디지털 인터넷 라디오 수신기 구현에 대한 연구

(The Study on Development of a Digital Internet Radio Receiver)

박 인 규 †
(In Gyu Park)

요 약 본 논문에서는 인터넷과 연결되어 사용되는 일반 PC가 아닌 소형의 임베디드 인터넷 라디오 수신 단말기의 구현에 대하여 논의한다. 이러한 시스템의 표준이 아직 정하여 있지 않으며 알고리즘 또한 비공개 상태이다. 따라서 PC의 인터넷 라디오 수신 방식을 분석하여 끊김이 없는 고음질의 임베디드 인터넷 라디오 시스템을 구현을 위하여 여러 업체의 PC 인터넷 수신 알고리즘은 분석하고 하드웨어는 자체 개발하였다. 본 인터넷 라디오 시스템은 실시간 인터넷 라디오의 멀티스트리밍 기능, 임베디드 프로세서, 플래쉬 메모리, TCP/IP 인터훼이스, MP3 디코더 등으로 구성되어있다.

키워드 : 인터넷, 서버, 이더넷

Abstract This paper explains the design and development of the stand-alone high sound quality Internet Radio system, which is aimed for a small embedded type audio device rather than a general PC type. This device is designed to work with an internet connection. This kind of system is not standardized so far, and also the related algorithm is not open to the public. So it is necessary to analyze several receiving algorithms of current radio receivers, and develop our own hardware in order to overcome these obstacles, finally to get the high quality of sound radio. The main electronic components of this Internet Radio are TCP/IP interfaces, an audio MP3 decoder, an I/O interface, and a Flash Memory Card with advanced audio multicasting for the next-generation Internet Radio. Basic structures and implementation issues of the next-generation most-versatile digital music player, and Internet Radio receivers, are discussed.

Key words : Internet, Radio, Shoutcast, TCP/IP, WAN, IP, DHCP, LAN, Server, Ethernet, UART

1. INTRODUCTION

Internet Radio is undergoing a revolution that will expand its reception from local radio stations to broadcasts anywhere, anytime, and will be able to deliver its content beyond traditional broadcasters, to specific individuals, organizations, and groups. It will have no geographic limitations, so a broadcaster in any place can be heard wherever people access the Internet. The potential for Internet Radio is as vast as cyberspace itself.

Faster Internet connections will cause Internet Radio to become very popular. Even if a listener moves to another country, s/he will still be able to listen to their favorite music, and, in some cases, their hometown radio station. Listeners can explore new types of music that might be difficult to find on local radio dials. Internet Radio stations can also unite geographically diverse listeners via private broadcasts.

The technological key that has made netcasting possible is "streaming." Through streaming, an audio file can be played as it is downloaded. The good news about streaming audio is that it offers some advantages over traditional radio broadcasting. Radio airwaves can only take a limited number of niche listening tastes, but Internet Radio can supply a channel for every music genre. For example, Spinner.com, according to its CEO David Samuel, has 128 channels. And listeners can create their own channels as well as personalize their

† 정 회 원 : 홍익대학교 전자전기공학부 교수
parkik@wow.hongik.ac.kr
논문접수 : 2005년 4월 18일
심사완료 : 2006년 1월 3일

Internet Radio. By pairing sophisticated Internet tracking tools and niche broadcasting, Web sites can deliver precise psychographics to Internet Radio advertisers and brand marketers. In turn, they can provide more targeted messages. Traditional radio is local; Internet Radio is international. Big-name national and global advertisers enjoy a greater reach on the Web.

Traditional radio advertising typically offers 20 percent national, and 80 percent local, ads. Increasingly, listeners of online streaming music will be able to buy the music when they hear it. On the other hand, radio listeners often aren't told the name of the song playing, perhaps fumble to find a pencil, and in the end may not be able to find the music when they get to a retail outlet. Internet Radio also creates its own market in the workplace.

