(12) UK Patent Application (19) GB (11) 2 315 190 (13) A

(43) Date of A Publication 21.01.1998

(21) Application No 9714058.6

(22) Date of Filing 04.07.1997

(30) Priority Data
(31) 2180685 (32) 08.07.1996 (33) CA

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(51) INT CL^6
H04L 12/66

(52) UK CL (Edition P)
H4K KTK KTM

(56) Documents Cited
EP 0238948 A2 EP 0147197 A1 WO 97/23078 A1
WPI Abstract Accession No.95-267646/199535 & JP7170288 A

(58) Field of Search
UK CL (Edition O) **H4K KTK KTM , H4P PPA**
INT CL^6 **H04L 12/64 12/66**
ONLINE: WPI

(54) **An internet telephony gateway for providing an interface between a circuit switched and packet switched network**

(57) An Internet Telephony Gateway is disclosed for permitting telephone calls to be made from an ordinary telephone to a subscriber on the Internet. The Gateway includes an apparatus for enabling the establishment of telephone calls between a packet switched network 27 carrying data packets and a circuit switched network 4 carrying telephone signals, and comprises a first interface unit (fig.5, 44) for connection to the circuit switched network; a second interface unit (fig.5, 45) for connection to the packet switched network; a device for receiving signals at said first interface and converting them to data packets for transmission over the packet switched network and vice versa; and a processor (fig.5, 40) for determining destination information from incoming signals or data packets on one network and setting up a call to a destination on the other network in response to the destination information.

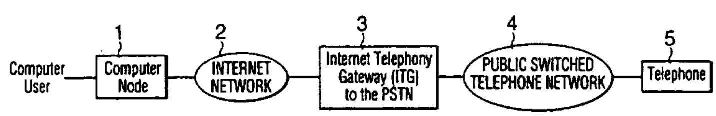
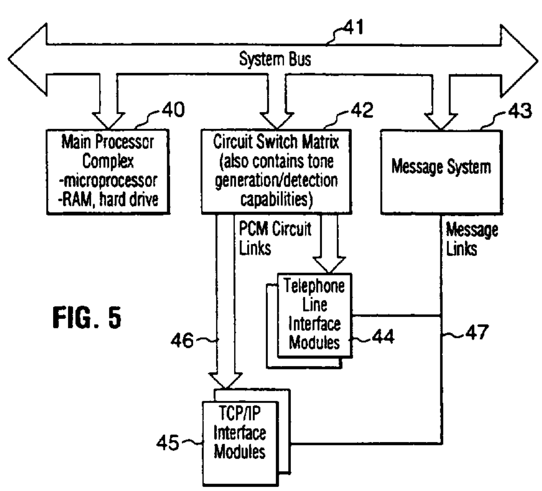


FIG. 1

FIG. 5



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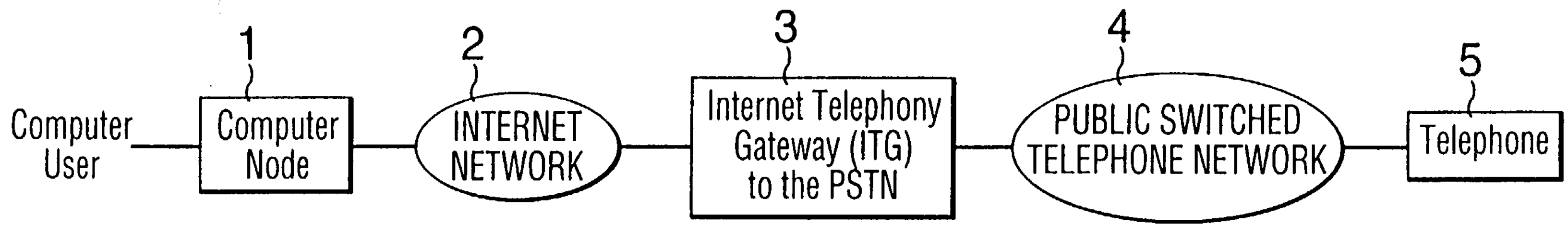
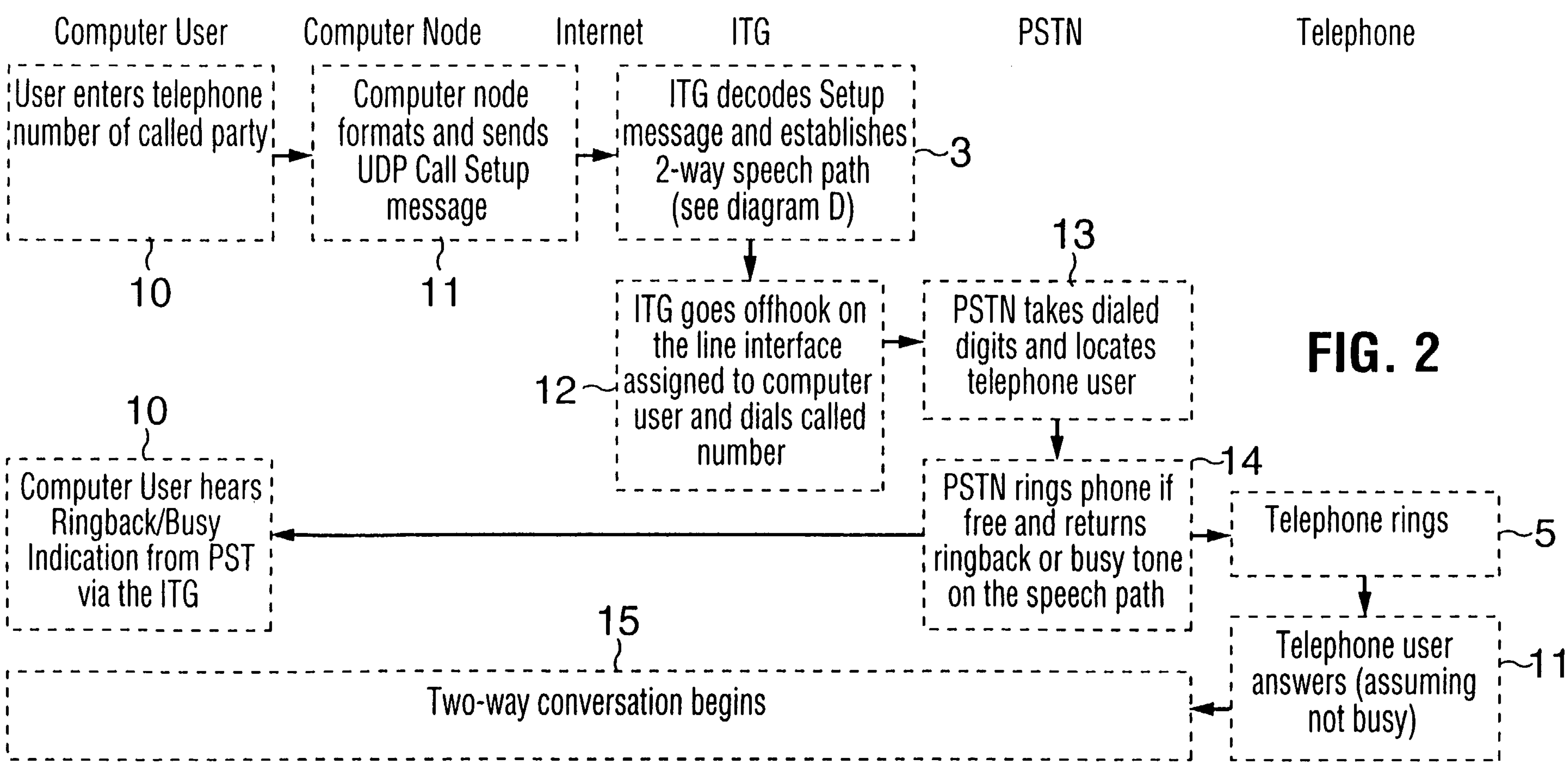
At least one drawing originally filed was informal and the print reproduced here is taken from a later filed formal copy.



