Development of mobile IP phone ”Mobile IP Talk”

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Abstract

The number of Internet telephone users is gradually increasing because of widespread broadband (ADSL, FTTH, CATV, etc.) Internet access services. We started our new Internet telephony service ”IP Talk” in 2002.

Recently Wireless LAN (WLAN), as one of leading network technologies, is introduced as Internet telephony for mobile users.

We have developed and evaluated mobile IP phone terminals, ”Mobile IP Talk” since 2001. ”Mobile IP Talk” could achieve continuous call waiting time performance of about one day by power managing function of WLAN interface, and moreover solve the IP address mobility problem by our original IP telephony protocol HTTP-based Conference Application Protocol (HCAP) to handle with DHCP, NAT and firewall problems of Voice over Internet Protocol (VoIP).

We evaluated basic telephony functions of ”Mobile IP Talk” and found it achieved good performance of practical use.

In this paper, we describe the concepts, hardware structure, the calling processes, protocol and basic evaluation.

keywords: Wireless LAN, Mobile IP Phone, VoIP, Internet telephony, HTTP, Firewall, NAT, DHCP

1. Introduction

Several Internet telephone services have been started since the beginning of 21th in Japan. They use specially designed VoIP adapters with each existing VoIP standard as H.323, SIP, and MGCP. We have been operating our Internet telephone network system ”IP Talk” [1][2] since June, 2002, which is independent from ISPs and broadband lines, and available even in private network. Our services are provided by small VoIP adapters. In our laboratory, we developed some VoIP adapters which can be connected to fixed IP network.

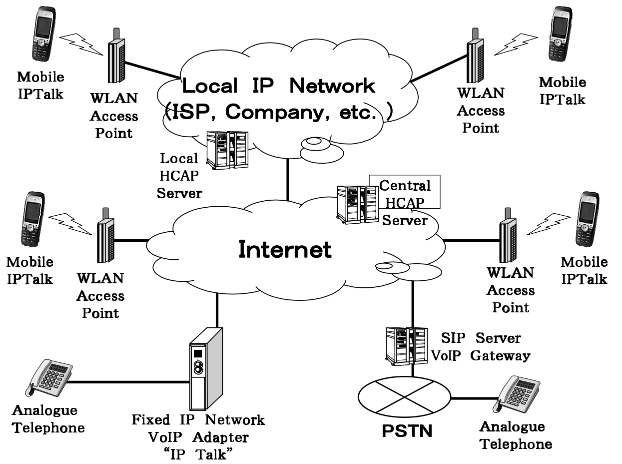


Figure 1. Mobile IP Talk Network

On the other hand, users can utilize WLAN (especially, IEEE 802.11 standard [3] [5] [4] [6]) equipments easily at home, offices, and public access points around train stations, coffee shops, hotels, and so on. These techniques created some of Internet access experiments for mobile users by PDAs, Notebook PCs, WLAN interfaces and WLAN access points connected to the Internet.

Several manufacturers provide mobile IP phone terminals with 802.11b WLAN standard interfaces[7][8][9]. These products are usually utilized by assignment of fixed IP addresses in restricted area as company buildings, because the WLAN interface’s power consumption is too large to use for as long a time as existing cellular phones, and the IP address mobility problem doesn’t allow wide-ranging movement.

After the above development, we developed a mobile VoIP terminal with a WLAN interface which is connected to IP network(Figure 1), called ”Mobile IP Talk”. We plan to provide ”Mobile IP Talk” terminals as extended wireless phones for company office users at first, because WLAN access points can be set up only in the restricted area. Though we don't estimate that mobile IP phone will take the place of cellular phone in a few years, we consider to make use of "Mobile IP Talk" around public WLAN access points.

We developed two versions of hardware which are made up of general integrated circuits modules without customized ones like existing cellular phone's parts. These have a WLAN card interface. The first version has the Compact Flash (CF) card type, and the second version has the Secure Digital Input/Output (SDIO) card type. Especially the second version accomplished more than 20 hours call waiting time performance by the power managing function of WLAN cards.

Our mobile IP telephony protocol for "Mobile IP Talk" is the same as our protocol for fixed IP connection telephony, which is called HCAP[10]. HCAP solved DHCP, NAT and firewall problems by applying HTTP [11] technology for signaling calls and transmitting voice signals over the Internet. Additionally HCAP solved the mobility problem of IP address for all-round Internet access environment, too.

2. VoIP over WLAN related work

2.1. VoIP standard

Multimedia conference standardization H.323 was established as 1st version[12] in 1996 and 2nd version[13] in 1998 for low bit rate communication network by International Telecommunication Union - Telecommunication Standardization Sector (ITU-T).

