Design and Implementation of 2.4GHz Wireless Skype Phone

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Summary

Recently, Internet phone is more and more popular, and is an emerging wireless technology. Among these, Skype is one of the most popular software at present. However, traditional internet telephony still works with wired earphones, microphones, and extremely limited range between the handset and the host running VoIP software. This paper proposes a new style Internet phone - wireless Skype phone. The core technology of the proposed Skype phone employed 2.4 GHz wireless network with hopping technology. The system consists of a mixed implementation of software and hardware. The system is divided into two parts: the base station and the handset. Three major parts are considered in the hardware: baseband circuit, RF circuit, and macro-controller unit. On the other side, in the software the third party application communicates with the device and transfers the human interface control parameters to Skype by the USB HID driver. Considering this 2.4GHz band is the same as industry, science and medicine (ISM) channel shared by a variety of other applications such as wireless network, bluetooth product and so on, we avoid the radio interference by designing a frequency hopping technology. Based on our results and analysis, our wireless Skype phone can save 13% power than bluetooth while talk time as well as anti-interference characteristics. Furthermore, unlike Blue Tooth, our wireless Skype phone can also dial out by its hand-set device.

Index Terms — Internet telephony, Skype, Bluetooth, VoIP,

Key words:
Wireless、*Internet phone* 、*low power*

INTRODUCTION

As the Internet and broad-band networks are developed rapidly, the demand of communication has a spectacular rise in recent years. The new trend of human communication is transporting voice over Internet Protocol (IP) based network known as Voice over IP (VoIP) or Internet telephony. VoIP applications are becoming increasing popular as VoIP can provide free or low cost calling worldwide [1][6]. While the principal application of these systems has been in providing communicational ability to portable and mobile devices, there has been a tremendous and growing interest in supporting isochronous services such as telephony service or streaming video [12].

VoIP has a number of benefits over traditional telephone services such as convergence of technologies which allows for a flexible and fuller communication experience. However, numerous VoIP applications suffer from a problem that the performance varies with the network state. Additionally, because of the firewall or other equipments, the connection fails constantly. We select Skype [9] as our platform with a kind of technique similar to STUN (Simple Traversal of UDP through NATs (Network Address Translators)). The technology assists applications to find out the public IP address and the type of NAT service. Further, Skype has better compatibility and the user need not modify the firewall setting.

Skype is a free P2P application based on Kazaa architecture for Internet telephony and instant messaging. Skype has various products such as SkypeOut, Skype Voicemail, Skype for PocketPC. Instead of making efforts in developing all-around application, Skype announced the Skype API (Application Program Interface) enabling hardware devices and software applications to seamlessly integrate with Skype's Internet telephony software. So far many applications have developed, but none of them play attention to an important factor – the mobility of physical internet phone.

At present, USB phones and bluetooth earphones are the representative applications of physical internet phones. Generally speaking, USB Skype phone has extremely short range away from the PC running Skype software. Users use bluetooth earphones can not dial, select target or do any other complex operations. In this paper, we propose a new internet phone application and implement a 2.4GHz band portable, low cost, and power-saving wireless Skype phone. In the proposed system, we combine the advantages of the USB phones and bluetooth earphones to enhance the convenience. In order to solve the problems above, we use the client-server architecture that a basestation is adopted as a communication interface between the handset and the PC host. 2.4GHz wireless transmission is used to transmit speech data and the control parameters between the basestation and the handset to extend the work range. Compared to bluetooth, the handset designed with a keypad can not only be used to receive a call but also dial out.

An essential requirement in a personal communication system is the low cost of the terminal. The IP (Intellectual property) of bluetooth is a kind of expensive cost for most developers or vendors. For this reason we exploit 2.4 GHz wireless network to implement our system and still keeps the advantages of bluetooth. As the bluetooth technology, our system avoids interference from other signals by hopping to a new frequency after transmitting or receiving a packet. Additionally, the system we proposed is also designed with a specific sleep mode to reduce the power consumption. From the physical power measurement result, our system consumes less power than the bluetooth earphones in the market by 13%.

