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Voice over Internet with text chat

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ABSTRACT

VoIP is a protocol designed to improve voice over IP and other networks based on packet switching. The birth of VoIP came as an alternative to the expensive public telephone network (PSTN) for voice transmission. Given QoS for VoIP, factors such as system capacity, frequency, packet delay, loss, and channel configurations are of paramount importance. These parameters with security issues, channel bandwidth allocation, and reliability issue represent major challenges facing VoIP systems. To master Internet voice communication systems, a clear understanding of Internet Protocol (IP) is mandatory. The audio is transferred by the RTP protocol that is added in the IP packet. In this paper, I will focus on the techniques used in VoIP systems and the challenges faced by these systems. As the voice transmission market in transition from traditional PSTN networks continues to VoIP, new technologies are being tested and tested to improve the quality of services provided by VoIP systems. The main focus is finding ways to deal with high traffic and booming demand for VoIP systems.

Keywords: *Voice over IP, PSTN, QOS, Jitter, Packet Loss, Packet Delay, IP, RTP, Channel Bandwidth*

1. INTRODUCTION

Interest and the need for VoIP have been noticed since the introduction of the first computer network. The main objective of VoIP systems is to provide either very difficult or too expensive services to implement using the traditional PSTN network. The VoIP system is primarily based on the provision of voice communications using existing Internet Protocol (IP). [1]

The Internet has proved to be a cheap way to send data, like emails, to the world over the years. As a result, VOIP was invented to transmit voice over these cheap media and reduce the high costs associated with conventional telephone lines. VoIP will

Enables people to communicate via voice around the world at a much cheaper price than people used to spend on regular phone lines. This significant difference in cost has highlighted VoIP systems to be an interesting area for researchers.

Conversely, cheap telecommunications costs do not cost the best service. Applications associated with using VoIP since its first use has shown far less performance than traditional telephony services. Given aspects such as quality of service, VoIP still needs a lot of improvements to stand up to traditional phone lines. Unsatisfactory data quality transferred through the Internet infrastructure may be due to the uneven nature of Internet services available in different parts of the world. [2]

Currently, there is a lot of interest in the use of VOIP over cellular networks. Given the increasing number of VoIP users on the Internet and the constant need for mobile operators to increase profits, the introduction of VoIP over cellular networks is worth cracking. By introducing VoIP into cellular networks, existing cell operators can easily and at the low-cost switch to all IP networks. This step will significantly reduce operational costs and thus increase profitability. Operators will benefit not only from this but also from users. Users will enjoy a much cheaper voice connection. This will be accomplished using devices such as truphone. This device carries programs that will enable the mobile phone to be converted to a VoIP phone while connected to the Internet. [3]

All this can be achieved if the quality of service for VoIP systems is maximized. To achieve this type of desired quality of service, things such as system capacity, jamming, packet delay, loss of channel configuration must be placed under serious monitoring.

2. HOW DOES VOIP WORK?

Due to the fact that VOIP uses regular Internet architecture, it works more or less like IE Other Internet service in the perspective of the basic level. At the end of the transmission, the data (sound) is subjected to analog compression and then divided into data packets with distinct serial numbers. The search table will then assist the server at the end of the transmission to determine the IP of the intended receiver. Once the server ejects the receiving IP, it initiates the data transmission in the same way as sending e-mail messages and other Internet data.

At the end of the future, the data is collected and arranged according to the sequence numbers that you carry, and then undergo digital to analog conversion to enable the receiver to hear the sound. Transport using VoIP can be between two computers, two phones, or even between a phone and a computer. Using VoIP, a person can even use a computer to create a call to a landline or vice versa. [Jain, 2004]. Figure 1 displays a network overview of VoIP functions.

