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Adaptive Strategy for Voice over Internet Protocol in New Zealand

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1. ABSTRACT

The cost effectiveness of voice communications over Internet Protocol based networks in general has been amply demonstrated. The Internet as the infrastructure for regional and global communications has also been accepted as a norm. However, the variability of this infrastructure in terms of its media bandwidth and networking component latency constraints imposes varying quality of voice communications for different regions.

This paper briefly covers issues of packet loss and delay and the pros and cons of alternative approaches like Automatic Repeat Request (ARQ) and Forward Error Correction (FEC) to achieve the desired quality in voice communications over Internet Protocol networks. The paper outlines the experimental constructs and the results obtained in a study of existing behaviour of IP network in the New Zealand region, conducted as part of a Master of Computing thesis investigation. The application of the strategy that evolves from the experiment can be extrapolated to cover other links in the NZ region in situations where the links under consideration are similar to the experimental link.

Keywords

Voice communications, Internet Protocol, Forward Error Correction, adaptive strategy

1. INTRODUCTION

Two definitions should help to clarify what Voice over Internet Protocol involves:

"Voice over IP means the transmission of voice traffic in packets." (Black, 2000, p.1)

"VoIP (Voice over Internet Protocol) is a term used in IP telephony for a set of facilities managing the carrier voice information through the Internet using the Internet Protocol (IP)." (Search Networking, 2001).

Voice over IP is an extensive subject, but at the core it comes down to trying to transport speech signals in an acceptable way from sender to designated receiver over an IP network. An Internet Protocol (IP) network is a computer network which uses the IP packet (or datagram) to transmit information. In order to transfer analog voice messages over a network, voice is digitized with an ADC (analog to digital converter), transmitted, and received with DAC (digital to analog converter) to transform the digital signal back to an analog voice message again.

During the VoIP transmission, voice and other audio signals are digitized and packetized into TCP/IP (Transmission Control Protocol/The Internet Protocol) packets. TCP/IP is the basic communication language or protocol of the Internet. Networks such as the Internet carry IP packets containing a header (to control communication and address destination) and a payload to transport data. When implementing VoIP technology, voices are digitized into data packets, transmitted over the Internet and reconverted into voice at their destination. This process is open to packet loss and delay, which can be reduced by using strategies like Forward Error Correction (FEC) - the primary focus of this paper. The paper explains why use of both VOIP and FEC is increasing significantly and outlines the results of experiments conducted to see whether loss rate and loss time in New Zealand are similar to those measured in other countries, particularly America.

1.2 WHY USE VOIP?

It is important to notice that IP is not a particularly suitable protocol for telephony. The IP network was originally designed to transport data traffic, and it is an unreliable mechanism. However, during the past several years, there has been a significant increase with the use of packetized audio over packet-switched network. This is induced by the almost universal presence of IP, associated protocols and equipment among users. "IP is chosen protocol for Internet telephony because, as the mountain climber says, 'it is there.'" (Black, 2000, p.7) At present, the Internet is indeed being used to carry voice conversations in certain specific applications.

Besides, VoIP technology could have a number of advantages over traditional circuit-switched PSTN (public switched telephone network). Firstly, Internet telephony could reduce much of the communication cost especially for long distance overseas telephone calls. "It has been estimated that packet voice networking costs only 20 to 30 percent of an equivalent circuit-based voice network. This is true for both carriers (service providers) and enterprise (private) users. Logically, this implies that enterprise users can operate long-distance voice services between facilities at less cost than purchasing long-distance voice services from a carrier, and it's often true." (Cisco Systems, 2001) Secondly, VoIP can use compression techniques (codec) to reduce the bandwidth consumption such as G.723 or G.729A. Thirdly, VoIP has the advantage of enabling better controlled (e.g. multicasting) efficient data storage, better noise tolerance, and providing value-added services (e.g. web whiteboard).

However, there are some problems with the integration between VoIP and the Internet. Voice data communication must be a real time stream, but this is in contrast with the Internet's heterogeneous architecture that can be made of many routers, and the routing method can result in a very high round trip time (RTT), so it is necessary to modify communication parameters to get the VoIP architecture properly working.

If we want to deliver real time speech over the Internet, we should realize that there are several differences between circuit switching networks and packet switching networks:

- Packet loss: the packets sometimes may be lost on a packet switching network but seldom on circuit switching networks.
- Packet delay: The packets suffer from much higher delay on packet switching
- networks than on circuit switching networks.
- Delay jitter: The delays of the packets are variable on packet switching networks but fixed on circuit switching networks. (Kostas, Borella, Sidhu, Schuster, Grabiec, and Mahler, 1997)

A major concern with VoIP is the Quality of Service (QoS). Factors that affect the QoS, such as packet loss, packet delay, and delay jitter, must be overcome to make the QoS better.