The rest of the paper is organized as follows. In the next section, we provide an overview of our system architecture and the working principles. We also discuss some design issues such as connection mechanism between the base station and the handset. Section 3, 4, 5 describe the details of the system design. Section 6 presents the integrated system, the analysis of system performance and energy consumption. Finally, in section 7 we conclude our work and enumerate future avenues of research.

2. SYSTEM ARCHITECTURE AND THE WORK PRINCIPLE

This section covers the overview of the proposed system architecture. We divide the system into two parts, the base station and the handset with three components – PC Client software, hardware and micro-controller.

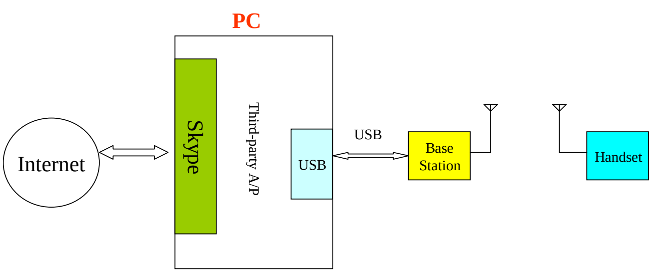

Fig. 1. The architecture of the proposed system

Fig. 1 shows the proposed block diagram of the different components in our Skype Phone. When the speech data is received from Internet, the Skype software transfers the data to the USB driver. The third-party application transfers the interface control parameters between the USB device and Skype by the USB HID driver. Then, PC transmits the speech data to the basestation and converts to digital data. The baseband circuit manages the speech in/out, including signal amplifier and low pass filter. The MCU converts the speech data by the ADC/DAC, enables the RF transceiver, and controls the system.

In our system we employ pulse code modulation (PCM) and FSK modulation to digitize and code the voice signal for wireless communication.

A. PCM

In modern communication system, pulse code modulation is used comprehensively in today's digital signal. It is a process in which analog signals are converted to digital form. The practical design of PCM always uses filtering, sampling, quantizing, and encoding to realize the physical implementation. Fig. 2 shows the PCM implementation in our system. In the system we use 8KHz sampling rate because of the general speech frequency is ranging from 300 to 3.4KHz.

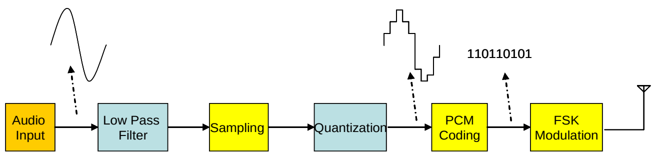

Fig. 2. PCM block diagram

B. FSK Modulation

FSK modulation (Frequency Shift Key) is a method of transmitting digital signals and commonly believed to perform better in the presence of interfering signal. Two different carrier frequencies are used to represent zero and one. The logic 0 is represented by a wave at a specific frequency $\omega_C - \omega_D$, and logic 1 is represented by a wave at a different frequency $\omega_C + \omega_D$. The full equation is as following:

$$V_{FSK}(t) = A\cos(\omega_C \pm \omega_D)t \quad (1)$$

Where ω_C is the carrier frequency and ω_D is the value of carrier frequency offset. The value of A is the amplitude of FSK signal.

C. Butterworth Low Pass Filter

Low-pass filter in our system will pass relatively low frequency portions of the signal but filter out the high frequency ones. The Butterworth Filter we used is the filter type that results in the flattest pass band and contains a moderate group delay.

3. PC Client Software Design

Most of VoIP applications either software or hardware phones need to work with the PC presently. In our system the PC client software is composed of Skype API, USB audio driver, and third-party application.

Skype is a pure VoIP application that we need a third party application to transfer message between Skype and device. Fig. 3 shows the relationship between each layer and component. In the proposed system the third party application use USB interrupt to transfer state messages and control device polling signals. The Skype API is divided into two separate parts: Skype Phone API and Skype Access API. Skype Phone API is an interface that Skype uses to access devices. This API is controlled by Skype and the device-side of the API can be viewed as a driver. Skype Access API is an interface that Skype publishes to third party applications to access Skype functionality.