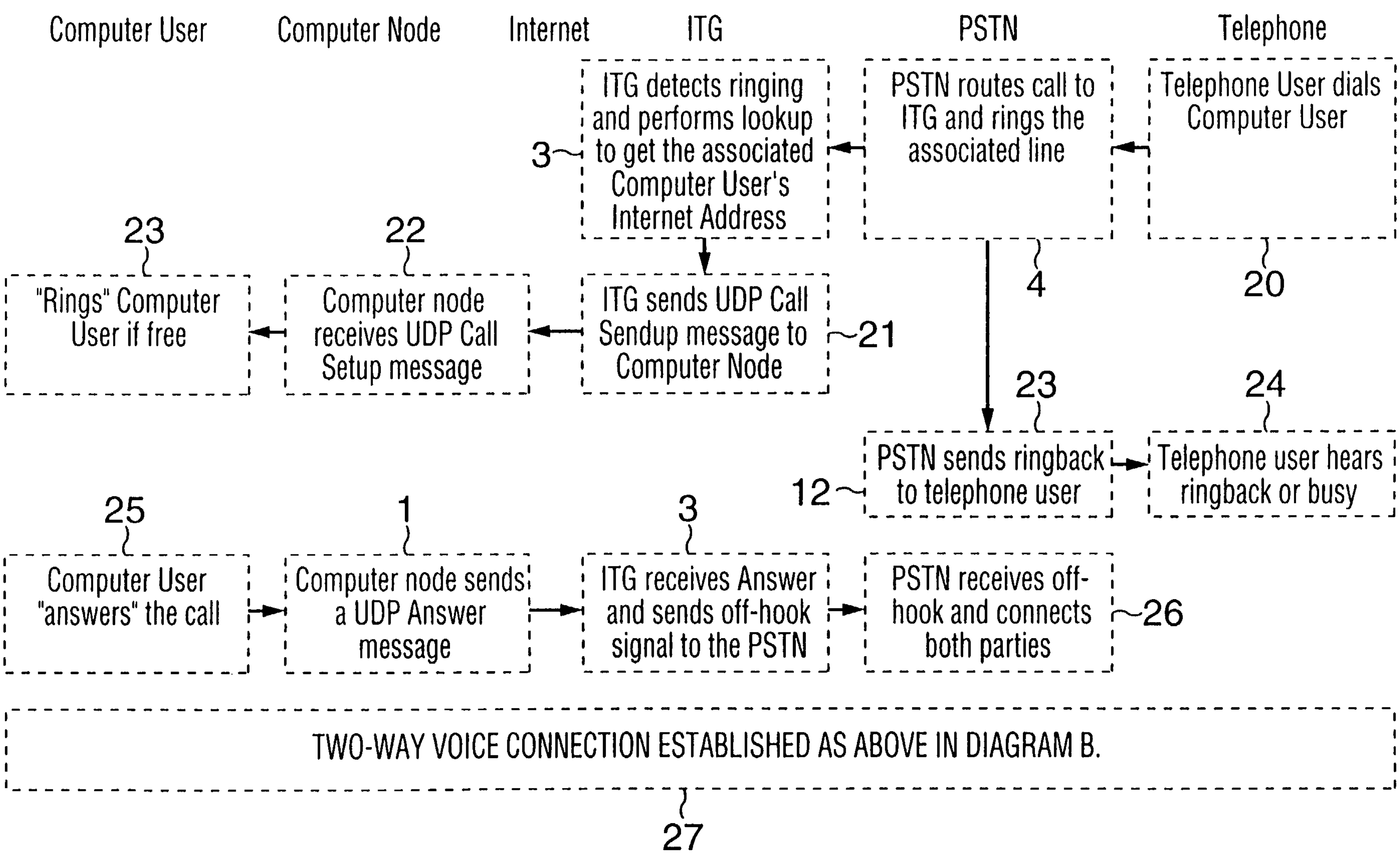
1
Computer User
Computer Node
2
INTERNET NETWORK
3
Internet Telephony Gateway (ITG) to the PSTN
4
PUBLIC SWITCHED TELEPHONE NETWORK
5
Telephone

FIG. 1



Computer User
Computer Node
Internet
ITG
PSTN
Telephone
User enters telephone number of called party
10
Computer node formats and sends UDP Call Setup message
11
ITG decodes Setup message and establishes 2-way speech path (see diagram D)
3
ITG goes offhook on the line interface assigned to computer user and dials called number
12
PSTN takes dialed digits and locates telephone user
13
PSTN rings phone if free and returns ringback or busy tone on the speech path
14
Computer User hears Ringback/Busy Indication from PST via the ITG
10
Telephone rings
5
Telephone user answers (assuming not busy)
11
Two-way conversation begins
15

FIG. 2



Computer User
Computer Node
Internet
ITG
PSTN
Telephone
Telephone User dials Computer User
20
PSTN routes call to ITG and rings the associated line
4
ITG detects ringing and performs lookup to get the associated Computer User's Internet Address
3
ITG sends UDP Call Sendup message to Computer Node
21
Computer node receives UDP Call Setup message
22
"Rings" Computer User if free
23
PSTN sends ringback to telephone user
12
23
Telephone user hears ringback or busy
24
Computer User "answers" the call
25
Computer node sends a UDP Answer message
1
ITG receives Answer and sends off-hook signal to the PSTN
3
PSTN receives off-hook and connects both parties
26
TWO-WAY VOICE CONNECTION ESTABLISHED AS ABOVE IN DIAGRAM B.
27

