Degradation of QoS of VoIP Traffic Through WiFi, WiMAX and WiFi-WiMAX Networks

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***Abstract*— Identifying those causes and parameters that affect the QoS of VoIP through WiFi, WiMAX and together are carried out using OPNET. Network optimization to mitigate the deterioration of the QoS are discussed. The average jitter of VoIP transiting the WiFi-WiMAX network was found to be higher than that of WiFi at some instances. It was expected to be less than traversing through the WiFi network only and higher than passing through WiMAX. Both the MOS and the packet end-to-end delay were found to be higher than expected. Optimization leading to a solution which can ameliorate the QoS are discussed.**

***Index Terms*—MOS; QoS; VoIP; WiFi; WiMAX.**

I. Introduction

Due to the enormous and ever increasing popularity of using the Internet, especially on mobile devices, it is becoming increasingly difficult to ignore the importance of voice chat and making calls using the Internet in our day-to-day life. Because there exists different types of networks (i.e. WiMAX, WiFi, 4G, LTE, 3G, CDMA, GSM, GPRS, EDGE etc.), in most cases the data has to traverse multiple networks. From the users’ experience, it has been observed that when VoIP (Voice over Internet Protocol) traffic travels from one type of network to another, the quality of it degrades. The main aim of the research is to find out to what extent the QoS (Quality of Service) of the VoIP traffic degrades while traveling through multiple networks. To achieve this aim, our objectives include: 1. Designing different network scenarios on Opnet modeler; 2. To simulate them to obtain the results and 3. Finally analyzing the results obtained through the simulation. The first two scenarios consist of several VoIP clients exchanging data through the same type of networks i.e. WiMAX-to-WiMAX, WiFi-to-WiFi. The QoS components of VoIP traffic such as Mean Opinion Score (MOS), Throughput, Availability, Cross-talk, Distortion, Attenuation, Link Utilization Distribution, Jitter, Loss and Echo etc. will be examined and analysed. In the third scenario, instead of passing through the same network, the VoIP traffic will be travelling through heterogeneous networks i.e. WiMAX-to-WiFi. The same parameters will then be analysed and compared with the results previously obtained from the homogeneous scenarios.

The paper has been organized according to the following parts: Section-1 provides a brief introduction to the research. Section-2 describes the background information and technological terms used in this paper. Section-3 comprises the background “Literature Review” covering a wide range of research articles and projects. Section-4 contains the research methodology including the simulation scenarios and required configuration to achieve them. Section-5 analyses the results which is then followed by the conclusion and outlines for future research directions.

II. Background Terminology

A. *VoIP (Voice over Internet Protocol)*

Voice over Internet Protocol or VoIP [1] is the transmission of digitized voice traffic over a data network to make telephone calls, This differs from using a traditional, analogue circuit switched public network as now the data has been split into packets. These packets can take any route to reach the destination. Packetised data travel through a virtual circuit which differs from a circuit switched network in that the circuit does not need to be reserved between the sender and the receiver with packet switching. This efficient use of channels also means that the packet can arrive out of sequence, experience delay or never arrive at all due to severe traffic congestion. This is one of the major disadvantages of sharing traffic across a virtual network that VoIP traffic has to contend with. Multiple routing of the VoIP traffic does ensure a cheaper and often free of cost flow of traffic between the different intra-packet network components such as the routers and switches. Sending digital data as packets mean that all types of digitized data such as voice, video, fax, music and telephony can now be transmitted over a common shared network.

VoIP has the clear advantage of inexpensive scalability compared to other telephony systems, being software packet based. This allows lines to be shared with other users and services thus helping to lower costs over circuit switched networks. However, being a mainly software based network - it is vulnerable to the increasingly growing threat of attacks from hackers in terms of worms and viruses. [2] discusses some security solutions to address this potential problem.

Convergence has been accelerated with the deployment of 3G [3], especially between the internet, fixed and mobile services. Global access to the internet regardless of the means of transportation is becoming more prevalent, especially with the spread of WiFi, WIMAX and femtocells. Demands for greater bandwidths to support multimedia broadband access is also increasing and being expected by the consumers. This was facilitated by the adoption of the IP Multimedia Subsystem (IMS) in the Rel. 5 version of UMTS (Universal Mobile Telecommunications System). The

IMS is a packet based control overlay network used for transporting user data and signaling.