Before streaming, audio files could not be listened to until the entire file had been downloaded. This technology allows for simultaneous "live" transmission on the Internet. The technology behind streaming is called compression-decompression technology. This technology compresses the signal, forces it into a small envelope, and when it gets to the receiver, jumps out like a jack-in-the-box. This is constantly repeated, over and over, creating a stream of information. There are a number of devices that allow users to receive these audio streams on their receivers or their computers. RealPlayer by RealNetworks, Shoutcast, and Microsoft's Media Player are the most popular receivers, as both are free downloads. Sonicbox is a newly developed device that allows users to harness a multitude of Internet Radio listening options with their FM radios.

Well-known among Internet Radio listeners are the problems associated with transmission. A slow connection, or too much simultaneous traffic on one site, can often cause the signal to break up. A connection speed of 28.8k is needed for decent reception, although 56k is much better for ensuring continuity.

Another problem with Internet Radio broadcasting is that most servers cannot support more than a few hundred listeners tuning in simultaneously. In order to accommodate a larger audience, stations have to invest in larger bandwidth, which is a costly solution. The cost-effective solution is a technology called IP Multicast. When this technology is implemented, servers will no longer have to accommodate each user individually, but instead can send information only once, reaching a large group of users.

Many new types of Internet Radio are beginning to be marketed, which are either comparable to, or even exceed, the Destiny Media Player (Destiny Media Technologies, Vancouver, Canada) in performance and cost. In the near future, they will be designed with many additional functions, such as external extractible storage (Flash Memory Card, Compact Flash, Memory Stick), Real Time Clock, Recorder/Player, Automatic Answering Machine, AM/FM radio, and Wakeup Morning Call. And Internet Radio is not limited to audio. Internet Radio Broadcasting may be accompanied by photos or graphics, texts, links, as well as interactivities such as message boards and chat rooms. This expanded media capability can also be used in other ways. For example, with Internet Radio it is possible to train, educate, and provide listeners with links to documents and payment options. Listeners can interact with a trainer or educator, and exchange other information on the radio broadcast site.

2. OVERVIEW OF INTERNET RADIO

Figure 1, below, illustrates the infrastructure of the network/multimedia radio system of the Internet, Audio Broadcasting Server, Client Receiver based on PC, and Client Receiver based on Embedded Internet Radio. Internet Radio can be accessed and controlled by three different communication interfaces. The first LAN interface is used for communicating with user Receivers in homes or offices through the Internet, and for delivering MP3 audio files. The second IR interface is used for the IR remote controller to deliver MP3 audio files, and to control data.

The USB interface communicates directly with a local PC. The system board consists of an ARM-7, 16/32-bit RISC processor, built-in program NOR Flash memory, Extractible data Flash Memory

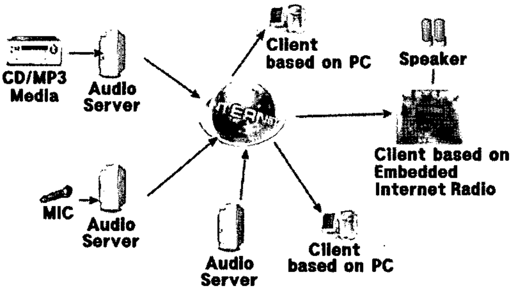

Fig. 1 The Structure of the Internet Radio Broadcaster and Receiver

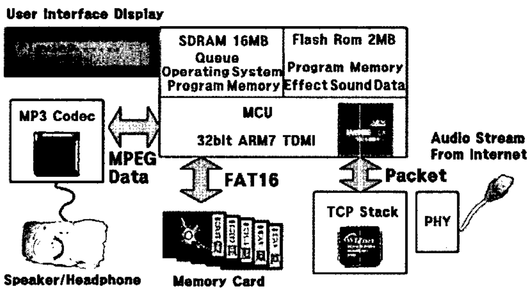

Fig. 2 Internet Radio Block Diagram

Card, MP3 Decoder/Encoder, a TCP/IP Stack Processor, a PHY chip, a SMPS Board, a LCD Display, an IR Remote Controller, and an Audio Power Amplifier.

The 16/32-bit RISC microprocessor[4] is designed to provide a cost-effective and high performance micro-controller solution for hand-held devices and SRAM, LCD controllers, 2-ch UART, and 4-ch DMA.

