Journal of Mobile Communication and Networking

Volume No. 13

Issue No. 1

January - April 2025

ENRICHED PUBLICATIONS PVT.LTD

JE - 18,Gupta Colony, Khirki Extn,
Malviya Nagar, New Delhi - 110017.
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Journal of Mobile Communication and Networking

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Journal of Mobile Communication and Networking

(Volume No. 13, Issue No. 1, January - April 2025)

Contents

PERFORMANCE COMPARISON OF K-MEANS &CANNY EDGE DETECTION ALGORITHM ON MRI IMAGES

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ABSTRACT

MRI segmentation plays a crucial role in many medical imaging applications .Various approaches are applied for the Segmentation of the MRI depending on the medical application, Image modality and other factors. The objective of this paper is to perform a segmentation process on MR images of the human Brain using Kmeans Algorithm and Canny Edge Detection Algorthm. K-means Clustering algorithm gives us the segmented image of an MRI having the same intensity regions. K-means Clustering segments all the three matters of the brain i.e. Grey matter, White matter and Dark matter. Also the edge detection algorithm is implemented that gives us the boundaries of the various regions of the MRI depending on scale and threshold values used for the segmentation. Implementation of each algorithm is then discussed. Finally, the experimental results of each algorithm are presented and discussed. Following algorithm are used in this paper. Clustering is an unsupervised way of data grouping with a given measure of similarity. Clustering algorithms attempts to organize unlabeled feature vectors into clusters, such as samples within a cluster, that are more similar to each other than to samples belonging to different clusters, in which a validity measure is computed for each set of clusters. The number of clusters, which optimizes this measure, is the optimum number of clusters in the data set. The purpose of multi-resolution image analysis is to decompose the image into multi-frequency representations to visualize contents of interest in variable resolutions. Multi-scale filtering such as the canny operator can detect the edges in the low contrast or low S/N images. They are good edge detectors because they follow the optimal filter design criteria: good localization and high S/N output.

KEYWORDS Clustering Analysis, Medical Imaging, Thresholding Techniques, k-Means, Canny edge detector.

1. INTRODUCTION

Segmentation refers to the process of partitioning a digital image into multiple segments (sets of pixels, also known as superpixels). The goal of segmentation is to simplify and/or change the representation of an image into something that is more meaningful and easier to analyze.[11] Image segmentation is typically used to locate objects and boundaries (lines, curves, etc.) in images. The result of image segmentation is a set of segments that collectively cover the entire image, or a set of contours extracted from the image.[15]

Medical imaging refers to the techniques and processes used to create images of the human body (or parts thereof) for clinical purposes (medical procedures seeking to reveal, diagnose or examine disease) or medical science (including the study of normal anatomy and function). As a discipline and in its widest sense, it is part of biological imaging and incorporates radiology (in the wider sense), radiological sciences, endoscopy, (medical) thermography, medical photography and microscopy (e.g.

for human pathological investigations). Measurement and recording techniques which are not primarily designed to produce images, such as electroencephalography (EEG) and magneto encephalography (MEG) and others [16], but which produce data susceptible to be represented as maps (i.e. containing positional information), can be seen as forms of medical imaging. In the clinical context, medical imaging is generally equated to radiology or "clinical imaging" and the medical practitioner responsible for interpreting (and sometimes acquiring) the image is a radiologist[1]. Magnetic resonance imaging (MRI) is an imaging technique used primarily in medical settings to produce high quality images of the inside of the human body. MRI is based on the principles of nuclear magnetic resonance (NMR), a spectroscopic technique used by scientists to obtain microscopic chemical and physical information about molecules. Magnetic resonance imaging (MRI) is a noninvasive medical test that helps physicians diagnose and treat medical conditions [2]. MR imaging uses a powerful magnetic field, radio frequency pulses and a computer to produce detailed pictures of organs, soft tissues, bone and virtually all other internal body structures. The images can then be examined on a computer monitor, printed or copied to CD. MRI does not use ionizing radiation (x rays).Detailed MR images allow physicians to better evaluate various parts of the body and certain diseases that may not be assessed adequately with other imaging methods such as x-ray, ultrasound or computed tomography MRI, or magnetic resonance imaging, is a means of “seeing” inside of the body in order for doctors to find certain diseases or abnormal conditions. MRI does not rely on the type of radiation (i.e.,ionizing radiation) used for an x-ray or computed tomography (CT) scan. The MRI examination requires specialized equipment that uses a powerful, constant magnetic field, rapidly changing local magnetic fields, radiofrequency energy, and dedicated equipment including a powerful computer to create very clear pictures of internal body structure.

MR imaging of the head is performed to help diagnose:

- Tumors of the brain. • Developmental anomalies of the brain.
- Disorders of the eyes and the inner ear.
- Stroke.
- Trauma patients (in selected patients).
- C+auses of headache.

The main advantages of MR imaging system are:.
- It has an excellent capability for soft tissue imaging
- It has very high resolution of the order of 1mm cubic voxels
- It has high signal to noise ratio
- Multi channel images with variable contrast can be achieved by using different pulse sequences; this can be further utilized for segmenting and classifying different structures.[12]

2. MRI SEGMENTATION ALGORITHM

2.1 MRI Segmentation using Clustering Algorithm

Clustering is an unsupervised way of data grouping with a given measure of similarity. Clustering algorithms attempts to organize unlabeled feature vectors into clusters, such as samples within a cluster, that are more similar to each other than to samples belonging to different clusters, in which a validity measure is computed for each set of clusters. The number of clusters, which optimizes this measure, is the optimum number of clusters in the data set. The flowchart of the clustering approach is shown in Figure 1. The critical part of the clustering approach is choosing the additional cluster center.[3,4]One of the most common clustering methods is the K means algorithm. In its first step, an initial mean vector iteration is arbitrarily specified for each of the K clusters. Each pixel of the training set is then assigned to the class of which the mean vector is closest to the pixel vector, forming the first set of decision boundaries. A new set of cluster mean vectors is then calculated from this classification, and the pixels are reassigned accordingly. In each iteration, the K means will tend to gravitate toward concentrations of data in nearby regions of the feature space. The algorithm iterates until there is no significant change in pixel assignments. The criterion for terminating the iterative process can be defined in terms of the net mean migration from one iteration to the next.

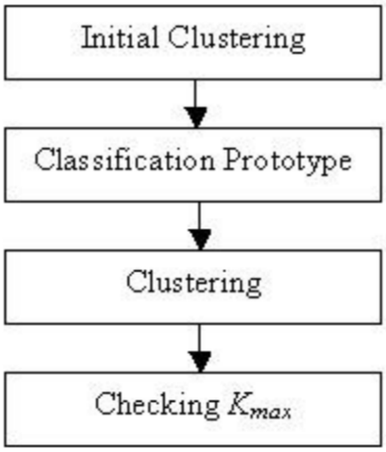


Figure 1: Flowchart of the clustering approach.

Kmeans clustering algorithm was developed by J. MacQueen (1967), k-means clustering is an algorithm to classify or to group your objects based on attributes/features into K number of group.K is positive integer number. The grouping is done by minimizing the sum of squares of distances between data and the corresponding cluster centroid. Thus the purpose of K-mean clustering is to classify the data.[14].

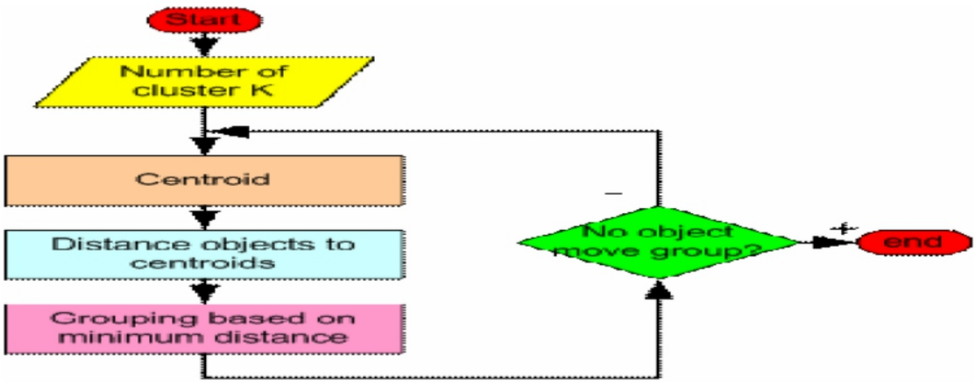


Figure 2 : flow chart for K- Means algorithm

2.1.1 Following are the algorithm of K Means

Step 1. Begin with a decision on the value of k = number of clusters

Step 2. Put any initial partition that classifies the data into k clusters. You may assign the training samples randomly, or systematically as the following:

• Take the first k training sample as single-element clusters

• Assign each of the remaining (N-k) training sample to the cluster with the nearest centroid. After each assignment, recomputed the centroid of the gaining cluster.

Step 3 . Take each sample in sequence and compute its distance from the centroid of each of the clusters. If a sample is not currently in the cluster with the closest centroid, switch this sample to that cluster and update the centroid of the cluster gaining the new sample and the cluster losing the sample.

Step 4 . Repeat step 3 until convergence is achieved, that is until a pass through the training sample causes no new assignments. [5,6,7]

2.2 MRI Segmentation using Canny Edge Detection Method

First of all, we have to clarify what is Edge Detection. Here are some definitions of edge detection: An edge is not a physical entity, just like a shadow. It is where the picture ends and the wall starts. It is where the vertical and the horizontal surfaces of an object meet. It is what happens between a bright window and the darkness of the night. Simply speaking, it has no width. If there were sensor with infinitely small footprints and zero-width point spread functions, an edge would be recorded between pixels within in an image. In reality, what appears to be an edge from the distance may even contain other edges when looked closer. The edge between a forest and a road in an aerial photo may not look like an edge any more in an image taken on the ground. In the ground image, edges may be found around each individual tree. If looked a few inches away from a tree, edges may be found within the texture on the bark of the tree. Edges are scale-dependent and an edge may contain other edges, but at a certain scale, an edge still has no width .Edges characterize boundaries and are therefore a problem of fundamental importance in image processing. Edges in images are areas with strong intensity contrasts – a jump in intensity from one pixel to the next. Edge detecting an image significantly reduces the amount of data and filters out useless information, while preserving the important structural properties in an image.

The Canny edge detection algorithm is known to many as the optimal edge detector. Canny's intentions were to enhance the many edge detectors already out at the time he started his work. He was very successful in achieving his goal and his ideas and methods can be found in his paper, "A Computational Approach to Edge Detection". In his paper, he followed a list of criteria to improve current methods of edge detection. The first and most obvious is low error rate. It is important that edges occuring in images should not be missed and that there be NO responses to non-edges. The second criterion is that the edge points be well localized. In other words, the distance between the edge pixels as found by the detector. and the actual edge is to be at a minimum. A third criterion is to have only one response to a single edge. This was implemented because the first 2 were not substantial enough to completely eliminate the possibility of multiple responses to an edge. [8] Canny specified three issues that an edge detector must address. They are:-

• Error rate:-The edge detector should respond only to edges, and should find all of them; no edges should be missed.
• Localization:- The distance between the edge pixels as found by the edge detector and the actual edge should be as small as possible.
• Response:- The edge detector should not identify multiple edge pixels where only a single edge exists. [9,10]

2.2.1 Steps:

Step 1. Apply derivative of Gaussian
Step 2. Non-maximum suppression .
• Thin multi-pixel wide “ridges” down to single pixel width

Step 3. Linking and thresholding
• Low, high edge-strength thresholds
• Accept all edges over low threshold that are connected to edge over high threshold

To improve current methods of edge detectors we must follow:

• The first and most obvious is low error rate. It is important that edges occurring in images should not be missed and that there be no responses to non-edges.

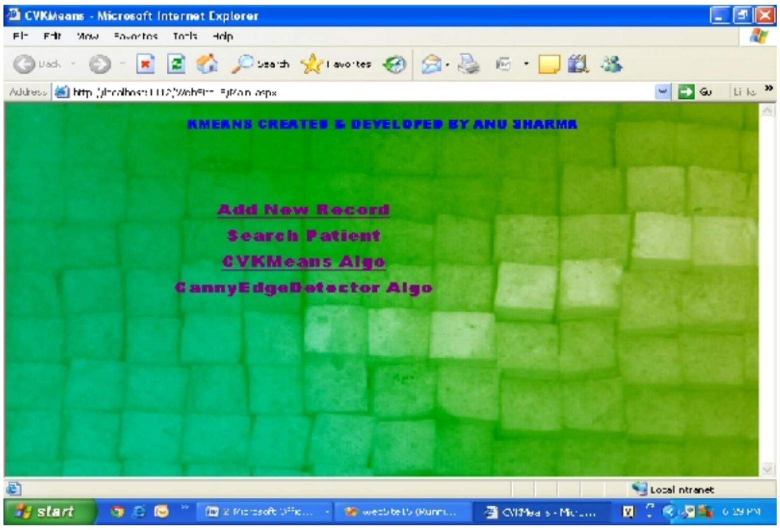
• The second criterion is that the edge points be well localized i.e., the distance between the edge pixels as found by the detector and the actual edge is to be at a minimum.