The SIP (Session Initiation Protocol) - a development of the IETF (Internet Engineering Task Force) was adopted by the 3GPP (Third Generation Partnership Project) for setting up IP-based multimedia sessions, this includes VoIP. The current IEEE 802.16 WiMAX and IEEE 802.11 WiFi networks fully support real-time services such as VoIP [4].

B. SIP (Session Initiation Protocol)

Making, maintaining and clearing a call requires signaling and control information to be exchanged between the network entities. This is actually a rather complicated process where internet mobility is involved across various types of devices with differing capabilities and network technologies. A protocol that has been chosen to manage these "sessions" is known appropriately as the "Session Initiation Protocol" or SIP [5]. SIP works alongside and in complement with the existing real-time protocols. The source and destination endpoints known as "user agents" discover each other and negotiate the parameters for efficient exchange of information by the use of SIP. The necessary user agents and intermediary nodes are handled by SIP by the creation of proxy servers. These proxy servers can then request and respond to invitation, registration and other requests. SIP is a transport protocol independent of the type of session being established. SIP is designed to be agile, flexible and to handle various types of multimedia data exchange.

SIP being an application layer control protocol can take care of the entire multimedia call set-up to termination process. It also includes the ability to handle multicast call set-up including the removal of the participant. SIP is designed for mobility with features such as redirection and name mapping. A powerful feature of SIP is the ability to maintain an externally visible identifier invariant of location [6]. For example, SIP supports these call set-up features: session set-up, user availability, user location, user capabilities and session management.

C. QoS Parameters:

The data networks being flexible in its ability to handle multifarious types of data services over the Public Switched Telephone Network (PSTN) puts the Plain Old Telephone Service (POTS) at a financial disadvantage [4]. The QoS for VoIP can be measured using several different types of metric, such as jitter, the end-to-end delay and the Mean Opinion Score (MOS), as shown in Table 1.

Jitter "is the variation in arrival time of consecutive packets" [10]. Jitter is calculated over a period of time [7]. It should be noted that the buffers can both under-fill and over-fill resulting in packets being dropped.

TABLE 1.
MEAN OPINION SCORE (MOS) [7].

Quality Scale	Score	Listening Effort Scale
Excellent	5	No effort required
Good	4	No appreciable effort required
Fair	3	Moderate effort required
Poor	2	Considerable effort required
Bad	1	No meaning understood with effort

Packet end-to-end delay "is measured by calculating the delay from the speaker to the receiver [including the] compression and decompression delays" [8].

The International Telecommunication Union – Telecommunication (ITU-T) gives the guidelines for the delay and jitter for different types of call quality, as shown in Table 2 [10].

TABLE 2.
GUIDELINE FOR THE VOICE QUALITY [8].

Network Parameter	Good	Acceptable	Poor
Delay (ms)	0-150	150–300	> 300
Jitter (ms)	0-20	20–50	> 50

D. WiFi™ (IEEE 802.11x)

The contention wireless networking technology, WiFi, evolved from the non-wireless IEEE Ethernet 802.3 Local Area Network (LAN) technology to become the IEEE 802.11 Wireless LAN or WLAN. The physical and data link layers are defined, operating over two frequency bands of 5 GHz and 2 GHz. Two popular WiFi standards are the 802.11b (11 Mbit/s) and 802.11g (54 Mbit/s) with an operating range of between 80-100 m. Being a contention based system the speeds quoted are a theoretical maximum. The contention causes the comparatively low bitrates and thus affects the QoS, especially for real-time services like VoIP. This is not helped by the large headers of the WiFi and VoIP protocols themselves. Its uptake and popularity has been due to the inexpensive price of the router and most network equipment coming with it built-in, including the WiFi antenna. WiFi has now become widespread covering: domestic, industrial, public spaces including on public transportation [9].

E. WiMAX™ (Worldwide Interoperability for Microwave Access) Technology

WiMAX, when it was first introduced ten years ago was meant to provide a global wireless high speed mobile Internet access. However, LTE (Long Term Evolution) has largely superseded this application. WiMAX, however, is not dead and there are around 580 operators in the world providing backhaul and rural access to fast broadband internet access, often in the less developed regions of the world. Typical application scenarios of WiMAX are shown in Fig. 1. WiMAX was designed to provide the same experience as that of fixed internet services, such as QoS, Service Level Agreement (SLA), interoperability with off course mobility, wide coverage and security [10]. It is ironic that WiMAX, once touted as the "4G of Wireless Technology" has now been superseded ahead of its time by LTE. WiMAX is still probably the first all IP mobile internet technology allowing true scalability to carry multimedia traffic [11]. WiMAX provides a coverage area of 50 km with data rates of 75 Mbps [12].