FIG. 3

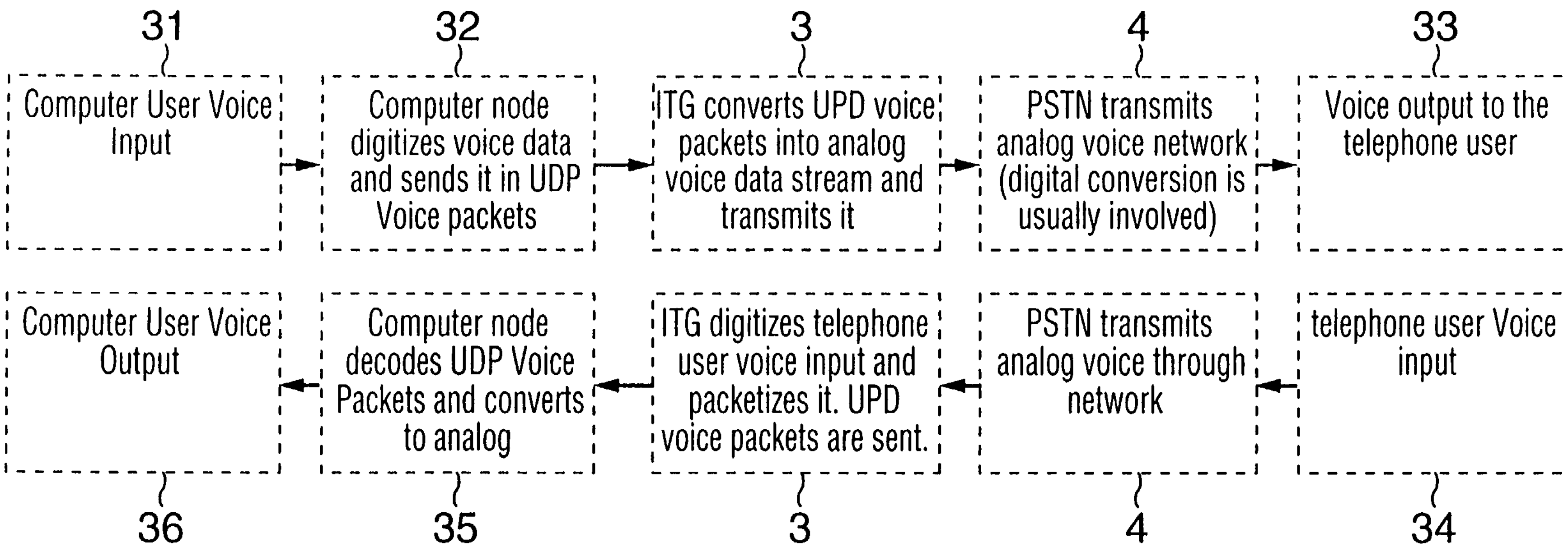

31
Computer User Voice Input
32
Computer node digitizes voice data and sends it in UDP Voice packets
3
ITG converts UPD voice packets into analog voice data stream and transmits it
4
PSTN transmits analog voice network (digital conversion is usually involved)
33
Voice output to the telephone user
Computer User Voice Output
36
Computer node decodes UDP Voice Packets and converts to analog
35
ITG digitizes telephone user voice input and packetizes it. UPD voice packets are sent.
3
PSTN transmits analog voice through network
4
telephone user Voice input
34

FIG. 4

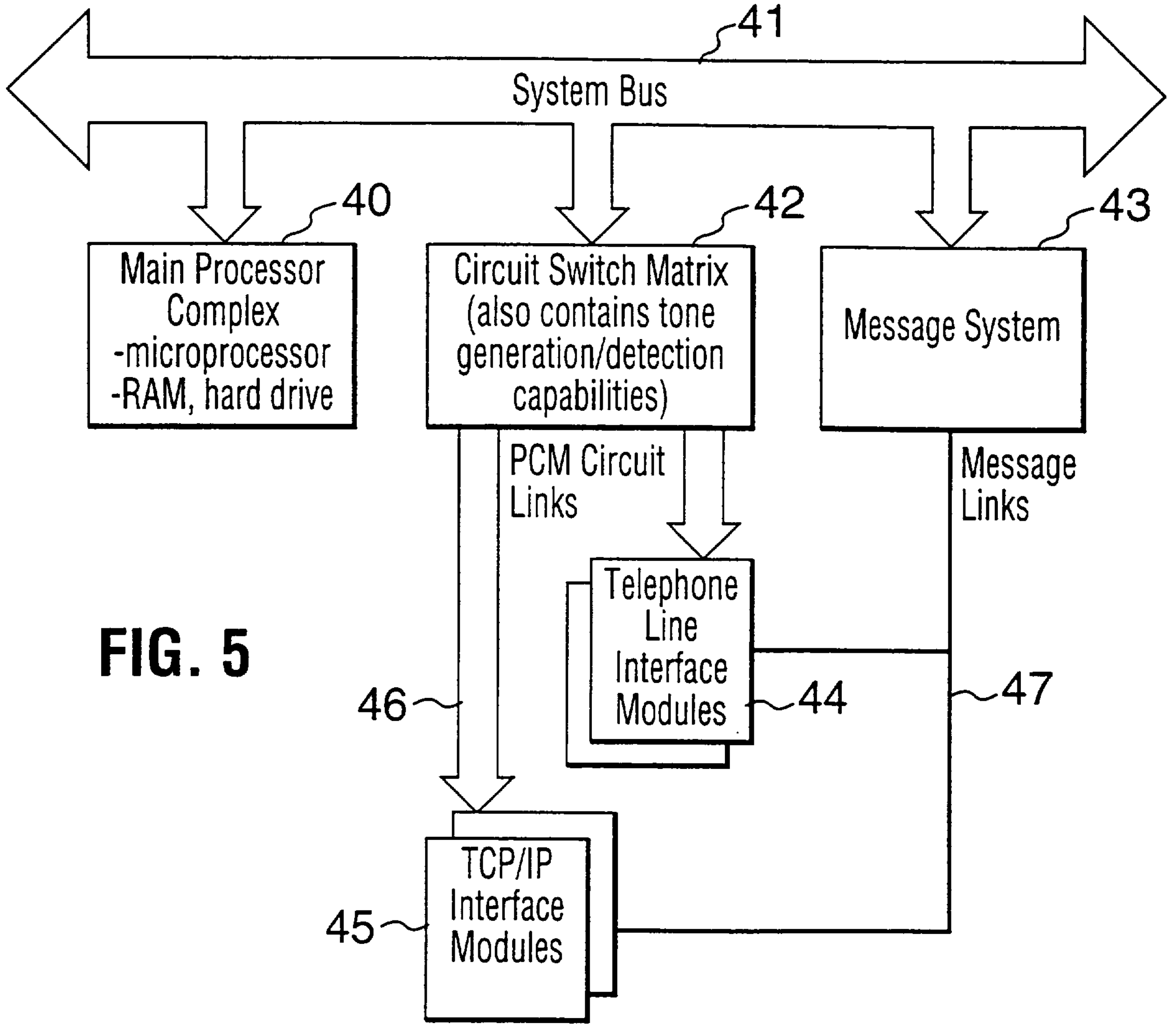

41
System Bus
40
Main Processor Complex
-microprocessor
-RAM, hard drive
42
Circuit Switch Matrix (also contains tone generation/detection capabilities)
43
Message System
PCM Circuit Links
Message Links
Telephone Line Interface Modules
44
46
47
TCP/IP Interface Modules
45