The USB Driver which plays an important role in our system includes control transfer, interrupt transfer, isochronous transfer, and bulk transfer. We use the isochronous transfer in voice data transmission and the interrupt transfer in volume and function control. Besides, since we use the USB interface to transfer voice and control

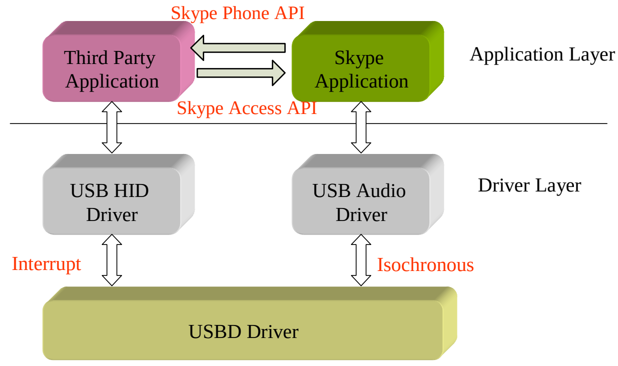

Fig. 3. System architecture of PC Client software

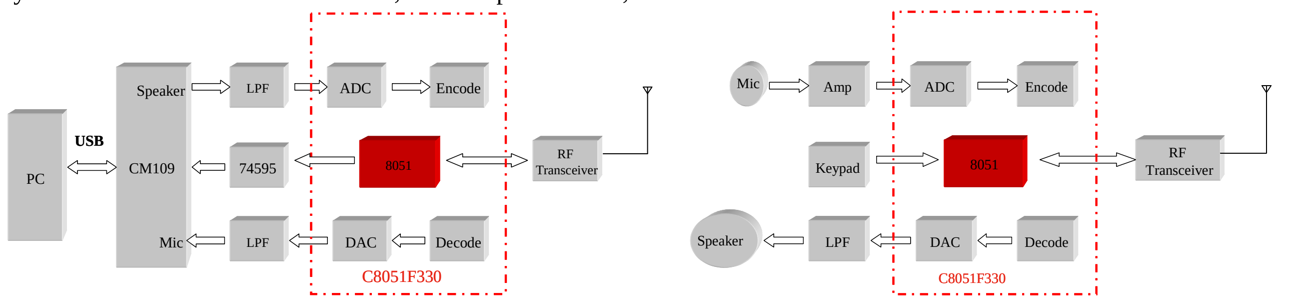

Fig. 4. (a)System architecture of the basestation;

(b) System architecture of the handset

the volume, we need to support HID Device Class and Audio Device Class [13][14].

4. Hardware Design

The system hardware design is composed of baseband circuit, RF circuit and MCU circuit. The speech coprocessor used here is a low-power, small size and high levels of integration (on-chip ADC, DAC and 25 MIPS peak CPU) 8051 micro-controller [10]. It can perform wireless, digital, full-duplex speech compression and decompression function [11]. To design an efficient transmission in the 2.4GHz wireless band and avoid the interference are a challenge work. Therefore, we selected Nordic nRF2401 [8] as 2.4GHz RF transceiver so as to achieve small-size, low cost, low current consumption and low voltage. The output power and frequency channels of the transceiver are easily programmable and the current consumption is very low, there is only 10.5mA at an output power of -5dBm and 18mA in receive mode.

CM109 [5] is a highly integrated single chip USB audio controller for embedded earphone driver, booster, USB transceiver and specifically for VoIP application. It is developed to enable a regular phone, handset, or headset which is interfaced to the USB port on the PC. Fig. 4 illustrates the integrated architecture of the basestation and the handset. As the Figure shows, we use the CM109 USB audio controller to be the interface between the PC and the base station.