• A third criterion is to have only one response to a single edge [13]

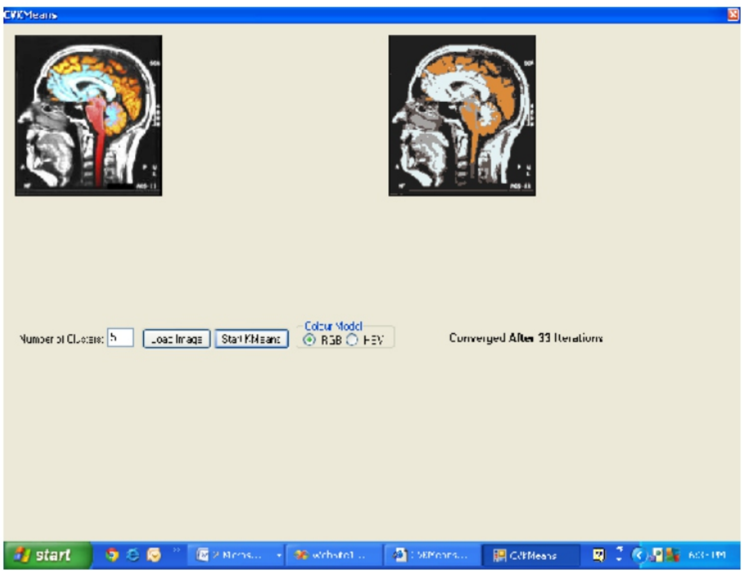
3. IMPLEMENTATION

Proposed algorithm is implemented in .Net using C# language. Here we have presented some snapshots of main page as well as comparisons of our algorithm with k means and canny edge detector.

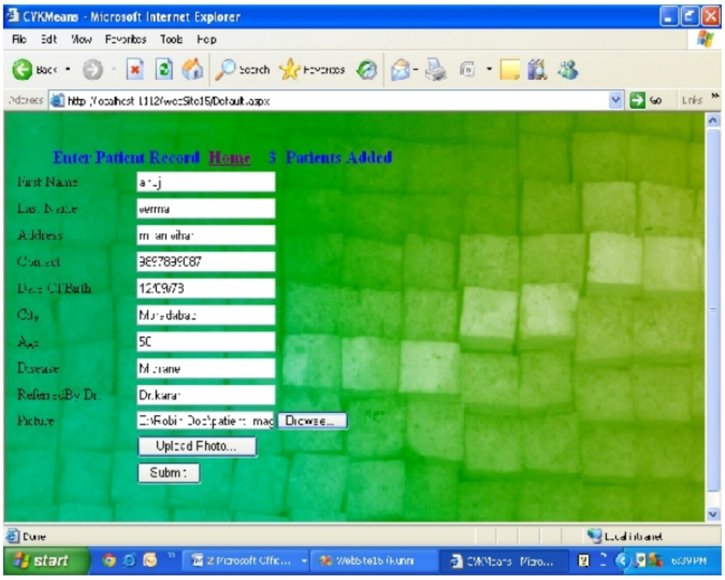
3.1Main Page



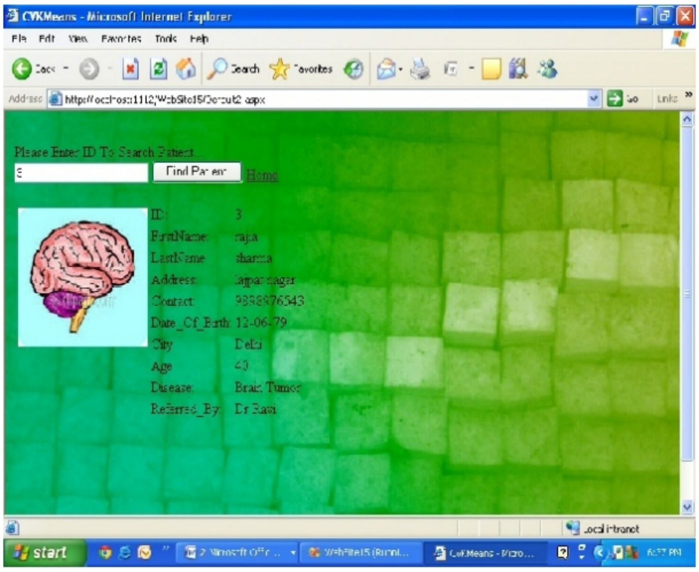
3.1Implementation of K-Means Clustering Algorithm.



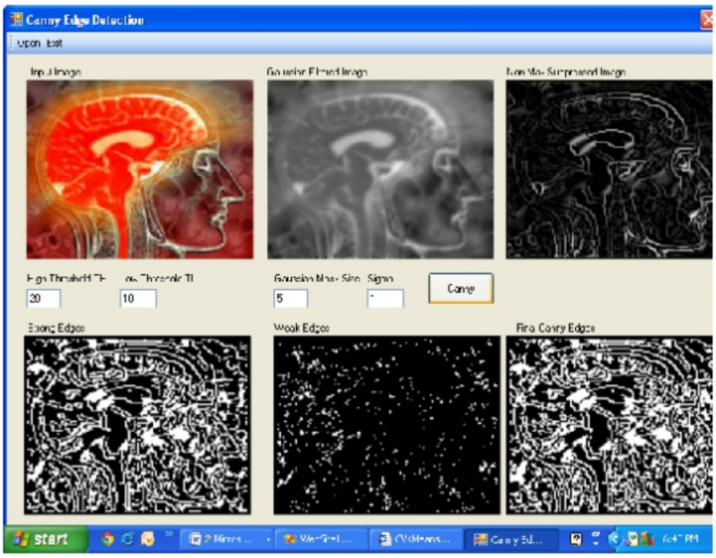
3.2IMPLEMENTATION OF SAVE PATIENT RECORD



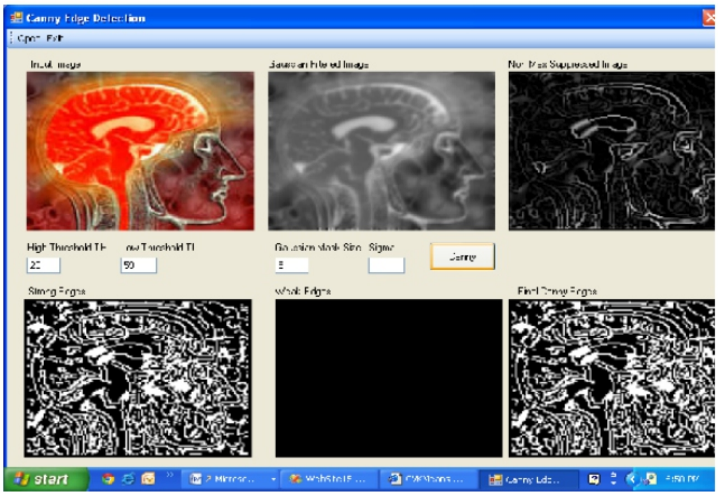
3.3Implementation of Search Patient Record



3.4Implementation of canny edge detection algorithm with T1= 20 and T2=10



3.5 Implementation of Canny Edge Detection Algorithm with T1= 20 and T2=59



4. RESULTS AND DISCUSSION

We have shown that Canny Edge Detection and K-means clustering algorithms are quite useful for retrieval of relevant images from image database. Our results indicate that the proposed approach offers significant performance improvements in retrieval of medical images.Further, by fine tuning of shape feature extraction and using other shape feature extraction methods, performance of the retrieval process can be improved more.

1) The k-mean clustering provides a lower localization error, and qualitatively, a dramatic improvement in edge detection performance over an existing edge detection method for speckled imagery.

2) The k-mean clustering meant to allow for balanced and well localized edge strengthmeasurements in bright regions as well as in dark regions.

3) The performance of the k-mean clustering has been demonstrated for edge-detection speckle reducing anisotropic diffusion.

4) This segmentation method can be develop by other project to get better view for medical image.

We conclude that following are the Disadvantages of Canny Edge Detector

1) The Canny algorithm contains a number of adjustable parameters, which can affect the computation time and effectiveness of the algorithm.

2) The size of the Gaussian filter: the smoothing filter used in the first stage directly affects the results of the Canny algorithm. Smaller filters cause less blurring, and allow detection of small, sharp lines. A larger filter causes more blurring, smearing out the value of a given pixel over a larger area of the image.

3) Thresholds: A threshold set too high can miss important information. On the other hand, a threshold set too low will falsely identify irrelevant information (such as noise) as important. It is difficult to give a generic threshold that works well on all images. No tried and tested approach to this problem yet exists.

4) Complex Computations,

5) Time consuming

4.1 Error Analysis .

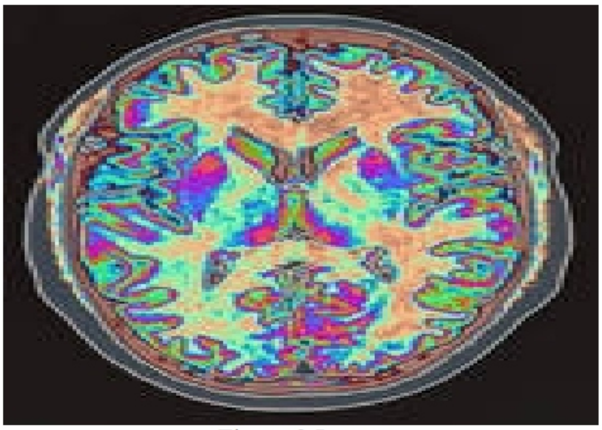


Figure 3:Image

TABLE 1: Noise Ratio of K-MEANS & CANNY Edge Detector

% ERROR RATE			
Noise Ratio	0	20	50
CANNY	4.1667	13.4766	21.5386
K-MEANS	2.8212	5.3277	20.4861

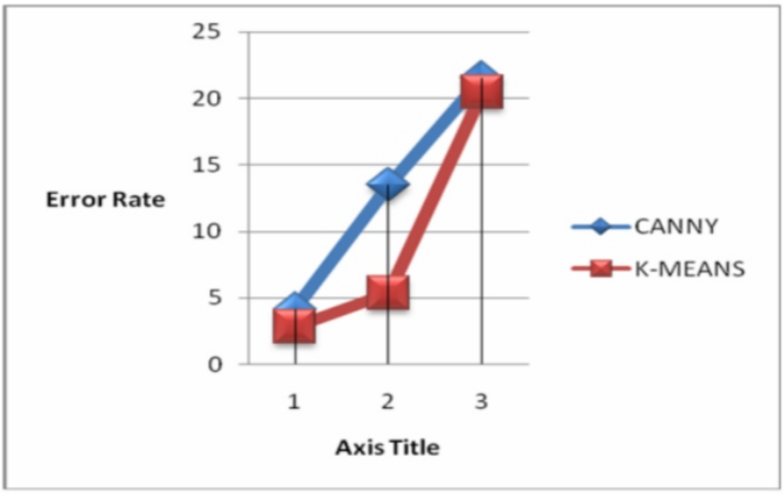


Figure 4: Noise Ratio Graph of K-MEANS & CANNY Edge Detector

TABLE 2: ENTROPY & MUTUAL INFORMATION (MI) OF CANNY EDGE DETECTORS & K-MEANS Algorithm

Edge detectors	MI	Entropy
Canny	0.0287	2.4095
Sobel	0.0302	1.7232
Roberts	0.0310	1.5261
Prewitt	0.0308	1.6631
K-Means	0.1369	5.6174

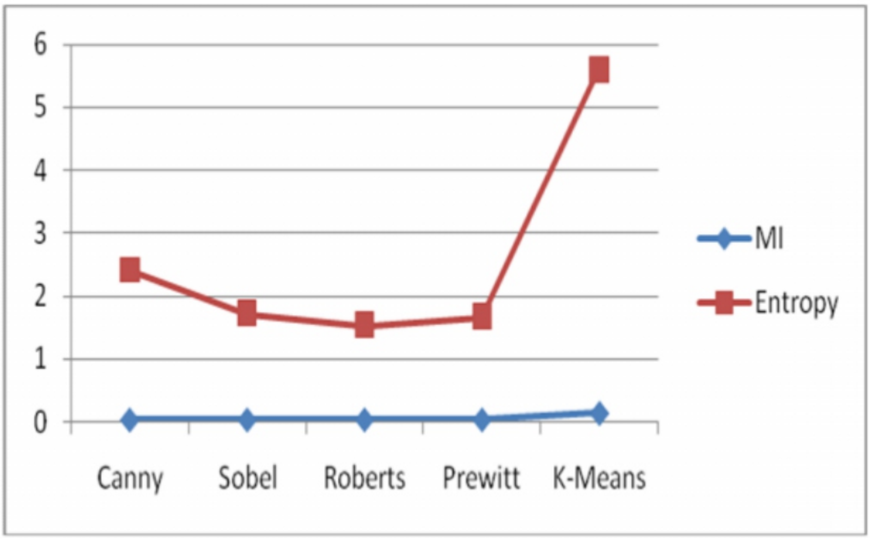


Figure 5: ENTROPY & MUTUAL INFORMATION (MI) GRAPH OF EDGE DETECTORS & K-MEANS Algorithm

The comparison of the segmented images are done by taking the entropy and mutual information measures.The entropy of an image can be defined as a measure of the uncertainty associated with a random variable and it quantifies, in the sense of an expected value, the information contained in a message. Entropy of an image E returns a scalar value representing the entropy of grayscale image I. Entropy is a statistical measure of randomness that can be used to characterize the texture of the input image. Entropy is defined as -sum (p.*log2(p)) where p contains the histogram counts.