WiMAX comes in two types of technologies: the fixed IEEE 802.16/a/d version and the wireless IEEE 802.16-2005 (16e) amendment [13]. The latest version is known as WiMAX rel 2 or IEEE 802.16m. The latest version allows download bitrates up to 1 Gbit/s through channel aggregation for low mobility users.

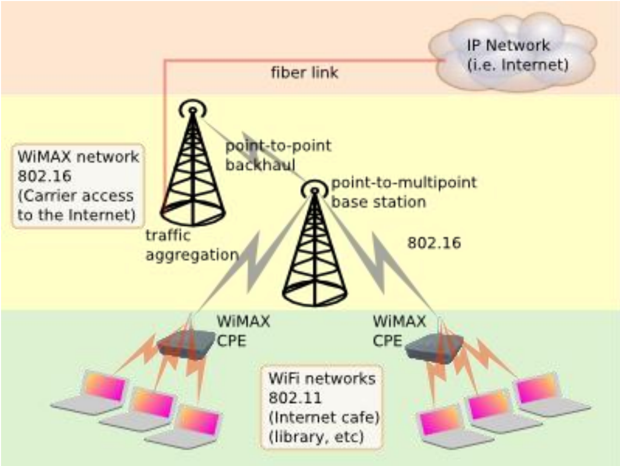

Figure 1. Application scenarios of WiMAX. (From: http://www.accessmillennium.com/images/wifi_vs_wimax.png)

III. LITERATURE REVIEW

A previous simulation study of VoIP over both WiFi and WiMAX [9] has shown that VoIP activity does impact negatively on the overall throughput of both technologies. However, only in the WiFi network is packet loss and jitter experienced. The parameters commonly used to study the performance of the network, for example a study of WiMAX and UMTS using the OPNET network simulation software include: “MOS, end-to-end delay, jitter, and packet delay variation” [7].

It would appear that not all software implementations of VoIP clients are equal as they vary in their effect on voice quality on a study performed over High Speed Packet Access (HSPA) [14].

To overcome the severe problems of VoIP calls over WiFi when approaching WiFi capacity and congestion, a new scheme, SQoSMA (Quality Assurance of Voice over WLANs) [14] was proposed. SQoSMA took the approach of combining the control and data and planes in order to detect and mitigate congestion events. This was achieved by choosing the appropriate adaptive audio codec with the suitable bitrate and then implementing a call stopping procedure where necessary to fix congestions.

An earlier similar scheme [15] was also explained with the use of edge VoIP gateway between the WLAN and the Internet Cloud. The task of the edge VoIP gateway was to select the appropriate variable speech coding rate (64, 40, 32, 24 and 16 Kbit/s) to lessen network congestion with a subsequent increase in the overall QoS of speech traffic.

A technique that reduced the transmission delay of VoIP traffic through a Transmission Control Protocol (TCP) - Friendly Rate Control (TFRC) algorithm based 802.11e network used the EDCF (Enhanced Distributed Coordination Function)/HCF (Hybrid Coordination Function) scheme [15].

[16] proposed using a routing and label based solution for transporting real-time VoIP traffic through WLAN which efficiently processed the procedures of: call QoS, mobility and call admission. Their procedure utilized a 15 node wireless mesh network to implement distributive packet aggregation utilizing MAC waiting without unbounded packet delays. The fully optimized procedure resulted in a performance gain of 13 times for six hops.

Since human voice is assessed by humans and is therefore purely subjective, a metric to assess this for VoIP traffic is needed that takes into account human subjectivity - which is lacking in the purely objective SNR (signal to noise) measure. A study [17] in this field was conducted to look at such metrics as the E-Model and the PESQ (Perceptual Evaluation of Speech Quality). The researchers studied the limitations of both measures and devised a new metric which combined the advantages of both, known as AdmPESQ (Advanced Model for Perceptual Evaluation of Speech Quality). AdmPESQ is especially applicable for heterogeneous networks with differing delay parameters and packet losses.

