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Quality Evaluation of VoIP Codecs in Cellular Mobile Networks

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Abstract

Maintaining the Quality-of-Service (QoS) of Voice-over-Internet Protocol (VoIP) is a major problem due to the instability of network conditions. One factor that influences VoIP call quality is the employed codec. In this research, the influence of codec on the QoS of VoIP calls is evaluated under Wi-Fi, 2G, 3G and 4G networks. VoIP calls are established using PCMA, PCMU, GSM and G.722 codecs and QoS parameters including jitter, round trip time and packet loss are measured. QoS measurements are compared against considered networks and codecs. Results illustrate that there exists a codec that better suits each network with considerable advantages over other codecs and minimum drawbacks.

Keywords: Audio Codec, Jitter, Quality-of-Service, Round Trip Time, Voice-over-IP

1. Introduction

Quality-of-Service (QoS) is considered a major factor that impacts the success and continuity of Voice-over Internet Protocol (VoIP). Traditional calls over PSTNs enjoy quality stability for the call duration provided by the guaranteed dedicated bandwidth [1] In contrast, VoIP calls rely on the Internet to route data. Consequently, the call quality is influenced by the network existing state in terms of number of users, congestion, and other conditions. The choice of codec used for audio compression and decompression also influences call quality, since these codecs vary in algorithm, bandwidth requirements and side effects. The dynamic nature of network conditions causes fluctuation in the performance of codec, and therefore must carefully be selected so as to minimize degradation in call quality. The problem of VoIP call quality remains an open research area highlighted by the use of IP to deliver voice calls in 4G.

A number of studies have compared the performance of audio codecs in literature [2-10]. However, no prior work has investigated the influence of different audio codecs on the performance of available cellular mobile networks. Different features and minimum/maximum data rates provided by cellular networks influence the performance of audio codecs. There is a need to carry out practical experiments to examine this impact. Therefore in this paper, the QoS of VoIP codecs is investigated for use in cellular mobile networks. Asterisk software is used to convert a personal computer into an advanced telephone exchange with multiple functions. It is used to make VoIP audio calls under a number of scenarios. In this evaluation, four codecs are considered: G722, GSM, PCMA and PCMU. Each codec is used to encode voice which is then transmitted over the wireless network. The evaluation is carried out considering three cellular mobile data networks: 2G, 3G and 4G networks. The results are compared with the QoS of VoIP calls over Wi-Fi. The evaluated QoS parameters are jitter, round trip time and packet loss. The quality of voice calls is analyzed using Wireshark software and the parameters are measured and compared under each codec and network scenario.

The remainder of this paper is organized as follows: Section 2 overviews related work. VoIP benefits, challenges and considered codecs are explained in Section 3. In Section 4, the VoIP system and QoS parameters are explained. Section 5 illustrates the experimental setup. Section 6 demonstrates the evaluation results which are discussed and summarized in Section 7. Finally, Section 8 concludes the paper.

2. Related Work

The authors of [2] compared the end-to-end QoS performance parameters of VoIP codec schemes against multiple traffic connections transmitted over the Internet. Background traffic was added to closely resemble real-world Internet. The performance of G.711, G.729A, G.723.1 and GSM.AMR codec schemes was compared using simulations. The results showed that G.729A was at least 2.81% better in terms of average accumulative end-to-end delay. G.711 resulted in at least 21.89% less average accumulative end-to-end jitter but produced the worst end-to-end packet loss ratio. In addition, GSM.AMR produced the best end-to-end effective transmission rate of up to 89.82%.

The work in [3] evaluated G.711, G.723 and G.729 codecs in terms of noise, jitter and packet delay variation. OPNET was used to design the network and two scenarios were considered for testing the performance of encoding techniques. The first scenario is a small-scale network to exemplify a company with a few branches in the same country. The other scenario is a large-scale network to simulate an international company. Results showed that G.729 experiences the least noise while G.711 network has less delay compared to the others. They also showed that both G.711 and G.729 have the same jitter and is less than the other codecs. In the second scenario, G.711 has the least jitter and packet delay variation.

