OPTIMIZING DSCP MARKING TO ENSURE VOIP'S QOS OVER HFC NETWORK

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ABSTRACT

Three major factors that can affect Voice over Internet Protocol (VoIP) phone services' quality, these include packet delay, packet loss, and jitter. The focus of this study is specific to the VoIP phone services offered to customers by cable companies that utilize broadband hybrid fiber coaxial (HFC) networks. HFC networks typically carry three types of traffic that include voice, data, and video. Unlike data and video, some delays or packet loss can result in a noticeable degraded impact on a VoIP's phone conversation. We will examine various differentiated services code point (DSCP) marking, then analyze and assess their impact on VoIP's quality of service (QoS). This study mimics the production environment. It examines the relationship between specific DSCP marking's configuration. This research avoids automated test calls and rather focuses on human made call testing. This study relies on users' experience and the captured data to support this research's findings.

KEYWORDS

QoS , VoIP, DSCP Marking , jitter, HFC Network, MOS.

1. INTRODUCTION

Voice over Internet Protocol (VoIP) is a newly adopted technology; its deployment continues to accelerate [1, 2]. The acronym VoIP refers to voice communication on the Internet using the Internet protocol (IP). Voice signals change into voice packets and get transmitted through the same network that providers use for data communications [3]. For voice traffic to travel in a Hybrid Fiber Coaxial (HFC) broadband network, the analog voice signal gets converted into digital signals. Additional hardware such as routers, switches, and servers are needed to transport voice traffic to its destination. IP broadband networks allow cable providers to combine voice traffic, data traffic, and video traffic to travel over a single communication link. Packet loss, jitter, and latency continue to be the major elements that impact the quality of VoIP [1, 4]. Packet prioritization can play an important role in VoIP's quality; where it differentiates between VoIP traffic and other traffic types that share the same networks links. The Differentiated Services Code Point (DSCP) marking is based on assigning different values for a various traffic types including VoIP [5]. Packet loss between 1% and 5%, and having less than 300 milliseconds (ms) of end-to-end delay are key elements of VoIP calls [6].

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As a result, finding methods that can address VoIP’s quality is the primary focus of many scholars and researchers in the telecommunications field. DSCP marking may allow HFC broadband network carriers to prioritize their traffic. If the carriers have the ability to mark the different types of traffic across the various devices in their networks, VoIP’s quality might improve. DSCP marking can be of a great benefit to VoIP. This research focuses primarily on providing and recommending the best DSCP marking that can be used in HFC network to ensure a better VoIP’s voice quality.

2. PURPOSE, PROBLEM STATEMENT AND HYPOTHESIS

This study focuses on our research purpose, the problem statement and our hypothesis.

2.1 Research Purpose Statement

This research primarily focuses on studying the quality of VoIP service delivery in an HFC broadband network when mixed with data and video traffic. The primary purpose of this research is to improve VoIP’s QoS. VoIP, video, and data share the same transport links in broadband HFC networks. VoIP traffic cannot tolerate packet loss or delay. Call instances where VoIP phone calls encounter considerable packet delays, both callers begin to talk over each other [6, 7].

2.2 Problem Statement

HFC networks carry multiple types of traffic that include voice, data, and video. Some of the elements that affect VoIP’s quality are packet loss, jitter, latency, and delay. Treating all three types of traffic equally in each element of the broadband HFC network, especially with over utilized capacity links, can degrade VoIP’s QoS.

2.3 Hypothesis Statement

Configuring VoIP traffic with a DSCP marking value of EF across all the equipment in a broadband HFC network would improve VoIP's QoS.

2.4 Research Question

This research focuses on addressing one primary research question: considering the routers’ links carry Internet, video, and VoIP traffic where traffic links are under-utilized, which of the DSCP marking has the best impact on VoIP’s QoS?

3. RELATED WORKS

The need for prioritizing voice traffic on IP networks became imminent. The implication of recent research studies on VoIP and QoS increased to improve VoIP services and make them affordable, cheaper, and a reliable for users’ daily needs [8, 9, 10]. The quality of VoIP traffic can be dependent on five factors: mean opinion score (MOS), jitter, latency, network load, and network throughput [11]. There is an imminent the need for QoS in VoIP calls while at the same time have the ability to support as many calls as possible [12].

