# 데이터 통신망에서 음성통신에 대한 연구

# A Study on Voice Communication over Data Communication Network

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## ABSTRACT

Voice and data are transmitted over a single packetized data communications network which is designed for data communications. The public switched telephone network for voice and the packet data network for data are merging into a single data network to get efficiency and to reduce operational cost. However, integrating voice and data transmission over a single data network is not easy because voice should be transmitted without delay but data should be transmitted without error. Advances in technology begin to overcome basic differences. Several integration methods in voice and data will be examined and reviewed here. Moreover, trends and problems on integration will be also discussed.

## I Introduction

The public switched telephone network(PSTN) has been developed for voice communications ver 100 years. A connected path through many switches is made when a person talks to another person. A fully dedicated path from a sender to a receiver is maintained during telephone calls. Thus a talk can be made without delay. Telephone network has well established and has become a global network in which one makes a call by dialing a phone number.

The data communication networks have also well established in many nations. The data communications are basically for exchanging digital data among computers. Thus, data accuracy is the most important. When there are errors, data are retransmitted again under a shake-hand protocol. Since the data are parcelled into packets and each packet can be travelled through a different path, the packet will be arrived with timing difference [1].

As computers are used at every business and many home, connections among computers are possible and data transfer rates are also increased tremendously. Moreover the internet with a web browser has contributed networking into the hottest topic in the computer industry. The high speed network can even support voice traffic with acceptable delay.

Two separate networks of PSTN and data network to be maintained need high costs, and PSTN cannot be easily maintained because of frequent moves. There will be advantages from integrating voice and data traffic. Voice traffics can be merged into data traffics in the data network. Integrated networks will reduce operational costs, maximize efficient use of resources, and provide flexibility in constantly changing and very competitive communications area.

The integration requires data networks that are optimized for data transmission but can support voice transmission to be ended with reasonable satisfaction of users. To integrate voice and data traffic or even to support multimedia traffic, the quality of service(QoS), functionality, and reliability should be at least the same as PSTN. We are in the transition period from separate networks developed from different backgrounds to a single packetized network for multiple media support.

## II Technologies in Voice and Data Integration

To smoothly transit from circuit switching network to packet switching network, it is essential to merge the signaling system of the PSTN in packet switching network. The signaling system provides connection path determination, call forwarding, and other advanced call services. Different technologies are developed for integrating voice and data for the different types of networks such as local area network(LAN), frame relay network, asynchronous transfer mode(ATM) network, and internet protocol(IP) network, because each network has different characteristics and is optimized for different applications.

### A. Voice over LAN

In most of business, LAN is well established and daily workings are done in networked computers. But telephone system has also played an active role in everyday business. In a private company, the private branch exchange(PBX) works as a single switching point for internal calls and as a gate for external calls. Even though the PBX system has been popular, there are a lot of problems, too. If the PBX fails, the entire phone network is dead. When you wants to expand the phone system above a certain limit, it cannot be acceptable. You have to buy a new large system. Programming a PBX askes specially trained man.

Beginning from 1997, the computer networks such as LAN started to route voice traffics. Completely new approaches to business telephone systems have been showed up for voice traffic. But voice over LAN offers many benefits over traditional PBX. Since there is no central point, extremely reliable, fault tolerant telephone networks can be deployed with packeted networks. Routing long distance calls over a wide area network(WAN), big budget saving is possible.

After a high speed LAN such as switched 100Mbit Ethernet is installed, almost unlimited capacity of telephone network can be designed. Phone network is built upon an open architecture such as H.323 standard. H.323 is for a multimedia conferencing standard that supports real-time conversational two-way video and audio in packet switched networks [2]. All these benefits cannot be obtained without user friendly telephones. The development of Ethernet phone as a replacement for traditional phone is very exciting so that the phone becomes a LAN appliance and it's very easy to use. Since Ethernet phone can have lots of functional buttons, many advanced call services such call forwarding, call identification, and recording call, and tracking call can be easily realized [2].

### B. Voice over Frame Relay

Frame relay has been designed to integrate the high speed of a dedicated line with the switching capability of a network. It is a specification for a digital network that provides virtual circuits with shared links. Because of lower overhead, frame relay networks have higher performance than traditional networks. A frame in frame relay networks is a basic parcel size of data that has a variable length with the maximum 4096 bytes.

The use of long, variable length frames is attractive to data transmission that has often bursty traffics. But this causes problems for voice traffic because the variable length means variable time delay that cannot be acceptable in voice communications. Frame relay networks provide a statistical average delay. Long delayed frames will cause gaps in the reproduced voice. When many frames arrive at a receiver, the receiver cannot handle them all. The receiver will drop excess frames, resulting pops, clicks in the recovered voice.