In 1999, Session Initiation Protocol (SIP)[14] was established as a new VoIP protocol by Internet Engineering Task Force (IETF), too. SIP is easier to be implemented and extended than H.323 generally because of text-base like HTTP. Recently SIP is becoming the mainstream of VoIP.

On the other hand, MGCP and megaco were established as VoIP protocols for large scale IP telephone network connected to PSTN by IETF. MGCP was adopted for VoIP on CATV.

2.2. VoIP problems

Recently, we can not get fixed global IPv4 addresses from ISPs easily. Most ISPs assign IP global or private addresses dynamically by Dynamic Host Configuration Protocol (DHCP) servers. It is the problem that caller terminals can not designate the IP address of the callee terminals when calling process starts with the H.323 or SIP Protocol, because IP address information for user's location must be registered in gatekeepers and location servers.

Another problem is related to private IP address like Local Area Network (LAN). There are Network Address Translation (NAT) routers in companies and homes. Accordingly some methods such as static IP masquerade configuration or Universal Plug and Play (UPnP)[31], etc. can solve this problem, but the cost to change the configuration is inevitable.

In H.323 and SIP, voice over User Datagram Protocol (UDP) is basically considered to integrate Internet telephone network system. Real-time Transport Protocol (RTP) is adopted as the upper layer on UDP to control real time performance, which is prescribed in H.225.0[32], [33] of H.323. A hurdle, therefore, to conventional VoIP system is that most firewalls do not allow UDP ports to be used as the means to be connected to outside.

Some solutions for these problems[34],[35], [36],[37],[38] were proposed in IETF SIP WG. Recently, IETF MIDCOM WG is researching architectures and communication protocols for policy control of the middle box, which has the functions of NAT and firewall[39],[40].

2.3. IP address mobility problems

It is easy to designate VoIP terminal's IP addresses in a sub-network with some WLAN access points, which can manage IP-based sessions even in cases of terminal's movement among themselves. 802.11 WLAN access points support internal sub-network roaming and hand-over functions, which is handled at the link layer (layer 2 in OSI model) by the 802.11 implementation. This mobility support is called Micro-mobility.

On the contrary, it is difficult to designate IP addresses over multiple sub-network. So, Mobile IP, H.323 and SIP try to support wide-range mobility in VoIP services [15]. This mobility support is called Macro-mobility. Mobile IP is an approach of the network layer (layer 3), [16] [17] [18] [19], while H.323 and SIP are approaches of the application layer (layer 7)[20] [21].

The 3GPP (Third Generation Partnership Project) is designing specifications for a third generation mobile system. The 3GPP is using IP technology end-to-end to deliver multimedia contents to mobile terminals by introducing SIP as the call control and signalling functions[22] [23].

2.4. Internet telephone quality regulation

We cannot guarantee quality of service (QoS) when the network system utilizes best effort public Internet. Criterions of QoS of VoIP have been discussed in standardizing organizations like ETSI/TIPHON. The regulation "TR101 329-2" of WG5 in ETSI/TIPHON[24] says that end-to-end delay for Internet telephony service should be less than 400 msec.

Referring to quality classification of IP telephone in ITU-T, ETSI/TIPHON, and Telecommunications Industry

Association (TIA), the simmilar report ”Towards Fundamental Growth in IP Telephony Services” was announced by the ”Study Group on IP Network Technology” in Ministry of Public Management, Home Affairs, Posts and Telecommunications of Japanese government on Feb. 22th, 2002[25].

In 2004, the Ministry and TTC organized the special working group to regulate the QoS of mobile IP phone over WLAN and published the first proposal[27]. This refers the standard of Association of Radio Industries and Businesses (ARIB)[26] to define the evaluation space of WLAN.

3. Basic design of ”Mobile IP Talk”

3.1. Hardware

”Mobile IP Talk” was desinged on the base of a small embedded terminal for ubiquitous network, which is called μ T-Engine [28] in Japanese TRON project. Figure 2 shows the hardware structure. The first and second version was developed in 2002 and 2003. The second one was improved for the mass-produce version in 2004.

The most important difference between the first and the second is WLAN interface. The both have an card slot for a general WLAN interface card on the market in order to exchange various types of wireless communication interface cards besides IEEE 802.11. The first adopted CF card type, and the second adopted SDIO card type.

The other circuits of the both except WLAN interface are almost same. The both load an embedded CPU (Mitsubishi M32R/E; 216MHz), a DSP (Texas Instruments C5409), a cellular phone type keyboard, a small color display(LCD), a microphone, a speaker, a rechargeable battery and so on, and support G.729A[29] and G.711 codecs with the silence suppression mechanism.

The CPU has sleep mode to decrease executing clock frequency. And WLAN chip module in SDIO WLAN card[30] also has sleep mode to reduce power consumption. The WLAN sleep mode is prescribed by IEEE 802.11. That’s why the second can control electric power consumption by changing to sleep mode for a call waiting time.