VoIP's degraded call quality, when compared with circuit-switched phone calls, became an issue that was not easy to tolerate or ignore. Since then, many research studies started to focus on improving the quality of VoIP's services. Due to VoIP's nature where it travels across various networks and equipment, many academics & practitioners continue to study the various aspects of VoIP to improve its service. VoIP's poor QoS through wireless fidelity (WiFi), universal mobile telecommunications system (UMTS) & WiFi UMTS networks led researchers to examine the MOS and the packets' end-to-end delay to improve VoIP's quality [13]. Researchers continue to focus on coming up with new solutions that improve VoIP's quality.

Some studies turned towards utilizing Machine Learning Quality of Experience (MLQoE) to improve VoIP traffic's quality of Experience. MLQoE selects the Machine Learning (ML) algorithm that shows the best performance and its elements in an automated process, considering the use of the dataset as input [14]. There is a recommendation to prioritize Voice traffic to handle packet losses in VoIP services. This can be achieved through providing a mechanism to drop first the least important content, in order to keep the best VoIP's quality signal for user perception using an Arduino platform [15]. Researchers investigated the performances of routing protocols Optimized Link State Routing (OLSR) and Training and Doctrine Command (TRADOC) Operations Research Agency (TORA) in a Mobile Ad hoc Networks (MANETs). The wireless mobile nodes group form a temporary network; this avoids the utilization of any centralized access-point management of the mobile networks. The research used the network simulator Optimized Network Engineering Tool (OPNET) 14.5 to analyse and evaluate some QoS metrics like end-to-end delay, Jitter, throughput and MOS. The OPNET simulation results showed that the OLSR protocol is a good candidate for VoIP application [16].

To ensure that mobile networks deliver improved VoIP QoS, each network element should follow the same single protocol. The investigation focused on WiFi and Cellular networks. Network Neutrality may become an obstacle for achieving the intended results of improving VoIP's conversations [17]. To address QoS in wireless networks regardless of whether they are heterogeneous or homogenous, the recommendation was to use a software defined network (SDN) architecture. The use of Smart Adaptive QoS for Heterogeneous and Homogeneous Networks (SAQ-2HN) architecture resulted in higher delay without the use of QoS and better results with the use of SAQ-2HN [18].

VoIP's QoS transition from an IP Version 6 (IPv6) network to an IP Version 4 (IPv4) network maintained its quality through the IP network tunnelling process. No substantial delay was noticed where it could have impacted the quality of VoIP phone calls [19].

4. RESEARCH DESIGN

This research uses an experimental testing design approach where its quantitative study investigates the impact of changing DSCP marking on VoIP phone calls. The experiments in this study rely on making manual VoIP phone calls. In each of the manual testing scenarios; one tester makes a call while the other tester answers. Primary elements of the test environment include a router and an MGC. The links between the router and the HFC broadband network carry all three types of traffic that include data, video, and VoIP. VoIP traffic marking changes take place at the MGC level. The routers pass VoIP traffic and perform prioritization dependent on the MGCF's DSCP marking. Iris tool captures the end-to-end call signalling. Iris traces will show the MOS, DSCP marking, and any packet loss or packet drops.

4.1 Phone Calls' Testing: VoIP to signaling system 7 (SS7) and SS7 to VoIP

The first testing scenario focuses on making calls from a VoIP network and terminate on SS7 network. The second testing scenario focuses on making calls from SS7 network and terminate in a VoIP network. Each of the testing scenarios is composed of five independent DSCP marking configuration changes that include class selector 0 (CS0), CS1, CS3, CS4, and expedited forwarding (EF). The process of making 10 calls takes place after each of the DSCP configuration changes.