Paper [4] evaluated the quality of VoIP calls made in an indoor environment through a Mobile Ad Hoc Network (MANET). The evaluation considered G.711, G.727 and G.723.1 codecs and compared the resultant Mean Opinion Score (MOS), jitter, delay and packet loss. Results revealed that G.711 provided the best performance. Similar results were obtained in [5] and [6].

The study presented in [7] compared VoIP call quality when using codec adaptation, rate adaptation and codec-rate adaptation. The assessment considered codecs G.711, G.723 and G.729. Comparative results showed that rate adaptation delivered the best performance under low to moderate traffic. In contrast, under high congestion, codec adaptation performed better. Codec-rate adaptation suited severe situations where codec adaptation alone could not resolve congestion.

In [8] the authors compared the listening quality of VoIP calls considering four codecs (GSM, Speex, PCMA and PCMU). They investigated the influence of packet loss, jitter and bandwidth on call quality. In addition, they proposed a dynamic codec switching scheme where VoIP calls can change from one codec to another while a call is in progress. This allows the application to select the codec that best suits current network conditions so as to maximize call quality. Results demonstrated that the proposed scheme improved call quality compared to a fixed codec.

The evaluation carried out in [9] uses simulations to compare G.711, G.729 and AMR-WB codecs under different network conditions. Results demonstrated that encoding using ARM resulted in less quality degradation compared to other codecs.

The study in [10] investigated the degradation in QoS for voice traffic across Wi-Fi and WiMax using simulations. Three scenarios for data transfer were considered: Wi-Fi to Wi-Fi, WiMax to WiMax and Wi-Fi to WiMax. All three scenarios were run for one hour and results were obtained for jitter, Mean Opinion Score (MOS) and packet end-to-end delay. Results revealed that WiMax outperformed Wi-Fi, which suffered from high jitter during the first five minutes. The average jitter of the Wi-Fi to WiMax scenario exceeded that of Wi-Fi at some points. In terms of MOS, both WiMax and Wi-Fi to WiMax had better performance compared with Wi-Fi. Regarding packet end-to-end delay, WiMax resulted in the best performance.

3. VoIP Benefits, Challenges and Codecs

VoIP is a technology solution that allows voice transmission over an IP network. It is utilized to make phone calls over the Internet by sending packets through the packet switched networks. This section explains the key benefits of VoIP, challenges facing VoIP calls and the main codecs used in VoIP.

3.1 VoIP Benefits and Features

VoIP has many benefits due to the use of Internet instead of traditional phone lines. Following are the main features and benefits [11]:

Cost: Compared to the traditional phone line, initial setup and operation costs are mainly less for VoIP systems, as no phone lines are required. VoIP calls from PC to PC over the Internet are free, and from PC to landline incur a cost, albeit significantly lower than traditional phone calls.

Accessibility: VoIP systems enjoy high accessibility and location independency. Internet connection is the only requirement for VoIP calls, regardless of the user location.

Flexibility: When employing a PBX (Private Box Exchange), the maximum number of phones that can be deployed in the system is bounded by the number of phone lines in the system. With VoIP, thousands of connections can be established, as the number of lines is limited by bandwidth.

Voice Quality: When the VoIP system uses a fast and reliable Internet connection, the voice quality is at least as good as that of a traditional phone connection. However call quality can be affected by low bandwidth and poor Internet.

Extra Features: VoIP systems offer a range of extra features including call forwarding, call waiting, voicemail, caller ID and three-way calling at no additional cost.

3.2 Challenges of VoIP

Similar to the features, VoIP faces of challenges mainly due to its communication over the Internet [10].

Bandwidth: The contention of data services over limited bandwidth results in congestion. Packet queuing causes latency and jitter, which directly influences the QoS of VoIP calls. Since data and voice share the same network bandwidth, it is challenging to optimally allocate bandwidth to ensure acceptable QoS for VoIP calls.

Low Quality of Voice Calls: In addition to a weak or unstable Internet connection, many other factors can influence the quality of VoIP calls including increased network load and signalling protocols incompatibility.

Power Failure: Traditional telephones operate on 48 volts supplied by the telephone line itself without the need for an external power supply. Therefore, a power failure does not influence service availability. However, since VoIP equipment such as PCs rely on an AC power source, a backup power system such as Uninterruptible Power System (UPS) is required for VoIP so that they can continue to operate during a power failure.