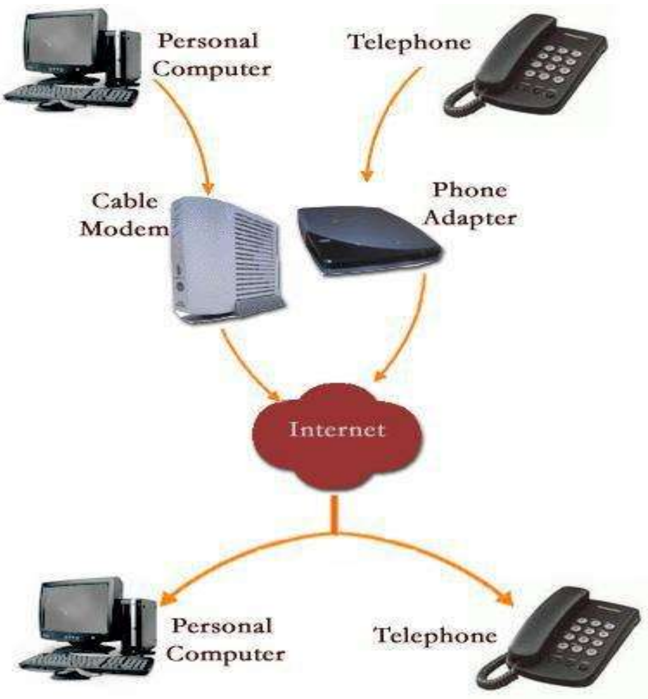

Fig. 1: Basic VoIP functionality

3. CARRYING VOICE OVER IP

As previously reported, in VoIP systems, the sound is made using the current IP. This command is done by adding the RTP header in the IP packet. Figure 2 shows Regular IP package and VoIP package where the RTP header is included.

RTP is encapsulated in UDP and IP. The audio range for each conversation depends on the CODEC and sampling rate.

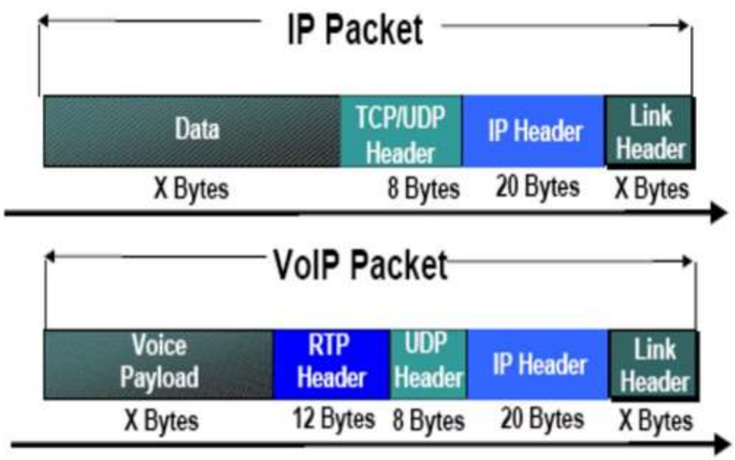

Fig. 2: RTP header added onto an IP packet

4. VoIP PERFORMANCE

In order for VOIP to stand against UMTS, the desired quality of service must be achieved. As we mentioned earlier, one of the reasons for the low performance of Voice over Internet Protocol is that the different regions of the world have a distinct nature in terms of Internet performance. To analyse the performance details of this system, we need to consider the following parameters:

• System capacity / bandwidth available
• Package loss
• Network delay / delay
• Cut off
• Sound echo
• Security

These factors are so far the main challenges facing VoIP systems. All these parameters are variables that can be changed to achieve the desired quality of service.

5. WHAT IS QUALITY OF SERVICE?

Quality of Service or Quality of Service is a comprehensive set of network standards and mechanisms that ensure that services are delivered with high performance. Network administrators typically use QoS mechanisms as a reference model to optimize the use of existing network resources to achieve the desired performance without having to expand or provide more resources for the network.

At first, network quality meant equal treatment for entire network traffic. This means that the best effort in the network is distributed equally across all visits. This condition does not provide any guarantee of network performance characteristics such as delays, changes, reliability, and security. QoS is here to change the situation with the idea that different applications have different requirements and that different users also have different needs. The main idea of the quality of service is that the network effort should not be evenly distributed over all traffic that needs to give some priority traffic over others.

The main objective of QoS is to deliver priority delivery services to the applications they require. This is done by ensuring adequate bandwidth, monitoring and monitoring delay and frequency, and reducing data loss.

In IP-based networks, one of which is VoIP, there are two main types of service quality identified by the Internet Engineering Task Force (IETF): Intserv and Diffserv. These models contain several mechanisms that ensure the provision of preferential services for specific traffic in the network.