Voice over Internet Protocol (VoIP) has become increasingly popular over the last few years. It is now commonly used by a wide range of population from all over the world. VoIP, with lower call rates, provides most of the facilities of the traditional Public Switched Telephone Network (PSTN) and incorporates some more value added features as well. As a result, there exists many companies providing VoIP services and hence the traffic has to run through different types of networks - often heterogeneous in nature. It has been observed through user experience that the Quality of Service (QoS) degrades while the traffic has to traverse through multiple networks. Materna [18] has rightly pointed out the four primary categories of VoIP attacks, namely:

1. Service Availability;
2. Service Integrity;
3. SPIT (Spam over Internet Telephony) and
4. Eavesdropping.

The successful availability without network outage is vital for the success on any well networked and connected corporation. Thus protection against any forms of “Service Availability Attacks” is of paramount importance. Downtime in the telephony network will mean: lost revenues for the enterprise and the service providers, unplanned maintenance costs and lost productivity. The IP Telephony network must be protected against all known forms of attacks which include: viruses, worms and specially the “Denial of Service” (DOS) types. The effects of these may range from the degradation of the QoS to the total loss of service. Degradation of the QoS is not just a minor nuisance but actually of a major concern as customers often request the highest voice quality when they subscribe to an IP Telephony service.

The effect of such an attack on VoIP is actually more sensitive and harmful as it has a lower threshold and immunity than computer data networks. Computer data networks are protected more securely and are usually affected well after the VoIP network. Thus a generic worm may adversely affect the VoIP network precisely because of these reasons, in advance of the computer network. The worm may at most, just slow down the computer data network. The worm may, however, totally bring the VoIP network down.

Our research will investigate to what extent the quality of service (QoS) of the VoIP traffic degrades while traversing multiple networks. To achieve this aim, we will design different network scenarios and simulate them on an Opnet modeler. The results obtained through the simulation will then be analyzed meticulously and reported and published in the literature.

IV. RESEARCH METHOD

Due to the financial constraints and equipment limitations, simulation of a sample network, especially in academic research, is very important in computer networking and telecommunication. Not only does it help to get the perspective view of a network, it also provides guidance for the future. Jack Burbank describes “Modeling and Simulation (M&S)” as an acute component in the “design, development and test and evaluation (T&E)” process. According to him, “It is almost always preferable to have insight into how a particular system, individual device, or algorithm will behave and perform in the real world prior to its actual development and deployment.” [19] The advantages of M&S take account of the capability of exercising scenarios and case-studies which are not easily achievable through any empirical methods such as: network scalability testing; the capacity to adapt models to test the systems’ sensitivity and to tune its performance [20] . In the case of two or more similar available technologies, it helps to compare and contrast in order to take deployment decisions. Our project make use of the OPNET Modeler because it integrates a wide range of technologies and protocols [21], as well as comprising a “development environment”. This helps to facilitate M&S for various types of networks and technologies including (but not limited to): VoIP, WiMAX, WiFi, 3G and LTE. Other networking technologies can be written in software or are available from third party sources.

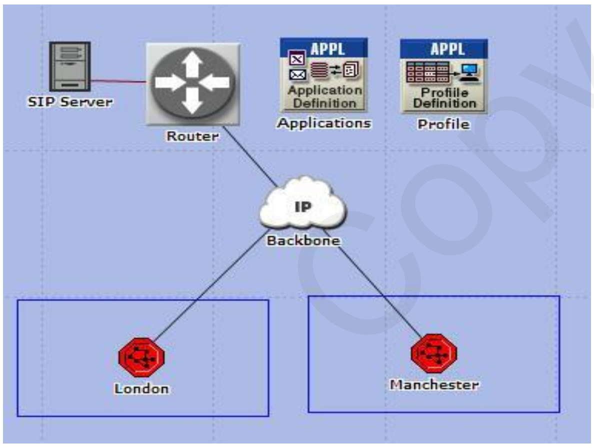

Figure 1: WiFi Network Scenario

In our first simulation scenario, two WiFi subnets, namely London and Manchester, were deployed. These are configured with a SIP server credentials connected through an IP cloud, as shown in Fig. 1.

In the second simulation scenario, two WiMAX subnets namely Cambridge and Bradford were used instead of the WiFi subnets. In the third scenario, one of the WiMAX subnet (Bradford) from the second scenario was replaced by a WiFi subnet (Manchester). The subnets implemented in the project are as follows:

TABLE 3.
DEVICES OF THE SUBNETS DEPLOYED.