3.2. Software

Figure 3 shows the software structure and the appearance. The operating system is μ ITRON for embedded computers, which supports multitasking. When the idle task on the CPU runs, μ ITRON changes it to the sleep mode.

The application software includes basic phone functions, graphical user interface, Phonebook, E-mail, Web browser, and IP telephony protocol stacks (HCAP and SIP). These

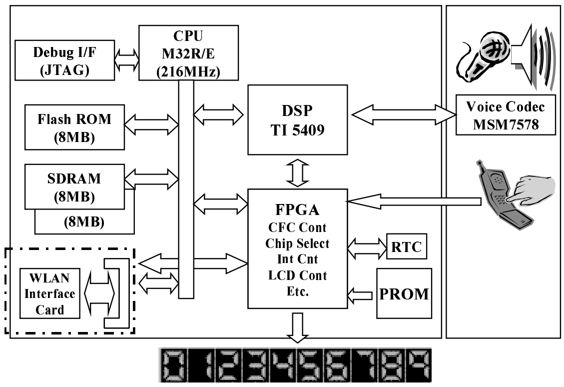


Figure 2. Hardware Structure

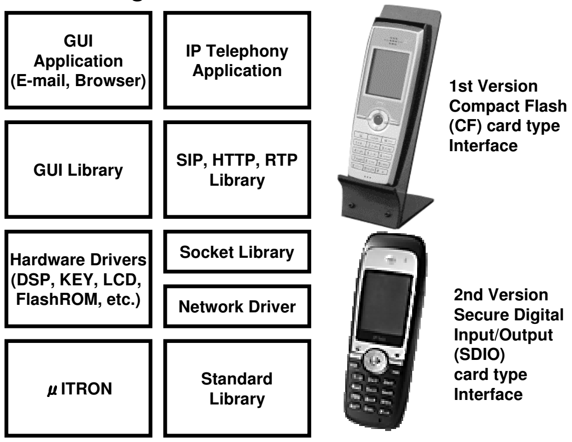


Figure 3. Software Structure and Appearance

stacks are almost same as the fixed connection terminal’s software. HCAP is described in the following section.

3.3. Connectivity control for various WLAN access points

The main purpose with mobile IP phones is to replace existing cellular phones. Mobile IP phones must be able to access any WLAN access points to surpass public cellular access. At first, the restricted access environments are like companies and homes, where we can setup static IP masquerade configuration of broadband routers, and mobile IP phones can surely transmit voice data by UDP. However, any access points might have different access identification. The next ones are public access points at train stations, coffee shops, hotels, and so on, where access permission and the transmission of UDP are restricted by circumstances. In conclusion, ”Mobile IP Talk” was designed to have several WLAN access identification and change automatically by referring to the personal identification database (Figure 4).

WLAN Access Point	SIP OK or NG	Access Identification	
Company			
Head Office (Tokyo)	OK	802.11a: ESS, WEP 802.11b: ESS, WEP	Com.User ID,Password ID unnecessary
Factory (Osaka)	NG	802.11b: ESS, WEP	ID unnecessary
Public			
Train Station (Vender A)	OK	802.11g: ESS, WEP	ISPUser ID,Password
Coffee Shop (Vender B)	NG	802.11b: ESS	ISPUser ID,Password
Home	OK	802.11b: ESS	ID unnecessary
Free	NG	802.11b: Nothing	ID unnecessary

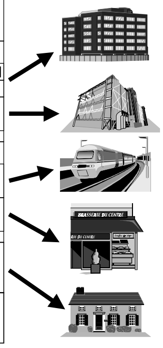


Figure 4. WLAN Access Points Identifications Management

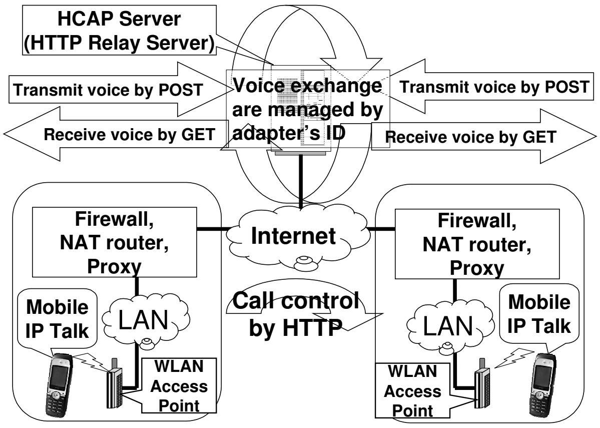


Figure 5. IP Telephony Structure with HCAP Server(HTTP Relay Server)