4.2 Answering the Research Question

The experiments' data for each of the test calls are collected and populated in identical tables. The data gets analysed and used to answer the research question including the research hypotheses. The analysis is based on the telecom industry and ITU's standards. The dependent variables of this research include the average MOS captured by the Iris tool, calling party's estimated MOS, called party estimated MOS, average jitter, average interpacket arrival time, latency, and packets lost. This research compares the experiment dependent variables' values with the VoIP industry's standards. MOS values between 4 and 5 refer to clear phone conversation, and anything below 4 as poor quality [20]. The acceptable jitter's value is less than 50 ms [21]. VoIP interpacket arrival time should be between 20 or 30 ms [22]. 1.5% or less are the acceptable data loss value [21]. The maximum tolerable latency value falls between 80 ms and 120 ms [23].

5. Experiments Results And Analysis

Each experiment will be composed of five tests and each of the tests is based on 10 phone calls. Each of the tests will focus on one DSCP configurations that include CS0, CS1, CS3, CS4, and EF.

5.1 VoIP to SS7 Experiments

This section focuses on VoIP to SS7 test scenrios and their related data.

5.1.1 VoIP to SS7 test calls with a DSCP marking value of CS0

10 random calls are made; testing results are captured in Table 1. The lowest MOS value of the 10 calls is 4.380 and the highest value is 4.400. The lowest average interpacket arrival time is 20.030 ms, while the highest value is 20.400 ms. The lowest jitter's value is 0.01 ms while the highest value is 5.30 ms. The highest packet loss is 0.27%, and the highest latency value is 54 ms. Examining the values of each of the dependent variables indicates that each of the values were within specifications which lead to good and clear phone conversations between both parties.

Table 1. VoIP to SS7 test calls with a DSCP marking value of CS0

Call No.	MOS Avg CQ	MOS Calling Party	MOS Called Party	Avg Inter-packet Time (ms)	Avg Jitter (ms)	Total Packets	Packets Lost	Packets Loss (%)	Lat-ency
1	4.380	4.400	4.400	20.030	1.00	3344	6	0.18	0
2	4.400	4.400	4.400	20.000	0.02	3438	0	0.00	35
3	4.390	4.400	4.400	20.000	0.05	12694	0	0.00	54
4	4.400	4.400	4.400	20.080	5.30	3534	0	0.00	32
5	4.390	4.400	4.400	20.000	0.02	13799	0	0.00	48
6	4.400	4.400	4.400	20.050	3.84	1987	0	0.00	33
7	4.400	4.400	4.400	20.400	2.97	2947	8	0.27	0
8	4.400	4.400	4.400	20.000	0.01	1582	0	0.00	32
9	4.400	4.400	4.400	20.000	0.02	23726	4	0.02	0
10	4.400	4.400	4.400	20.000	0.02	40522	6	0.01	0

5.1.2 VoIP to SS7 test calls with a DSCP marking value of CS1

10 random calls are made; testing results are captured in Table 2. All 10 calls have the same MOS value of 4.400. The lowest average interpacket arrival time is 20.000 ms while the highest is 20.070 ms. The lowest jitter's value is 0.02 ms while the highest is 7.83 ms. Packet loss is 0%, and the highest latency value is 20 ms. Examining the values of each of the dependent variables indicate that each of the values were within specifications that lead to good and clear phone conversations between both parties.

Table 2. VoIP to SS7 test calls with a DSCP marking value of CS1

Call No.	MOS Avg CQ	MOS Calling Party	MOS Called Party	Avg Inter-packet Time (ms)	Avg Jitter (ms)	Total Packets	Packets Lost	Packets Loss (%)	Lat-ency
1	4.40	4.400	4.400	20.000	0.02	1820	0	0.00	14
2	4.40	4.400	4.400	20.010	7.09	2195	0	0.00	9
3	4.40	4.400	4.400	20.000	4.04	4799	0	0.00	20
4	4.40	4.400	4.400	20.000	7.83	4438	0	0.00	8
5	4.40	4.400	4.400	20.010	7.49	1445	0	0.00	19
6	4.40	4.400	4.400	20.000	0.02	2944	0	0.00	8
7	4.40	4.400	4.400	20.000	0.02	1747	0	0.00	12
8	4.40	4.400	4.400	20.020	4.03	1755	0	0.00	12
9	4.40	4.400	4.400	20.070	7.13	1999	0	0.00	12
10	4.40	4.400	4.400	20.000	0.02	1350	0	0.00	11