3. FORWARD ERROR CORRECTION

3.1 WHAT IS FORWARD ERROR CORRECTION?

Forward error-correction (FEC) is a type of digital signal processing that improves data reliability by introducing a known structure into a data sequence prior to transmission or storage. This mechanism enables a receiving system to detect and possibly correct errors caused by corruption from the communication channel. As the name implies, this technique enables a receiver to correct errors without requesting retransmission of the original information. (Wang, Sklar, and Johnson, 2002) "FEC is an essential building block of any satellite or IP multicast based content distribution system" (Onion Networks, 2002)

The data communication industry has devoted considerable effort to research and development of forward error-correction techniques, with particular emphasis on improving the quality of packet-based multimedia communications systems such as VoIP. This work has played an important role in supporting real time interactive communication media over the Internet, including Internet telephony, Internet video conferencing, and long distance teaching.

3.2 How does FEC work?

The purpose of FEC schemes is to anticipate errors and provide information redundancy, allowing the receivers to reconstruct the information without asking the sender to retransmit. "Forward error correction is a methodology that uses error correction coding for transmission. The concept of FEC is the opposite of ARQ (automatic repeat request) which uses retransmission of data." (Shacham and McKenney, 1990). The FEC scheme requires a sender to incorporate error-correction information into each datagram in a data stream. If one datagram is lost, the correcting code contains sufficient redundant information to allow a receiver to reconstruct the missing datagram without requesting a retransmission.

According to Biersack (1993), it has indeed been shown that, even for a small ratio, FEC can be very effective and reduces the loss probability by several orders of magnitude. The trade off of adding redundancy is increasing network traffic. But in general, it is very difficult to guarantee a bandwidth in The Internet for VoIP application, and implementing an FEC scheme would consume network bandwidth. Therefore, it is important to notice that it is possible that adding an inappropriate amount of redundancy may result in a higher packet loss rate.

3.3 Why use FEC?

A lot of schemes were proposed to control the error in the audio transmission system. Some of these schemes could be used in some fields but may not be used in other fields. In general, these error-control techniques for communications are classified into two groups: FEC and ARQ (retransmission). Is retransmission a suitable method to recover the lost real time speech information packets? The answer is negative, because of delay constraint. The retransmission will make QoS poor, make the playback strategy complex and consume more network bandwidth. Due to these reasons, packet retransmission is not a suitable and feasible solution to the packet loss problem.

4. THE EXPERIMENTS

Because there are no reasonable mathematical expression to simulate the loss rate, loss burst size and round trip delay, end-to-end measurement is necessary to assess what occurs while a session is proceeding. Because previous papers did not discuss every factor completely, and the packet switching network characteristics mentioned in these papers may not be the same as those in New Zealand, it is necessary to measure the status again. In this way, the measurement results will be more suitable for development of VoIP systems in New Zealand.

There are many convenient built-in tools, such as 'ping' and 'tracert', that are used to show the status of the networks, but their functionality and usability are not complete and sufficient for the measurement of parameters in the experiment of this research. For each scenario in our measurement, some parameters had to be set to satisfy our requirements; for example packet inter-departure time and packet size. To meet this requirement, tools were developed using C and C++ programming language. They measure the end-to-end status, such as round trip delay and packet loss burst size, that affect the QoS while delivering voice over the Internet.

5. CONCLUSIONS

Full results and detailed discussion will be reported in further papers. The intention here is to give an overview of the conclusions:

- ♦ The loss rate increases as the packet interdeparture time decreases.
- ♦ Smaller packet interdeparture time make the loss more "bursty".
- ♦ Total loss time decreases as the packet interdeparture time increases.
- ♦ The loss rate, probability of the loss burst size, and total loss primary time are not affected by the number of redundancies.

Since the environment of networks in New Zealand is different from America or other places, the characteristics of the networks may be different from each other. There are a lot of papers that discuss the considerations of systems that deliver voice over the Internet, but all of the measurements were conducted in America or other places. In view of the differences in the environments of the networks, a series of measurements were conducted in New Zealand.

The results show that the effect of the packet interdeparture time, i.e. packet size, is not the same as in America. Su, Srivastava, and Yao (1999) stated that the packet loss rate increases as the packet interdeparture time increases. The results of the measurement in New Zealand show that the packet loss rate decreases as packet interdeparture time increases. This difference may be caused by differences in the environments of the networks or just by the differences of the link. The difference between these two results demonstrates that the effect of the packet interdeparture time varies from case to case.

Another result showed that the number of the redundancies does not affect the QoS of the system for delivering voice over the Internet. Based on this result, a strategy can be developed by concentrating only on the packet interdeparture time, which affects the probability of the loss burst size, total number of the bursts and the total loss time.

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