5. CONCLUSION

The Canny Edge Detection Algorithm contains a number of adjustable parameters, which can affect the computation time and effectiveness of the algorithm. The size of the Gaussian filter used in the first stage directly affects the results of the Canny algorithm. Smaller filters cause less blurring, and allow detection of small, sharp lines. A larger filter causes more blurring, smearing out the value of a given pixel over a larger area of the image. A threshold set too high can miss important information.

On the other hand, a threshold set too low will falsely identify irrelevant information (such as noise) as important. It is difficult to give a generic threshold that works well on all images.

On the other hand, In K-Means Algorithm, the effect of noise is less than canny edge detector algorithm.Also in K-Means, quality of segmentation increases with increase in the number of clusters but here the value of K is decided by us according to the nature of the MRI and the according to the application used.

6. FUTURE SCOPE OF K-MEANS ALGORITHMS

Kmeans algorithm segments the MRI on the basis of region intensity into K clusters. Hence all the three matters of the Brain i.e. Grey matter, White matter and dark matter can be detected from the Segmented image. Also we note that quality of segmentation increases with increase in the number of clusters but here the value of K is decided by us according to the nature of the MRI and the according to the application used. For example we can decide by seeing the image that how many clusters should be there in the particular image. For example by seeing the number of colors in a colored MRI we can easily guess the value of k. In the snapshot a MRI is segmented with different value of k which will give different results each time.

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THE VPQ SCHEDULER IN ACCESS POINT FOR VOIP OVER WLAN

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A B S T R A C T

The Voice over Internet Protocol (VoIP) application has observed the fastest growth in the world of telecommunication. VoIP is seen as a short-term and long-trem transmission for voice and audio traffic. Meanwhile, VoIP is moving on Wireless Local Area Networks (WLANs) based on IEEE 802.11 standards. Currently, there are many packet scheduling algorithms for real-time transmission over network. Unfortunately, the current scheduling will not be able to handle the VoIP packets with the proper manner and they have some drawbacks over real-time applications. The objective of this research is to propose a new Voice Priority Queue (VPQ) packet scheduling and algorithm to ensure more throughput, fairness and efficient packet scheduling for VoIP performance of queues and traffics. A new scheduler flexible which is capable of satisfying the VoIP traffic flows. Experimental topologies on NS-2 network simulator wereanalyzed for voice traffic. Preliminary results show that this can achieve maximum and more accurate VoIP quality throughput and fairness index in access point for VoIP over WLANs. We verified and validated VPQ an extensive experimental simulation study under various traffic flows over WLANs.

KEYWORDS VF, NVF, VoIP, WLAN, VPQ, WLAN

1. INTRODUCTION

This research Voice over IP over Wireless Local Area Networks (VoIPWLAN) is in the developing field of wireless broadband Internet technologies which has the great potential to provide a low-cost high-speed Internet voice calls with user mobility that can profoundly impact our lives in a positive way. VoIP over WLAN environment allows users to make IP-based calls over global networks. VoIP transmits IP-based telephony calls over packet on WLANs. In IPbased networks, analogue voice signals are digitized and move on real-time transmission over network. Then, find the efficient way to reach the proposed destination. Normally, they out of the order from original order and receiver side the packets rearranged in the proper way before convert into aging analogue voice signals for voice conversion over networks [1], [2], [3], and[4].

Currently, there are approximately 1 billion fixed telephony lines and 2 billion mobile-phone lines in the world. Now, we are moving ahead to IP network based protocols known as Voice over Internet Protocol (VoIP) [5]. VoIP over Wireless Local Area Network (WLAN) might be a leading application in collaboration with 3rd Generation (3G) mobile network [6]. A variety of new multimedia applications such as VoIP, video on demand (VoD), Internet Protocol TV (IPTV) and teleconferencing are based on network traffic scheduling algorithms. A number of research solutions have been proposed to satisfy different Quality of Service (QoS) requirements [7], [8], [9], [10], [11] and [12].

1.1 VoIP Protocol Architecture

Figure 1 shows the fundamental of VoIP protocol stack architecture to implement a VoIP network system over WLANs [13], [14]. VoIP is a real-time application and transmit the voice using Real-Time Transport Protocol (RTP), User Datagram Protocol (UDP) and Internet Protocol (IP) (RTP/UDP/IP) over WLANs [15], [16], [17] and [18]. Each VoIP packet has the headers, RTP (12 bytes), UDP (8 bytes), and IP (20 bytes) headers. Lastly, data-link layer Medium Access Control (MAC) has a (34 bytes) header. These all bytes headers calculate as 74 bytes of overhead in VoIP protocol.

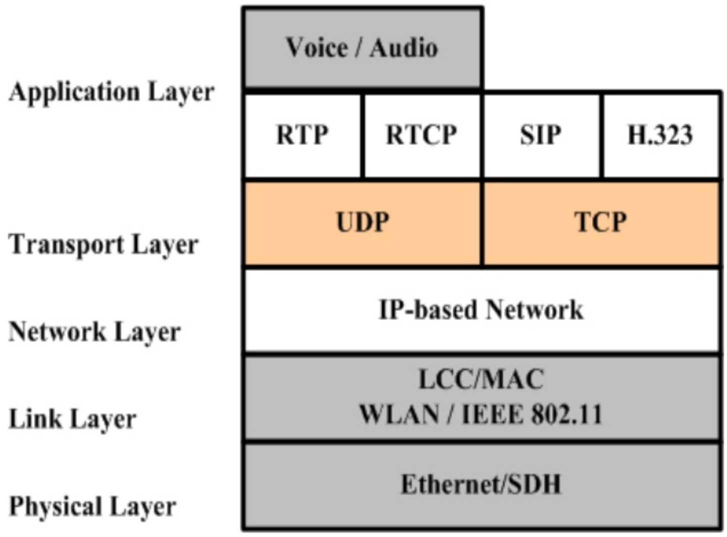

Figure 1. Voice processing Signals in a VoIP Network System

The Session Initiation Protocol (SIP) was considered for handles a multimedia call setup and H.323 is considered by ITU to allocate IP-based phones on the public telephone network to talk to PC-based phone over IP-based networks [19] and [20].ITU is a standard that specifies the components, protocols and procedures for multimedia communication services such as real-time audio, video, and data communications over IP-based packet networks [21], [22], [23], [24] and [25].

1.2 VoIP over WLANs Networks

VoIP over WLANs have observed a fastest growth in the World of communication. WLAN is most guaranteeing technologies among wireless networks, which have been facilitated high-rate voice services at very less cost and flexibility over IP-based networks [26], [27], [28] and [29]. Main source for such adaptation is that VoIP real-time application over WLANs is more flexible than traditional public switched telephone networks systems (PSTN) [30].Moreover, VoIP can support multiple infrastructure environments IP-based-Phones (IP-Phone, PC-based soft-Phone, IP-based Packet-

Phone), Soft-Phones (PC-to-PC Phones), Traditional and mobile Phones (Telephone, Cell-Phone). Detail are as below in table 1.

Traditional Phones	**IP-based Phone**	**Soft and Hard Phone**
Dialup Phone	IP-Phone	PC-to-PC Phone
Telephone	PC-based Phone	PC-to-Phone
Cell-Phone	PC-based Soft Phone	Phone-to-PC

VoIP provides mix mode communication with PC-to-PC, PC-to-IP-Phone and PC-to-CellPhone commutation over WLANs. They are moving on campuses, hotels, airports, health care, commercial, education, and industries to provide voice traffic. WLAN also provide audio, voice and video conferences over IP-based networks [31], [32], [33], [34], [35] and [36].In WLANs, there are two essential kinds of the services architectures: ad-hoc architecture and infrastructure architecture. In ad-hoc architecture, station (STA, a mobile node) can be able to connect with IP-based network without the connectivity to any wired backbone network and without the need of an Access Point (AP) [37]. In infrastructure, the STA can be able to connect with IP-based network with the connectivity to any wired backbone network and with the need of an AP. In this paper we will focus on infrastructure architecture network where VoIP traffic is transmitted signals via an AP. WLANs provide number of industries standards of AP. Each AP can maintain a restricted number of parallel voice nodes [38], [39], [40], [41], [42], [43], [44], [45] and [46].

1.3 VoIP, IEEE 802.11 MAC and WLANs Standards

The IEEE 802.11 WLANs, we called as a wireless Ethernet and play an important function in the future-generation networks. WLAN based on Link Layer (LL). LL divided into Logical Link Control (LLC) and Medium Access Control (MAC) sub-layer categorizes with two functions, Distributed Coordination Function (DCF) and Point Coordination Function (PCF) [47], [48] and [49]. The IEEE 802.11 WLANs support both contention-based DCF and contention-free PCF functions. DCF uses Carrier Sensing Multiple Access/Collision Avoidance CSMA/CA as the access method. [50], [51], [52], [53], [54], [55], [56] and [57].

1.4 Problem Statement

IP-based networks are managing voice, data, web browsing, email, and video applications on the same network flow over WLANs. They were not mainly designed for real-time transmission over WLANs support both contention-based DCF and contention-free PCF functions. DCF uses Carrier Sensing Multiple Access/Collision Avoidance CSMA/CA as the access method. [50], [51], [52], [53], [54], [55], [56] and [57].

1.4 Problem Statement

IP-based networks are managing voice, data, web browsing, email, and video applications on the same network flow over WLANs. They were not mainly designed for real-time transmission over WLANs and it can be a deadlock in the traffic flow over WLANs. We are showing a bottleneck topology of mix mode traffic over WLAN as below in figure 2 [58].

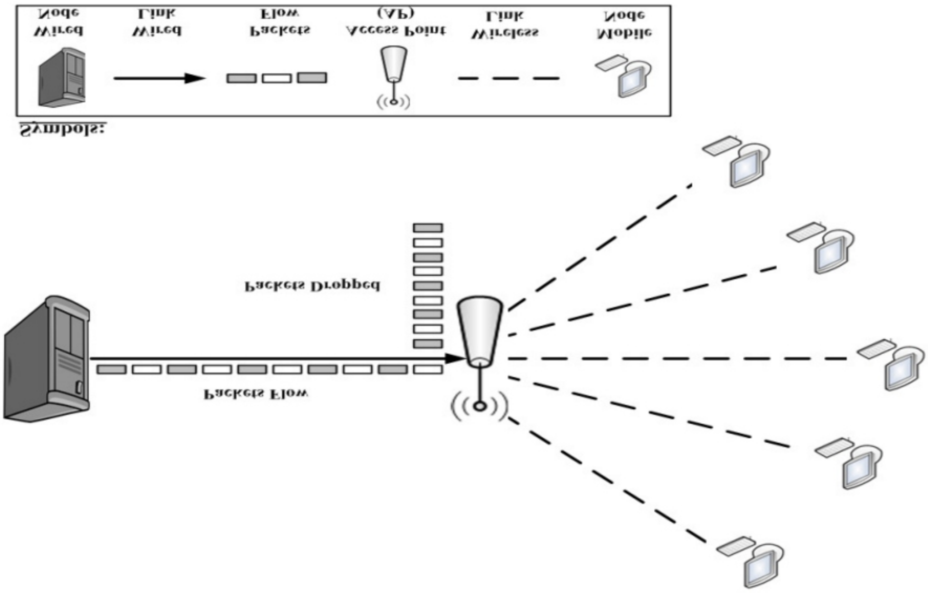

Figure 2. Bottleneck Topology of mix mode Traffic over LAN networks

Quality of Services (QoS) is considered the main issue in VoIP system. Due to its importance, following research focuses on solving the VoIP scheduling algorithm problem. This research tries to compare with some well know real-time scheduling algorithms over WLANs. The proposed method tries to achieve better acceptable results for VoIP high-speed real-time application. In figure 1.6, we implement a topology, it has bottleneck problem due to attacker on node 2 and node 5. VoIP is a real-time application that needs timely techniques to enhanced traffic over networks. This is a challenging task on VoIP networks.