5. MCU Firmware Design

Including the base station and the handset, the MCU design plays an important role in our system and the firmware is even more the core of the system design. In the Fig. 4 the MCU part is framed by a dotted line and we can easily find out the

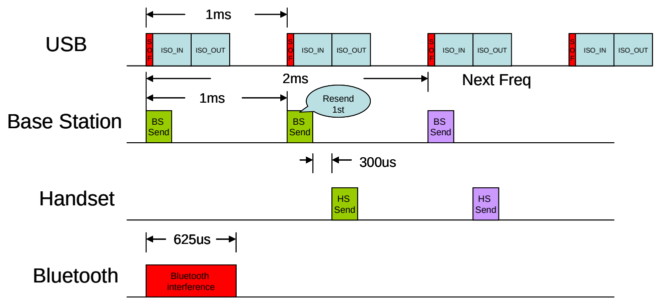

Fig. 5. The communication protocol of the proposed system

position in the whole system. The MCU controls the data transmission, enables the RF transceiver, ADC/DAC, and the communication with the PC and the handset.

Most of wireless communication systems operate in half-duplex mode to avoid interference between transmitter and receiver. In our system the base station is the host and the handset is the terminal. The base station can send data on the line and then immediately receive data on the line from the same direction in which data was just transmitted. The communication protocol is shown in Fig. 5. We design the hopping mode like Bluetooth with 80 different channels. Considering the RF transfer/receive switch time 130us, we set the rate at 500 hopping per second. In order to transfer data correctly, synchronizing the handset to the basestation is crucial to the hopping system. This system exploits a parameter called *TimeSlotCnt* to count time in both sides. When the data is transmitted, the *TimeSlotCnt* of the basestation is used to synchronize the hopping counter in the handset.

We also design a data retransmission function in the proposed system for the speech performance. For the example in Fig. 5, when the data transmission is interfered with bluetooth signal in 2.4GHz, the handset will not receive the message from the basestation. After 1ms, if the basestation still do not receive the acknowledge message from the handset, then the basestation will transmit the data again. When the data loss exceeds 2ms, the basestation and the handset will

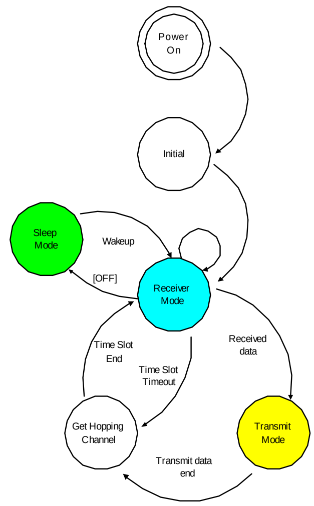

Fig. 6. The state graph of the handset

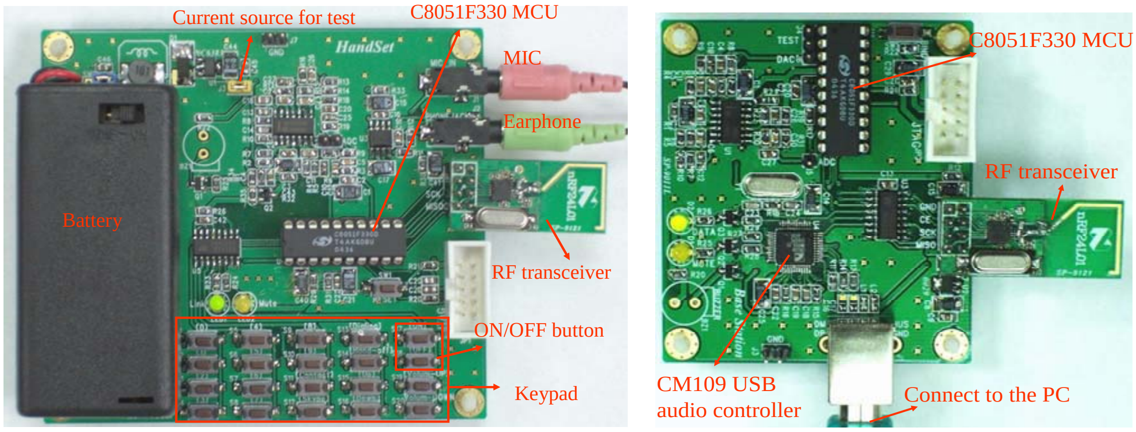

Fig. 7. (a) System prototype of the basestation; **(b) System prototype of the handset**

hop to the next frequency and try again.