4. HTTP-based Conference Application Protocol

Everyone can use mail related protocols(SMTP,POP3) and Hyper Text Transfer Protocol (HTTP)[11] without any configuration changes to routers and firewalls. We invented a solution for call management and stream data transfer by HTTP. In our solution, our VoIP terminals work as HTTP clients managed by our HTTP relay servers, because the VoIP terminals themselves can not transmit call control data and stream data each other directly over NAT routers and firewalls without their configuration change. Even if a VoIP terminal's IP address is changed by DHCP, HTTP relay servers can recognize the terminal's location as a HTTP client. We named this protocol as HTTP-based Conference Application Protocol (HCAP) as an application layer VoIP protocol. Figure 5 shows the basic structure of HCAP.

4.1. HTTP relay server

Every VoIP terminal sets up logical sessions, and transmits call control data and stream data to another terminal via HTTP servers, which are called HTTP relay servers. All communications between terminals and HTTP relay servers are carried out mostly by HTTP, whereas in the case of stream data transmission by UDP like RTP. There are three different kinds of servers involved in the process.

Location managing server is similar to a SIP proxy, manages calling procedures between terminals. Each terminal has one, single session managing server to support all of its calling process, both inbound and outbound. It is placed in DeMilitarized Zone (DMZ) of local domains.

Session managing server relays streaming data between terminals. It is placed in DMZ of local domains, too.

Stream relay server assigns each terminals to a session managing server in the setup process of the terminal, and is similar to a DNS server.

Call control communications among domains are processed by session managing servers, whose global IP addresses are registered in database of location managing servers.

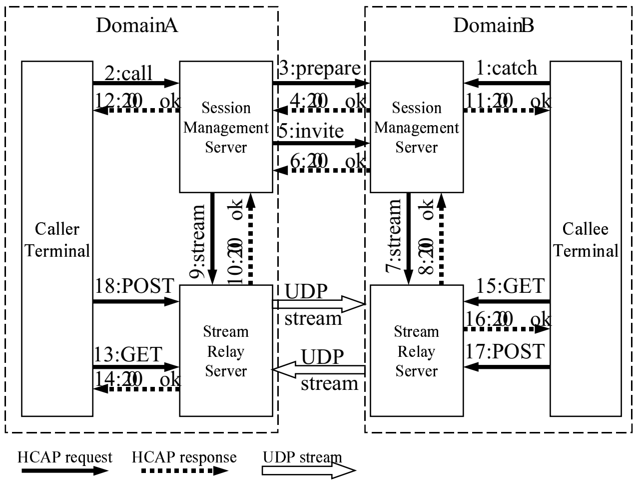


Figure 6. Basic Example of HCAP Call Control

4.2. Basic call control and Basic Method CGI

Terminals transmit call information to the session managing server by GET method of HTTP, which is a normal

BMCGI	GET from	GET to	Message
register.cgi	CTM	LMS	registration of terminal
call.cgi	CTM	t_SMS	call from caller
prepare.cgi	t_SMS	r_SMS	exchange information
invite.cgi	t_SMS	r_SMS	transmit call information
catch.cgi	CTM	SMS	make catch session
cancel.cgi	CTM	t_SMS	cancel call
bye.cgi	CTM	SMS	terminate call
stream.cgi	SMS	SRS	control stream communication
signal.cgi	CTM	SMS	signal teminal information
dbcom.cgi	all nodes	all nodes	operate database

Table 1. Basic Method CGI (BMCGI)

GET message with Common Gateway Interface (CGI) defined originally on HCAP. We call it Basic Method CGI (BMCGI) to do call control.

Figure 6 and Table 2 shows a basic call flow. Table 1 shows typical BMCGI definition (t_ : caller-side, r_ : callee-side, CTM : Communication Terminal, LMS : Location Managing Server, SMS : Session Managing Server, SRS : Stream Relay Server).

Every terminal transmits a GET method with catch.cgi to the SMS to receive calling information from another terminal at its start up time. Keeping permanent HTTP sessions with each terminal alive, the SMSs can identify terminals' current IP addresses used in transmitting and receiving stream data.

When a terminal calls to another terminal, it transmits a GET method with call.cgi to the SMS. In the case that SMSs manage the caller terminal and the callee terminal belong to different domains, the caller SMS transmits a GET method with invite.cgi to the callee SMS. In the next step, the callee SMS transmits a status response of the GET method with catch.cgi to the callee terminal to inform that it is being called, which starts ringing on the callee terminal. When it is off-hooked by the user, it transmits a GET method with signal.cgi. The status information is relayed via SMSs to the caller terminal. Then stream transmission starts between terminals. If terminals are located in different domains, stream data is relayed via SRSs which are to be assigned by the SMSs.