Through the past decades many schedulers were introduced to solve real-time traffic application issues. These schedulers can be divided into three groups and these groups are as following, packet-based schedulers, frame based-packet schedulers, and regulative packet schedulers. These above problems degrade the QoS of VoIP over IP-based networks. We need to introduce a new voice scheduling algorithm to solve above VoIP traffic issues. New method should be an efficient, fair, high throughput, bandwidth guarantee and that will enhance performance of VoIP over WLANs

1.5 Aim and Objectives

The aim of this paper is to introduce an efficient schedulers and algorithms that support the VoIP application over WLANs. We will assume a fundamental related work to examine the available schedulers with their outcomes and their drawbacks. We will introduce new scheduler and algorithms to enhance the performance of VoIP over WLANs using IEEE 802.11 standards. We will evaluate, examine, and simulate our techniques with related algorithms for real-time applications. To improve the real-time traffic scheduler algorithm it is possible to resolve many of these problems. In this research the specific objectives are as following:

• To develop a new Voice Priority Queue (VPQ) scheduler architecture and algorithms for VoIP traffic that can be proficient to fulfil the scheduling requirements over WLANs.
• To classify VoIP-Flow (VF) traffic and Non-VoIP-Flow (NVF) traffic over WLAN using IEEE 802.11 standards.
• To compare VPQ scheduler and algorithms with more related work, to evaluate, validate and verify our scheduler with other schedulers.
• To enhance the scalability of traffic over WLAN, using our Test-Bed in VoIP.

The paper is organized as follows. In section II we discuss the related work with different scheduling algorithms and initiate their limitation for multimedia application. In section III, we proposed a new VoIP scheduling algorithm and methodology. In section IV we describe simulation experimental Setup that compares the efficiency between new VoIP scheduling algorithm and other related scheduling algorithm. Section V, we describe the results and in section VI we conclude this paper with future research work remarks.

2. RELATED WORK

The high-speed packet-switched networks are essential research areas. Voice over IP (VoIP) over Wireless Local Area Network (WLAN) network is one of the most applying technologies to utilize high-speed packet-switched networks. The IEEE 802.11 standard has expanded with an importance of researchers in the last decade.There are some scheduling algorithms to support packet scheduling in wire and wireless networks. Few of them are as following: Class Based Queue (CBQ), Faire Queue (FQ), Weight Faire Queue (WFQ), Generalized Processor Sharing (GPS), Worst-case Fair Weighted Fair Queuing (WF2Q), Deficit Round Robin (DRR), Deficit Transmission Time (DTT), Low Latency and Efficient Packet Scheduling (LLEPS), Credit Based SCFQ (CB-SCFQ), Controlled Access Phase Scheduling (CAPS), Queue size PredictionComputation of Additional Transmission (QP-CAT), Temporally-Weight Fair Queue (T-WFQ), Contention-Aware Temporally fair Scheduling

(CATS), and Decentralized-CATS (DCATS).

We will also study a general discussion of related research work in this section for real-time applications.

2.1 Class Based Queue (CBQ)

Floyd et. al. [59] introduced the Class Based Queue (CBQ) for hierarchical bandwidth linksharing and resource technique of packet-switch networks. CBQ is a class based algorithm and share the bandwidth for each class with well organize manner over IP-based networks. CBQ manages bandwidth link-sharing rations for all classes. It maintains each queue and provide fairly link-sharing. We have noticed that CBQ best solution for data traffic. It is implementing with gateway technique and fulfils the range of service and link-sharing. CBQ is a combination of classifier, estimator, selector, and over limit process to schedule the traffic flow classes those extended link sharing limits.

In figure 3, CBQ provides the solution of multiple types of traffic flow over IP-based networks. CBQ divides the bandwidth or allocates the link-sharing according to the traffic requirement. In below figure, CBQ has 6 Mbps bandwidth and it needs to sharing this bandwidth in three different kind of traffic flows like audio 1Mbps, video 2Mbps and File Transfer Protocol (FTP) 3Mbps over wireless nodes.

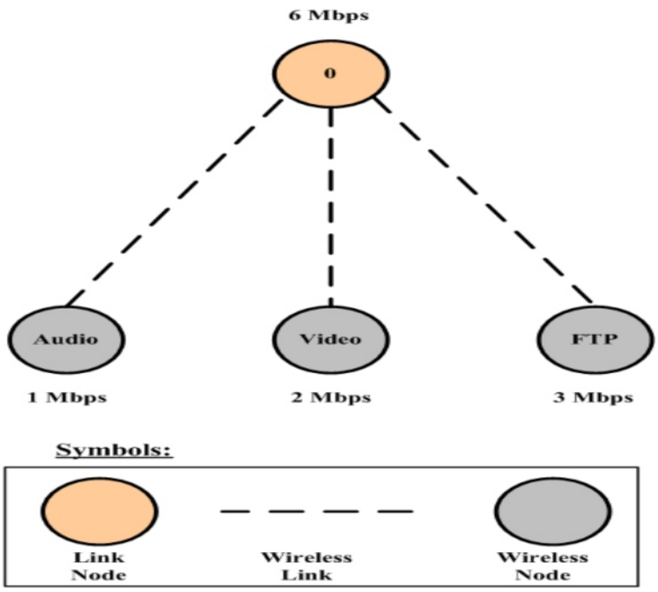

Figure 3. Priority, Structure of Link-Sharing Allocation [59]

The drawbacks of CBQ are as following: If bursty traffic on real-time application and it will become the cause of the delay. CBQ does not assume the delay of packets and the dealy of packets is needed less and scalable for VoIP application. The real-time application requires a specific buffer to control the bursty traffic on IP-based networks but the bulky buffer introduces the delay of playback longer.

2.1 Weighted Fair Queue (WFQ)

WFQ is a sort-based packet scheduling algorithm with latest updating of Generalized Processor Sharing (GPS). WFQ offers N queues same time with different bandwidth service rate by providing each queue a weight and arranges different percentage of output port bandwidth. WFQ calculates the departure time of each packet and manages multiple sizes of packets. The traffic flows of bigger packets are not assigned extra bandwidth. WFQ is inefficient to calculate the timestamps. WFQ will invite the work complexity of log , where task is the number of flows. Furthermore, due to slow process of sorting among the timestamps, WFQ is not suitable of real-time applications.

2.2 Contention-Aware Temporally fair Scheduling (CATS)

Seoket. al. introduced [61] the Contention-Aware Temporally fair Scheduling (CATS) is packet based algorithm and offered fairness traffic flow over WLANs IEEE 802.11a/b standards. CATS introduced for equal time sharing for each flow over IP-based networks. CATS decide packets scheduling order after virtual finish time. In addition, the scheduler is proficient for performance in multi-rate WLANs. CATS provides solution for Carrier Sense Multiple Access / Collision Avidness (CSMA/CA) technique. Figure 4 shows the flowchart of QP-CAT algorithm.

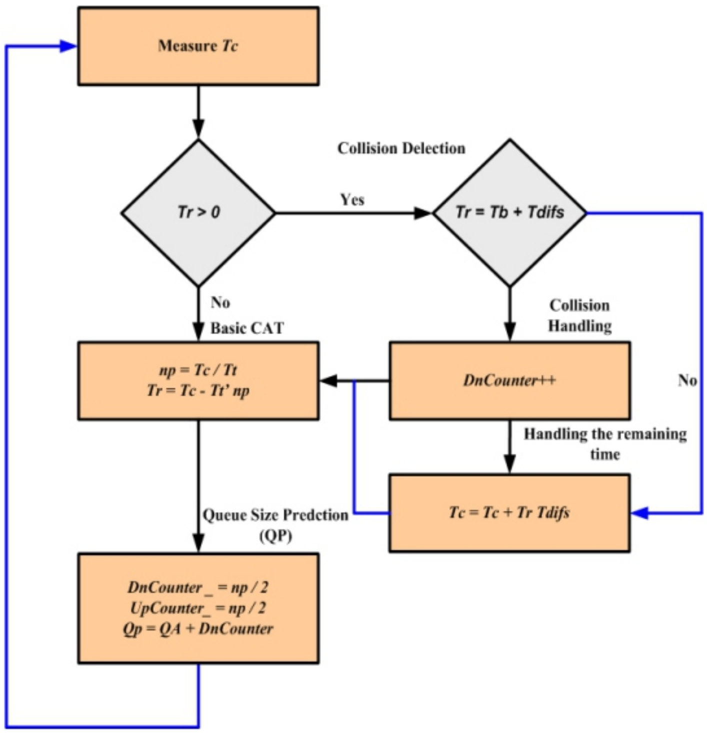

Figure 4. Flowchart of QP-CAT Algorithm [60]

The drawbacks of Contention-Aware Temporally fair Scheduling (CATS) are as following. The CATS based on Generalized Processor Sharing (GPS) that applied on wired based networks and based on

GPS and less performance than CATS.

3. METHODOLOGY

The scheduling architectures provide major role in Wireless LAN (WLAN) networks to fulfill the essential requirements of Voice over Internet Protocol (VoIP) traffic flow over IP-based network. To guarantee the requirements of the new VoIP over WLAN applications, a Quality of Service (QoS) requires responsible scheduling architecture, algorithms, efficient traffic, and enhances voice traffic flow over WLANs. Being an emerging technology, WLAN offers VoIP, Internet Protocol TV (IPTV) and High-Performance Video-Conferencing (HP-VC) traffic over IP-based networks. The goal of this section is to propose an efficient traffic scheduler for VoIP traffic flow over WLANs. The numbers of related schedulers have been proposed to support traffic flow over IP-based networks. We noticed, these related schedulers support traffic flows with limited services, the special requirements of real-time application cannot meet with fully support over WLANs.

The scheduling architecture an important technique to achieve efficient throughput and fairness over WLANs IEEE 802.11 Standards. Scheduling technique illustrated the voice traffic over WLANs. It will be able to offer link-sharing of bandwidth, to tolerate the status of changing traffic queues and to be scalable over IP-based networks. The Weighted Round Robin (WRR) [62] scheduler idea is to increase and allocate the different quantity of traffic flow in a round for each non-empty queue over networks. The Round Robin (RR) scheduler is one that keeps a separate queue for every flow with packets waiting flow, and RR serves one packet from each queue in turn [63].

3.1 Voice Priority Queue (VPQ) Scheduler Architecture

VPQ is pre-packet delay bounds and provide both bounded delay and fairness over WLANs IEEE 802.11 Standards. VPQ is making it easy to provide both delay-guarantees and fairness concurrently over networks. VPQ provides throughput guarantees for error-free flows, long term fairness for error flows, and short term fairness for error-free flows, and graceful degradation for flows that have received excess service. VPQ scheduler architecture is as following in figure 5.

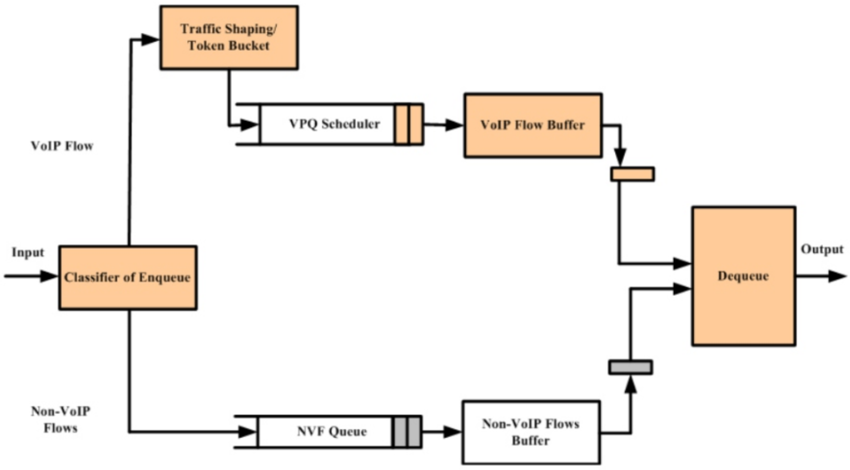

Figure 5. Voice Priority Queue (VPQ) Scheduler Architecture

VPQ scheduler architecture based on initializes traffic flows, classification of enqueue traffic flows, VoIP-Flow (VF) and Non-VoIP-Flow (NVF), traffic shaping, token bucket, Voice Priority Queue (VPQ) management system, VoIP-Flows buffer, Non-VoIP-Flow buffer and dequeue traffic flows for end user over WLANs IEEE 802.11 Standards. In Figure 5, classification of enqueue traffic flows sorted-based and checks the index of the incoming packet. The classification architecture supports a mechanism that work as like Differentiated Services (DiffServ) and dispatched into different traffic flows depending on their destination mobile stations. After that, classification send traffic flows to the VF traffic shaping and token bucket flow. The shaping controls the amount of flow and sent traffic flow to the token bucket flow. The token bucket applies on bursty traffic as regulated maximum rate of traffic over WLANs IEEE 802.11 Standards. The traffic shaping and token bucket send VF flows to main Voice Priority Queue (VPQ) scheduler component for processing over network. VPQ forward these packets into Voice- -Flow-Buffer (VFB). The VFB is temporary buffer flow for VF and NVF traffic. For easily understand we would like to inform you that Base Station (BS) has instantaneous and perfect understanding of VF, NVF channel's position due to their weight,energy and priority queue. Finally, VF and NVF flows dequeue traffics in well manner to the destination over WLANs.