The world's first Ethernet based TCP/IP hardwired chip contains all necessary Internet protocols (TCP, IP, UDP, ICMP, ARP, DLC and MAC) for Internet connectivity. Since The protocol stack is processed by hardwired logic, it provides high performance, and eases the speed problem of Internet access with related TCP/IP software and memory access. In addition, since OS is not mandatory, W3100A requires minimal sized ROM and g RAM for MCU power, and saves OS licensing fees. In addition, users can save engineering resources and development cost in TCP/IP and networking programming because of reduced marketing time. W3100A simultaneously supports 4 independent channels, and can be interfaced with any Physical

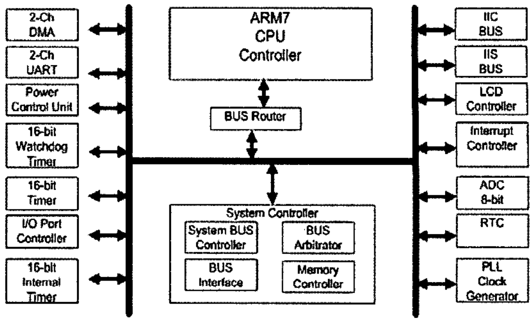

Fig. 3 RISC Processor Block Diagram

-System Board Features:

1. Microcontroller: 16/32-bit RISC MCU
2. BootROM: four Flash Memory (512K × 32 bit), support byte, half-word, word size boot ROM
3. SDRAM: Two SDRAM Chips (4M × 32)
4. Data NAND Flash Memory Card (521Mbit × 2)
5. LAN interface[6]: + 3.3V PHY, 10M/100M bps
6. Serial Communication Port: USB 1.1
7. MP3 Encoder/Decoder
8. Audio Power 100Watt Amplifier
10. Power supply: 5V, 3.3V, and 2.5V

Layer (PLC, Optical, Wireless, etc.) support MII (Media Independent Interface), and runs on 3.3 V supply and 5 V tolerant I/O.

The CGI web server chip is a LSI of hardware protocol stack that provides an easy, low-cost solution for high-speed Internet connectivity for digital devices, by allowing simple installation of TCP/IP stacks in the hardware. It contains TCP/IP protocol stacks such as TCP, UDP, DHCP, ARP and ICMP protocols, as well as Ethernet protocols such as Data Link Control and MAC.

The Audio Decoder[1] is a single-chip, low-power MPEG layer 2/3 and MPEG2-AAC audio stereo decoder. It also contains the G.729 Annex A speech compression and decompression technology for use in memory based or broadcast applications. Additional functionality is achievable via download software (e.g., CELP voice decoder, Micronas SC4 [ADPCM] encoder/decoder). The Audio Decoder block accepts compressed digital data streams as serial bit streams, or in parallel format and provides serial PCM and S/PDIF output of decompressed audio. In addition to the signal processing function, the IC incorporates a high-performance stereo D/A converter, headphone amplifiers, a stereo A/D converter, a microphone amplifier, and two DC/DC converters with supply compression rates up to 16:1. It defines several profiles for different applications. This IC decodes the 'low complexity profile' that is especially optimized for portable applications. The Audio Decoder also implements a voice encoder and decoder, which is compliant to the ITU Standard G.729 Annex A. SC4 is a proprietary Micronas speech codec technology that can be downloaded to the Audio Decoder, to allow recording and playing back speech at various sampling rates.

The performance of Internet Radio is determined by the combined characteristics of the audio/music quality, the associated hardware reliability, measured no-break-up of audio in terms of audio resolution, audio response time, transmission rate, volume control, audio quality (cut off frequency and response uniformity), size, and weight of the main body. Therefore, the performance of an Internet Radio Receiver is limited by the trade-offs made between those features. To compensate audio distortions and breakup of streaming, a package of sophisticated software will be introduced.

The device's controls are limited and awkward. Five preset buttons, along with a backlight button for the display, which shows the song's name and artist. A Memo button makes a record of the songs listeners prefer.

Unfortunately, there's no easy way to flip between stations; listeners need to use the switch to get to any station that's not pre-set. The problem lies in the fact that these switch controls have too many functions. They are used to scroll through stations, station categories, and controller settings. Required station scrolling will take four or five seconds to load, and cannot be changed again until that channel has finished loading.

