31. The connection software 23 is executed by the computer 50 and comprises a communication client to establish the call using a communication protocol, and to route the call via a communication server 120 accessible via the Internet 120. The communication client may be a VoIP client.

Preferably, the device 10 is a memory stick device with a Universal Serial Bus (USB) plug 15 to be plugged into the USB port/socket of the computer 50. Power for the device 10 is drawn from the computer 50 and data between the device 10 and computer 50 is communicated via the USB plug 15. The audio recorder and reproducer 21 is a sound card operatively connected to the audio signal connector 22. The sound card 21 converts analog audio to digital audio data and vice-versa. The speaker 30 (an earpiece) and microphone 31 is provided together via single cable for convenience. The earpiece 30 and microphone 31 and releasably operatively connectable to an input jack 24 of the audio signal connector 22. In another embodiment, the audio signal connector 22 may have a wireless transceiver to wireless communicate with a wireless earpiece 30 and microphone 31, for example, using Bluetooth™ or Radio Frequency (RF).

The communication client of the connection software 23 is a Session Initiation Protocol (SIP) client that uses SIP as a communication protocol to communicate audio data received from the microphone 31 and transmit audio data to the earpiece 30. SIP is an application-layer control protocol and is a signaling protocol for Internet conferencing, telephony, events notification and instant messaging. The protocol initiates call setup, routing, and authentication to endpoints within an IP domain. SIP can establish sessions for features such as audio/videoconferencing and call forwarding to be deployed over IP networks.

Referring to Figure 3, a server-side communication system 100 is provided to enable proper routing of calls made from the device 10 or a SIP IP phone 60. The system 100 generally comprises a web server 110, a SIP server 120 and a TDM switch 130. The web server 110 provides the user interface for the authentication

of the user and to enter the numbers they wish to call. The web server 110 communicates with the SIP server 120 by transmitting the destination phone number where the SIP server 120 determines whether it is a virtual number (the call is to another SIP enabled device) or is a PSTN number. If the number is a
 5 PSTN number the call is routed to the TDM switch 130 which is then eventually routed to A-Z Termination 140 to reach PSTN devices such as a PSTN phone 150 or mobile phone 155. A payment gateway may be provided to enable users to top-up their account balance. Alternatively, users may purchase top-up cards, which are scratched to reveal a code to top up their account balance.

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Referring to Figure 4, in a typical scenario, the device 10 is plugged 200 into a computer 50 to initiate a telephone call over SIP. Next, the connection software 23 is automatically installed onto the computer 50 and/or executed by the computer 50. The connection software 23 may be copied to the computer 50 and the local
 15 copy of the connection software 23 on the computer 50 may be executed by the computer 50. The user is directed to a web site of a web server 110 of a service provider. The user must verify 203 their identity by entering their login and password at a login page of the web site. Once the user is authenticated, they are permitted 204 to make a phone call. The authenticated user is directed to a "Make
 20 a Call page" of the web site. A graphical web dialer or an input box is provided for the user to enter the phone number to call. The user inputs the phone number or selects an entry from their phonebook stored in the web site. The phone number is then transmitted to a SIP server 205. Apart from facilitating communication from the device 10, the SIP server 205 can also handle calls from a SIP IP phone 60.
 25 The SIP server 205 supports: user authentication, redirect and registration services, and traditional telephony features such as personal mobility, time-of-day routing and call forwarding based on the geographical location of the person being called. The SIP server 120 determines 205 whether the number is a virtual number (to a device that is able to communicate using SIP) or a Public Switched
 30 Telephone Network (PSTN) number. If the number is a SIP number, the SIP server 205 routes 207 the call to the SIP number and the call is free for the user. If the number is a PSTN number, the balance in the user's account is checked 208 to determine if there is sufficient credit to make the call. If there are insufficient funds, the call is terminated and the user is advised to top up their account. If there are
 35 sufficient funds, the SIP server 205 routes 210 the call to a tandem routing (TDM) switch 130. The TDM switch 130 connects one trunk to another and is an intermediate switch or connection between an originating telephone call or location

and the final destination of the call. The TDM switch 130 eventually routes the call via an A-Z Termination 140 to connect the call to the destination party to their PSTN phone 150 or mobile phone 155. The A-Z Termination 140 is a call termination service to terminate VoIP call traffic to any telephone directory number
 5 in A-Z destination worldwide. Once the call to the PSTN number is finished 212, the call charges are calculated and deducted 213 from the user's account. A billing scheme is provided which enables different billing regimes according to calls to various countries, flat rates, volume discounts, special offer periods, and loyalty programmes to be implemented.