4. EXPERIMENTAL SETUP & SIMULATION

In this section, we will discuss experimental setup and simulation of our methodology. We will perform validation and verification of the developed simulation modules. We will explains some of the main component that used in this research paper, this include: A Novel Voice Priority Queue (VPQ) Scheduler Architectures over WLANs IEEE802.11 Standards. Furthermore, we provide an experimental setup of the scheduler architectures considered in our research. We will explain all three stages of the VPQ and simulate VPQ scheduler components. VPQ based on the two types of traffics flow named as VoIP-Flow (VF) and Non-VoIP-Flow (NVF) as discussed in the section three . VPQ traffic will initialize from classification of enqueue traffic flows to dqueue traffic flows for end user.

4.1 NS-2 Simulations and Results Analysis Process

NS-2 is based on OTcl scripts to setup network topologies for VoIP over WLANs IEEE 802.11 Standards. Normally, NS-2 processes consists of the following steps: The Tcl simulation codes, Tcl interpreter, simulation results, pro-processing and finally, it will fall into two types of results formats trace file analysis and Network Animator (NAM). The NAM will perform graphically and interpreted into the trace file (.tr) and then analysis is shown in X-graph or graph tool. NS2 supports the real-time, VoIP traffic schedulers over WLANs.

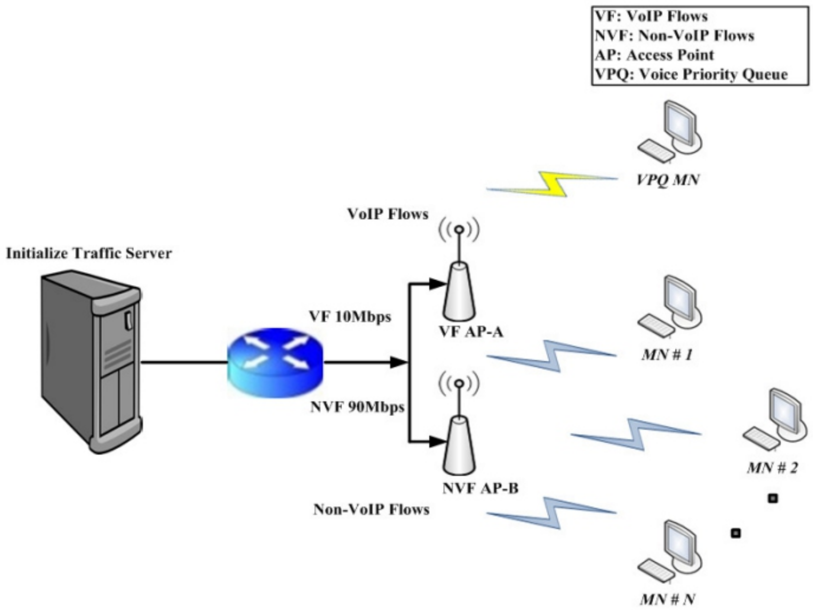

Figure 6 Simulation Setup of VPQ Scheduler Algorithm

We can create multiple topologies using nodes and packet forwarding technique. We can also connect the nodes to form links. NS-2 allows queue management and packet scheduling and queues shows the locations where packets may be held or dropped over IP-based networks. Its can supports is included for drop-tail First-In-First-Out (FIFO), Class Based Queue (CBQ), RED Queue management, Fair Queue (FQ), Stochastic Fair Queue (SFQ) and Deficit Round Robin (DRR) as we discussed in detail in the section two related work. Furthermore, NS-2 supports differentiated traffic services as like classification of traffic over WLANs. We implement our Novel Voice Priority Queue (VPQ) Scheduler Architectures over WLANs IEEE802.11 Standards. As above,Figure 6 shows the simulation setup. We have two type of traffic flow like VoIP flow and Non-VoIP flow. We have VoIP flow 1Mbps and Non-VoIP flow 10 Mbps traffic flow. The bottleneck for this set of simulation is the link between the two types of flow. The bandwidth of the links between nodes and routers is 11Mpbs and the propagation delay in 1ms.

5. RESULT & DISCUSSION

In this section, includes our achieved results those based on various types of topologies simulations in Network Simulation – 2 (NS-2). We have shown stages of Voice Priority Queue (VPQ) using NS-2 simulation and test-bed experimental setup. We have described the topologies in simulation. We have compared the VPQ with Contention-Aware Temporally fair Scheduling (CATS), Temporally-Weighted Fair Queuing (T-WFQ) and controlled access phase scheduling (CAPS) traffic schedulers. Furthermore, we have also compared the VPQ Test-Bed with CATS, T-WFQ and CAPS schedulers. We will discuss in detail these results are as following. Figure 7 shows the total throughput (Mbps) of proposed VPQ, CATS and T-WFQ algorithms over WLAN network in the simulation. If we compared with previous all 4 flows the difference in the throughput measurement. In the all pervious flows we measured in (Kbps) and in the total throughput we measured in (Mbps) to evaluate the throughput results.

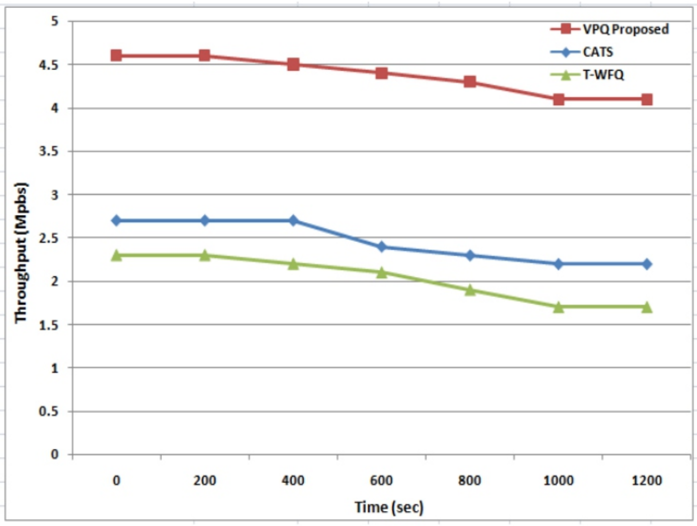

Figure 9. Total Throughput According to the Packet Size of Flow

The T-WFQ algorithm shows the lowest throughput among and T-WFQ throughput stated from 2.6 (Mbps) and its reached 0.7 (Mbps) on 1200 (sec). The CATS was higher than T-WFQ algorithm due to better performance over WLAN. The CATS throughput started from 2.3 (Mbps) and step by step decreased to 1.6 (Mbps). Our proposed VPQ algorithm better then both algorithms over WLANs and VPQ has higher throughput due to, classify the VF, NVF traffic, high date-rate flow and facilities more packets. The proposed VPQ started the throughput 4 (Mbps) and ended to 3.2 (Mbps) against 1200 (sec).

Figure 10 shows the fairness index according to the packet size of the flow. The fairness measured from

0 to 1 and above than 0 considered best. We also compared proposed VPQ, CATS and T-WFQ algorithm. We noticed in the T-WFQ algorithm start its fairness from 0.88 and it's moved to down 0.3 against 1200 (sec).

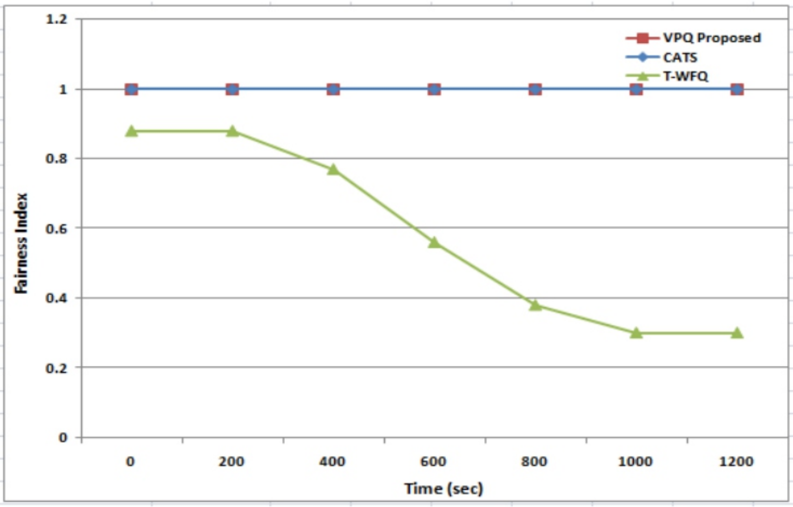

Figure 10 Fairness Index according to the Packet Size of Flow

As compared VPQ and CATS both shown best fairness index and they reached to same 1 index from started and ended time of simulation.We explained our simulation and experimental results in the graphs and tables. We also evaluated our Voice Priority Queue (VPQ) scheduler's results with most related scheduler and algorithms. We evaluated VPQ with Contention-Aware Temporally fair Scheduling (CATS), Decentralized-CATS, Decentralized-CATS+ and controlled access phase scheduling (CAPS), and Temporally-Weighted Fair Queuing (T-WFQ).

6. CONCLUSION

In this paper, we have presented a novel scheduling discipline called Voice Priority Queue (VPQ) scheduling algorithm. VPQ is simple, fair, and efficient with voice flow. In addition, it satisfies the unique requirement imposed by voice flow. In comparison to other scheduling algorithms of similar efficiency, VPQ has better throughput properties, as well as a higher efficiency. We expect that our proposed algorithm will be able to be offer fairness and delay guarantee in the future VoIP over WLANs. VPQ has satisfied these two requirements in CSMA/CA based 802.11 WLANs. The main function of VQP is that time should be fairly allocated despite the variable data rate and packet size of access point. In future work, we will investigate VPQ with Video traffic over WLANs on IEEE 802.11 Standards.

ACKNOWLEDGEMENTS

The authors would like to thank you for Universiti Utara Malaysia for support my research work.

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Patch Antenna Performance investigations

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A B S T R A C T

Micro strip antennas are gaining popularity with miniaturization of communication. gadgets. Research is going on to discover new substrate materials for improved performance of the patch antenna. The antenna performance, as a function of substrate thickness over wide range for two different dielectric materials of relative permittivity in the ratio 1:2, is examined. The evaluation is carried for two different resonant frequencies namely 1GHz and 2. 5GHz.. Further the performance of patch antenna is evaluated using a mixed dielectric substrate of two different materials. The design and performance analysis is carried out using High Frequency Structure Simulation (HFSS) Software. The results are reported and discussed.

Keywords: 3db Bandwidth, Directivity, Micro strip Patch, Probe feed and VSWR.

I. INTRODUCTION

Micro strip antennas are low profile and less weight antennas and are generally used when conformal low profile antennas are required [1,3]. With the rapid development of Integrated circuits and wireless communication systems compact wideband antenna design has become a challenging task [1, 2]. Printed antennas are widely used in variety of wireless communication applications. Patch antennas are used where size, weights, performance, ease of installation are major constraints. Many researchers are working on exploring suitable substrate materials for improving performance of patch antennas[4-10].

1. Literature Review

Deepak et al. [4] tested micro strip antenna with substrate thickness 1. 589 mm for four different substrates namely RT/ Duroid 5880, Fused Quartz, Alumina and Epsilam. They observed that Alumina exhibited better performance. [4].

Debashis Sarmah et al. [5] carried out the study of Graded composite materials (LDPE/TIO2) as Substrate for Micro strip Patch Antennas in X-band.