Thus, the hardware is less than ideal, but how about the stations? In our testing we found the content to be mostly excellent. Our control group of 100 stations was grouped into 15 categories, including Country, Rock, Urban, Jazz & Blues, Latin, News, and Comedy. Each category housed several stations. For example, the Classical category had stations for traditional classical, eclectic classical, opera, and classical pop. We found that the music stations offered good variety, but that talk channels were more repetitive.

The system is intended to function as a standalone audio system, making it quick and easy to store music and create an essentially unlimited number of customized play lists. A large display, intuitive interface, and advanced search features are meant to make it easy to find the music listeners want -- instantly. Plus, the system optionally offers suggestions based on tracking listening habits.

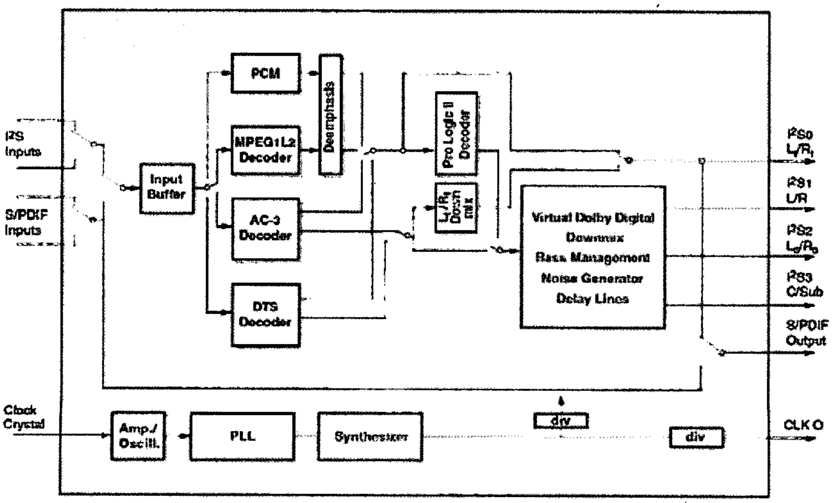

Fig. 4 Audio Decoder Processor Block Diagram - Control Panels

3. SOFTWARE

3.1 Block Diagram

The main elements of this software consist of a MP3 Decoder Driver, TCP/IP Chip Driver, Socket Program, Circular Buffer Handling Program, User Interface, DHCP Server Driver[2], and Script Parser for linking to an IP address.

The internal block diagram of the mechanism of linking hardware and Micro-C Operating System shown in Fig. 5 is made of a Stream Receiver, Buffering, MP3 Decoding Algorithm and uC/OS. The Stream Data Receiver offers the flexibility of good quality by receiving audio data being steamed continuously by Shoutcast Broadcast Server format. The routines required for storing data on Buffer are stored as following: the interrupt routine, and the sending routine via TCP/IP W3100A chip. These data are divided into packets with a fixed size and stored on the circular Queuing Buffer. The size of these circular buffers determines the quality of the streaming audio.

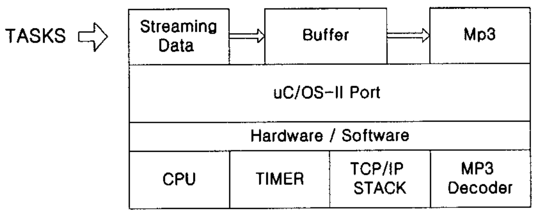

Fig. 5 Software Block Diagram

The data format that is broadcasted from the Broadcaster Server is called meta data. Meta data consists of MP3 data and title data. Meta data received by the Buffer should be transformed into MP3 data, thereby eliminating the title data.

Since the uC/OS is one of the preemptive OS, during the developing of programming, the developer should observe carefully where the current state-of-task is placed. The MP3 decoding procedure is designed to have priority over other routines, so that audio decoding may not be broken up in case CPU time is incorrectly allotted. Since the maximum transmitting speed from the Internet Radio Broadcasting Server is around 128Kbps, it is not necessary that the streaming procedure time be longer than the CPU time.