4.3. Call control and telephone handling

In the call process, "Mobile IP Talk" exchange the terminals' identification numbers by SMSs. "Mobile IP Talk" transmits GET method and receives server response when it starts calling and confirms being called. Figure 7 shows the basic call process by HCAP.

(1) A caller terminal starts calling by transmitting a calling message with the terminal's identification number to a SMS.

Step	Transmitter	Receiver	Message
1	r_CTM	r_SMS	catch.cgi
2	t_CTM	t_SMS	call.cgi
3	t_SMS	r_SMS	prepare.cgi
4	r_SMS	t_SMS	200 OK for prepare.cgi
5	t_SMS	r_SMS	invite.cgi
6	r_SMS	t_SMS	200 OK for invite.cgi
7	r_SMS	r_SRS	stream.cgi
8	t_SRS	t_SMS	200 OK for stream.cgi
9	t_SMS	t_SRS	stream.cgi
10	t_SRS	t_SMS	200 OK for stream.cgi
11	r_SMS	r_CTM	200 OK for catch.cgi
12	t_SMS	t_CTM	200 OK for call.cgi
13	t_CTM	t_SRS	GET for receiving stream
14	t_SRS	t_CTM	200 OK for GET
15	r_CTM	r_SRS	GET for receiving stream
16	r_SRS	r_CTM	200 OK for GET
17	r_CTM	r_SRS	POST for transmitting stream
18	t_CTM	t_SRS	POST for transmitting stream

Table 2. Basic process of HCAP

(2) A callee terminal has already transmitted a GET method to confirm being called to the SMS. When the server transmits the GET status response to the callee terminal to show being called, it makes the terminal start ringing.

(3) The terminal's ringing makes a user off-hook the terminal, whereby it transmits a GET method with off-hook information to the SMS.

(4) The caller terminal has already transmitted a GET method to confirm being off-hooked. It receives that information as the GET status response.

(5) Both terminals start transmitting voice data to each other.

When the callee user doesn't off-hook, the caller terminal transmits the GET method of cancel.cgi to the SMS by the caller user's on-hook. That cancel information is transmitted to the callee terminal via the SMS.

After some trials, we found that HTTP sessions sometimes tend to be disconnected regardless of the HTTP keep-alive mechanism. However, the SMSs transmit dummy data to the respective "Mobile IP Talk" in order to keep the HTTP sessions for callee confirmation.

4.4. Macro-mobility for sub network cross over

HCAP can solve Macro-mobility problems as the application layer protocol like DHCP problems, because all terminals' network positions are managed by their identification. Figure 8 shows the IP address maintenance mechanism. The process is discribed briefly as follows.

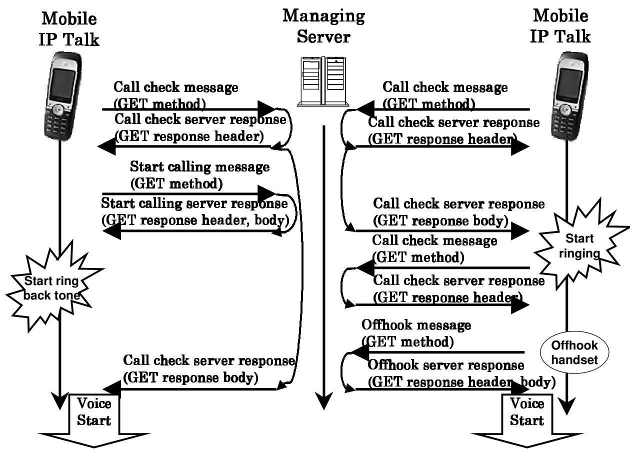


Figure 7. Call Control with Session Managing Server

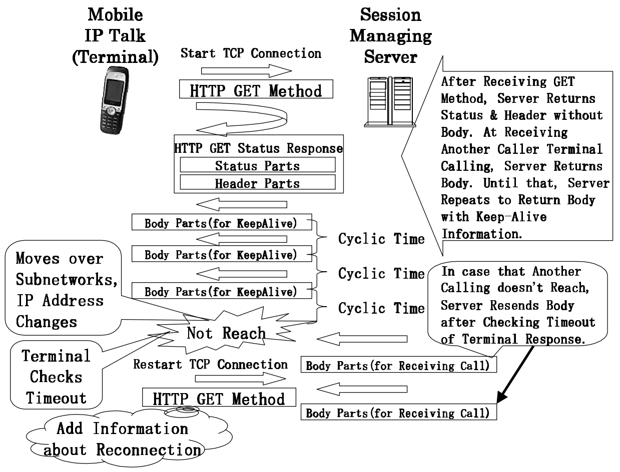


Figure 8. Keeping HCAP Session in changing IP Address

(1) A ”Mobile IP Talk” terminal always keeps the session by the GET method of catch.cgi, and the SMS repeats transmitting dummy data as the body of GET response periodically.