The Variation of impedance bandwidth and return losses with different substrate materials are investigated in Micro strip-Coupled Rectangular Dielectric Resonator Antennas by Shok Theng Low et al. [7].

In case of a Patch the resonant frequency depends on the length of the patch and length is a crucial parameter. The thickness (h) of substrate is a function of resonant frequency and lies generally in the range 0. 003 $\lambda 0$ to 0. 05 $\lambda 0$ [1,2].

In order to explore the effect of substrate permittivity and thickness on antenna performance micro-strip antennas with two different dielectric materials of relative permittivity in the ratio 1:2 and varying thickness over wide range are designed and tested. Further this exercise is carried for two resonant

frequencies namely 2. 5 Ghz and 1GHz. For both substrate materials thickness is varied over range 1:12 for 2.
5GHz and 1:16 for 1GHz.
From the results for 2. 5 GHz it is observed that RT Duroid showed better gain of 5. 5dB,while FR4 provided better bandwidth of 61. 9MHz. It is felt by using a combination of these materials for substrate both parameters may show better values.
Accordingly a combination of RT/Duroid and FR4 dielectric materials of thickness 0. 24 cms each is taken as substrate and the antenna units are designed and tested. In first arrangement RT/ Duroid 5880 is kept as substrate above ground plane, then Fr4.
In second arrangement FR4 is kept as the lower substrate and then RT/ Duroid5880. The performance parameters namely Return loss, VSWR, Peak Gain, Directivity, 3dB Bandwidth and Radiation Efficiency are evaluated through simulation with the help of HFSS software. The results are presented and discussed.

II. DESIGN OF PATCH ANTENNA

The dimensions of patch are given by the following relations [1].

Width of the patch $W = \frac{c}{2fr}\sqrt{\frac{2}{\epsilon r+1}}$ (1)

Due to fringing effects the effective length of patch L eff appears to be more than the actual length of the patch L. The effective length is depends on substrate permittivity and resonant frequency

$$L_{eff} = \frac{c}{2fr\sqrt{\epsilon_{eff}}} \quad (2)$$

Actual length of patch L is $L = L_{eff} - 2\Delta L$ (3)

The change in length due to fringing effects is given by [1-2]

$$\frac{\Delta L}{h} = 0.412\,\frac{(\epsilon_{eff}+0.3)\left(\frac{W}{h}+0.264\right)}{(\epsilon_{eff}-0.258)\left(\frac{W}{h}+0.8\right)} \quad (4)$$

ϵ_{eff} is the effective relative permittivity and

$$\epsilon_{eff} = \frac{\epsilon r+1}{2} + \frac{\epsilon r-1}{2}\left[1+\frac{12h}{W}\right]^{-0.5} \quad (5)$$

ϵr is Dielectric constant of substrate. h is substrate thickness. c is free space velocity of e m wave and f_r is resonant frequency of patch.

The Ground plane dimensions are [1]

Length of the Ground plane $Lg \geq 6h+L$ (6)

Width of the Ground plane $Wg \geq 6h+W$ (7)

The co-ordinates of feed point location with origin at middle of patch are given as

$$Y_f = \frac{W}{2}; \text{ and } X_f = X_0 - \Delta L \quad (8)$$

$$X_0 = \frac{L}{\pi}\cos^{-1}\sqrt{\frac{50}{Z0}};\ Z_0 = \sqrt{50 * Zin} \text{ and} \quad (9)$$

$$Zin = 90 * \frac{\epsilon r^2}{\epsilon r-1}\left(\frac{L}{W}\right)^2 \quad (10)$$

These equations give an approximate X and Y co-ordinates of feed point. The exact location of feed point can be obtained after iterative trial process for better impedancematching.

The Ground plane Length Lg and Width Wg are taken as 6cm and 8cm respectively for resonant frequency 2. 5GHz and as 19cm and 21cm respectively at 1GHz For the analysis the two substrates used are Retardant-4(FR4) with εr 4. 4 and RT/Duroid 5880 with εr 2. 2 are used.

The dimensions of the patch antennas for different cases are given in Tables I to IV. The results of parameters, Return loss, VSWR, Peak Gain, Directivity, 3dB Bandwidth and Radiation Efficiency are evaluated for patch with substrates FR4 and RT duroid and are given in tables V to VIII The results for patch with combined material substrate are given in Tables IX and X..

III. Results:

The performance evaluation parameters estimated through simulations using HFSS software are given below. The results are discussed in next section. In all the tables h is substrate thickness in cm.

Table I

Retardant-4(FR4) f_r = 2. 5GHz			
Substrate Thickness in cm	Width of the patch in Cm	Length of the patch in cm	Feed point Location in cm
0. 05	3. 6	2. 84	0. 83
0. 15	3. 6	2. 82	0. 7
0. 25	3. 6	2. 78	0. 7
0. 32	3. 6	2. 73	0. 7
0. 48	3. 6	2. 6	0. 7
0. 6	3. 6	2. 56	0. 7

Table II

RT/Duroid5880 f_r= 2. 5GHz			
Substrate Thickness cm	Width of the patch cm	Length of the patch cm	Feed point Location cm
0. 05	4. 734	3. 96	0. 71
0. 15	4. 734	3. 93	0. 71
0. 25	4. 734	3. 9	0. 71
0. 32	4. 734	3. 8	0. 94
0. 48	4. 734	3. 73	1
0. 6	4. 734	3. 63	1. 2

Table III

Retardant-4(FR4) f_r = 1GHz			
Substrate Thickness cm	Width of the patch cm	Length of the patch cm	Feed point Location cm
0. 09	9. 1	7. 1	2. 3
0. 372	9. 1	7	1. 72
0. 654	9. 1	6. 9	1. 72
0. 934	9. 1	6. 7	1. 72
1. 21	9. 1	6. 6	2. 3
1. 5	9. 1	6. 3	2. 3

Table IV

RT/Duroid5880 f_r= 1GHz			
Substrate Thickness in cm	Width of the patch in Cm	Length of the patch in cm	Feed point Location in cm
0. 09	11. 9	10	2. 2
0. 372	11. 9	9. 95	2. 3
0. 654	11. 9	9. 72	2. 0
0. 934	11. 9	9. 6	2. 5
1. 21	11. 9	9. 1	3. 5
1. 5	11. 9	8. 7	3. 5

Table V

RT/Duroid5880 f_r= 2. 5GHz						
h Cm	0. 05	0. 15	0. 25	0. 32	0. 48	0. 6
Return Losses dB	-32. 7	-24. 5	-23. 8	--25. 6	-27. 3	-23. 4
Gain dB	5. 44	5. 74	5. 57	5. 01	5. 5	5. 435
VSWR	1. 047	1. 13	1. 239	1. 2	1. 09	1. 144
3dB Bandwidth MHz	3	15	17	24	28. 5	37
Peak Directivity	5. 89	5. 87	5. 66	5. 05	5. 55	5. 4
Radiation Efficiency	0. 9	0. 97	0. 98	0. 986	0. 989	0. 99
Resonant frequency GHz	2. 49	2. 46	2. 41	2. 45	2. 47	2. 5

Table VI

Retardant-4(FR4) f_r = 2. 5GHz						
h Cm	0. 05	0. 15	0. 25	0. 32	0. 48	0. 6
Return Losses dB	-15. 5	-34. 6	-20	-22. 1	-17. 0	-17. 2
Gain dB	1. 36	1. 87	2. 91	3. 26	3. 64	3. 61
VSWR	1. 4	1. 03	1. 21	1. 16	1. 04	1. 31
3dB Bandwidth MHz	20	25. 1	30	45. 9	61. 9	77. 4
Peak Directivity	4. 79	4. 82	4. 79	4. 75	4. 62	4. 38
Radiation Efficiency	0. 2	0. 52	0. 67	0. 76	0. 81	0. 82
Resonant frequency GHz	2. 48	2. 43	2. 41	2. 5	2. 46	2. 44

Table VII

RT/Duroid5880 f_r= 1GHz						
h Cm	0. 09	0. 372	0. 654	0. 934	1. 21	1. 5
Return Losses dB	-12. 7	-18	-20. 2	-20. 36	-16. 4	-10. 3
Gain dB	5. 3	5. 6	5. 62	5. 49	5. 5	5. 4
VSWR	1. 6	1. 28	1. 15	1. 21	1. 35	1. 9
3dB Bandwidth MHz	4. 7	7. 6	11. 6	15. 5	31	82. 4
Peak Directivity	5. 88	5. 78	5. 69.	5. 55	5. 56	5. 45
Radiation Efficiency	0. 88	0. 971	0. 984	0. 9883	0. 989	0. 992
Resonant frequency GHz	0. 982	0. 975	1	0. 975	1. 03	1. 01

Table VIII

Retardant-4(FR4) f_r = 1GHz						
h Cm	0. 09	0. 372	0. 654	0. 934	1. 21	1. 5
Return Losses dB	-17. 1	-28. 22	-17. 8	-19	-21. 4	-18. 4
Gain dB	2. 05	2. 06	3. 12	3. 54	3. 76	3. 81
VSWR	1. 33	1. 07	1. 03	1. 02	1. 18	1. 26
3dB Bandwidth MHz	6. 7	12. 4	14	17. 8	19. 3	29
Peak Directivity	4. 91	4. 84	4. 84	4. 73	4. 67	4. 71
Radiation Efficiency	0. 20	0. 52	0. 67	0. 76	0. 81	0. 813
Resonant frequency GHz	0. 99	0. 988	0. 97	0. 98	0. 98	1. 02

Table IX-substrate arrangement ground plate, RT/ Duroid5880, Fr4.

h Cm	RT/Duroid 5880 (0. 48 Cm)	RT/Duroid 5880 (0. 24 Cm) +FR4 (0. 24Cm)	FR4 0. 48cms
Return Losses	-27. 3	-23. 4	-17. 0
Gain in dB	5. 5	5. 03	3. 64
VSWR	1. 09	1. 14	1. 04
3dB Bandwidth in MHz	28. 5	30. 6	61. 9
Peak Directivity	5. 55	5. 53	4. 62
Radiation Efficiency	0. 98	0. 89	0. 81
Resonant Frequency	2. 47	2. 44	2. 46

Table X-substrate arrangement ground plate,FR4, RT/Duroid 5880.

h Cm	RT/Duroid 5880 (0. 48 Cm)	FR4 (0. 24 Cm) + RT/Duroid (0. 24 Cm)	FR4 0. 48cms
Return Losses	-27. 3	-21. 63	-17. 0
Gain in dB	5. 5	5. 36	3. 64
VSWR	1. 09	1. 18	1. 04
3dB Bandwidth in MHz	28. 5	44. 1	61. 9
Peak Directivity	5. 55	5. 68	4. 62
Radiation Efficiency	0. 98	0. 93	0. 81
Resonant Frequency	2. 47	2. 47	2. 46

IV Discussion:

The results obtained in present work indicate that with increase in substrate thickness impedance bandwidth increases which is in line with literature findings[1-2]. Furtherresults obtained by the authors for RT/ Duroid 5880 with thickness 1. 5 cm are in good agreement with those reported by[4] for 1. 589cms thickness.

At resonant frequency of 1GHz, as thickness of substrate is varied from 0. 09 cm to 1. 5 cm; in case of RT/ Duroid the 3dB bandwidth increases from 4. 7 MHZ to 82. 4 MHz, a sixteen fold increase but little variation in gain. In case of Retardant-4(FR4) substrate the bandwidth increase is only four fold 6. 7 MHZ to 29 MHz and gain varies from 2dB to 3. 8 dB. Regarding the VSWR it is initially decreasing and then showing increasing trend for both materials. Similar trend is observed with these materials at resonant frequency of 2. 5 GHz also. At both the frequencies RT/Duroid 5880 is showing better directivity than Retardant-4(FR4).

With regard to the radiation efficiency, with increase in substrate thickness the Efficiency is increasing in case of both materials However RT Duroid 5880 is showing better efficiency of around 90% for all thickness values while with FR4 the efficiency has wide variation from 20% to 82%.

With regard to RT/Duroid 5880 at 1 GHz, better overall performance is observed with substrate thickness of 0. 654 cm resulting in lowest value of 1. 15 for VSWR, gain of 5. 62dB, bandwidth of 11. 6MHz and efficiency of 98. 4%. However at 2. 5 GHz the performance is better with substrate thickness of 0. 48 cm with VSWR of 1. 09, gain of 5. 5dB, bandwidth of 28. 5 MHz and efficiency of 98. 9%. Thus it can be observed that though there are variations in substrate thickness and bandwidth the values of gain and efficiency remain more or less same at both the frequencies.