5.1.3 VoIP to SS7 test calls with a DSCP marking value of CS3

10 random calls are made; testing results are captured in Table 3. All 10 calls have the same MOS value of 4.400. The lowest average interpacket arrival time is 20.000 ms while the highest is 20.070 ms. The lowest jitter's value is 0.02 ms while the highest is 7.83 ms. Packet loss is 0%, and the highest latency value is 20 ms. Examining the values of each of the dependent variables indicate that each of the values were within specifications that lead to good and clear phone conversations between both parties.

Table 3. VoIP to SS7 test calls with a DSCP marking value of CS3

Call No.	MOS Avg CQ	MOS Calling Party	MOS Called Party	Avg Inter-packet Time (ms)	Avg Jitter (ms)	Total Packets	Packets Lost	Packets Loss (%)	Lat-ency
1	4.40	4.400	4.400	20.010	2.72	2889	0	0.00	8
2	4.40	4.400	4.400	20.000	0.04	2897	0	0.00	0
3	4.40	4.400	4.400	20.000	1.53	797	0	0.00	0
4	4.40	4.400	4.400	20.000	0.11	3588	0	0.00	0
5	4.40	4.400	4.400	20.000	4.01	24077	0	0.00	0
6	4.40	4.400	4.400	20.000	0.03	10704	0	0.00	0
7	4.40	4.400	4.400	20.000	0.91	1681	0	0.00	0
8	4.40	4.400	4.400	20.000	0.19	6499	0	0.00	0
9	4.40	4.400	4.400	20.000	1.54	457	0	0.00	0
10	4.40	4.400	4.400	20.000	0.01	12619	0	0.00	0

5.1.4 VoIP to SS7 test calls with a DSCP marking value of CS4

10 random calls are made; testing results are captured in Table 4. All 10 calls have the same MOS value of 4.400. The lowest average interpacket arrival time is 20.000 ms while the highest is 20.010 ms. The lowest jitter's value is 0.01 ms while the highest is 0.02 ms. Packet loss is 0%, and there is no indication of any latency. Examining the values of each of the dependent variables indicate that each of the values were within specifications that lead to good and clear phone calls between both parties.

Table 4. VoIP to SS7 test calls with a DSCP marking value of CS4

Call No.	MOS Avg CQ	MOS Calling Party	MOS Called Party	Avg Inter-packet Time (ms)	Avg Jitter (ms)	Total Packets	Packets Lost	Packets Loss (%)	Lat-ency
1	4.40	4.400	4.400	20.000	0.02	11929	0	0.00	0
2	4.40	4.400	4.400	20.000	0.01	1023	0	0.00	0
3	4.40	4.400	4.400	20.000	0.02	9106	0	0.00	0
4	4.40	4.400	4.400	20.000	0.01	8171	0	0.00	0
5	4.40	4.400	4.400	20.010	0.02	11843	0	0.00	0
6	4.40	4.400	4.400	20.000	0.01	5507	0	0.00	0
7	4.40	4.400	4.400	20.000	0.02	90026	0	0.00	0
8	4.40	4.400	4.400	20.000	0.01	5484	0	0.00	0
9	4.40	4.400	4.400	20.000	0.02	2904	0	0.00	0
10	4.40	4.400	4.400	20.000	0.02	5502	0	0.00	0

5.1.5 VoIP to SS7 test calls with a DSCP marking value of EF

10 random calls are made; testing results are captured in Table 5. All 10 calls have the same MOS value of 4.400. The average interpacket arrival time for each of the 10 calls is 20.000 ms. The lowest jitter's value is 0.00 ms while the highest jitter's value is 0.01 ms. Packet loss is 0%, and there is no indication of any latency. Examining the values of each of the dependent variables indicate that each of the values were within specifications that lead to good and clear phone calls between both parties.

