DESIGNED TERMINAL DEVICE OF MICROCONTROLLER BASED IP TELEPHONY SYSTEM

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ABSTARCT

In this study IP Telephony system is investigated. Packet transfer and system signaling protocols which uses in this system and digital transmission of speech signal are given. Delay sources which effects to voice data packets in real networks are examined. Some solution methods are given for decreasing these packet delay values.

Microcontroller based with low cost IP Telephony terminal device design is proposed and realized. ADPCM coding algorithm is used for digital speech data transmission. Speech packet architecture which uses low cost IP Telephony terminal device is described.

Protocol architectures which describe IP Telephony system are explained. In this study, design of low cost terminal device supports H.323 protocol architecture which uses IP Telephony system.

In conclusion some problems and solution methods which encountered design of low cost IP Telephony terminal device are given.

I. INTRODUCTION

In the last years, development of The Internet technologies has been augmented to using areas on human life. High bandwidth demand of the switching system of the internet is revealed. Companies which generating and developing internet technologies are researching new systems which merge

IP Telephony eliminates the cost of maintaining multiple networks each dedicated to the traffic of a single medium.

By utilizing IP as a common transport layer, network dedicated to voice, video, and data effectively merge into a single unified network. In a packet-switched network, real-time streaming data such as voice and video are compressed into packetized data and sent out over physical links to be reassembled and re-sequenced on the other end. This process results in an effective use of network bandwidth. In order to deliver high quality voice solutions, IP Telephony incorporates families of multimedia specifications.

Large corporations typically have separate offices or supply chains spread out over many geographical locations over the world. The cost of telephone communications with these offices is much higher. Depending on the configuration of the company is existed intranet network between headquarter of the company to offices. The company billing cost of intranet to network suppliers. [1,2]

Bypassing the PSTN calls using IP Telephony system, companies saves costs of telephone calls. IP Telephony system scheme is shown in figure 1.

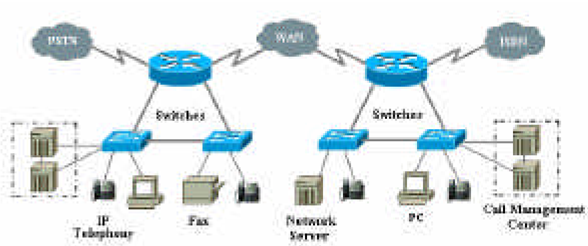

Figure 1

II. THE BENEFITS OF IP TELEPHONY

Today's businesses employ IP Telephony networks to reduce costs as well as to broaden the means of communication. Communication across IP networks has been elevated from simple Web and email access to multimedia conferencing sessions and real-time document collaboration. As a wealth of features available with circuit switched voice systems are added to future Internet Telephony products and services, international and domestic callers alike will turn to IP Telephony providers for an integrated solution. This integration across networks environments increases reliability and interoperability while offering greater economies of scale.

III. OPERATING OF THE SYSTEM

Terminal device in the IP Telephony system obtains supply voltage from switching equipment by means of special software or external power supply system. While user switches on the terminal device, software in terminal device is sending broadcast packets on the network. These broadcast packets are shown in figure 2.

Frame Indicator	SFD	Source Address	Target Address	Length of Packet	DATA	FCS
62B	2B	6B	6B	2B	46B - 1500B	4B

Figure 2

As shown in figure 2, broadcast packets have target address of the call management server and data segment which requests message to join the system. Only call management server could process and reply these broadcast packets. Call management server have database which contains terminal devices information related IP Telephony system. This information includes telephone number, Mac address, IP address, gateway address and open-close situation of the network. Initially, this IP Telephone information must be notified the database of call management server.

Call Management server updates its database with sent data by IP Telephony terminal which requests to join IP Telephony system. After this update process, the IP Telephony terminal device is switched from close situation to open situation in call management server database. Call management server will send some network data to IP Telephony terminal after this update process. This network data contains IP address, default gateway address etc.