From the results in tables IX and X for patch with substrate of combined materials the following observations can be made. Though the materials are same and thickness same for both cases the position of them do influence the parameters Higher gain and better bandwidth are obtained when high permittivity material is near to ground plane.

V. CONCLUSIONS

Few rectangular patch antennas with different dielectric strength substrates and with different thicknesses are designed and their performance is evaluated. RT /Duroid 5880 material with lower dielectric constant of 2. 2 is exhibiting improved performance over FR4 with dielectric constant 4. 4 at both frequencies. Using combination of dielectric materials for substrate the performance of the antenna is improved compared to that made from single material substrate.

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Study of Packet Loss Prediction using Machine Learning

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A B S T R A C T

The importance of providing guaranteed Quality of Service (QoS) cannot be overemphasised in the Next Generation Network (NGN) environment. NGNsupports converged services on a common IP transport network. Call Admission Control (CAC) mechanism provides QoS to class-based services in a proactive manner. We use Machine Learning (ML) techniques for providing autonomous CAC due to the factors of complexity, scale and dynamicity of NGN. Packet loss prediction is one of the metrics for provisioning QoS. This paper is an effort to measure the packet loss prediction using machine learning approach. Two ML based model are used to predict packet loss, Decision Tree model and Logistics Regression model. Performance measure is based on experiments and observations. The outcome of the comparative study states that Decision Tree model gives better result compared to the LogisticsRegression model for prediction of packet loss.

Keyword: QoS, NGN,CAC, Decision Tree, Logistics Regression, prediction, packet loss, machine learning

I. INTRODUCTION

QoS is a prime concern for both the service providers and subscribers. The problems of guaranteed QoS arise due to the advances in the network architectures, the demand for multimedia services and applications. IP-based NGN promises guaranteed QoS [1][4][5]. CAC is a major preventive technique to provide guaranteed QoS to various class-based services as recommended by the ITU-T for NGN [2]. Guaranteed QoS was first witnessed in the ATM networks supporting class-based services [3]. CAC mechanism plays a proactive role in providing QoS by limiting the entry of traffic at the edges of the network. Its job is to make a decision to admit or deny a new call into the network based on the condition that the QoS of the existing calls and also the new calls are satisfied. CAC approach becomes difficult and intractable to solve through conventional analytical methods due to growing of number of services, their classes and size of the network [6][7]. We use Machine Learning (ML) approach to solve traditional CAC approach. ML helps the system behavior through the process of learning which is based on observation of performance data over a period of times [8].Once appropriately trained, they are able to estimate and predict future system behavior and subsequently make admission decisions with high accuracy and speed. ML approaches are applied in the telecommunications domain to solve various network management problems [9]. Neural network (NN) offers approach learning for traffic and service quality [10]. Combination of Particle Swarm Optimization and Fuzzy logic for next generation mobile communication networks provide better CAC scheme [12]. Support Vector

Machine (SVM) based CAC algorithm utilizes service vector and network vector to predict admission state for admission decisions [13]. This scheme accelerates calculation speed with lower call delay. Lower call blocking probability and call dropping probability is also achieved here. Bayesian network (BN) based CAC framework implements delay prediction based on call admission decisions [14]. There occurs a comparison between NN and BN for response time modeling in service-oriented systems [15]. Packet loss prediction is one of the metrics for provisioning QoS. Our paper is an effort to measure the packet loss prediction using machine learning. Two ML based model are used to predict packet loss, Decision Tree model and Logistics Regression model. Performance measure is based on experiments and observations. The outcome of the comparative study provides some interesting insights into the behaviour of Decision Tree model and Logistics Regression model for prediction of packet loss. The rest of this paper is structured as follows. In section II we present the details of machine learning model. Section IIIprovides software used with variables. Simulated results with graphical representations are provided in section IV. Section V concludes the paper by suggesting possible future works.

II. MACHINE LEARNING MODEL

CAC system generally resides on an edge router, whose function is to allow controlled traffic into the core network. A generic CAC framework based on the ML approach is shown in figure 1. The input to such a system consists of customer call requests and the output is a decision to either admit or deny the call. The call request consists of traffic descriptors and desired QoS in the network. Traffic descriptors include parameters like peak rate, average rate, maximum burst duration and type of application which are supplied by the caller. QoS requirements include some measure of metrics like packet loss, average delay or delay variation (jitter). Available link bandwidth and buffer occupancy are also inputs to CAC. Based on the choice of inputs and outputs, the ML module is trained offline with a set of data which is observed in the system over a period of times. The training data set consists of cases where both the inputs and outputs are known. However, when the trained model is in the online mode, it provides the estimate of the output for a particular input combination. It is clear that the overall CAC performance is dependent on the prediction accuracy of the model. Prediction accuracy depends on how well the model estimates the unknown output when presented with unseen cases not present in the training set. In our paper, we use two ML based model to predict packet loss, Decision Tree model and Logistics Regression model. A decision tree is a decision support tool which provides decisions and their possible consequences. Logistic regression is the appropriate regression analysis to conduct when the dependent variable is binary. The logistic regression is a predictive analysis. Logistic regression is used to describe data and to explain the relationship between one dependent binary variable and one or more nominal, ordinal, interval or ratio-level independent variable.

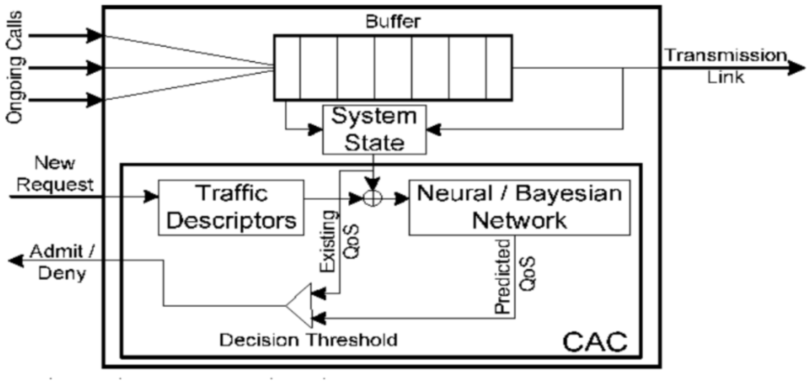


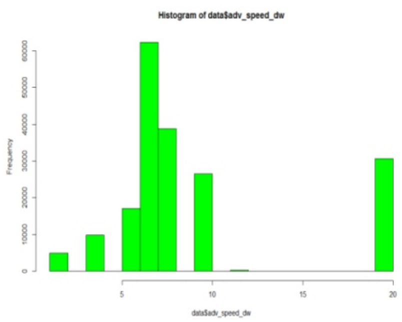
Figure 1: Machine Learning based CAC

III. SOFTWARE USED WITH VARIABLES

Here, the environment is R Studio. We use R language. We collect the data set from internet. We get total number of observations 190166 from 12 variables. Continuous variables are ad_speed_dw, adv_speed_up, latency, jitter, downstream_throughput, upstream_throughput. Categorical variables are packet_loss, technology, isp, id, postcode, timestamp.

IV. SIMULATED RESULTS WITH GRAPHICAL REPRESENTATIONS

It is clear from the figure 2, figure 3 and figure 4 that the medians of "adv_speed_dw" in two categories of "packet_loss" are significantly different. It means the variable "adv_speed_dw" has a role in deciding "packet_loss".



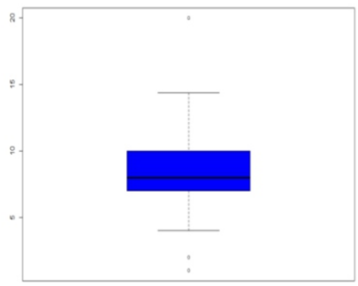


Fig 2. hist(data$adv_speed_dw, col='green') Fig 3. boxplot(data$adv_speed_dw, col = "blue")

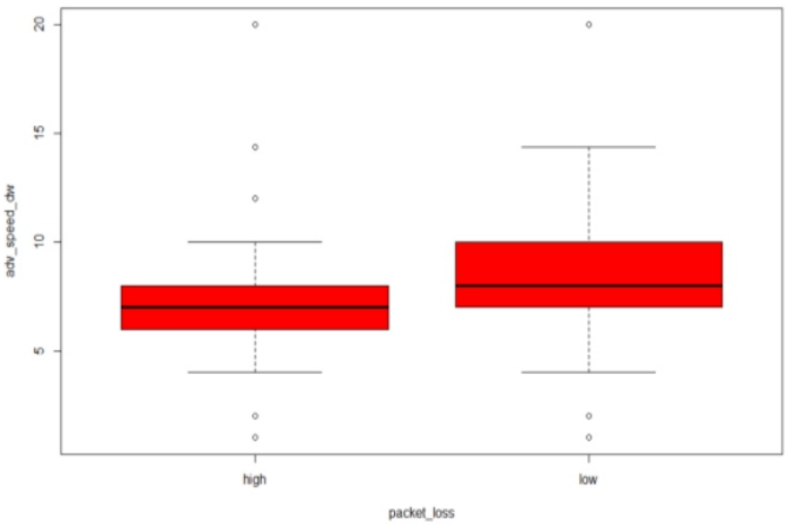
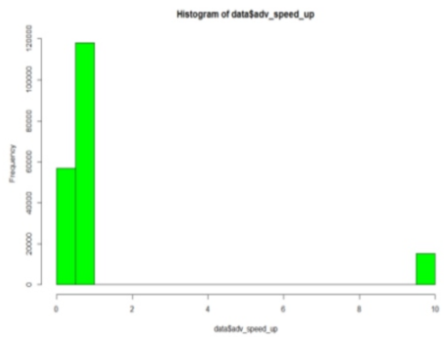


Fig 4. boxplot(adv_speed_dw~packet_loss, data = data, col= "red")

It is clear from the figure 5, figure 6 and figure 7 that the medians of "adv_speed_up" in two categories of "packet_loss" are not different. It means the variable "adv_speed_up" has no role in deciding "packet_loss".



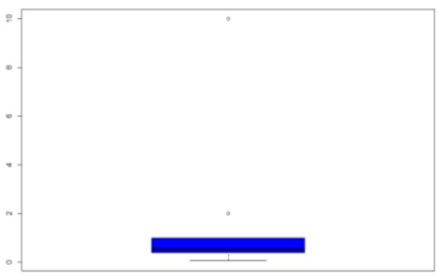


Fig 5. hist(data$adv_speed_up, col='green') Fig 6. boxplot(data$adv_speed_up, col = "blue")

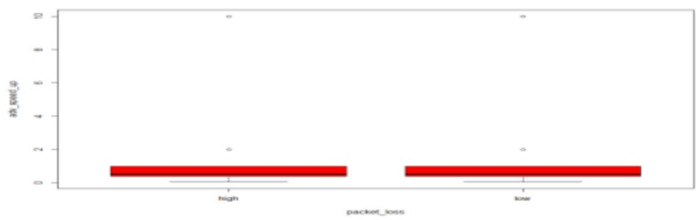


Fig 7. boxplot(adv_speed_up~packet_loss, data = data, col="red"

For the variable latency, it is difficult to visualize difference in median in two categories due to presence of extreme values which are shown in figure 8, figure 9and figure 10.

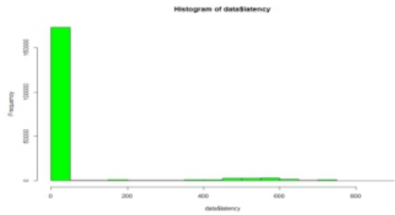


Fig 8. hist(data$latency, col='green')

Fig 9.boxplot(data$latency, col = "blue")

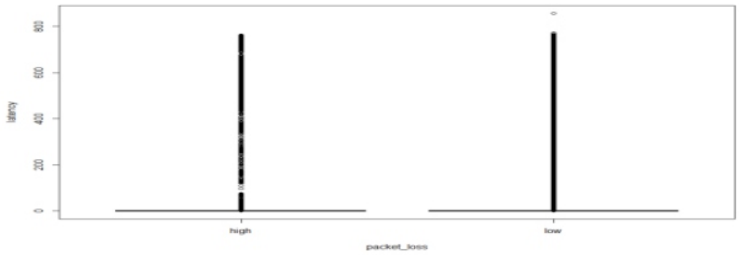


Fig 10. boxplot(latency~packet_loss, data = data, col= "blue")

For the variable jitter, it is difficult to visualize difference in median in two categories due to presence of extreme values which are shown in figure 11, figure 12 and figure 13. We observe that variable 'jitter' has a significance value more than 0.05. So we can confirm that this variable does not have any effect in deciding the 'packet_loss'.