Table 5. VoIP to SS7 test calls with a DSCP marking value of EF

Call No.	MOS Avg CQ	MOS Callin g Party	MOS Called Party	Avg Inter-packet Time (ms)	Avg Jitter (ms)	Total Packets	Packets Lost	Packets Loss (%)	Lat-ency
1	4.400	4.400	4.400	20.000	0.01	1286	0	0.00	0
2	4.400	4.400	4.400	20.000	0.00	730	0	0.00	0
3	4.400	4.400	4.400	20.000	0.01	4618	0	0.00	0
4	4.400	4.400	4.400	20.000	0.02	8118	0	0.00	0
5	4.400	4.400	4.400	20.000	0.02	13543	0	0.00	0
6	4.400	4.400	4.400	20.000	0.02	1313	0	0.00	0
7	4.400	4.400	4.400	20.000	0.02	24123	0	0.00	0
8	4.400	4.400	4.400	20.000	0.02	1969	0	0.00	0
9	4.400	4.400	4.400	20.000	0.01	6799	0	0.00	0
10	4.400	4.400	4.400	20.000	0.02	3561	0	0.00	0

5.2 SS7 to VoIP Experiments

This section focuses on SS7 to VoIP test scenrios and their related data.

5.2.1 SS7 to VoIP test calls with a DSCP marking value of CS0

10 random calls are made; testing results are captured in Table 6. The lowest MOS value of the 10 calls shows a value of 4.360 and the highest value is 4.400. The lowest average interpacket arrival time is 20.000 ms while the highest is 20.400 ms. The lowest jitter's value is 0.01 ms while the highest is 5.47 ms. The highest packet loss is 0.49%, and the highest latency value is 54 ms. Examining the values of each of the dependent variables indicate that each of the values were within specifications that lead to good and clear phone conversations between both parties.

Table 6. SS7 to VoIP test calls with a DSCP marking value of CS0

Call No.	MOS Avg CQ	MOS Calling Party	MOS Called Party	Avg Inter-packet Time (ms)	Avg Jitter (ms)	Total Packets	Packets Lost	Packets Loss (%)	Lat-ency
1	4.39	4.400	4.400	20.000	0.03	12698	0	0.00	54
2	4.40	4.400	4.400	20.000	0.02	4791	0	0.00	20
3	4.38	4.400	4.400	20.400	0.26	1760	4	0.23	0
4	4.40	4.400	4.400	20.040	3.68	3442	0	0.00	35
5	4.36	4.400	4.400	20.100	0.01	1235	6	0.49	8
6	4.40	4.400	4.400	20.010	5.47	1591	0	0.00	32
7	4.39	4.400	4.400	20.000	0.06	13800	0	0.00	48
8	4.40	4.400	4.400	20.000	0.02	3540	0	0.00	32
9	4.40	4.400	4.400	20.000	0.02	3211	0	0.00	0
10	4.40	4.400	4.400	20.000	0.20	26927	4	0.01	0

5.2.2 SS7 to VoIP test calls with a DSCP marking value of CS1

10 random calls are made; testing results are captured in Table 7. All 10 calls had the same MOS value of 4.400. The lowest average interpacket arrival time is 20.000 ms while the highest is 20.120 ms. The lowest jitter's value is 0.01 ms while the highest is 6.69 ms. Packet loss is 0%, and the highest latency value is 19 ms. Examining the values of each of the dependent variables indicate that each of the values were within specifications that lead to good and clear phone conversations between both parties.

Table 7. SS7 to VoIP Test Calls with a DSCP Marking Value of CS1

Call No.	MOS Avg CQ	MOS Calling Party	MOS Called Party	Avg Inter-packet Time (ms)	Avg Jitter (ms)	Total Packets	Packets Lost	Packets Loss (%)	Lat-ency
1	4.40	4.400	4.400	20.000	0.02	1351	0	0.00	12
2	4.40	4.400	4.400	20.010	6.69	2582	0	0.00	8
3	4.40	4.400	4.400	20.020	2.71	1830	0	0.00	14
4	4.40	4.400	4.400	20.000	0.01	2884	0	0.00	8
5	4.40	4.400	4.400	20.000	0.02	1436	0	0.00	19
6	4.40	4.400	4.400	20.120	4.96	1232	0	0.00	8
7	4.40	4.400	4.400	20.000	0.02	1998	0	0.00	12
8	4.40	4.400	4.400	20.080	5.39	1352	0	0.00	11
9	4.40	4.400	4.400	20.010	3.14	1354	0	0.00	12
10	4.40	4.400	4.400	20.000	0.02	2188	0	0.00	9