Furthermore, considering the tight power consumption of the handset we design the power saving mode. When the user presses the OFF button, the micro-controller switches off all the external current and then enters into sleep mode. The micro-controller wakes up periodically by the external RC charging and detects any button that has been pressed. Then the micro-controller sets the RF transceiver into receive mode in order to synchronize with the basestation. At last, the micro-controller examines the control field of received data and decides the next state. The complete state figure is shown in Fig. 6.

6. Integrated System Outlook and Performance

The photograph of overall system prototype is shown in Fig. 7. The figure illustrated the outlook of the hardware structure of the basestation and the handset. The power consumption of the handset is measured by the NI DAQ (Data Acquisition) system. We placed a 10 ohm resistor in series with the test current source of the handset and sample the voltage drop across the resistor at 1000 samples/second. Then the NI Labview [16], a GUI-based data acquisition, measurement analysis, and presentation software processes the data and shows the computed current value dynamically on the screen.

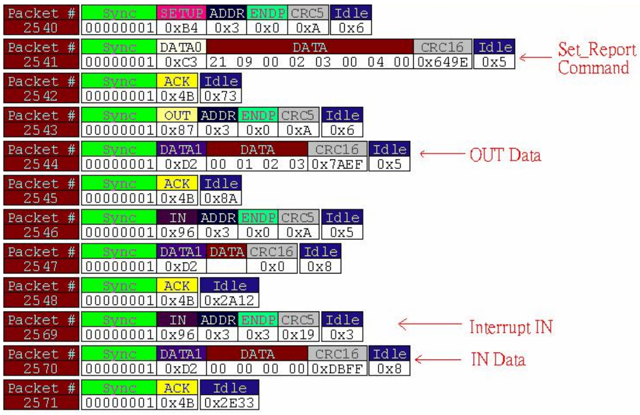

Fig. 8. We use the USB analyzer to observe the USB interrupt transfer

We also compare our result with a bluetooth earphone. From the results, our system works in 27.34mA and the Bluetooth works in 31.41mA that we achieve about 13% energy saving.

In the Fig. 8, we use the USB analyzer to acquire the data transferred in the USB bus. The third party application successfully uses the *Set_Report* command in the HID Device Class to transfer the data to the device. We can observe that the device using the USB interrupt mode to response the current state.

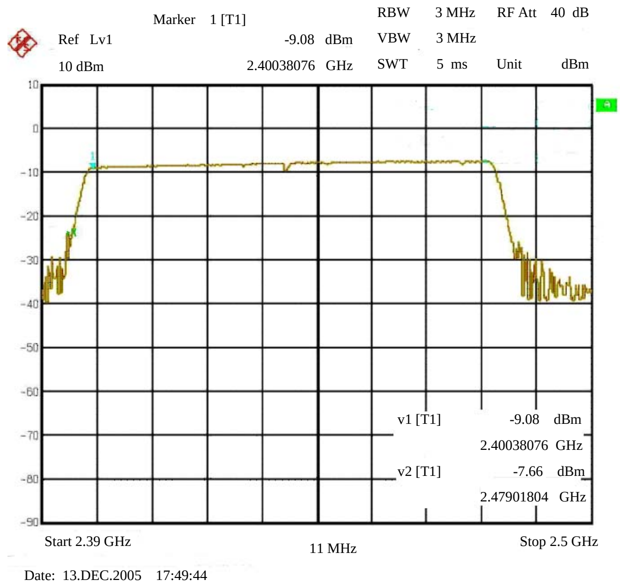

Fig. 9. The frequency and the power measurement

In the Fig. 9, the spectrum analyzer is used to measure the RF power and the hopping technology in the proposed system. We set the trace at the Max Hold mode in this measurement. According to the result, it should be noted that the hopping frequency range covering completely from 2400MHz to 2479MHz and the output power is -7.66dBm (0.171mW). Furthermore, the work range can be extended to 10 meter that is compatible to most of Skype users.