Security: VoIP systems are more vulnerable to security attacks compared to traditional phone lines for a number of reasons. The use of softphones makes it easier to install harmful software. VoIP servers and gateways can be physically accessed and an attacker may perform a range of tasks including sniffer software insertion, identification of communicating parties and call interception. The use of a wireless network also introduces further security challenges.

Emergency calls: In traditional telephones, each connection is associated with a physical location, which makes it easy to track the caller's location in the case of an emergency. In contrast, a VoIP connection is not linked with a specific location, making it very complicated for VoIP users to be tracked.

3.3 VoIP Codecs

There are many codecs mainly formalized by the International Telecommunication Union (ITU), which vary in mathematical scheme and bandwidth requirements. In this work, we consider four famous voice codecs: PCMA_8kHz, PCMU_8kHz, GSM_8kHz and G.722_16kHz. Following is a brief description of each.

ITU-T G.711: G.711 is a Pulse Code Modulation (PCM) codec that was introduced by ITU in 1972 for use in digital telephony. With respect to coding algorithm, it has two variants: A-law (PCMA) and μ-law (PCMU). G.711 is used in PSTN networks and Integrated Services Digital Network (ISDN) lines. It is simple to implement but utilizes high bandwidth [9][12].

ITU-T G.722: G.722 is a license-free codec that was approved by the ITU in 1988. It is widely used in VoIP and in radio broadcast and provides improved voice quality due to its sampling spectral features while consuming the same bandwidth compared to G.711. [13].

ETSI GSM: GSM is a Regular Pulse Excitation–Long Term Prediction (RPE-LTP) codec that was standardized by the European Telecommunications Standards Institute (ETSI) in 1992. It was the first codec to be used by the GSM (Global System for Mobile Communications) network in addition to its wide use in VoIP software [14][15].

4. VoIP System Design and QoS Parameters

4.1 VoIP System Design

In order to carry out the evaluation, a telephone exchange that allows users to make VoIP calls over the Internet is set up. This section describes the system design in addition to network scenarios under which VoIP calls are made. Additionally, it explains the QoS parameters that are used for the evaluation.

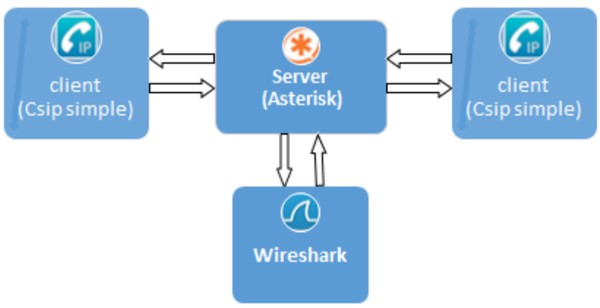

Fig. 1 VoIP System Components

The system consists of a number of connected components as shown in Fig.1. Asterisk software is used to convert a personal computer into an advanced telephone exchange with multiple functions. More than one client can be registered in the server. The client (soft phone application) registers the account name, username, Server/ domain (IP), and the password. Then the client selects the appropriate codec. The application searches for the domain (IP), if it is found, it begins communication by sending the request to the server. When a client makes a call to another client, the server receives the request which contains the username and password and tests their values with the stored ones. If they match, it establishes the connection, alternatively, it waits for a new request. Wireshark works with Asterisk to collect data regarding the QoS of VoIP calls through packet and network analysis measured throughout the call duration. The collected data is used to evaluate the QoS of VoIP calls for various scenarios.

VoIP call quality is evaluated under two network modes:

Wi-Fi Network: In this scenario clients are connected to the server using the IEEE802.11g standard operating in the 2.4GHz frequency band and a channel bandwidth of 20 MHz. VoIP calls are made using mobiles with enabled Wi-Fi interfaces. Four VoIP calls with considered codecs are made under this scenario and the QoS is assessed for each call.

Cellular Mobile Network: A public IP is used in this scenario to connect clients to the server, each client using a mobile equipped with a mobile data interface. There are three options for mobile data: 2G network which provides the least data rates, 3G and the high-speed high-capacity 4G network. The four considered codecs are selected to make VoIP calls under each network option and the QoS is measured for each call.