Subnet Name	Scenario	Base Station Type	Work Station Type	Number of Work Stations
London	WiFi	Wifi	Mobile	4
Manchester	WiFi	Wifi	Mobile	4
Cambridge	WiMAX	WiMAX	WiMAX Workstation	4
Bradford	WiMAX	WiMAX	WiMAX Workstation	4
Manchester	WiMAX_ WiFi	WiFi	Mobile	4
Cambridge	WiMAX_ WiFi	WiMAX	WiMAX Workstation	4

The workstations in both of the WiMAX and WiFi networks models are configured to run the VoIP application. This VoIP Applications is defined to generate one voice frame per packet and to run as an ‘Interactive Voice’ service. This application is defined in the application profile to run in serial mode. Calls to workstations are based on random generation and are exponentially distributed with an average duration of three (3) minutes. Call inter-arrival time are exponentially distributed. In addition to the application profile and application configuration, the WiMAX network model contains a WiMAX profile. In this profile, a service class of ‘Gold’ with UGS (Uncorrelated Gaussian Sources) distribution for VoIP Application is created and this service is deployed and classified on all subscriber stations.

V. RESULTS ANS DISCUSSION

The average jitter graphs, as shown in Figs. 2a and 2b, were obtained from simulating all three scenarios for one hour. They revealed that WiMAX always has better performance over WiFi.

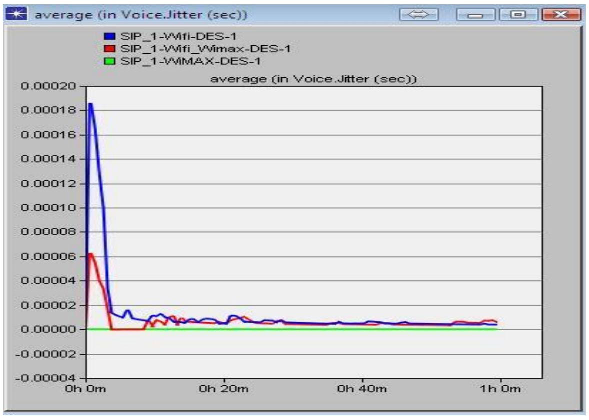

Figure 2a. Average VoIP Jitter (Overlaid). (Top curve is WiFi, middle curve is WiFi-WiMAX, bottom line is WiMAX)

WiFi also suffered from an extreme level of jitter during the first five minutes, this was likely due to the convergence period. Although WiMAX, on the other hand, suffered from a similar hike, it was much lower than that observed for WiFi.

The most interesting result we have found is that the average jitter of WiFi-WiMAX scenario, at some points, exceeds that of WiFi. It should ideally always remain somewhere in-between WiFi and WiMAX. Because the simulation was run on the basis of making random calls and there is no direct handover associated, this result is very intriguing. However, further research is required to find out the reason(s) behind such behaviour of the WiFi-WiMAX scenario.

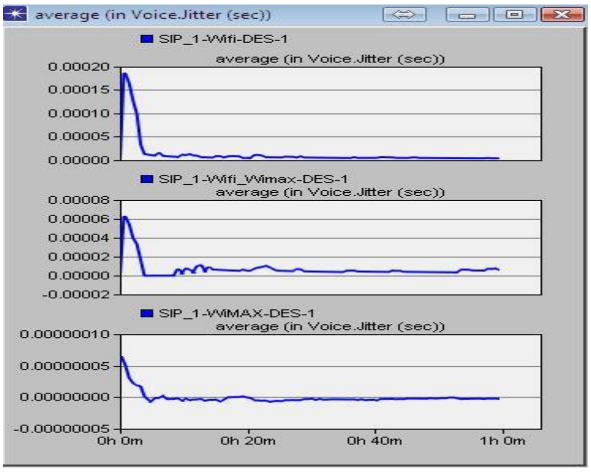

Figure 2b: Average VoIP Jitter (Top: WiFi; middle: WiFi-WiMAX; bottom: WiMAX)

In terms of the MOS, both WiMAX and WiFi-WiMAX observe similar levels of performance, as shown in Fig.3. Although call generation was exponentially distributed, the MOS performance of these two networks remains very steady over the whole simulation period.