FIG. 5

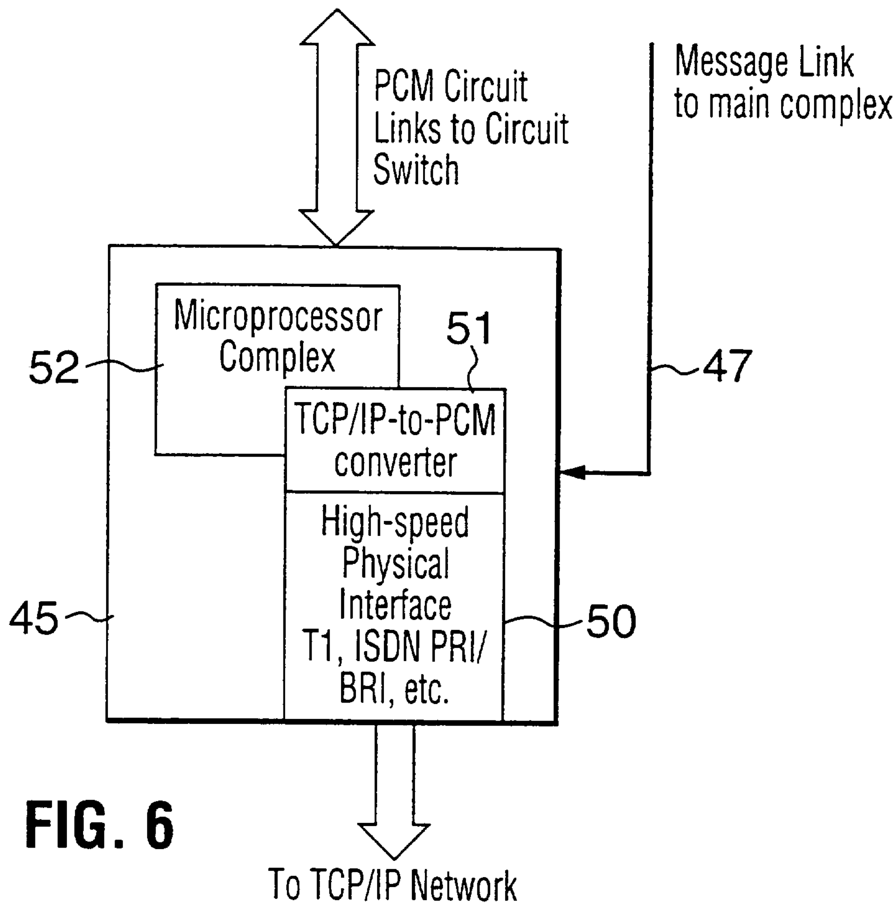

PCM Circuit Links to Circuit Switch
Message Link to main complex
Microprocessor Complex
51
52
47
TCP/IP-to-PCM converter
High-speed Physical Interface T1, ISDN PRI/ BRI, etc.
45
50
To TCP/IP Network