3.2 Streaming Buffer

The main influencing factor in audio quality in the streaming procedure is QOS. The TCP in this system, of course, does not support the fully satisfactory QOS. However, it has been verified that QOS is sufficiently satisfied by using the Buffers shown in Fig. 6. This Figure shows the structure that transforms incoming broadcasting data into MP3 decoding data.

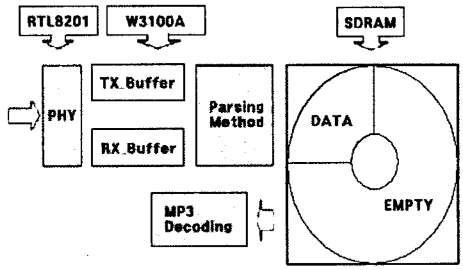

Fig. 6 Streaming Block Diagram

The incoming raw data to the PHY chip is passed and stored with a built-in dual port SDRAM. Two buffers are formed using TX- and RX- Buffers; sizes can be 1KB, 2KB, 4KB, or 8KB.

As soon as data is received in the RX-Buffer, the interrupt signal is sent into MCU; MCU then processes the decoding, and stores the transformed data into Circular Queue. And then, almost simultaneously, MCU extracts data from the Circular Queue, and starts to play music.

3.3 Circular Queue

Figure 7 shows the Queue as it is being broadcast in 32Kbps speed from the Broadcasting Server. Thus, the extracting speed is designed not to be lower than 32Kbps; in the event that the incoming speed is lower than 32Kbps, the data that have not been extracted, should be temporally buffered here. The warning pointer and the safe pointer can be pre-set. The decoding procedure is activated if the

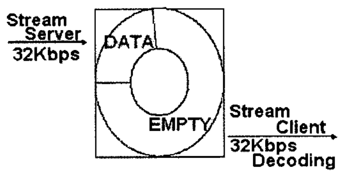

Fig. 7 Circular Queue Block Diagram

current pointer is located under the safe pointer, and is de-activated if the current pointer is located under the warning pointer.

3.4 Broadcasting Data Format

Figure 8 shows the streaming data format. The broadcasting data format is not formed using MP3 data, MP3 data size, title data, or title data size. Since the total packet length is not fixed, the starting point with any specific data should be carefully located in every incoming packet. The procedure of receiving the stream data is as follows: first, Internet Radio attempts connection with the Broadcasting Server; next, generate the Header and attach it with HTTP using "Method Type GRT," which is involved with data such as Host, User-Agent, Icy-Meta Data, and Accept. The last procedure is to receive the Header information. If the received data has a different URL on the HTTP location, it means the connection may be not available. In this case, another connection should be tried.

3.5 QOS

As stated, the main influencing factor in audio quality during the streaming procedure is QOS. The streaming data is transmitted in a packet sized at around 1.4Kbyte of segment data. Network loading continually increases, and, at times, might become overloaded, or the internal buffer could overflow. If this occurs, music can be discontinued. To overcome congestion, the following method is proposed:

• TCP SlowDown

This method increases the size of the Window. The size is incremented with an exponential curve from the bottom segment. When TIMEOUT or ACK signal is entered, the increment should be discontinued.

• AIMD (additive increase, multiplicative decrease)

The Window size continues to be incremented exponentially from the bottom. Then, after the Window size is reached at a threshold value, the

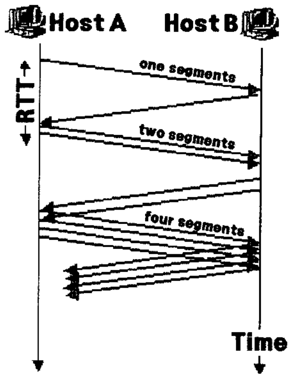

Fig. 9 TCP Slow Down

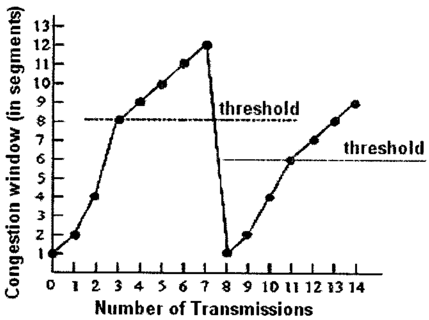

Fig. 10 AIMD

Window size should be incremented linearly rather than exponentially. After that, if congestion occurs, threshold values are divided in half, and the Window size is set to one.