5.2.3 SS7 to VoIP test calls with a DSCP marking value of CS3

10 random calls are made; testing results are captured in Table 8. All 10 calls have the same MOS value of 4.400. The lowest average interpacket arrival time is 19.990 ms while the highest is 20.000 ms. The lowest jitter's value is 0.01 ms while the highest is 1.10 ms. Packet loss is 0%,

and the highest latency value is 8 ms. Examining the values of each of the dependent variables indicate that each of the values were within specifications that lead to good and clear phone calls between both parties.

Table 8. SS7 to VoIP Test Calls with a DSCP Marking Value of CS3

Call No.	MOS Avg CQ	MOS Calling Party	MOS Called Party	Avg Inter-packet Time (ms)	Avg Jitter (ms)	Total Packets	Packets Lost	Packets Loss (%)	Lat-ency
1	4.40	4.400	4.400	20.000	0.01	2576	0	0.00	8
2	4.40	4.400	4.400	20.000	0.07	1104	0	0.00	0
3	4.40	4.400	4.400	20.000	0.24	8526	0	0.00	0
4	4.40	4.400	4.400	20.000	0.04	6304	0	0.00	0
5	4.40	4.400	4.400	19.990	0.04	26959	0	0.00	0
6	4.40	4.400	4.400	20.000	0.06	1035	0	0.00	0
7	4.40	4.400	4.400	20.000	0.18	3584	0	0.00	0
8	4.40	4.400	4.400	20.000	0.11	23136	0	0.00	0
9	4.40	4.400	4.400	20.000	0.18	45362	0	0.00	0
10	4.40	4.400	4.400	20.000	1.10	52977	0	0.00	0

5.2.4 SS7 to VoIP test calls with a DSCP marking value of CS4

10 random calls are made; testing results are captured in Table 9. All 10 calls have the same MOS value of 4.400. The average interpacket arrival time for each of the test calls is 20.000 ms. The lowest jitter's value is 0.01 ms while the highest is 0.03 ms. Packet loss is 0%, and there is no indication of any latency. Examining the values of each of the dependent variables indicate that each of the values were within specifications that lead to good and clear phone calls between both parties.

Table 9. SS7 to VoIP test calls with a DSCP marking value of CS4

Call No.	MOS Avg CQ	MOS Calling Party	MOS Called Party	Avg Inter-packet Time (ms)	Avg Jitter (ms)	Total Packets	Packets Lost	Packets Loss (%)	Lat-ency
1	4.40	4.400	4.400	20.000	0.02	17400	0	0.00	0
2	4.40	4.400	4.400	20.000	0.01	3346	0	0.00	0
3	4.40	4.400	4.400	20.000	0.02	23142	0	0.00	0
4	4.40	4.400	4.400	20.000	0.01	6308	0	0.00	0
5	4.40	4.400	4.400	20.000	0.04	1659	0	0.00	0
6	4.40	4.400	4.400	20.000	0.02	26464	0	0.00	0
7	4.40	4.400	4.400	20.000	0.02	955	0	0.00	0
8	4.40	4.400	4.400	20.000	0.02	1697	0	0.00	0
9	4.40	4.400	4.400	20.000	0.02	45362	0	0.00	0
10	4.40	4.400	4.400	20.000	0.03	1763	0	0.00	0

5.2.5 SS7 to VoIP test calls with a DSCP marking value of EF

10 random calls are made; testing results are captured in Table 10. All 10 calls have the same MOS value of 4.400. The average interpacket arrival time for each of the test calls is 20.000 ms. The lowest jitter's value is 0.01 ms while the highest jitter's value is 0.02 ms. Packet loss is 0%, and there is no indication of any latency. Examining the values of each of the dependent variables indicate that each of the values were within specifications that lead to good and clear phone calls between both parties.