FIG. 6

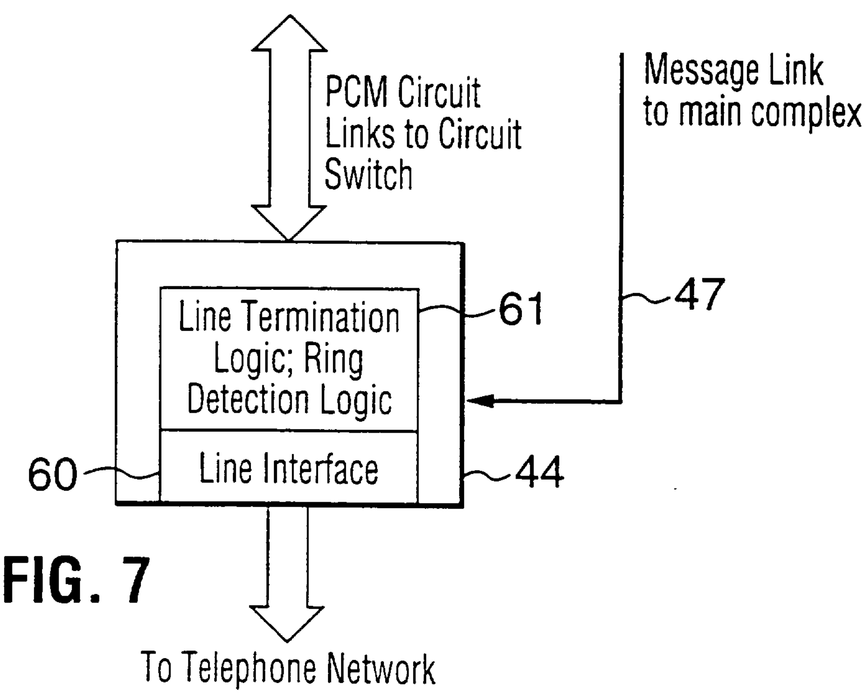
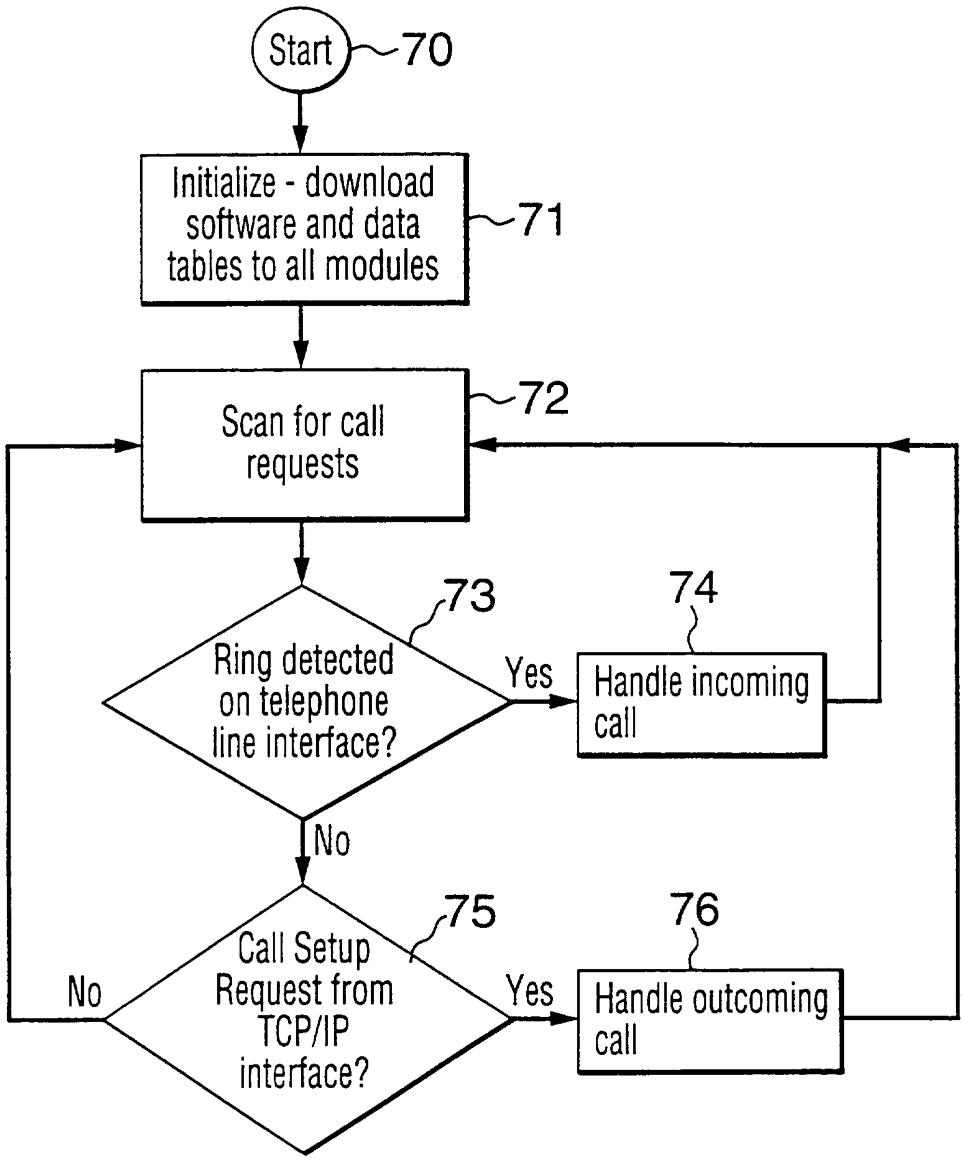
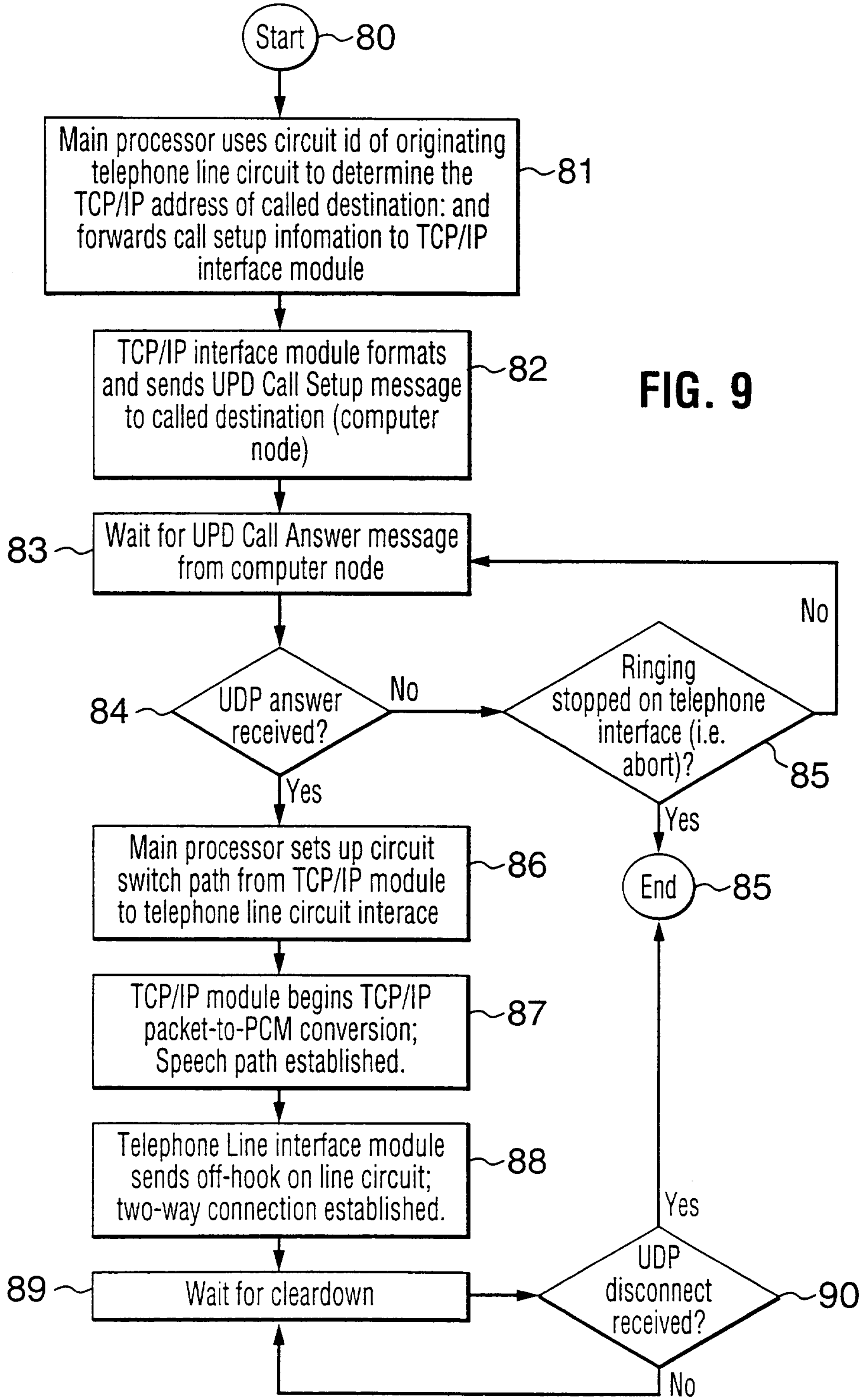

PCM Circuit Links to Circuit Switch
Message Link to main complex
Line Termination Logic; Ring Detection Logic
61
47
Line Interface
60
44
To Telephone Network

FIG. 7



Start
70
Initialize - download software and data tables to all modules
71
Scan for call requests
72
Ring detected on telephone line interface?
73
Yes
Handle incoming call
74
No
Call Setup Request from TCP/IP interface?
75
No
Yes
Handle outcoming call
76

FIG. 8



Start
80
Main processor uses circuit id of originating telephone line circuit to determine the TCP/IP address of called destination: and forwards call setup infomation to TCP/IP interface module
81
TCP/IP interface module formats and sends UPD Call Setup message to called destination (computer node)
82
Wait for UPD Call Answer message from computer node
83
UDP answer received?
84
No
Ringing stopped on telephone interface (i.e. abort)?
85
No
Yes
End
85
Yes
Main processor sets up circuit switch path from TCP/IP module to telephone line circuit interace
86
TCP/IP module begins TCP/IP packet-to-PCM conversion; Speech path established.
87
Telephone Line interface module sends off-hook on line circuit; two-way connection established.
88
Wait for cleardown
89
UDP disconnect received?
90
Yes
No