Based on QoS evaluation, it is possible to identify the ideal codec for each network scenario. The selected codec should provide the best possible quality for VoIP calls with least QoS degradation.

4.2 Quality-of-Service Parameters

The QoS of VoIP calls is measured using a number of parameters. Following is a brief description of delay, jitter and packet loss, since these parameters are considered in this work to evaluate the influence of codec on call quality [10][16][17].

Latency: Latency in VoIP is the time it takes for the caller's voice to reach the receiver. It is mainly caused by slow network connections. High latency degrades the quality of VoIP calls by slowing the conversation, intensifying echo and causing callers to interrupt each other. The codec also influences the resultant latency as different codecs use different packet interval times. Acceptable latency values are up to 150ms.

Jitter: Jitter is defined as the delay between two arriving packets and is caused by network conditions. High jitter in VoIP calls causes voice to arrive choppy, jumbled or disrupted. Acceptable jitter values are up to 50ms.

Packet loss: Packet loss is the failure of packets reaching their destination and is caused by network congestion. It is considered a critical parameter for delay-sensitive services such as VoIP. Acceptable packet loss values are up to 5%.

5. Experimental Setup

The system consists of the sever side and two clients that communicate with each other through the server. Following are the details of the network components and setup.

Server settings: The server side uses a computer with Core i5 7th gen, 8GB RAM and HDD 1T. Asterisk is installed in the server having two configuration files which are (sib.conf) and (extensions.conf).

Mobile phone settings: The first phone was a Huawei MT7-TL10 mobile and the second was a Xiaomi Mi 8 phone. For Wi-Fi, Csipsimple softphone was employed in both phones to provide voice calls.

Wi-Fi network setup: The two clients were connected to the server using single-hop Wi-Fi. Both clients resided with the sever in the same network.

Cellular network setup: When running the system using mobile data network, the first client was in Bahri city and the second client was in Khartoum city. The server was in a separate network in Khartoum and the three types of cellular mobile networks (2G, 3G and 4G) were tested.

Call settings: All experiments took place between 1:00 and 3:00 pm, and an average of three calls were attempted per test. The duration of each call was 1 minute. All users were stationary throughout the duration of calls.

When a VoIP call is established, the data passes from the sender to the server, which passes the data to the receiver's address. During the call, Wireshark works with Asterisk to analyze packets and measure the call QoS. The codecs considered in this project are PCMA (8kHz), PCMU (8kHz), GSM (8kHz) and G.722 (16kHz).

Four network environments were evaluated, and under each considered network four VoIP calls were made, one call for each codec. The QoS was measured for each call, and results for Jitter, Round Trip Time (RTT) and Packet Loss were obtained.

6. VoIP QoS Evaluation

This section demonstrates by results the evaluation of the QoS of VoIP calls. First, codecs are evaluated under each network to demonstrate the variation of maximum, average and minimum values of jitter and RTT. Then the average obtained jitter and RTT are of each codec are contrasted to evaluate their performance. In addition, packet loss are also compared.

6.1 QoS Evaluation under Wi-Fi

Two softphones were utilized to make four VoIP calls connected via Wi-Fi. Each call used a specified codec and Jitter and RTT values were obtained for each codec.

6.1.1. Jitter

Fig.2 shows the resulting minimum, average and maximum jitter for considered codecs. Based on results, similar jitter values were obtained under all codecs. Jitter using Wi-Fi is considerably low, with average values of 31, 27, 3.6 msec for maximum, average and minimum jitter, respectively.