If VoIP applications achieve a desired quality of service, they will have the following advantages:

• Administrators will have good control over the use of network resources that will enable them to operate the network in a business perspective to maximize profits.
Time and sensitivity applications and users will be provided with the resources they need at the same time, and other users can access the network.
• The user experience will be improved as a result of improved system performance.
• Given the fact that existing resources will be used optimally, the overall operating cost will be reduced. This also ensures that there is a minimum need for expansions and upgrades. [5]

VI. CHALLENGES TO IP

A. System capacity/bandwidth available.

Providing sufficient bandwidth for voice transmission is the first important step towards achieving the desired quality of service. The challenge here is that the available bandwidth is a limited resource. This means that VoIP systems must be designed to take advantage of the available bandwidth without exceeding the maximum, while efficiently providing the real-time efficient voice. Table 1 shows the availability of the VoIP bandwidth.

Compressed RTP Bandwidth Calculations			
CODEC	PPP 6 Bytes of Header	ATM 53 Bytes Cells with a 48 Byte Payload	Frame-Relay 4 Bytes of Header
G.711 at 50 pps	68 kbps	N/A	67 kbps
G.711 at 33 pps	44 kbps	N/A	44 kbps
G.729A at 50 pps	12 kbps	N/A	11.2 kbps
G.729A at 33 pps	8 kbps	N/A	7.4 kbps

Table 1: Bandwidth provisioning for VoIP

A more accurate way is included to provide the second layer header in the bandwidth accounts. Results as shown in Table 2.

CODEC	Sampling Rate	Voice Payload in Bytes	Packets per Second	Bandwidth per conversation
G.711	20 msec	160	50	80 kbps
G.711	30 msec	240	33	53 kbps
G.729A	20 msec	20	50	24 kbps
G.729A	30 msec	30	33	16 kbps

Table 2: Bandwidth provisioning for VoIP with the header 2 included in calculations.

CODEC	Ethernet 14 Bytes of Header	PPP 6 Bytes of Header	ATM 53 Bytes Cells with a 48 Byte Payload	Frame-Relay 4 Bytes of Header
G.711 at 50 pps	85.6 kbps	82.4 kbps	106 kbps	81.6 kbps
G.711 at 33 pps	56.5 kbps	54.4 kbps	70 kbps	54 kbps
G.729A at 50 pps	29.6 kbps	26.4 kbps	42.4 kbps	25.6 kbps
G.729A at 33 pps	19.5 kbps	17.4 kbps	28 kbps	17 kbps

Table 3: RTP bandwidth allocations CD-ROM

VoIP services use symmetric bandwidth in the uplink and downlink. The main problem is that there may be an imbalance in the uplink and downlink bandwidth even at the HSDPA system stage.

The ability of the system also affects bandwidth traffic for each subscriber, thereby reducing the number of subscribers. In GSM and UMTS systems, the 12.2 Kbps audio codec is widely used in CS audio services. In the case of a VoIP protocol stack, the Routing Table Protocol (RTP) and the User Datagram Protocol (RDP) are used. These two protocols are implemented by an IP packet. Because the IP packet carries RTP, UDP, and IP addresses, the audio requires a 32 kbps / 64 Kbps data rate for successful transmission. [6]

B. Loss of loss

In IP networks, voice is treated as normal data. As a result, voice packets are vulnerable to unfortunate situations that are dropped when traffic is high and the network is congested. Re-sending lost data packets can solve the problem in data transfer, but this is not a solution for voice data. These solutions fail to transmit audio data because audio packets can contain a range of 40 to 80 milliseconds of speech information. Package loss significantly reduces the quality of service in systems. In systems such as Vocoder ITU-TG.711, a standard for fee quality, the packet loss rate can lead to a serious degradation of the user experience. Other types of programmers who implement more severe data compression tend to deteriorate more severely. [13]

In the jitter calculation, which I will discuss later in this article, missing packets are usually ignored because they are considered infinitely late and used in accounts that will distort the accounts. Package loss can be compensated in the endpoint with algorithms such as Hide Package Lost (PLC) or Recover Package Recovery (PLR). Netload frequency can be applied to meet packet loss, but its use requires additional bandwidth. [7]

In order to secure sufficient bandwidth for packets in a VoIP channel, the network device must be able to select VoIP packets. This means that VoIP packages must be selectable from all other IP movements. Network devices implement this definition by pointing to source and destination IP headers or Datagram Protocol (DAT) headers. The package definition process is called classification, which is the basic basis for achieving the required quality of service.