Table 10. SS7 to VoIP Test Calls with a DSCP Marking Value of EF

Call No.	MOS Avg CQ	MOS Calling Party	MOS Called Party	Avg Inter-packet Time (ms)	Avg Jitter (ms)	Total Packets	Packets Lost	Packets Loss (%)	Lat-ency
1	4.40	4.400	4.400	20.000	0.02	3551	0	0.00	0
2	4.40	4.400	4.400	20.000	0.02	90024	0	0.00	0
3	4.40	4.400	4.400	20.000	0.02	1291	0	0.00	0
4	4.40	4.400	4.400	20.000	0.02	8526	0	0.00	0
5	4.40	4.400	4.400	20.000	0.01	1754	0	0.00	0
6	4.40	4.400	4.400	20.000	0.01	1041	0	0.00	0
7	4.40	4.400	4.400	20.000	0.02	10694	0	0.00	0
8	4.40	4.400	4.400	20.000	0.02	3364	0	0.00	0
9	4.40	4.400	4.400	20.000	0.02	17395	0	0.00	0
10	4.40	4.400	4.400	20.000	0.02	8107	0	0.00	0

5.3 Calls' Data Analysis

We rely on our analysis of the data to answer the research question and the research hypotheses. Research Question: Which of the DSCP marking have the best impact on VoIP QoS? Our testing data show that the tests associated with a DSCP marking of EF have the best results and the best positive impact on VoIP QoS. EF test cases are the only tests that have the best results. Each call has a MOS score of 4.400. Each of the test calls of both tests show a 0 packet loss and a 0 latency. Each of the test calls of both tests show 20.000 ms average interpacket arrival time. Our research's hypotheses statement states that cconfiguring VoIP traffic with a DSCP marking value of EF across all the equipment in a broadband HFC network improves VoIP's QoS. VoIP traffic that have a DSCP marking of EF kept its level of good quality and the mixture of video and Internet traffic have no negative impact on VoIP phone calls. When comparing the test data associated with a DSCP marking of EF with the other eight different test cases, the data show a clear difference where VoIP traffic with a DSCP marking of EF improved. None of the test calls associated with the DSCP marking of EF have any packet loss or latency. EF DSCP marking test calls have the same MOS value of 4.40. The interpacket arrival time associated with each of the EF DSCP marking test calls is 20.00.

6. CONCLUSIONS AND FUTURE WORKS

This study found that the higher VoIP traffic's DSCP marking value assignment, the less the chances of higher latency or packet loss in the VoIP traffic. Also, the study found that lower DSCP marking values have a negative impact on VoIP quality where the probability of running into higher latency or packet loss increases. This study also highlighted that VoIP traffic's DSCP marking is beneficial and have value even when the links between the MGC and the routers are underutilized. This research concludes that it is important to use DSCP marking in IP networks where various traffic share the same links. The assignment of EF DSCP marking to VoIP prioritizes VoIP over other types of traffic which eliminates packet loss, packet delay and improves VoIP's quality.

Future research should consider studying an additional three of the DSCP marking that include CS2, CS5, and CS6. This experiment tested five of the DSCP marking while the links between the MGC and the adjacent routers were not at full capacity. Future research studies should consider making the same tests but while the links between the MGC and the routers are at full capacity. Other recommendations include making the same tests that were executed in this experiment while changing the DSCP marking either in the adjacent routers or in the media gateway (MGW). This research did not test any VoIP-based calls to VoIP-based calls within the same network or between two different VoIP networks. Those are some of the options and ideas that researchers can consider and take into considerations to perform future research studies that are based on this research. Such studies can further strengthen this research and its results.

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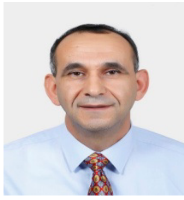

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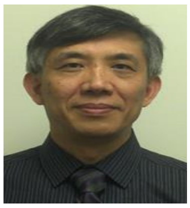

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