On the other hand, although at the beginning of the simulation the WiFi network observes similar level of MOS. However, as time passes, with the increased level of VoIP traffic due to the higher number of calls generated, the MOS decreases. So, considering MOS, it can be concluded that: 1) both WiMAX and WiFi-WiMAX networks surpass the WiFi network and 2) although the MOS of WiFi-WiMAX network should ideally be somewhere near the mid-point of the WiFi and WiMAX MOS graphs, it observes much higher performance.

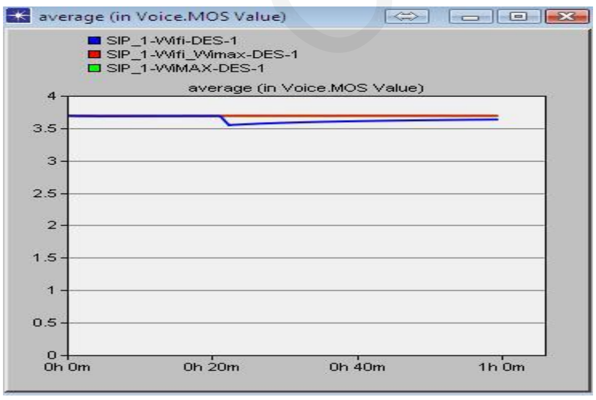

Figure 3: Average MOS (Overlaid) of 3.7. (Top line is WiFi, bottom line WiFi-WiMAX)

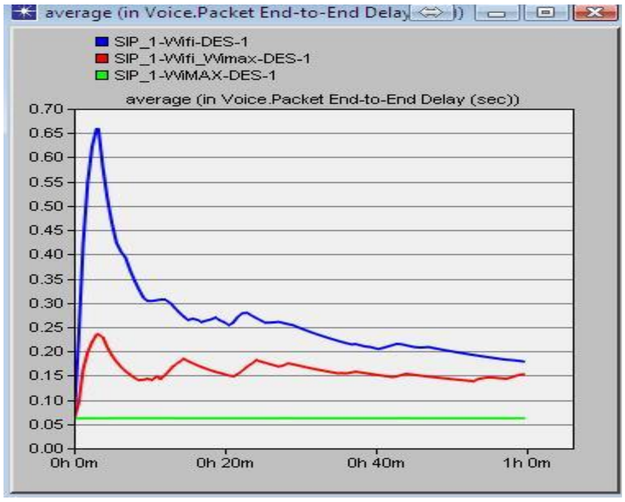

Figure 4: Packet End-to-End Delay. (Top curve is WiFi, middle curve is WiFi-WiMAX, bottom curve is WiMAX)

In terms of the packet end-to-end delay, WiMAX provides a better services than either using just WiFi or WiFi-WiMAX, as shown in Fig. 4. In fact, WiMAX constantly remains in the “Good” range, as outlined in Table 2. Although WiFi observes a high level of packet end-to-end delay at the initial setup phase, it reaches and remains within the “Acceptable” band after the network has converged. The WiFi-WiMAX network remains within the “Acceptable” band even during the convergence period.

VI. Conclusion

The paper has presented the initial findings of traffic through WiFi and WiMAX. Initially two scenarios where designed where both generation and termination of the VoIP calls take place in the homogenous networks such as WiFi and WiMAX. An additional scenario was later added where calls are generated using the WiFi network and terminated using WiMAX networks and vise-versa.

The most thought-provoking finding of our research is regarding the average jitter of the WiFi-WiMAX scenario of not being in-between WiFi and WiMAX. Our research shows that it does not always perform as expected; even, at some points in time, it exceeds that of WiFi.

The MOS of the WiFi-WiMAX network should ideally be somewhere halfway of the WiFi and WiMAX MOS graphs. Our research has found that it exhibits much higher performance than that. Similarly, the packet end-to-end delay of WiFi-WiMAX remains close to that of WiFi and is much higher than expected.

Future work will include other networks covering: GSM, GPRS, EDGE, UMTS (3G), CDMA, LTE and 4G. The effect of handover covering, soft, softer and hard on the network traffic will be of one particular major focus of this ongoing research initiative.

The study will be further broadened to investigate the effect on other parameters such as on the: throughput, queuing delay and the packet drop rate. The reason for the behavior of the network parameters will be carefully scrutinized with the goal of their optimization in order to improve the overall efficiency of the network.

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