FIG. 9

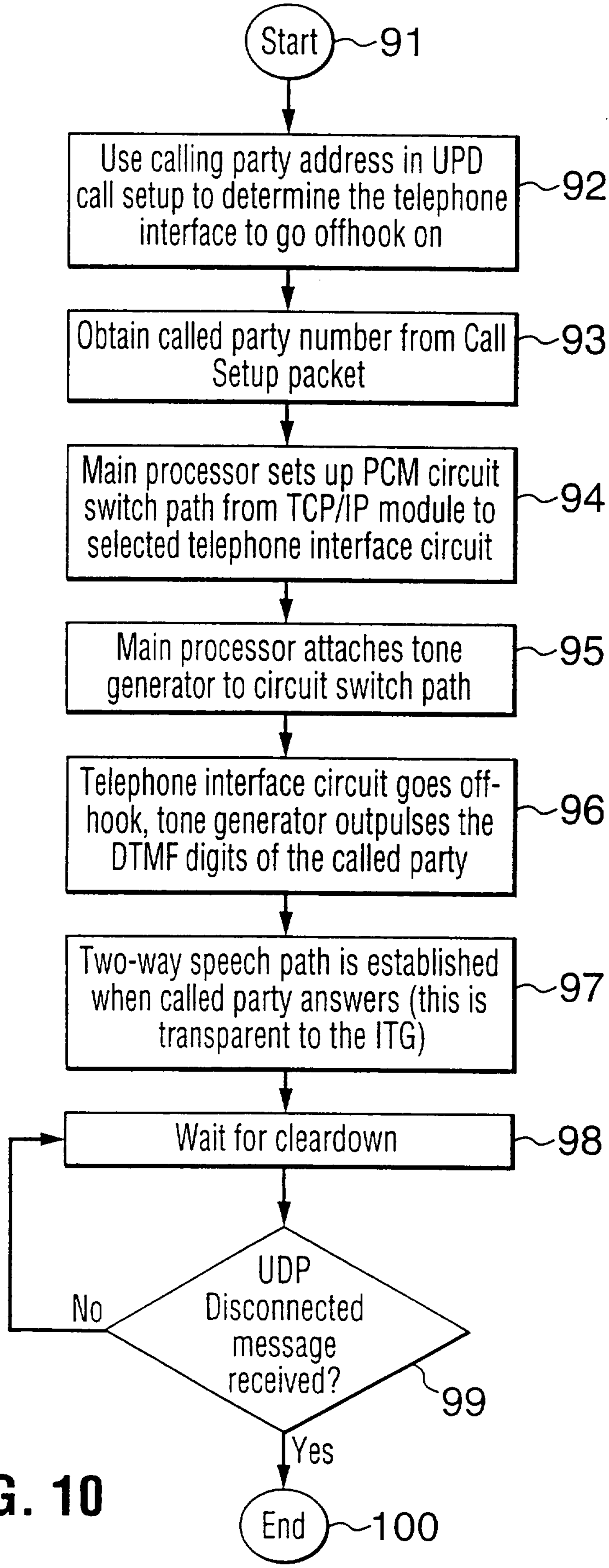

Start
91
Use calling party address in UPD call setup to determine the telephone interface to go offhook on
92
Obtain called party number from Call Setup packet
93
Main processor sets up PCM circuit switch path from TCP/IP module to selected telephone interface circuit
94
Main processor attaches tone generator to circuit switch path
95
Telephone interface circuit goes off-hook, tone generator outpulses the DTMF digits of the called party
96
Two-way speech path is established when called party answers (this is transparent to the ITG)
97
Wait for cleardown
98
UDP Disconnected message received?
99
No
Yes
End
100

FIG. 10

INTERNET TELEPHONY GATEWAY

This invention relates to an Internet Telephony Gateway, and more particularly to a gateway for permitting telephone calls to be made from an ordinary telephone to a subscriber on the Internet.

The PSTN (Public Switch Telephone Network) has enabled subscribers to make telephone calls over switch circuits for many years. The circuits are established at the time of call setup and remain dedicated to the communicating subscribers for the duration of the call.

More recently, the Internet has become a popular means of communication. The Internet consists of a multitude of interconnected networks each conforming to the TCP/IP protocol suites, so that users on the network can communicate. Unlike the PSTN, the Internet is a packet switched network, i.e. one in which data is carried in individually addressed packets from the near end to the far end.

The Internet is convenient for non-time critical data, such as file transfer and E-mail, although recently it has become popular for real time applications. It has long been possible to communicate in real time over the Internet using "chat servers". In such an arrangement, users communicate by entering data on their computer keyboards in real time. More recently, software has become available that in conjunction with an audio card permits audio to be exchanged in real time over the Internet. This enable subscribers to carry on "telephone conversations" in real time over the Internet.

The disadvantage of such software, however, is that it only enables communication between Internet subscribers. A receiving subscriber must have his computer connected to the Internet and running the appropriate software in the receive mode. An Internet subscriber cannot communicate with an ordinary POTS (Plain Ordinary Telephone Service) subscriber on the PSTN.

An object of the invention is to alleviate this disadvantage.

According to the present invention there is provided an apparatus for enabling the establishment of telephone calls between a packet switched network carrying data packets and a circuit switched network carrying telephone signals, comprising a first interface unit for connection to the circuit switched network, a second interface unit for connection to the packet switched network, means for receiving signals at said first interface and converting them to data packets for transmission over the packet switched network and vice versa and processing means for determining destination information from incoming signals or data packets on one said network and setting up a call to a destination on the other said network in response to said destination information.

The invention thus provides a gateway through which, for example, Internet users can call subscribers on the public switch telephone network and vice versa. In the case of a telephone subscriber wishing to call someone on the Internet, the subscriber calls the number of the gateway, which then establishes a virtual connection over the Internet with the user. In one embodiment each incoming circuit at the gateway is mapped to a user's TCP/IP address. Alternatively, the gateway can determine call party information from incoming signals on a trunk.

In the reverse direction, the computer user sends a message to the gateway requesting the establishment of a call to a telephone subscriber on the PSTN. The gateway sets up the call and the two can then communicate.

According to a second aspect of the invention there is provided a method of establishing calls between a packet switched network carrying data packets and a circuit switched network carrying telephone signals, comprising the steps of providing a gateway between the packet switched network and the circuit switched network, receiving data packets from the packet switched network, sending a call setup packet to said gateway to identify the called party on the circuit switched network, extracting the number of the called party from said call setup packet at said gateway, establishing a call from said gateway to the called number over the circuit switched network, and converting in real time signals from said circuit switched network to data packets for transmission over said packet switched network and vice versa while the call is in progress.

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

Figure 1 is a general block diagram of a system employing a gateway in accordance with the invention;

Figure 2 is a block diagram showing the establishment of a call from a computer user to a telephone subscriber;

Figure 3 is a block diagram showing the establishment of a call from a telephone subscriber to a computer user;

Figure 4 is a block diagram showing the speech paths;

Figure 5 is a functional block diagram of a gateway in accordance with the invention;

Figure 6 is a more detailed block diagram of a TCP/IP module;

Figure 7 is a block diagram of a telephone interface;

Figure 8 is a top level system flow chart;

Figure 9 is a flow chart showing the handling of an incoming call; and

Figure 10 is a flow chart showing the handling of an outgoing call to a PSTN.