(2) When the terminal’s IP address is changed by moving between different sub networks, the terminal finds out the timeout of receiving dummy data from the server by reconnection of WLAN.

(3) The terminal reconnects the session by resending another GET message on the new WLAN route.

(4) Then, the server can recognize the terminal’s new IP address.

(5) If the calling data from another terminal is transmitted to the terminal when reconnecting to the SMS, the SMS cannot transmit the acknowledgment to the caller terminal. Then the caller terminal can retransmit the calling data by detecting timeout of acknowledgment.

The response performance of changing IP address depends on the interval of dummy data transmission.

4.5. Communication to SIP proxy server and gateway

Some Internet telephone carriers support SIP proxy servers and gateways to be connected to PSTN, in order to shorten the communication route with calling time charge. We developed SIP connecting functions of ”Mobile IP Talk”, fot the case that the recipient of your call does not have the ”Mobile IP Talk”. Our SIP client functions are related with HCAP functions. Especially the Internet access environment with global IP addresses assigned dynamically by DHCP gives a difficult problem for ”Mobile IP Talk” to connect under NAT broadband routers. A SIP terminal must insert global IP address in the URL of SIP messages, but one under NAT broadband routers cannot recognize outside global IP address. We solved this problem by HCAP communication, in which a ”Mobile IP Talk” can get the information from a SMS (Figure 9).

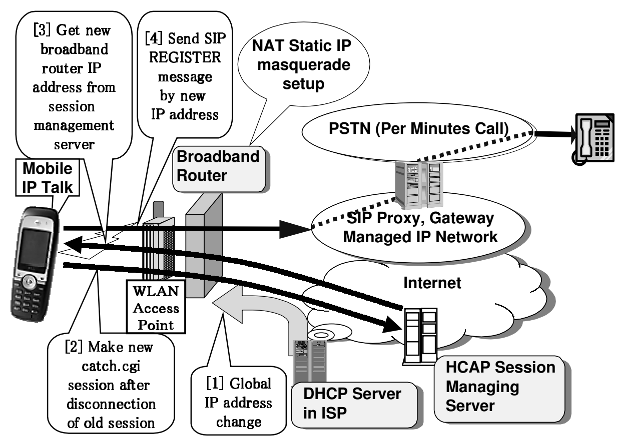


Figure 9. HCAP Solution for SIP Communication with Dynamic Global IP Address

4.6. Voice over HTTP

HCAP contains two stream data transmission protocols, which are UDP like RTP, and our original one by HTTP. We call the latter Voice over HTTP (VoHTTP). Stream data like voice is transmitted by the body of POST method from a terminal to a SRS and received by the body of GET status response from the SRS to the terminal.

VoHTTP is worse than RTP in the real time performance because VoHTTP has TCP communication overhead like retransmission. PSQM [41] of VoHTTP and RTP were the same, in the case of our VoHTTP prototype in a closed LAN without transmitting data to and from the outside [42]. PSQM measured value of VoHTTP became worse, compared with RTP, when other network data are transmitted.

Packet loss created by a network simulator makes the voice quality worse in a closed LAN. While UDP communication increases noise level, HTTP communication tends to cause increased voice delay, voice interruption and even voice cut off by TCP retransmission as typical phenomena. When packet loss ratio achieves 10% in VoHTTP case of 20 msec as the packet interval time, it is almost impossible to hear voice because of congestion.

Though the packet interval size of VoHTTP can be shortened upto 20 msec in a closed LAN, we can use the size of 100-200 msec for public Internet after evaluation[43]. We have encountered that voice is cut off very rarely on the public broadband Internet , and that shows extreme reduction of window buffer size and retransmission by TCP doesn't occur. Figure 10 shows our implementation of VoHTTP.

4.7. Optimization of stream data relay

Even though we have VoHTTP method to support voice communication between private/local domains, we would like to utilize RTP as many cases as possible because of the better real time performance of UDP. In order to identify the cases clearer where RTP should be used, we defined terminal connection types. Utilizing the terminal connection type information in the process of call control, SMSs determine which to use, VoHTTP or RTP, based on the combinations of the type information of the caller and callee terminals. In accordance with the idea that RTP should be used as many cases as possible for better real time performance, we designed the system so that VoHTTP is used only when firewalls and/or NAT routers are involved.

Figure 11 shows basic HCAP network structure for exchange of UDP and HTTP communications.

5. Evaluation

We evaluated the basic G.729a voice quality of "Mobile IP Talk" mainly from a viewpoint of IEEE 802.11b WLAN and plan to evaluate the total network system performance.