3.6 File System

Built-in Flash Memory Cards are for recording streaming audio data. After recording, it should be possible to replay it in any other MP3 player, or in any kind of Window's based PC, which is why a uC/File system is used as OS. This uC/FS provides a FAT file System, which can be used with any media, and which provides basic hardware access functions.

Micro-C/FS also provides a high performance library that has been optimized for speed, versatility, and memory footprint. uC/FS is written in

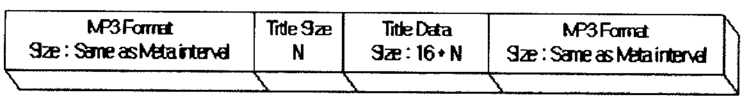

Fig. 8 Broadcasting data Format

ANSI C, and can be used on virtually any CPU. Some features of uC/FS are:

- MS-DOS/MS-Windows compatible FAT12 and FAT16 support.
- Multiple device driver support. This allows the use of different device drivers with uC/FS, which facilitates access to different types of hardware, with the file system at the same time.
- Multiple media support. A device driver allows access to different medias at the same time.
- OS support. uC/FS can easily be integrated into any OS. In that way file operations can be made in a multithreaded environment.
- ANSI C stdio.h like API for user applications. An application using standard C I/O library can easily be ported to use uC/FS.
- Very simple device driver structure. uC/FS device drivers need only very basic functions for reading and writing blocks. Therefore it is very simple to support custom hardware.
- Generic device driver for SmartMedia cards, which can easily be used with any kind of card reader hardware.

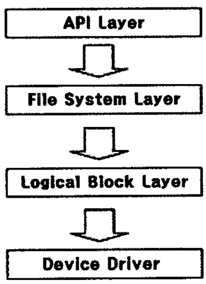

Fig. 11 File System Layers

4. Getting Audio over the INTERNET

There are two ways to deliver audio over the Internet: downloads, or streaming media. In downloads, an audio file is stored on the user's computer. Compressed formats like MP3 are the most popular form of audio download, but any type of audio file can be delivered through a Web or FTP site. Streaming audio is not stored, but only played. It is a continuous broadcast that works through three software packages: the encoder, the server, and the player. The encoder converts audio content into a streaming format, the server makes it available over the Internet, and the player retrieves the content. For a live broadcast, the encoder and streamer work together in real-time. An audio feed runs to the sound card of a computer running the encoder software at the broadcast location, and the stream is uploaded to the streaming server. Since that requires a large amount of computing resources, the streaming server must be a dedicated server.

Getting audio over the Internet is pretty simple:

① The audio enters the Internet broadcaster's encoding computer through a sound card.

② The encoder system translates the audio from the sound card into streaming format. The encoder samples the incoming audio and compresses the information so it can be sent over the Internet.

③ The compressed audio is sent to the server, which has a high bandwidth connection to the Internet.

④ The server sends the audio data stream over the Internet to the player software or plug-in on the listener's computer. The player software translates the audio data stream from the server and translates it into the sound heard by the listener.

To accommodate variables of where an Internet Radio Receiver is located, and the communication networks with which a home network is interfaced, three Internet remote control applications can be applied:

First, Internet Radio can provide some level of wakeup morning call. For example, listener's time is coordinated with broadcast time, allowing the Internet Radio to rouse a sleeper. A presenter who is about to speak should check the projector's status to insure that all is in order.

The second application is the real time lecture system. As in the first application, a presenter who is about to speak should check the projector's status to insure that all is in order. The JPEG video files are sent to all Radio sets which are located at different rooms via the Internet, and stored in internal NAND flash memory. Then, as a lecturer speaks, the server encodes his words into MP3 format data, and this MP3 encoded data is sent to all Radios in real time through the Internet. A presenter can have the option of delivering presentations either locally or remotely with a live MP3 audio link.

The third application is to receive and replay message data through the Internet. For implementing these applications, and in order to have their own IP address, hardware must be added.