If IP Telephone wants to call another IP Telephone, it sends packets to network. These packets have target address of call management server and data segment which number of target IP Telephony. After call management server received these packets, will be query from databases. If target IP Telephony registered in database and status of the target IP Telephony is appropriate for calling (open and non-busy situation), call management server send packets to source IP Telephony which contains permission of calling process and IP and MAC address of target IP Telephony. These process are shown in figure 3

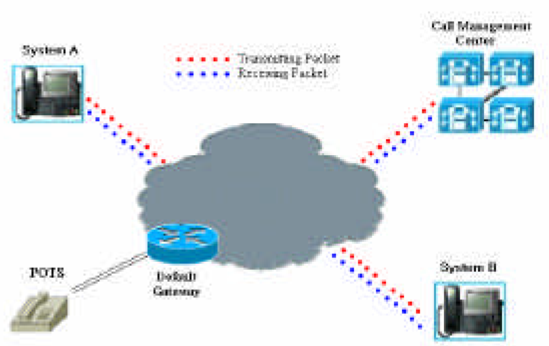

Figure 3

If target IP Telephone terminal device is exists in same network, source IP Telephone terminal device starts connection to target IP Telephony by using IP and MAC address. Otherwise target IP Telephone terminal device is exists in another network, source IP Telephone terminal device starts connection to target IP telephony by using default gateway addresses.

While connection process, source and target IP Telephony terminal device exchanges their information data which includes speech coding algorithm, protocol number of connection type and data process speed of microcontroller.

After connection process, starts signalization which includes telephone ringing and handset control etc. between source and target IP Telephone terminal devices. This signalization processes is executed by software in the source and target IP Telephony devices. These connection processes diagram using H.323 protocol is shown in figure 4.

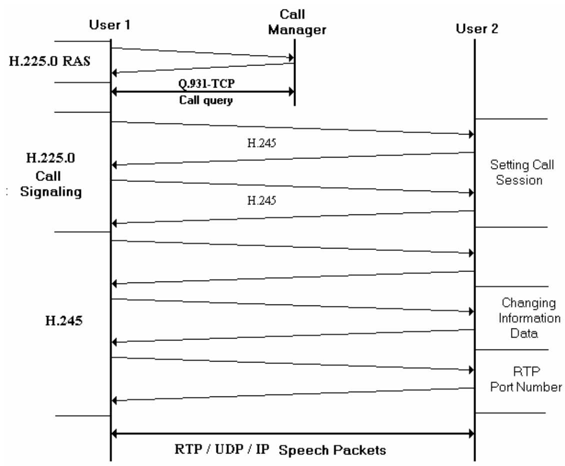

Figure 4

While connection process, IP Telephony devices are uses TCP protocol algorithm for packet deliver controlling. After finishing connection process, IP Telephone terminal devices are obtain digital speech data and exchanges speech packets. While this process, packet control algorithm is changed to RTP from TCP. Because of this RTP is designed to support real-time traffic which requires playback at the receiving application in a time sensitive mode such as for video and voice systems.

On the sending side, initially analog speech signal is converted digital PCM signal by using telecom codec integrated circuits. PCM signal bandwidth is 64 kbit/s. This bandwidth value is not appropriate for transmitting digital speech signal by using packets on the non-reliable networks. Therefore, some voice coder algorithms is used for decreasing digital speech signal bandwidth. This digital speech data are segmented. By adding RTP header, segmented digital speech data is converted RTP packets. These packets are sending to target on the network by adding IP and data link headers.

On the receiving side, arrived RTP packets are converted segments of digital speech data by taking off data link, IP and RTP headers. These segments are converted PCM data by using decoding algorithms. PCM data is transformed analog signal and conducted to handset. [4]

IV. VOICE CODERS USED IN THE IP TELEPHONY SYSTEM

Speech coding is the process of preparing a stream of speech data for transmission. In IP Telephony system, speech signal has to be divided into time-frames where each time frame may represent a single data packet. Often, the data is compressed, in order to reduce the rate from about 128 kbps to rates in the order 2-4 kbps.

Many coders apply silence suppression to achieve an extra coding gain of about 50%. Silence, however, is not very comfortable to listen to and may cause the other user to believe that the connection has been lost. In order to get a more pleasant sound, comfort noise that resembles the background noise can be generated by the receiver. [5]

The International Telecommunication Union - Telecommunications standards sector (ITU-T) has published standards voice coders. The result is a list of recommendations known as the G.xxx series. Especially G.711, G.726, G.729 and G.723 are popular in IP telephony, since they are all part of ITU's H.323.

A G.711 standard describes the 64 kbit/sn PCM voice coding technique. G.711 encoded speech signal is already in the appropriate format for digital transmission of voice.