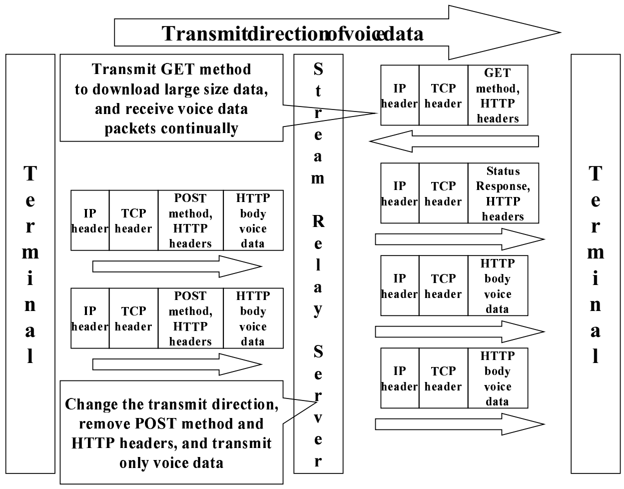
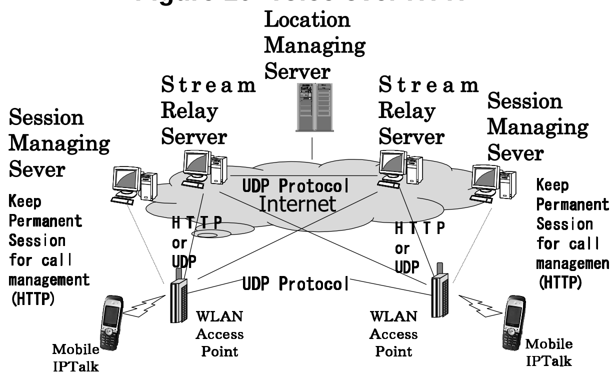


Figure 10. Voice over HTTP



Mobile IP Talks communicate pier to pier as many cases as possible, using UDP.

Stream Relay Server is used when at least one of the two Mobile IP Talk has local IP address.

Session Managing Server solves DHCP problem, using permanent session.

Figure 11. HCAP Relay Server

5.1. Experiment Environment

Figure 12 shows our basic voice quality measurement environment to collect voice quality data by a VoIP analyzer[45]. Two terminals must be placed closely to be connected to the analyzer. In this measurement, we utilized PESQ LQ[44] as a voice quality score. Generally speaking, the best PESQ LQ in the case of G.729a is said to be from 3.0 to 3.1. The worst PESQ LQ is 1.0 when people can't talk at all. Table 3 shows basic measurement result in the case that the distance between two terminals and WLAN access points are 2m.

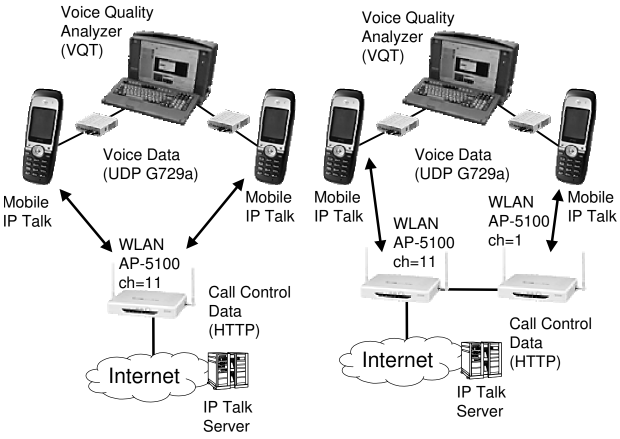


Figure 12. Basic voice quality measurement

No. of AP	WEP	Other WLAN Interference	Other Terminal Talking	PESQ LQ Mean	PESQ LQ Max	PESQ LQ Min
2	ON	None	None	2.98	3.08	2.80
2	ON	None	Talking	3.01	3.14	2.94
2	ON	802.11g	None	2.96	3.11	2.53
2	ON	802.11g	Talking	2.77	3.03	1.96
2	OFF	None	None	2.95	3.12	2.86
2	OFF	None	Talking	2.92	2.99	2.70
2	OFF	802.11g	None	2.61	3.04	1.40
2	OFF	802.11g	Talking	2.65	3.06	1.92
1	ON	None	None	2.97	3.02	2.91
1	ON	None	Talking	2.99	3.08	2.92
1	ON	802.11g	None	2.93	2.96	2.87
1	ON	802.11g	Talking	3.00	3.03	2.97
1	OFF	None	None	2.97	3.05	2.69
1	OFF	None	Talking	3.00	3.10	2.79
1	OFF	802.11g	None	2.84	3.06	2.46
1	OFF	802.11g	Talking	2.78	3.09	2.25

Table 3. Basic Measurement of Voice Quality

5.2. Distance from Access Point

In the experiment of basic WLAN performance, the distance effects the throughput performance of data transmission very clearly[46]. In the case of 2-15m distance between a terminal and an access point, the PESQ LQ is from 2.8 to 3.1 when the WLAN condition is stable. The longer the distance is, the worse PESQ LQ tends to be, when a lot of wireless apparatuses except IEEE 802.11 work. That shows the throughput performance of the terminal which is distant from an access point tends to be influenced by the interference of other wireless apparatuses.