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Preferably the user's phonebook is stored on the web site, which offers almost unlimited storage capacity remotely. Also, if the device 10 is lost, the user's list of phone numbers is retained elsewhere. Alternatively, the phonebook may be stored locally on the device 10.

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The local copy of the connection software 23 on the computer 50 may be upgraded by the web site 110. Burner software and associated instructions may be provided on the web site 110 to facilitate the upgrade and to synchronise the version of the connection software 23. This facilitates efficient distribution of patches, upgrades of
 20 newer drivers to be loaded onto the device 10 for enhanced performance.

The device 10 is assigned a Direct Inward Dialing (DID) number. This means that a user using a device 10 can be contacted by calling the DID number assigned to the device 10. For example, if an Australian visited Europe carrying their device 50
 25 with an Australia DID number, their family in Australia could call him or her by calling the DID number. The rates charged to the family would be local call rates rather than IDD call rates. Even more cost effective is if the family also had their own device 10 to call him or her, and the call would be free.

30 To secure the call and connection software 23, the connection software 23 and the call may be encrypted. Thus, if the device 10 is lost or stolen, sensitive information stored on the device 10 would remain secure and protected.

Advantageously, the user is able to carry their device 10 to any computer 50 they
 35 are using to be instantly capable of making a call to anyone around the world. There is no need to download software via the Internet and wait for several minutes for the download to complete and then for the entire software application

to install on the computer. The user's phonebook is also conveniently stored on a remote site or locally on the device 10. The installation of the necessary USB drivers and execution of the connection software 23 occurs within a period of time depending on the processing power of the computer 50.

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Although a sound card 21 is described, a software audio codec may be used where the software audio codec is used by the processor of the computer 50 to record and reproduce audio signals.

10 Although an earpiece 30 and microphone 31 have been described, a speaker phone for party conference calls may be connected to the audio signal connector 22.

15 Although a phone call is described, it is envisaged that video calls may also be made using the device 10 and the communication system 100. In such cases, a web cam and a display device are required to be operatively connected to the computer 50 and also at the destination to record and reproduce video data.

20 Although SIP has been described, any VoIP protocol may be used such as H.323 protocol or Media Gateway Control Protocol (MGCP).

25 Although coupling of the device 10 to a computer 50 is described, it is envisaged that this covers wireless coupling. Wireless coupling may, for example, be via wireless USB ultrawideband. Although USB has been described, other interface standards may be used such as IEEE 1394 FireWire or i.LINK.

It will be appreciated by persons skilled in the art that numerous variations and/or modifications may be made to the invention as shown in the specific embodiments without departing from the scope or spirit of the invention as broadly described.

30 The present embodiments are, therefore, to be considered in all respects illustrative and not restrictive.

WE CLAIM:

1. A portable communications device to be coupled to a computer for enabling a user to call at least another user via a network, the portable
5 communications device comprising:
 - a non-volatile memory device to store connection software for communicating via the network;
 - an audio recorder and reproducer to record and reproduce audio signals;
 - an audio signal connector for communicating the audio signals between
10 the audio recorder and reproducer and a speaker and microphone;
 - wherein the connection software is executed by the computer and comprises a communication client to establish the call using a communication protocol, and to route the call via a communication server on the network.
- 15 2. The device according to claim 1, wherein the connection software is preloaded onto the non-volatile memory device.
3. The device according to claim 1, wherein the connection software is copied to the computer and the copy of the connection software on the computer is
20 executed by the computer.
4. The device according to claim 1, wherein the communication client is a Session Initiation Protocol (SIP) client and the communication protocol is Session Initiation Protocol (SIP).
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5. The device according to claim 1, wherein the connection software is automatically executed when the portable communications device is coupled to the computer and the connection software connects to a web server, the web server having a web site for the user to verify their identity before calls from the user are
30 permitted.
6. The device according to claim 5, wherein the web site stores a phonebook for the user.
- 35 7. The device according to claim 6, wherein the user enters a phone number on the web site or selects a user stored in the phonebook to make the call, the web server forwarding the number to be called to the communication server.