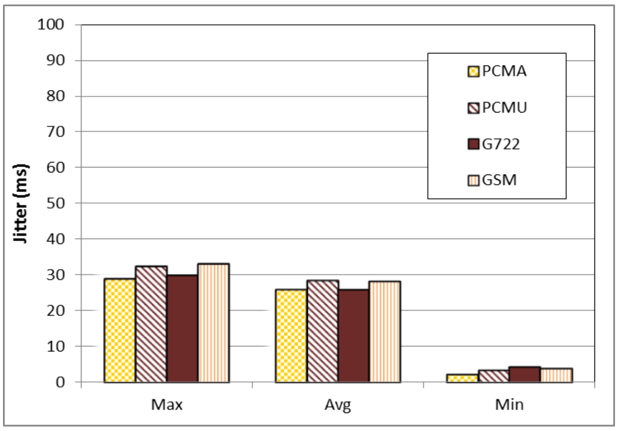

Fig. 2 Wi-Fi Maximum, Average and Minimum Jitter

6.1.2 Round Trip Time

RTT was obtained for each codec and the results for minimum, average and maximum RTT are demonstrated in Fig.3. As can be seen in the figure, G.722 recorded the lowest value of RTT which is approximately 4msec while the highest value is under GSM codec, though it is yet considered low (under 20msec).

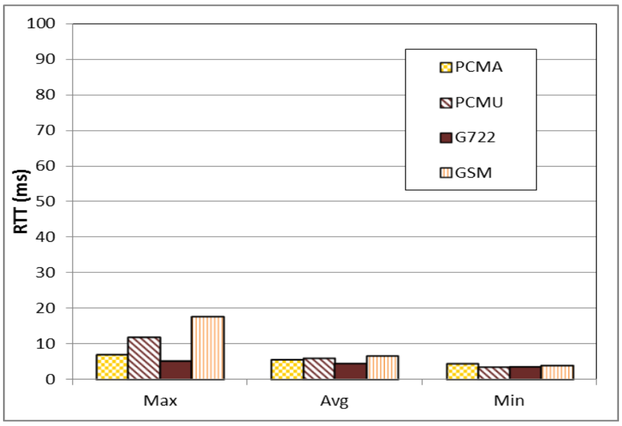

Fig. 3 Wi-Fi Maximum, Average and Minimum RTT

6.2 QoS Evaluation under 2G

Jitter and RTT measurements were obtained for each VoIP call under Second Generation (2G) network where each call used a different codec.

6.2.1 Jitter

Fig.4 shows the resulting minimum, average and maximum jitter for considered codecs. Based on the obtained results, the lowest value of average jitter was achieved when PCMA was used for coding audio, with similarly close values under G.722 and PCMU (ranging from 14 to 30msec). GSM had the worst jitter, reaching up to a maximum of 380msec.

6.2.2 Round Trip Time

In contrast to jitter, GSM outperformed other codecs when evaluating RTT. PCMA and PCMU both suffered high RTT. These results are shown in Fig.5.

6.3 QoS Evaluation under 3G

The Third Generation (3G) network is considered, and QoS parameters were measured for each codec.