A frame relay network can only detect the transmission error but cannot correct errors or does not retransmit the error frame. Frame relay thus does not guarantee end-to-end delivery of data packets. A frame can be lost or be discarded during transmission. Generally voice system compensates for missing frames by copying previous frame to approximate the lost data. Because of inherent delay problem, frame delay network will be an interim solution for voice and data integration.

### C. Voice over ATM

Asynchronous transfer mode(ATM) is a high-performance, cell-oriented switching, and multiplexing technology that utilizes fixed-length packets to carry multimedia data including voice. In order to handle delay sensitive data such as voice, ATM uses the fixed size packet, so called, the cell. As each cell has a 5 byte head and a 48 byte payload and is passed through hardware switches, a high performances are obtained. A 53 byte cell is a compromise between the longer length cell for data transmission and the shorter cell for voice transmission. The delay sensitive voice is packeted more often in the input queue of the ATM to meet the real time requirement of voice. The ATM can thus handle a bursty traffic with a kind of dynamic bandwidth.

The circuit-based technology for real time voice is mapped to ATM in the permanent virtual circuit(PVC) with a constant bit rate(CBR). This is possible using circuit emulation and ATM adaptation layer 1 [3]. This approach must have a dedicated bandwidth for traffics, which is clearly significant disadvantage. Real time voice can be handled using the variable bit rate(VBR) real-time connections. VBR for voice increases bandwidth efficiency exploiting silence periods of voice. Note that there is about 50% silence periods in a common talk. Adapting voice coding, more bandwidth reduction is possible. If variable rate coding is acceptable, network congestion can be also avoided by reducing the bit rate of voice coding.

### D. Voice over IP

Internet Protocol(IP) supports connections in the different types of networks. A networked computer can be connected through the internet to another networked computer worldwide. IP phone system has commercially introduced by VocalTec in 1995 [4]. The IP phone initially was not easy to use comparing to traditional telephone because a call setup should be arranged using a e-mail or a standard phone. But the opportunity for low-cost long-distance or international call excited telephone user community but traditional telephone industry faced a big challenge.

The devices connected to the end of IP telephone system may be standard phones, fax machines, multimedia personal computers(PCs), or IP digital phones. The devices does voice/fax compression, packets generation, and applications interface. The figure 1 show the overall functional diagram.

In figure 1, gateways provide a routing service and servers does some advanced applications in addition to connecting.

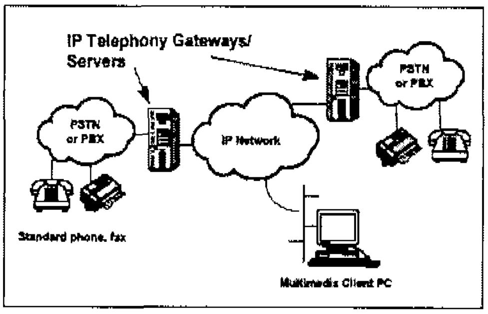


Fig. 1 Overall functional diagram of IP phone system.

Delay in ATM is also a key element for high QoS. Toll quality voice service requires delay not greater than about 0.25 sec. Even though ATM is designed for real time data such as voice, networks should be managed carefully to avoid congestions or burst error generations.

## III Current Trends and Problems

Voice over LAN will replace the function of PBX and remove telephone lines in a company. PSTN will be replaced with a voice over IP(VoIP) system. All the networks including PSTN are merging into a single high speed data network under internet protocols. VoIP is now possible in the communications environment such as PC-to-PC, PC-to-Phone, Phone-to -Phone, and PC-to-Cell Phone.

The most promising areas for VoIP are intranets in a company and commercial extranets. Gateways in VoIP should be able to handle hundreds of calls simultaneously. As even PC-based gateways can handle these amount of calls. Considering these facts, IP networks will work as a unifying agent for all types of data [5].

IP networks are often based upon public wide area networks of which performances cannot be easily improved significantly even after investing a lot of money. Moreover, video traffics are rapidly increasing in the internet and they requires much greater bandwidth than voice traffics. All these factors can degrade reproduced voice quality. Since there are still many households that do not have a PC, or do not connected to a network, VoIP cannot cover those homes.

However, sales related to IP phone are growing rapidly. In U.S., it is expected that they will reach to 2 billion dollar in 2004. In Korea, there are several millions of IP phone users in 2000. IP phone users are making long distance, international, or cell phone calls without paying a cent.

## IV Conclusions

Voice communications have done through PSTN based upon circuit switching. Data communications have designed for non-delay sensitive traffics. But voice traffics are integrated into data traffics as data networks can handle delay problems. future IP phone service will depend on high speed and reliable WAN, advanced voice coding, and user friendly IP phone.

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