A G.726 standard describes ADPCM coding which encodes using 4-bit samples, giving a transmission rate of 32 kbit/s. ADPCM coding algorithm is encode differences in amplitude, real signal samples and prediction values. PCM and ADPCM coding algorithms is referred waveform coding algorithms.
G.729 standard describes CELP coding algorithm that enables voice to be coded into 8 kbit/s. CELP algorithms is referred voice coder algorithms.

G.723.1 standard describes MP-MLQ coding technique that enables voice to be coded 6.3 kbit/s. MP-MLQ technique is hybrid voice coding algorithms which combine benefits of the waveform coding algorithms and voice coder algorithms.

V. CALL SIGNALING PROTOCOLS IN THE IP TELEPHONY SYSTEM

In IP Telephony system, several protocols provide the control and management of telephony session in an internet connection media. These protocols are H.323, Megaco, MGCP and SIP. The principal job of these protocols is to set up, management, routing and clearing calls in an IP network. Nowadays, H.323 and SIP are more using than other protocol structures in the IP Telephony system. H.323 protocol defines in considerable detail the operations of user devices, gatekeepers, gateways and other nodes which use system. H.323 defines user terminal device that could provide real-time, two way audio or video and data communications with another user terminal device. Call management server is defined as gatekeepers. Call control and transmission protocols which determines in this protocol structure are shown in figure 5.

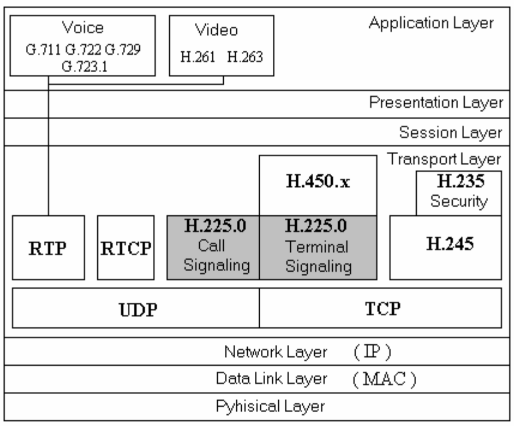

Figure 5

SIP protocol defines set up, modify and tear down sessions between session users. This protocol structure widely uses between mobile users or users which are join to this system via computer. SIP protocol use proxy server instead of the gateway controller or gatekeeper concepts. SIP Proxy server receives a request from client and decides how delivered to target address. SIP protocol structure is shown in figure 6. [7,8]

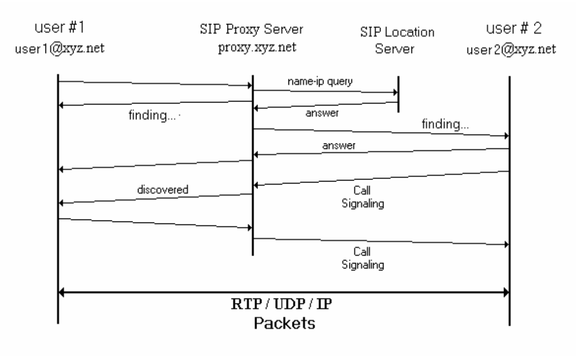

Figure 6

VI. DESIGN OF THE IP TELEPHONY TERMINAL DEVICE

IP Telephony terminal device uses internet connection media for transmitting speech packets. Therefore terminal device have network interface chip (NIC) which is used to realizing needed process on the network.

This chip controls network media for transmitting packet to the network, receiving packet from the network or collision detection on the network which obtains reliable transmitting or receiving.

This hardware is managed and controlled by micro controller in the terminal device. In addition to NIC control, microcontroller is manages and controls application tasks such as generating of call tones, handset control, displaying numbers etc. of the terminal device. For this reason, high speed micro controllers should be selected.

In this study, we have used 8900 A NIC which is produced by Cirrus Logic, to realize network process of the IP Telephony terminal device. This hardware supports easily access from 8-bit microcontroller systems. PacketPage architecture of the NIC enables controlling of packets and faults debugging of packets quite easily.

The microcontroller used in this study works at 20 MHz frequency and contains 512 bytes RAM memory which has 8-bit memory architecture and five input-output ports for controlling other units in the terminal device. Designed IP Telephony terminal device in this study is shown in figure 7.