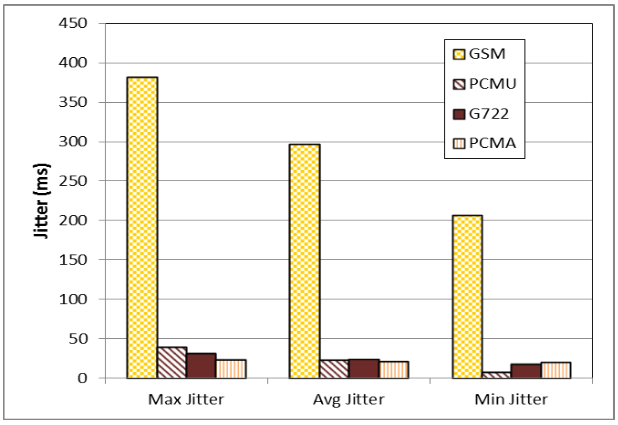

Fig. 4 2G Maximum, Average and Minimum Jitter

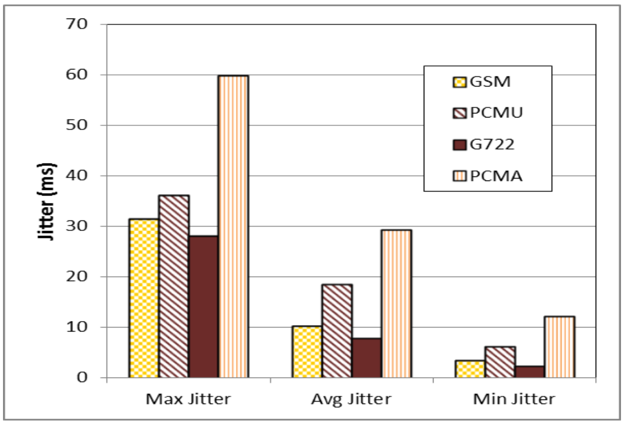

Fig. 6 3G Maximum, Average and Minimum Jitter

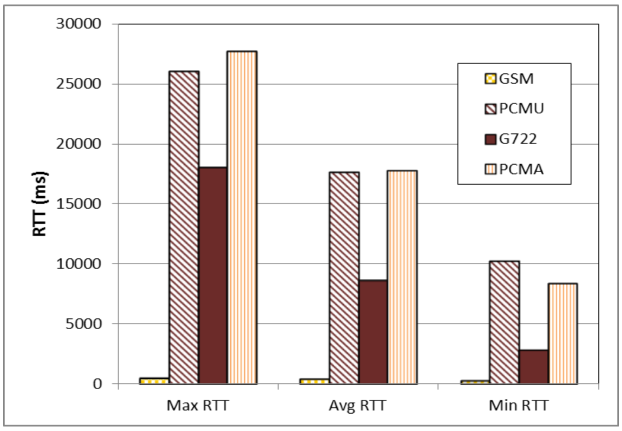

Fig. 5 2G Maximum, Average and Minimum RTT

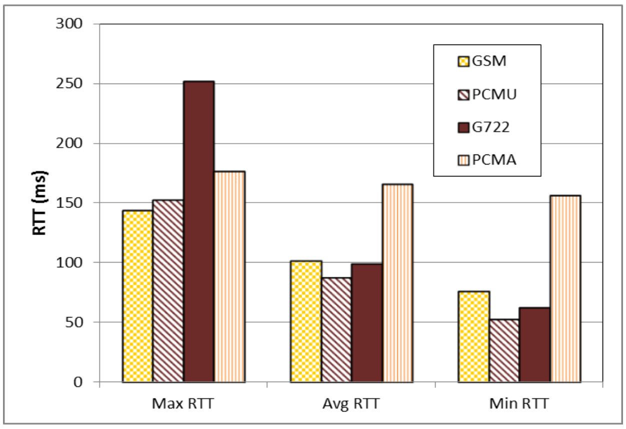

Fig. 7 3G Maximum, Average and Minimum RTT

6.3.1 Jitter

Fig.6 shows the resulting minimum, average and maximum jitter for considered codecs. It is apparent that G.722 achieved the best jitter compared to other codecs. The highest value of jitter was achieved when using PCMA.

6.3.2 Round Trip Time

Fig.7 shows the minimum, average and maximum RTT under each codec. As can be seen in Fig.9, GSM and PCMU maintained lower values for minimum, average and maximum RTT. However, maximum RTT under G.722 is high. The highest value for average RTT is obtained under PCMA.

6.4 QoS Evaluation under 4G

To evaluate the QoS under Fourth Generation (4G) network, four VoIP calls are made using the four considered codecs and the resultant Jitter and RTT were obtained for each codec.

6.4.1 Jitter

Fig.8 shows the resulting minimum, average and maximum jitter for considered codecs. It clearly shows that no jitter was obtained under PCMA in addition to a close resemblance in average and minimum jitter under other codecs. In addition, PCMU suffered the highest maximum jitter.

6.4.2 Round Trip Time

As can be observed in Fig.9, PCMA resulted in RTT=0 for minimum, average and maximum RTT. Similar to jitter, other codecs resulted in comparable values of average and minimum RTT. The highest value of maximum RTT is obtained under GSM.