5.3. WEP

In the above experiment conditions, there was no influence for PESQ LQ whether WEP is processed or not. That shows the WLAN environment has enough throughput performance for a few voice data streams, added WEP processing overhead.

5.4. Other WLAN access interference

Recently IEEE 802.11a and 802.11g apparatuses are increasing following after 802.11b. When a PC with a 802.11g WLAN interface is downloading by FTP close to the ”Mobile IP Talk”, in the case of about 2m distance between a terminal and an access point, PESQ LQ is not influenced so much. On the other hand, the longer that distance is, the worse PESQ LQ tends to be by the 802.11g WLAN access interference.

5.5. Hand-over (Micro-mobility)

We confirmed the hand-over function of ”Mobile IP Talk” by moving between 2 WLAN access points. We can listen to the received voice continuously. The roaming time is shorter than 1 second. In the case of WEP processing, the time increases.

5.6. Simultaneous calling

We experimented simultaneous calling with about 30 terminals and 6 WLAN access points in a office building. The LAN is connected to the Internet by FTTH(100MBps), whose available bandwidth is more than 5Mbps. In the experiment, all the terminals call by G.711 voice codec to the SIP proxy and gateway on the Internet, because G.729a voice data is too small to consume the full bandwidth of the FTTH. We confirmed 28 terminals could call to the other kinds of external terminals simultaneously.

5.7. Battery consumption

The electric power consumption of 1st ”Mobile IP Talk” is about 450mA.

When the 2nd ”Mobile IP Talk” doesn’t manage electric power consumption by a CPU and a WLAN card of sleep mode, it can work for 3-4 hours whether in calling or call waiting. In the case of changing to sleep mode in call waiting, it can work for longer than 20 hours.

The battery capacity of the 2nd ”Mobile IP Talk” is 2,000 mAh. The electric power consumption is 190-240mA in normal mode, and 60-120mA in sleep mode. (The maker’s data sheet shows the electric power consumption of the WLAN card is about 260mA in normal use like web browsing, and 1.5mA in sleep mode.)

Figure 13(Axis X: Time[Hours], Axis Y: Electric current[mA]) shows a electric current log of the 2nd ”Mobile

IP Talk”. In the upper case, the terminal continued to output log througn the use of WLAN. In the lower case, the terminal didn’t. In the bose cases, terminal continued to receive the dummy data by catch.cgi response. In our implementation, the dummy data size is one byte per 30sec - 1min.

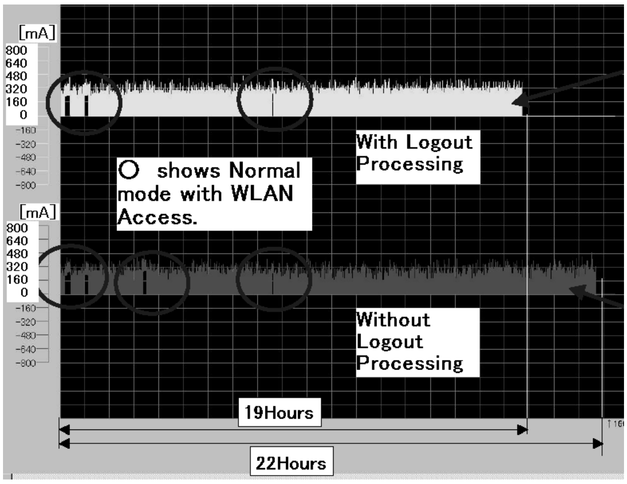


Figure 13. Battery consumption

6. Conclusion

We developed two hardware versions of mobile IP phone; ”Mobile IP Talk” with a WLAN interface, and are evaluating the availability, utility, and performance of them as practical business tools.

After several experiments including field studies with potential partners/customers, we have confirmed that the performance of the system has reached to the level which is acceptable for the most of cost conscious consumers and business customers.

We are sure now that our mobile IP phone service based on HCAP is the most suitable for Japanese Internet environments to reduce high PSTN and cellular phone communication cost by utilization of best effort Internet infrastructure.

We will continue to improve our system about availability, performance, QoS, various kinds of wireless network obstacles, security and new functions by getting feedbacks from many customers.

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