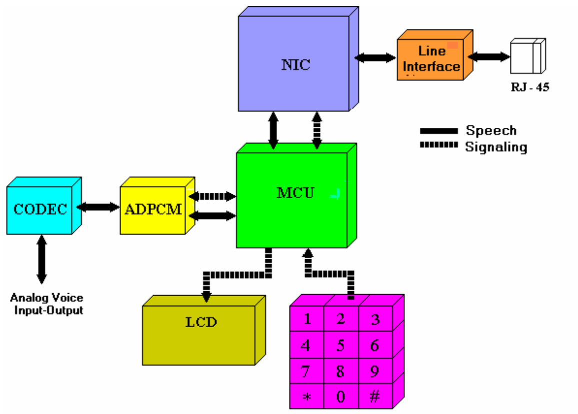

Figure 7

Analog speech signal is converted to PCM signal which has 64 kbit/s bandwidth by telecom codec's. After this process PCM signal is coded to 32 kbit/s ADPCM signals with using software in the microcontroller.

These ADPCM data is segmented into portions cover 80 bytes width. By adding network headers, segmented ADPCM data is converted packets. This process is shown in figure 8.

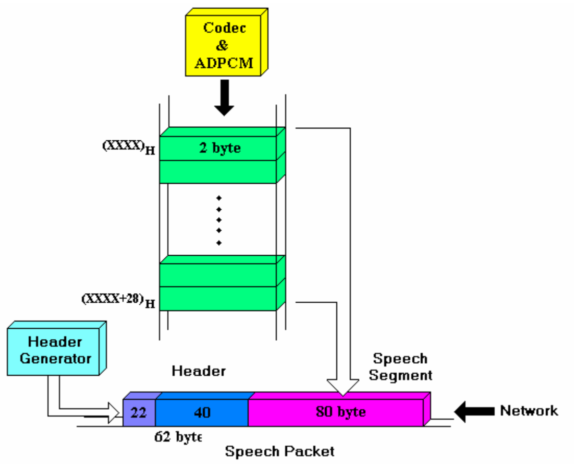

Figure 8

As shown in figure 8, appended network headers (Eth./IP/UDP/RTP) to segment is 64 byte length. Result of this, total packet length will exists 142 byte length. Bandwidth of these packets is given by.

$$BW = \frac{Codec \quad sample \quad rate * Speech \quad packet \quad length}{Total \quad speech \quad packet \quad length} \quad [1]$$

$$BW = \frac{16.10^3 * (142*8)}{(80*8)} = 28{,}4 kbit/s$$

By adding network headers which have 64 kbit/s length to 80 bytes speech segments is inefficiently for transmitting packets on the network. For resolve this problem header compression techniques is used.

However these compression techniques are run local switching nodes and terminal devices. A result of this, microcontrollers which works high frequencies strong and consists process architectures as same as a digital signal processors is should be used in the switching devices and terminals. [8]

In this study, is decided that a large scale of IP Telephony calls are exist same network. If target terminal device exist same network adding headers which use local switching nodes (ethernet) is sufficiently.

In this condition, total bandwidth of speech packets is given by

$$BW = \frac{16.10^3 * ((80+22)*8)}{(80*8)} = 20{,}4 kbit/s$$

Using switching units in the local networks processes 100 Mbit/s digital signals. If 100 IP Telephony calls appears in the network at the same time, total bandwidth consumption is 4,08 Mbit/s. This value is not great for 100 Mbit/s switches.

VI. CONCLUSION

IP Telephony system supports to combine our separate data and voice into single transmission media. The potential benefits for corporate and home users include the reduced cost of only needing to buy a single transmission line.

However, there are significant barriers to acceptable QoS that substitute form existing telecom networks. These barriers are excited by using dsp based microcontrollers. Therefore IP telephony terminal device is become high costs. [12]

In this study, as shown in figure 7, microcontroller based with low cost IP Telephony terminal device design is purposed. ADPCM coding algorithm is used for digital speech data transmission. Designed terminal supports H.323 protocol architecture which is used in IP Telephony systems.

Using IP Telephony systems instead of public telephony network is depending on using smart houses prevalently. [13]

However IP Telephony system has barriers which using protocols in the system. Its expected that publishing new protocol architectures which related IP Telephony systems.

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