The system shown in Figure 1 comprises a computer node 1 connected through the Internet 2 to a telephony gateway 3. Gateway 3 is connected through the public switched telephone network (PSTN) 4 to a individual telephone subscriber 5. Computer user at node 1 and telephone subscriber at telephone 5 can establish two-way voice communications in a manner that will described in more detail.

Figure 2 shows the initiation of a call from a computer user 10 to a telephone subscriber 11. Firstly, the computer node 1 formats and sends a UDP (User Datagram Protocol) message at 11 to the ITG 3. This UDP contains information about the number of the called party on the PSTN. The ITG 3 decodes this message, goes off hook at 12 and calls the user 11 by dialling digits on the PSTN as shown in block 13. At block 14, the PSRN rings the telephone 5, which if not busy, is answered by the user 11. From this point, two-way conversation can be established through the ITG as indicated by block 15. Of course, if the telephone 5 is busy, a ring back flash busy indication is sent from the PSTN 4 via the

ITG 3 to the user 10. From this point on, the system on the Internet side operates in the manner of a conventional Internet telephone, and on the PSTN side, operates in the manner of a conventional PSTN telephone. From the Internet side, the computer uses voices digitized and sent in packets to the ITG 3, where it is depacketized and forwarded over the PSTN to the telephone user 11, either in the form of analog or digital signals, for example PCM.

The establishment of a call from the telephone subscriber to the computer user is represented in Figure 3. As shown at block 20, the telephone dials the number of the ITG 3 and the call is routed by the PSTN 4. Based on the incoming circuit, or called party information contained within the incoming signals, the ITG 3, as shown at block 21, sends a packetized UDP call setup message which is received by the computer node 1 at block 22 and, in accordance with the computer node's software, "rings" the computer if the user is free (block 23). If the computer user is unavailable, a message is sent back through the system, which is picked up by the PSTN. As indicated by block 23, this sends a busy signal back to the user (block 24).

If the computer user answers the call (block 25), computer node 1 sends a UDP answer message to IDTG 3, which in turn sends an off hook signal to the PSTN, which is received by the PSTN 4 (block 26). Two-way voice connection is then established as represented by block 27.

Figure 4 is a representation of the system once two-way voice communication has been established. Block 31 shows the computer user inputting voice, which is digitized at the computer node at block 32 and send in UDP packets to the ITG 3 which converts the packets into an analog voice data stream (although it could be digital) and transmits it to the PSTN 4 which is then outputted at 33 to the telephone user. In the reverse direction user input 34 is transmitted through the PSTN to the ITD 3, which packetizes the voice input and sends it as UDP voice packets (block 35) to the computer node which outputs it as voice output 36.

Referring now to Figure 5, the ITG (Internet Telephony Gateway) comprises a main processor complex 40, for example a personal computer, connected to a system bus 41. The system bus is also connected to a switching matrix 42 and a messaging system 43. The switching matrix interconnects telephone line interface modules 44 and TCP/IP into phase modules 45. The switching matrix 42 is connected to the module for PCM (Pulse Code Modulator) links 46. The messaging system 43 is connected to the interface modules over message links 47.

The main processor 40 is responsible for the high level control of the ITG. It issues instructions over the system bus to the messaging system 43, which in turn sends control message to the interface modules 44 and 45 to direct the operations of these modules in order to setup and tear down calls.

The interface modules 44, 45, serve as slaves to the main processor 40. Changes in call state (for example, originations, disconnects) are reported to the main processor 40 over the message links 47.

Calls originating on the Internet are routed to TCP/IP interface module 45. The incoming data packets are depacketized and routed through the switching matrix to the appropriate outgoing telephone line module 44, where they are sent over the PSTN as either analog or digital signals. The system works in a similar manner in reverse.

The TCP/IP module is shown in more detail in Figure 6. This consists of a high speed physical interface, for example T1, ISDN, PRI/BRI, etc. interface 50 to the TCP/IP network, which would normally be the Internet. This is connected to a TCP/IP-to-PCM converter 51, the entire unit being controlled by a microprocessor 52. The microprocessor includes local RAM and runs software supporting TCP/IP and the connection protocols to the network. As shown in Figure 7, the telephone interface unit comprises a line interface 60, which can be conventional telephone line interface, and a line termination logic unit 61, which is a standard interface unit that detects ringing and the like.

The operation of the system will be understood better by referring to the flow charts shown in Figure 8 to 10. Referring to Figure 8, after a start 70 the system initializes and downloads software and data tables to all modules 71. The system then scans for call request 72. Block 73 detects a ring on the telephone line interface and if a ring is detected control is handed to block 74, which will be described in more detail with reference to Figure 9. If no call is detected on the line interface, the system looks for a call setup request from the TCP/IP interface at 75, and if a call is found, control is handed over to block 76 (which will be described in more detail with reference to Figure 10). If no call request is detected, the system loops back to block 72, and the cycle repeats.

Block 74 is shown in more detail in Figure 9. After start-up 80, the main processor 40 uses the circuit idea of the originating telephone line circuit to determine the TCP/IP address of the call destination, and forwards call setup information to the TCP/IP interface as represented by block 81. The TCP/IP interface module formats and sends a UDP (User Datagram Protocol) call setup message to the call destination computer 82. The establishment of a call across the Internet in this manner is *per se* known.

The ITG then waits for a UDP answer message from the computer node 83. Decision unit 84 then determines whether an answer is received. If not, control is passed to unit 85 which determines whether ringing should be stopped (based on the time period). If no, the system loops back to block 83. If yes, the call setup mode terminates as shown at 85. If an answer is received, decision unit 84 passes control to the main processor as represented by block 86, which then sets up a circuit switch path through the switching matrix 42 (as shown by block 86) and the TCP module begins TCP/IP packet to PCM conversion at shown at 87. As shown at 88, the telephone line interface module sends an off hook signal to the subscriber telephone, and two-way connection is established. The system then goes into a wait mode for teardown as shown at 89. The system loops through 90 to determine whether a UDP disconnect message is received from the computer user. If yes the system terminates as shown at block 85.

Figure 10 illustrates block 76 in more detail, which handles a call setup request from the TCP/IP interface. The system starts at 91 and uses the calling party address contained in the UDP call setup message to determine the telephone interface that it is desired to go off hook on 92. The called party number is obtained from the call setup packet 93, and the main processor sets up a PCM circuit switch path from the TCP/IP module to the selected telephone interface circuit at block 94. The main processor attaches a tone generator to the circuit switch path at 95. The telephone interface circuit then goes off hook at 96, and two-way speech is established when the called party answers at 97. The system then goes into a wait for teardowm routine at 98. When a UDP disconnect message is received, decision unit 99 terminates the call at 100.