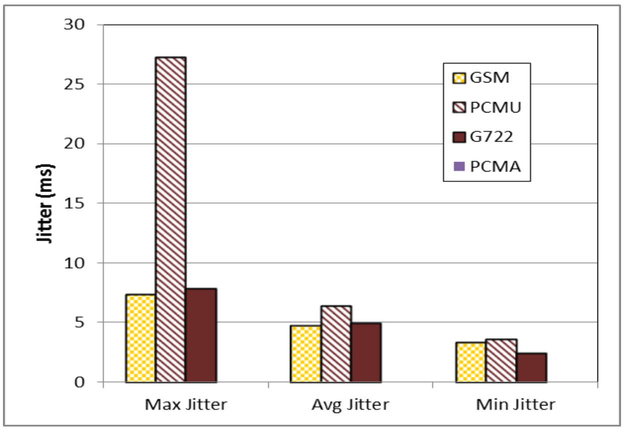

Fig. 8 4G Maximum, Average and Minimum Jitter

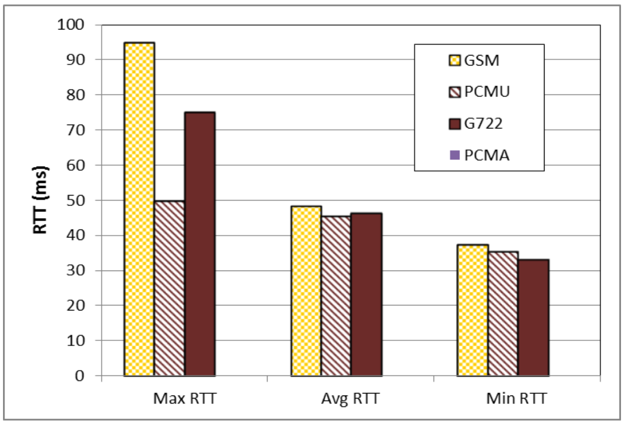

Fig. 9 4G Maximum, Average and Minimum RTT

6.5 Comparison between Different Network Scenarios

To contrast attained results among different network scenarios, we plot the average jitter, RTT and packet loss obtained from the experiment and compare the QoS parameter values under each network.

6.5.1 Jitter

The average jitter is obtained under Wi-Fi, 2G, 3G and 4G and the results are shown in Fig.10. Based on the results we got, the average jitter is high under 2G. at the same time, the jitter was better under 3G compared to Wi-Fi.

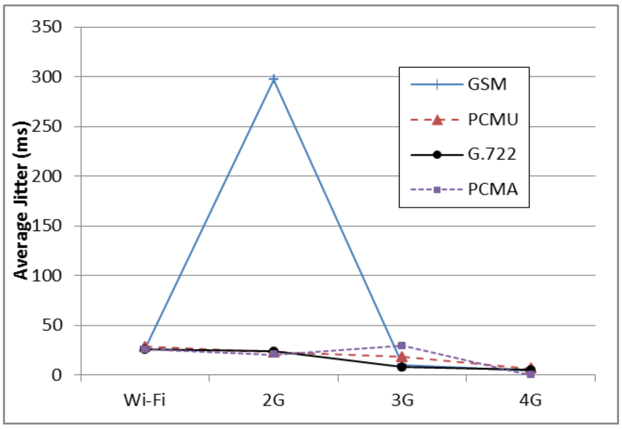

Fig. 10 Average Jitter under Wi-Fi, 2G, 3G and 4G

6.5.2 Round Trip Time

Average RTT is compared under Wi-Fi, 2G, 3G and 4G and the results are shown in Fig.11. Under all codecs, RTT follows a trend where the highest value is recorded under 2G. In contrast, RTT values under Wi-Fi, 3G and 4G were comparably low, with the lowest being under Wi-Fi.

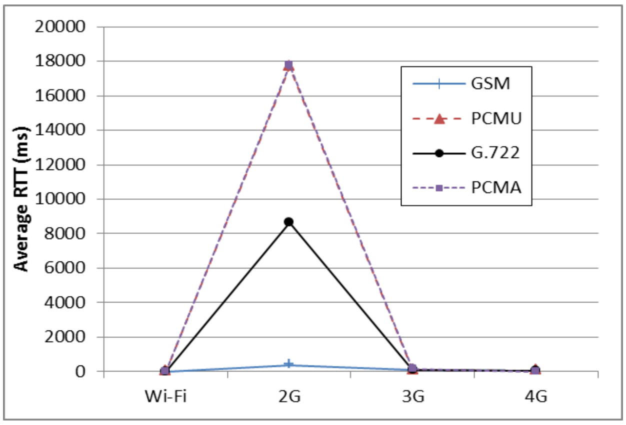

Fig. 11 Average RTT under Wi-Fi, 2G, 3G and 4G

6.5.3 Packet Loss

The resultant values of packet loss under Wi-Fi, 2G, 3G and 4G are shown in Fig.12. According to the results, the highest value of packet loss is recorded in 3G with much less packet loss under 2G and 4G. No packet loss was obtained under Wi-Fi.