8. The device according to claim 1, wherein the communication server determines whether the call is to a virtual number or to a PSTN number.
- 5 9. The device according to claim 8, wherein if a PSTN number is called, the call is routed to a PSTN switch.
- 10 10. The device according to claim 8, wherein if call is to a virtual number, the communication server routes the call to the another user over a communication protocol from any one from the group consisting of: SIP, H.323 protocol and Media Gateway Control Protocol (MGCP).
- 15 11. The device according to claim 4, wherein the communication server is a SIP (Session Initiation Protocol) server and has a predetermined IP (Internet Protocol) address.
12. The device according to claim 1, wherein the audio recorder and reproducer is a sound card or a software audio codec.
- 20 13. The device according to claim 1, wherein the audio signal connector is an external device connector or a wireless device connector for wireless connection with external audio devices.
- 25 14. The device according to claim 1, wherein the speaker is an earpiece connected to the microphone via a cable.
15. The device according to claim 1, further comprises a coupling device for coupling the portable communications device to a socket on the computer.
- 30 16. The device according to claim 15, wherein the coupling device is a USB plug and the socket is a USB socket, and the USB plug is directly coupled to the USB socket.
- 35 17. The device according to claim 1, wherein the network is the Internet.
18. The device according to claim 1, wherein the call is a telephone call or a video call.

19. The device according to claim 1, wherein the connection software is upgradeable via the network.
- 5 20. The device according to claim 8, wherein the user is billed according to a predetermined billing scheme if a PSTN number is called.
21. The device according to claim 20, wherein the user has a prepaid balance for making calls to a PSTN number, the prepaid balance being deducted by a
10 predetermined amount for each call to a PSTN number.
22. The device according to claim 21, wherein the prepaid balance is increased via credit card payment, direct debit payment or a top-up card.
- 15 23. The device according to claim 22, wherein the top-up card is scratched to reveal a secret code to be entered which causes the prepaid balance to be increased by a predetermined amount.
24. The device according to claim 1, wherein the portable communications
20 device is assigned a Direct Inward Dialing (DID) number.
25. The device according to claim 1, wherein the connection software and the call is encrypted.
- 25 26. A communication server for enabling a user to call at least another user via a network, the communication server comprising:
a receiver module to receive a telephone number for the call from a web server;
a communications module to determine whether the call is to a virtual
30 number or a PSTN number;
wherein if the call is to a virtual number, the communication server routes the call to the at least another user over a communication protocol to establish the call over the network; and
wherein if the call is to a PSTN number, the communication server routes
35 the call to the at least another user via a tandem routing switch to establish the call over the network and a PSTN network.

27. The server according to claim 26, wherein the communication protocol is Session Initiation Protocol (SIP), H.323 protocol or Media Gateway Control Protocol (MGCP).

5 28. The server according to claim 26, wherein the call is originated from or received by a portable communications device to be coupled to a computer, the portable communications device comprising:

a non-volatile memory device to store connection software for communicating via the network;

10 an audio recorder and reproducer to record and reproduce audio signals;

an audio signal connector for communicating the audio signals between the audio recorder and reproducer and a speaker and microphone;

wherein the connection software is executed by the computer and comprises a communication client to establish the call using a communication
15 protocol, and to route the call via the communication server.

29. A communication system for enabling a user to call at least another user via a network, the system comprising:

a communication client to establish the call using a communication
20 protocol, and to route the call via a communication server on the network;

a communication server to route calls initiated by the communication client according to whether the call is to a virtual number or to a PSTN number; and

a payment gateway to bill the user for calls to PSTN numbers.

25 30. The system according to claim 29, wherein the communication client is stored in a non-volatile memory device of a portable communications device to be coupled to a computer, wherein the portable communications device further comprises:

an audio recorder and reproducer to record and reproduce audio signals;

30 and

an audio signal connector for communicating the audio signals between the audio recorder and reproducer and a speaker and microphone.

31. The system according to claim 29, wherein the communication server
35 comprises:

a receiver module to receive a telephone number for the call from a web server;

a communications module to determine whether the call is to a virtual number or to a PSTN number;

wherein if the call is to a virtual number, the communication server routes the call to the at least another user over a communication protocol to establish the call over the network; and

wherein if the call is to a PSTN number, the communication server routes the call to the at least another user via a tandem routing switch to establish the call over the network and a PSTN network.

32. The system according to claim 31, wherein the communication protocol is any one from the group consisting of: Session Initiation Protocol (SIP), H.323 protocol and Media Gateway Control Protocol (MGCP).

33. A package for enabling a user to call at least another user via a network, the package comprising:

a portable communications device according to claim 1;

an earpiece connected to a microphone via a cable, to be operatively connected to the audio signal connector;

a card device to provide authentication information to the user for authentication with a web site before the user is permitted to make a call.

34. The package according to claim 31, wherein the card device provides the user with a predetermined balance to make calls to PSTN numbers.

35. The package according to claim 32, wherein the balance is credited via online payment or funds transfer, or using a top-up card.