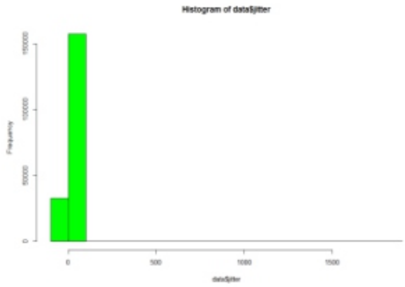


Fig 11.hist(data$jitter, col='green')

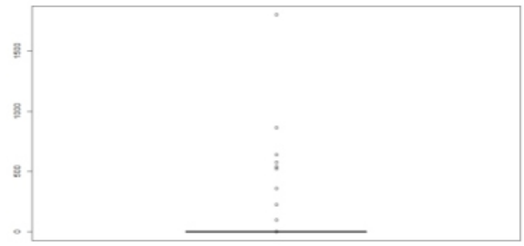


Fig12. boxplot(data$jitter,col="blue")

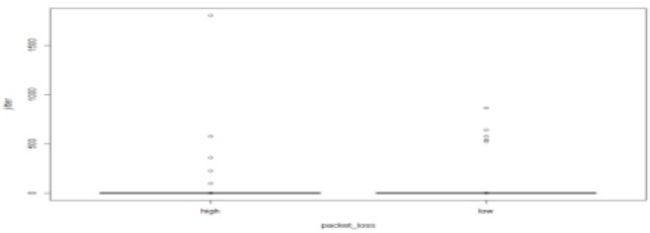
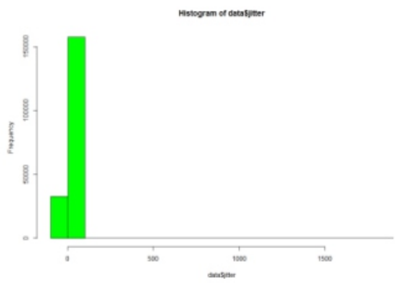


Fig 13. boxplot(jitter~packet_loss, data = data, col= "blue")

For the variable jitter, it is difficult to visualize difference in median in two categories due to presence of extreme values which are shown in figure 11, figure 12 and figure 13. We observe that variable 'jitter'

has a significance value more than 0.05. So we can confirm that this variable does not have any effect in deciding the 'packet_loss'.



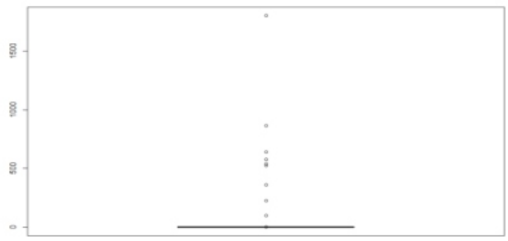


Fig 11.hist(data$jitter, col='green') *Fig12. boxplot(data$jitter,col="blue")*

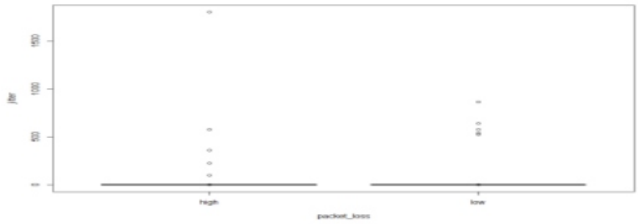


Fig 13. boxplot(jitter~packet_loss, data = data, col= "blue")

It is clear from the figure 14, figure 15 and figure 16 that the medians of "downstream_throughput" in two categories of "packet_loss" are significantly different. It means the variable "downstream_throughput" has a role in deciding "packet_loss".

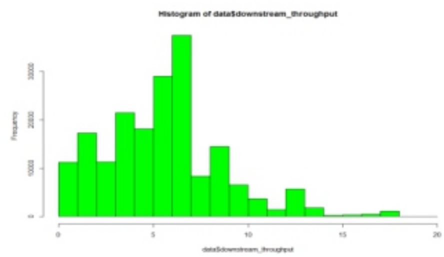


Fig 14. hist(data$downstream_throughput col='green')

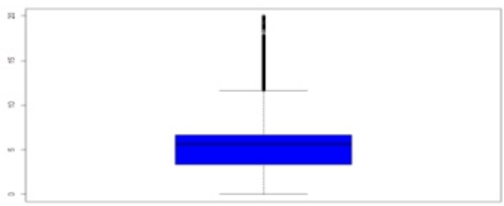


Fig 15.boxplot(data$downstream_throughput, col = "blue")

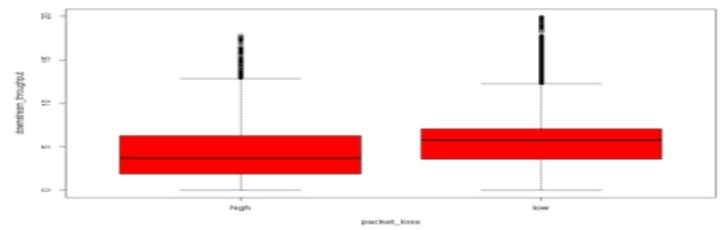


Fig 16.boxplot(downstream_throughput~packet_loss,data = data, col= "red")

It is clear from the figure 17, figure 18 and figure 19 that the medians of "upstream_throughput" in two categories of "packet_loss" are slightly different. It means the variable "upstream_throughput" has some role in deciding "packet_loss

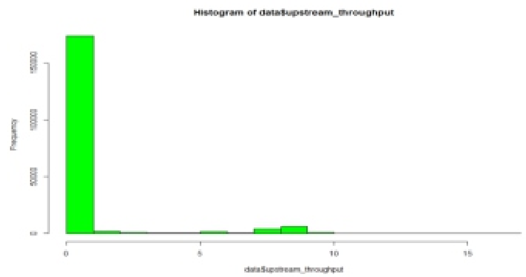


Fig 17. hist(data$ upstream _throughput col='green')

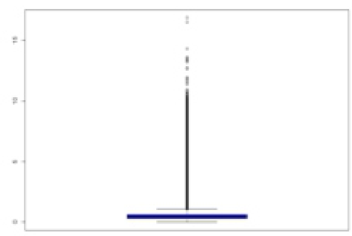


Fig 18.boxplot(data$ upstream _throughput, col = "blue")

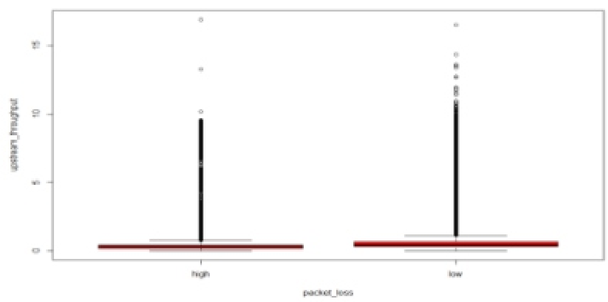


Fig 19. boxplot(upstream_throughput~packet_loss, data = data, col= "red")

Categorical variables packet_loss is shown in figure 20.

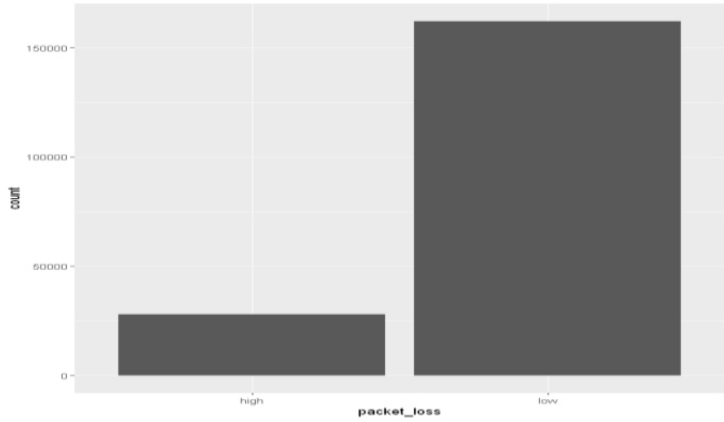


Fig 20.variable "packet_loss" ggplot(data, aes(x = packet_loss)) + geom_bar()

We consider the variable "technology" having highest number of occurrence among all. They are depicted in figure 21 and figure 22.

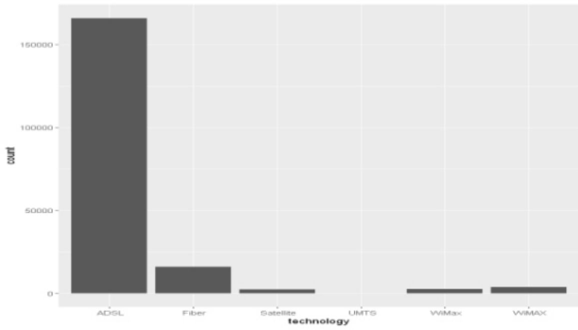


Fig 21. ggplot(data, aes(x = technology)) + geom_bar()

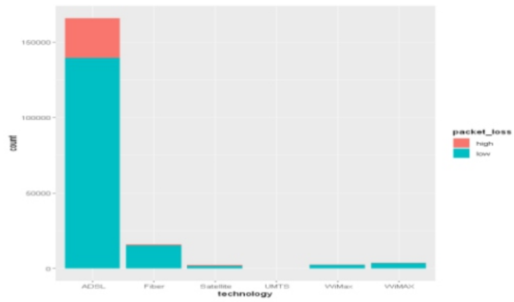


Fig 22. ggplot(data, aes(x = technology, fill = packet_loss)) + geom_bar()

We consider the variable "isp" having highest number of occurrence among all. They are depicted in figure 23 and figure 24.

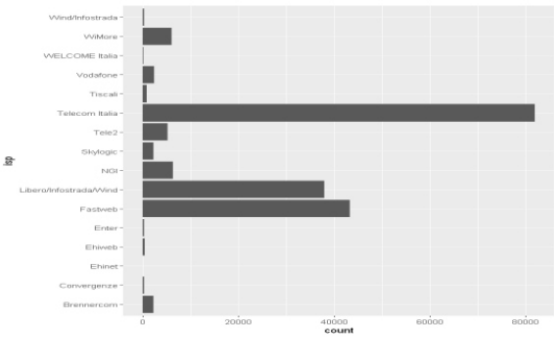


Fig 23. ggplot(data, aes(x = isp)) + geom_bar()+coord_flip()

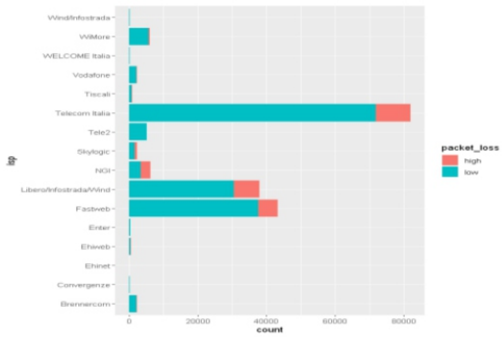


Fig 24. ggplot(data, aes(x = isp, fill = packet_loss)) + geom_bar()+coord_flip()

We drop 3 categorical variables "postcode", "timestamp" and "id" because postcode have missing values and timestamp have not appropriate format for taking data. id are not important variable for measuring packet loss.

We split the dataset in ratio 80:20 as train and test sets and check missing values. We get from observations over a period of times that there is no missing value. These results are obtained using the process of 10-fold cross validation, where the collected data is partitioned into training and test sets and the prediction accuracy averaged over 10 such iterations. Table 1 shows the parameters and their values. Table 2 shows the observations from prediction of LR model and table 3 shows the observation from Decision Trees model. We observe that LR model gives 85% accuracy on packet loss prediction and mis-classification error is 0.15. We also observe that DT model gives 89 % accuracy on packet loss prediction.

Table-1: Parameters and their values

Parameter	Value
Flow generation rate (sec)	5
Average duration (sec)	2.0
Packet generation rate (packets/sec)	Exponential (4)
Packet size (bits)	Exponential (1024)
Type of service	Best Effort

Table 2: Observations from prediction of LR model

Observations get from prediction of LR model		Confusion matrix	
Total dataset dimensions	190166 9	high	291 5264
Training dataset dimensions	152346 9	low	316 31949
Test dataset dimensions	37820 9		
High	0. 7784285		
Low	0.8653945		

Table 3: Observation from Decision Trees model

Observations get from Decision Trees model	
High	3166 1512
Low	2389 30753
Output: 0.8968535	

V. CONCLUSIONS WITH FUTURE WORKS

In our paper we go through all the experiments and observations. We conclude that Decision Tree model gives 89% accuracy on packet loss prediction and Logistic Regression (LR) model gives 85% accuracy on packet loss prediction. So it is clear from comparative study that Decision Tree model gives better result compared to the Logistics Regression model. In future works we will try to get 100% accuracy on packet loss prediction using Machine Learning.

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