7. Conclusion

We proposed a new style wireless 2.4GHz Skype phone to enhance the extremely limited range of USB internet phones. This system is implemented by using hopping technology, specific USB audio controller and RF transceiver. The 8051 microcontrollers are adopted as the coprocessor of the proposed system and control the client-server architecture to lower cost than bluetooth. Unlike Bluetooth earphones the system is designed with a keypad that we can dial out by the handset device. The performance of the whole system is demonstrated. It is shown that the proposed design can produced good quality with anti-interference characteristics. Furthermore, the work range is convenient enough to most of users. By means of the client-server architecture and the wireless operation feature this system can be used to derive many applications especially for convenience and mobility. In the future, we will focus on reducing the power consumption of the system and increasing the work range.

REFERENCES

[1] Kundan Singh and Henning Schulzrinne, "Peer-to-Peer Internet Telephony using SIP," *Proceedings of the international workshop on Network and operating systems support for digital audio and video NOSSDAV'05*, Stevenson, Washington, USA, pp. 63-68, 2005

[2] Ye Chen, Natt Smavatkul and Steve Emeott, "Power Management for VoIP over IEEE 802.11 WLAN," IEEE Wireless Communications and Networking Conference, Vol3, pp.1648-1653, March 2004.

[3] Alan V. Oppenheim and Ronald W. Schafer, "Discrete-Time Signal Processing, Prentice Hall," 1989.

[4] Ruei-Xi Chen, Liang-Gee Chen, Mei-Juan Chen, and Tsung-Han Tsai, "An I-Phone System Design and Implementation with a Portable Speech Coding Coprocessor," *IEEE Transaction on Consumer Electronics*, Vol. 43, No.4, pp.1262-1269, November 1997.

[5] CM109 Datasheet, "CM109 USB Audio I/O Controller," C-Media Electronics Inc, 2003.

[6] S. Kaplan and J. R. Davies, "Micro-controller based Internet Phone," *Proceedings of the IEEE Africon 2004 on Digital Object Identifier*, Gaborone, Botswana, , Vol.1, pp. 307-311, 15-17 September2004

[7] Bernard Sklar, "Digital Communications, Prentice Hall," 1988.

[8] Nrf24L01 Datasheet, "Single Chip 2.4GHz 2.4GHz Transceiver," Nordic, 2005.

[9] Skype, http://www.skype.com.

[10] C8051F330 Datasheet, "Mixed Signal ISP Flash MCU," Silicon Laboratories, 2005.

[11] AN147 Reference Design, "Wireless Digital Full-Duplex Voice Transceiver", Silicon Laboratories, 2005.

[12] Jiang Li, Keman Yu, Gang Chen, Yong Wang, Hanning Zhou, Jizheng Xu, King To NG, Kaibo Wang, Lijie Wang, and Heung-Yeung Shum, "Portrait Video Phone," *Proceedings of the ninth ACM international conference on Multimedia*, Ottawa, Canada, pp. 597-598, 2001.

[13] "Device Class Definition for Audio Devices revision1.0," USB Implementation Forum, 1998

[14] "Device Class Definition for Human Interface Device (HID) revision1.1," USB Implementation Forum, 1998

[15] Luca Negri, Mariagiovanna Sami, David Macii, and Alessandra Terranegra, "FSM-Based Power Modeling of Wireless Protocols: the Case of Bluetooth," *Proceedings of the 2004 international symposium on Low power electronics and design*, Newport Beach, California, USA, pp. 369-374, 2004.

[16] National Instruments Corp. http://www.ni.com.

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