Another way to implement classification is using the RSVP mechanism. This mechanism is dynamically classified as opposed to the above, a static classification method.

Once the evaluation process is completed by each hop in the network, each VoIP package is provided with the desired quality of service. In this range, special techniques can be set to achieve the priority queue. The priority queue ensures that any large embedded data packets do not interfere with continuous voice transmission and reduce bandwidth requirements in the fact that they push the 40-byte IP and UDP with RTP headers to only 2 or 4 bytes.

C. Network Delay / Delay

Network delays are a condition that arises when audio packets take longer than expected to reach their destinations. This condition eventually causes distortions in sound quality.

When you move audio packets, some of them are delayed and arrive at the destination later and then expect. This delay may be caused by many factors and the main reason is the basic network. Late packets usually arrive at the destination late or never at all. The quality of service tends to be more tolerant with packet loss than text.) [9]

The main known causes of delay are:

• Encoding
• Waiting
Wait for the package to move
• Sequencing
• Insulation buffer

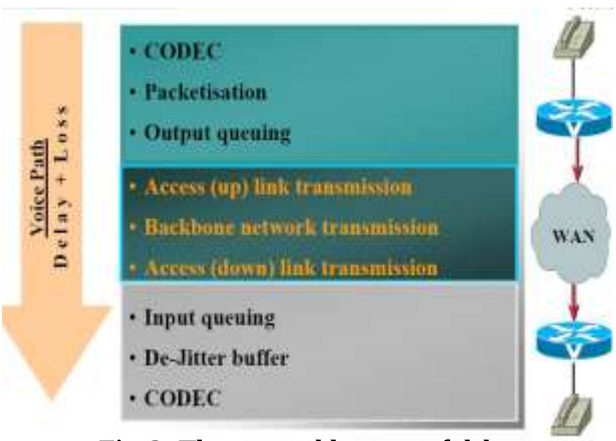

Fig. 3: The acceptable range of delay

Figure 3 shows the acceptable delay for various applications. Some of the causes of delay can be dealt with but some of them have no solution. Figure 4 shows the delay components at different transmission levels. [10]

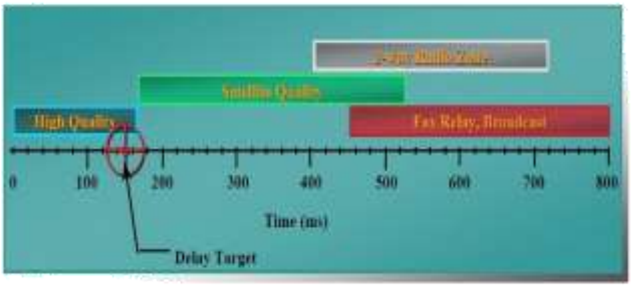

Fig. 4: delay components from source to destination.

Delay can be classified into two categories: static delay and variable delay also known as tremor. Figure 5 shows the reasons for the persistent delays in the network and its causes. Fixed delays are due to propagation, sequencing, and processing as shown in Figure:

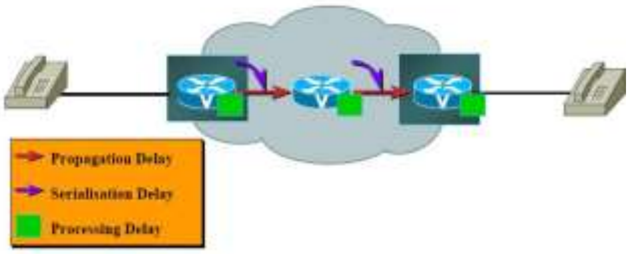

Fig. 5: fixed delays in a network

The propagation delay is usually about six microseconds per kilometer. The sequence delay in the buffer occurs to a serial link. The processes that impose delay are such as codecs, compression, packets, decompression, and decryption.