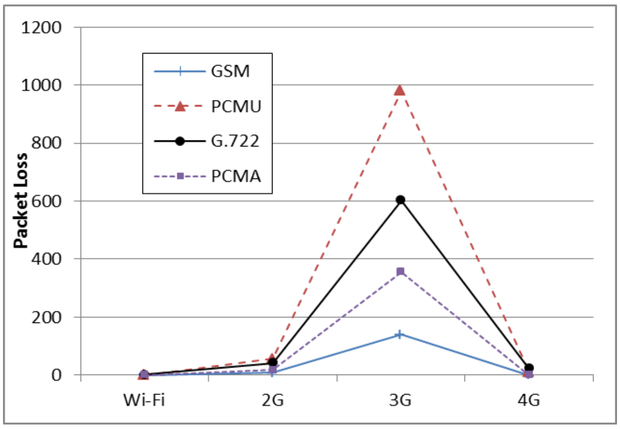

Fig. 12 Packet Loss under Wi-Fi, 2G, 3G and 4G

7. Discussion

According to obtained results, a single codec that provides the best performance under all networks does not exist. The resemblance in the characteristics of codecs resulted in comparable results for some experiments. This is more apparent when using codecs from the G.711 family (PCMA and PCMU). Under a network with high bandwidth such as Wi-Fi, G.722 in general outperformed other codecs in terms of jitter and RTT, even though all codecs maintained acceptable values.

VoIP experiments for cellular networks were conducted under real-world network conditions. All codecs followed the expected trend where the best performance was recoded under 4G and the worst was when evaluating 2G. Overall, GSM was the codec with the best performance.

It was observed that 3G network suffered high packet loss compared to other networks. In 3G, packets that actually arrived at the destination arrived with low RTT, though a considerable number of packets were lost. This is explained by the fact that no retransmission was used. Therefore, dropped packets did not increase delay due to packet retransmission, but network congestion contributed to the high packet loss. Observing attained results, GSM outperformed other codecs by maintaining the least packet loss under all networks, while PCMU suffered from the highest packet loss.

It is worth mentioning that the evaluation carried out in this work measures QoS parameters at the server. RTT, for example, measured at the server is not the same as end-to-end delay experienced by end hosts. While RTT indicates delay from the sender to the receiver and back, end-to-end delay measures the latency from source to destination. Therefore, end-to-end delay is calculated from the total RTT measured for both hosts. In the case when uplink and downlink conditions are similar, it would be safe to calculate end-to-end delay as half of total RTT. But that is not exact for wireless communication in general, as uplink and downlink channels experience different delays.

In this evaluation, the best way to estimate end-to-end delay from caller *c* to recipient *r* D_{cr} using only the network design implemented in this work would be: $D_{cr} = (RTT_c + RTT_r)/2$, where RTT_c is the round trip time of the caller measured at the server and RTT_r is the round trip time of the recipient measured at the server. The accuracy of this equation is higher when the caller and recipient experience comparable variances between uplink and downlink delay. The precision of the equation is also improved by calculating the average end-to-end delay taken over a long time interval.

8. Conclusions

This paper has evaluated the QoS of VoIP calls under Wi-Fi, 2G, 3G and 4G networks. It has compared jitter, round trip time and packet loss under four codecs: G.722, GSM, PCMA and PCMU. According to obtained results, under Wi-Fi, G.722 has outperformed other codecs in QoS parameter values. When 2G network was considered, GSM has the best QoS values for RTT and packet loss, but with increased jitter. GSM also had acceptable QoS values under 3G, which generally experienced high packet loss. In 4G, PCMA had the best call quality compared to other codecs. Our future work involves evaluating the QoS of video calls and examining the influence of codec on video call quality. In addition, it would be interesting to record quality indicators for calls while on the move and investigate quality behavior under different networks for the same mobility patterns.

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