If ADSL or Cable Modem is used in the home, the network projector can be used with ADSL or Cable connections. It is known that ADSL/Cable services do allow this even if there is no fixed IP address. The user's home server does not have to have a fixed IP address, but does need an address which is endowed differently by the DSLM, which is placed in the telephone exchange office at the very moment an ADSL/Cable modem is switched on. Therefore, even in the home, it is easy for customers to send data remotely through the Internet; these network projectors can be used in the home or in small offices that are connected with ADSL/Cable modems.

5. Test

There are a large number of broadcast servers on the web. But we choose two servers which have their own bit rate, site location as shown in the below table in this test. As several tests results as shown in this table, it is concluded that for a minimum good quality in replaying under the environment of no re-buffering, the warning point should be at least over 10kbyte, and the starting point should be over 100kbyte.

Broadcast Server	Location	Bit rate	Result
1. JazzRadio Bein http://195.88.140.11:8000	Overseas	32kbps	not bad
2. MUSE KPOP http://211.239.153.149:9100	Domestic	128kbps	good
3. Korean-POP http://210.120.247.73:8110	Domestic	128kbps	good
4. Gospel http://211.47.128.235:8100	Domestic	32kbps	very good
5. GCN http://209.17.76.226:8010	Overseas	8kbps	good
6. WOLF FM http://205.188.234.34:8020	Overseas	32kbps	not bad

6. CONCLUSION

The prototype of an Internet Radio Receiver has been developed. Good quality performance in decoding audio, and an excellent QoS, has been obtained. Moreover, by improving steaming algorithm and buffering schemes, a sharp and fine audio throughout the entire audible frequency spectrum has been achieved.

Internet Radio can support several different features, such as a NAND Flash Memory Card (512Mbyte) for storage, a Bell or Wakeup Morning Call, Real Time Clock, AM/FM radio, Recorder/Player, Lecture Receiver, and even an Automatic Answering Machine. Audio quality is determined by a continuous decoding ratio, as well as volume level uniformity through an entire audible frequency that can be optimized through the trade-offs between the combined characteristics of the audio quality, as determined by data flow mechanism, Buffer size, QoS, and broadcasting speed. One critical design problem with this type of discontinuity in decoding is Buffer size, due to an inequality between incoming speed into the Buffer, and outgoing speed out of the Buffer. To reduce discontinuation in decoding audio, a buffering mechanism and QoS were studied and developed. Internet Radio can be accessed by three different interfaces. The first LAN interface is used for communication with user PCs in home or office through the Internet. Meta data and MP3 audio files are remotely sent through the Internet. The second IR interface is used for remote control through IR communication. The final USB interface provides communication directly with a local PC. Internet Radio is capable of receiving messages from any user PC over the Internet, and it is expected that Internet Radio will be used in the near future in many applications, particularly in real time lecture systems, and video conferencing. Furthermore, recent improvements in data services over the Internet have made it possible to offer Internet Radio.

Our program was coded with C-language based on microC/OS-ii and is composed of firmware and four application software including Ethernet, File System, MP3 Decoding and USART.

REFERENCES

[1] *Documentation*, www.ronas.com/products/documetation/consumer/mas35x9f/downloads/mas35x9f_1pd.pdf, Micronas, 2002.
[2] *Literature*, www.cse.fau.edu/~borko/Research.ppt, 2000.
[3] *Technology Talk*, 2002 ww.msba.org/ Departments loma /techtalk/2002/0602.htm.
[4] *User manual*, S3C44BOX RISC Microprocessor, 2000.
[5] *Color management System* for 3-Panel LCOS Projectors, SID 2001 Digest.
[6] *Technical Data Sheet*, i2Chip W3100, I2Chip Inc. 2001.

박 인 규

In Gyu Park was born in Seoul, Korea, on December 5, 1948. He received his B.S degree in Department of Electronics. Engineering form Seoul National University. He received also his MS and Ph.D from The Ohio State University in 1982 and from Purdue University in 1985 respectively. In 1975 he joined the Samsung Electronics Company, Suwon, Korea, where he works as a development engineer in digital development project team. In 1987/1988, he led the computer design Lab. in Research Center of Samsung Electronics, Suwon, Korea. At present, he is a professor in Department of EE, HongIk University, Seoul, Korea