The other type of delay is the variable delay, known as jitter. Figure 6 shows the variable delay in the network. The main component here is the queue delay that occurs throughout the network and is greatly affected by the size of the package. The other thing that contributes to the variable delay is the temporary attenuators that lead to a variable delay in order to get rid of the sound. [4]

Fixed delays are out of our control but delays can be reduced by practicing the learning of sound packets as being sensitive to delays. Another solution is to mitigate the effects of anger in the jitter buffer once they reach the destination. This process has a side effect to increase delays. [1]

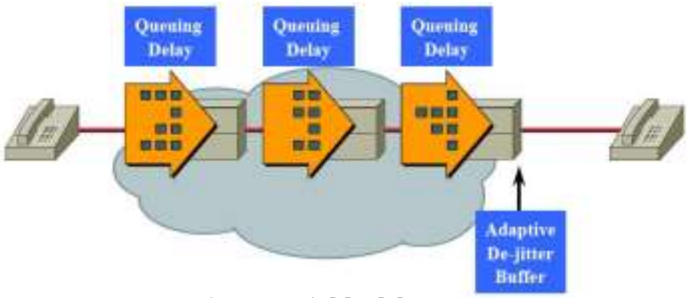

Fig. 6: variable delays

D. Jitter

Recall earlier, anger is a variable delay mainly caused by queuing, contention and sequencing along the grid. In general, jitter occurs more often in slow or overcrowded joints. QoS-based queues based on separation, bandwidth reservation, and faster connections can significantly reduce future jitter problems. Until then, oscillation remains an infamous flaw to the voice over Internet channels.

Real-time audio jitter can be classified into 3 types;

Type A: This type is classified as continuous anger. Package integration to packet delay is almost constant.

Type B: This type is called transient tremor. The main feature of this type of jitter is that it has a significant gradual delay that may affect a single package.

Type C: This is a jitter that consists of short-term delay differences. Here increases the delay that affects many packages. Apart from that, there may also be a package for packet delay change. This type of jitter usually results from congestion and road changes.

The transmission time jitter can occur in mobile phones because the processes involved in VoIP systems must compete for CPU time with other processes. This jitter is due to scheduling delays.

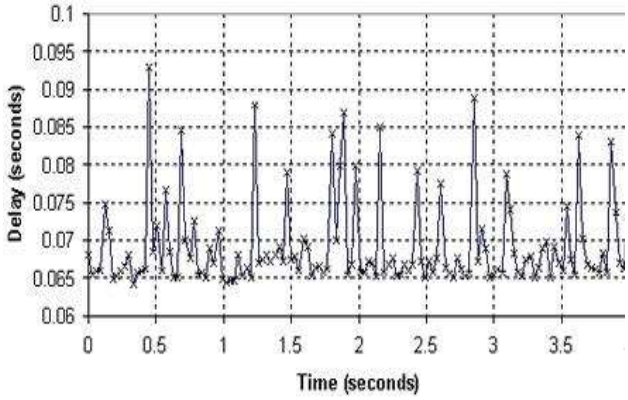

Fig. 7: Experimental results of the effect of packet congestion on delay (x = packet)

Figure 7 shows how severe congestion in the network increases delay time. X represents the beam graph and we see that in the busiest places for packets, the delay is more.

In Figure 8, the other side is investigated affects the delay, which is congestion access link. Other major causes of delay are;

• Sharing load between multiple access connectors or IP service providers
• Sharing the load within the IP service
• Sharing load-routing
• Routing table updates
• The road is fluttering
• Drift timing

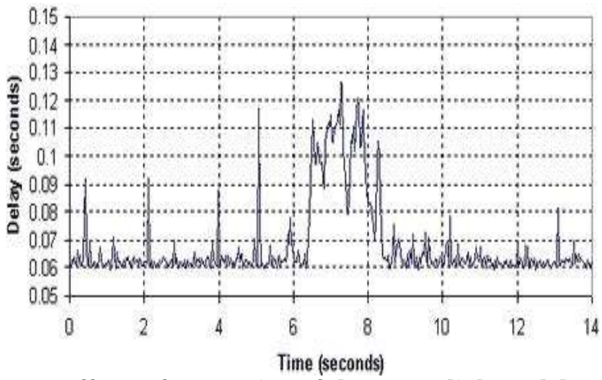

Fig. 8: Effects of congestion of the access link on delay

The most commonly used treatment for the removal of jitter effects is the use of buffers. The jitter buffers are designed to erase jitter effects from the encoded audio stream. This process is done by buffering each individual packet for a short interval before the recipient hears it. As a result, additional delays were introduced and some packets were lost but resolved. Adaptive jitter stores are more preferred than shuffle buffers because they are able to adjust their size, improve delay and avoid swaps.