In the embodiment described, there is a one-to-one mapping between each telephone line interface circuit and a computer user's TCP/IP address. In other words, each computer user is allocated a telephone line at the ITG. The computer user can thus give out his ITG telephone number, and any party calling that number will be connected to the computer user associated with that number. In the reverse direction, however, the computer user will be connected to the associated telephone line, but the ITG will dial any number requested by the computer user.

In an alternative embodiment, the ITG can be connected to a trunk, in which case call party information received on the trunk can be used to determine the TCP/IP address of the destination computer user.

It is assumed in the above description that the computer has a software sub-system that implements some form of Internet telephony capability. Such software is currently conventionally available and translates the actions of the computer user into UDP messages. The User Datagram Protocol is a convenient method of exchanging messages, but a similar low overhead transmission technique (such as XTP/Express Transfer Protocol), can be employed instead.

In the above, if the above system is described in a situation where the computer user wishes to terminate the conversation, the computer node sends a UDP disconnect message to the ITG 3. If the telephone user terminates the conversation, the telephone network terminates the connection and the computer user just hears the dial tone coming from the network. The computer user can then disconnect himself from the network or initiate another call.

In effect, the system allows the Internet to serve as a means of permitting a computer user to establish two-way communication with an assigned remote telephone line interface. For example, conceivably, a computer user in Ottawa, Canada who has reason to make a substantial number of calls in the Los Angeles area could subscribe to an ITG service in the Los Angeles area. This would in effect give the computer user in Ottawa access to a local line for Los Angeles over the Internet. The computer user could dial out from his line in Ottawa as if he were physically present with a telephone in Los Angeles.

It has been mentioned that it is possible to substitute a digital trunk interface for the line interface described above. This approach has the advantage that the conversion of digital voice data to analog voice data is no longer necessary, and it is also no longer necessary to have a dedicated line interface per computer user.

On the other hand, the call setup procedures are different and require the ITG to generate ring and busy tones for calls going to the computer user, depending on the state of the user. On a trunk interface, the ITG must also be prepared to received dial digits and must be able to send calling party directory numbers to the public network. For a call from a telephone subscriber to a computer, the ITG has to do the following (assuming a trunk interface). The ITG receives call party digits over the trunk and must translate these into the associated computer user TCP/IP address. He must then attempt to establish voice communication with the computer user and place a busy or ring back tone on the trunk depending on whether the computer user is free or already involved in another call.

Disconnect procedures are essentially the same as described above. The computer user signals a disconnect to the ITG using a UDP and the ITG forwards a disconnect instruction to the PSTN.

For a call from the computer user to the telephone subscriber, the computer user sends the called party's telephone number to the ITG in a UDP setup message. The ITG initiates the call over the trunk to the public telephone network using standard trunk procedures. This involves sending the calling party's directory number to the network so that the correct billing procedures can be followed. A voice pass is established at the same time to the network so that the computer user can hear the call progress tones (ring/busy) being supplied by the network. An end-to-end path is set up by the telephone network when the subscriber answers the calls.

The described system thus offers significant advantages over the prior art, in that it allows effective voice communication between computer users on the Internet and conventional telephone users on the PSTN.

The system has been described with reference to voice signals, although of course it would work with any signals capable of being carried over the PSTN, such as fax or even data signals.

Claims

1. An apparatus for enabling the establishment of telephone calls between a packet switched network carrying data packets and a circuit switched network carrying telephone signals, comprising:

a first interface unit for connection to the circuit switched network;

a second interface unit for connection to the packet switched network;

means for receiving signals at said first interface and converting them to data packets for transmission over the packet switched network and vice versa; and

processing means for determining destination information from incoming signals or data packets on one said network and setting up a call to a destination on the other said network in response to said destination information.

2. An apparatus as claimed in claim 1, wherein said first interface unit comprises a TCP/IP interface unit for sending and receiving data packets over the packet switched network and said second interface unit is a telephone line interface for connection to the public switched telephone network.

3. An apparatus as claimed in claim 2, further comprising a message system for sending and receiving control messages to said interface units under the control of said processing means.

4. An apparatus as claimed in claim 3, further comprising a circuit switching matrix connected between said first and second interface units.

5. An apparatus as claimed in claim 1, wherein said processing means sets up a call over the packet switched network based on the incoming circuit on said circuit

switched network, the addresses on said packet switched network being mapped to one-for one to circuits on said circuit switched network.

6. An apparatus as claimed in claim 1, wherein said processing means extracts the destination information from the called party information carried on the incoming signals over the circuit switched network.

7. An apparatus as claimed in claim 2, wherein said signals are PCM (pulse coded modulation) signals.

8. An apparatus as claimed in claim 7, wherein said second interface unit is a primary rate TDM interface.

9. A method of establishing calls between a packet switched network carrying data packets and a circuit switched network carrying telephone signals, comprising the steps of:

providing a gateway between the packet switched network and the circuit switched network;

receiving data packets from the packet switched network:

sending a call setup packet to said gateway to identify the called party on the circuit switched network:

extracting the number of the called party from said call setup packet at said gateway;

establishing a call from said gateway to the called number over the circuit switched network; and

converting in real time signals from said circuit switched network to data packets for transmission over said packet switched network and vice versa while the call is in progress.

10. A method as claimed in claim 9, wherein said packet switched network is a TCP/IP network, and said circuit switched network is the public switched telephone network

11. A method as claimed in claim 9, wherein the call progress is controlled by a main processor directing operations by control messages.

12. A method as claimed in claim 9, wherein said signals are PCM signals.

13. An apparatus for enabling the establishment of telephone calls, substantially as herein described with reference to the accompanying drawings.

14. A method of establishing calls, substantially as herein described with reference to the accompanying drawings.

15

Application No: GB 9714058.6
Claims searched: 1-14
Examiner: Peter Slater
Date of search: 25 September 1997

Patents Act 1977
Search Report under Section 17

Databases searched:

UK Patent Office collections, including GB, EP, WO & US patent specifications, in:

UK Cl (Ed.O): H4K (KTM , KTK); H4P (PPA)

Int Cl (Ed.6): H04L 12/64 , 12/66

Other: ONLINE: WPI

Documents considered to be relevant:

Category	Identity of document and relevant passage		Relevant to claims
X	EP 0238984 A2	(AT & T) - See whole document	1 & 9 at least
X	EP 0147197 A1	(AT & T) - See whole document	1 & 9 at least
X,P	WO 97/23078 A1	(MCI COMMUNICATIONS) - See whole document	1 & 9 at least
X	WPI Abstract Accession No. 95-267646/199535 & JP7170288 A (Hitachi) 04.07.95 (see abstract & US5604737 A)		1 & 9 at least

X	Document indicating lack of novelty or inventive step	A	Document indicating technological background and/or state of the art.
Y	Document indicating lack of inventive step if combined with one or more other documents of same category.	P	Document published on or after the declared priority date but before the filing date of this invention.
&	Member of the same patent family	E	Patent document published on or after, but with priority date earlier than, the filing date of this application.