With regard to delay, both fixed and adaptive shock valves are capable of making automatic adjustments in accordance with changes in delays. For example, if there is a delay in changing the step by 19 milliseconds, some packets may be discarded because of the change, but the jitter buffer will be rearranged quickly.

The timed buffer is usually seen as a time window with the early side alignment of the minimum delay and the delayed side representing the maximum allowable delay before the package is considered abandoned [11]

E. Echo

Sometimes when VoIP users make calls they can hear their voice reflected on their phone speakers after a few milliseconds. This disturbing phenomenon is known as an echo. The interval between talking and listening to your voice varies with different causes of echo. The short interval echo does not cause much damage but one can destroy the entire conversation. Delayed delays are delays and delays. PSTN is echoed but not as much as VoIP systems. This is because the PSTN has a much lower delay compared to VoIP. The maximum allowable delay for the PSTN network is about 10 milliseconds while VoIP can be Up to 400 milliseconds. This means that VoIP is more susceptible to echo.

When a part of the speaker's voice is played back, it is known as the loudspeaker echo. The listener's echo occurs when a portion of the speaker's voice is echoed by the listener and followed by a second echo that causes the part of the signal to reflect the listener. The end result is that the listener hears the speaker's voice twice, that is, he repeats.

The other type of echoes is the echo of convergence. This occurs at the beginning of the call and occurs because of the delay in the echo cancellation affinity.

To solve echoes, VoIP portals utilize line echo residues to eliminate or reduce echo levels from analog rings. The identification and verification of an echo source is an important process involving echo removal. [9]

Echo sellers typically face the back-end of the PSTN and remove the echo in the tail circuit on its side of the network.

F. Security

Although it is much easier to secure your phone using VoIP than PSTN, only a large number of VoIP consumer solutions do not support encryption yet. This makes it easy to do eavesdropping in VoIP and even change the contents of the data.

There are many open source solutions that make it easy to sniff VoIP conversations. A small degree of security is provided through the use of patented, proprietary audio codecs that are not easily accessible to open source applications. The use of this method of security has not proved effective.

The pressure in use is put in place by some vendors to counter eavesdropping. This method also makes it difficult to eavesdrop only but does not prevent it. Encryption and encryption are necessary to ensure proper VoIP security. There are possibilities to use IPSec to secure VoIP transmission by using opportunistic encryption. [12]

7. CONCLUSION

Voice over Internet Protocol (VoIP) is an advanced step in voice communications that benefits from widespread deployment and establishes the core Internet. Voice over Internet Protocol (VOIP) service has been able to provide a less expensive way to communicate but is still not fully embraced by everyone. This may be because of its low-cost swap for the poor quality of service.

The main reason behind this type of service quality in VOIP service is that it is because the Internet is not designed for voice transmission. This is due to the fact that the performance of Voice over Internet Protocol (VoIP) is greatly hampered by factors such as delays and packet loss. Delay has a much greater impact on VOIP performance due to the sensitivity of the audio data to the delay.

The nature of sending voice data over the Internet will always cause packet loss. The techniques used to address packet loss need to be monitored closely as most of them cause loss of synchronous packet delays.

Apart from delay, loss, and loss of firmness, the issue of security and reliability often arises because of the fact that the voice is transmitted through a wide-spread public media; the Internet.

In conclusion, the PSTN system is designed for the sole purpose of carrying audio. Will the use of the Internet as a backbone be used to transmit voice to PSTN standards? So far, I can say that VoIP can only be used with PSTN and not replaced. The VoIP service may have an opportunity to replace the PSTN if and only if specific communication standards are set for VoIP, solutions are determined for compatibility queries and the system is developed for cross-platform communications.

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