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QoS and Objective Performance Analysis of Triple Play Services over ADSL2+ (Asymmetric Digital Subscriber Line 2+)

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Abstract:

This paper will test the ADSL2+ broadband access network that support Triple Play Services (voice, video and data) on last mile. Initial applications presented are entertainment video (video on demand VoD and multicast video IPTV), voice (VoIP), and best-effort data (e.g. web browsing, File sharing /downloading). The QoS guidelines and the objective performance recommendations are presented in this paper in the context of an end-to-end delay, jitter, and delay variation. The networks were tested for the cases when the network availability is 85.5%-93% (services is full at each homes at a time, 7-14.5% traffic in core links) and 25% (services is not full at each homes at a time, 75% traffic in core links) and are agnostic to access technology based on ADSL2+, services architecture, and implementation. As result, the advantages of the ADSL2+ can facilitate the local content management and reduce the content access cost by minimizing traffic loading of the outgoing link.

Keywords: Triple Play Network Architected, ADSL2+, IP Technology (IPTV, VoIP, and Internet), QoS (Quality of Services), Network Availability.

1. Introduction

Over the last few years, obvious that traditional tele-communication companies that have been providing voice and/or data services are being interested in provisioning TV entertainment too [1]. Triple play is a marketing term used for the provisioning of three services: high-speed Internet, television (Video on Demand or regular broadcasts), and telephone over a single broadband connection. High speed Internet includes the bandwidth of 256 kbps (0.256 Mbit/s) or more. In terms of triple play, video is commonly referred to as Internet Protocol Television (IPTV), which describes a system where a digital television service is delivered using Internet Protocol over a network infrastructure. Finally, the voice part of triple play is commonly known as Voice over Internet Protocol (VoIP) or Internet Protocol Telephony (IP Telephony). This service involves the routing of voice conversations over the Internet or through any other IP-based network. [2] Triple play services require broadband with high-bandwidth capability. Since, all broadband technologies were not designed for this usage and, only some support the transmission of data, voices and video. The crucial question is whether DSL, the broadband technology with the largest subscriber base worldwide, can be used for triple play services? Since most DSL plants are of good quality; enabling 512 Kbit's to 90% of subscribers and 6Mbit/s to 60% of subscribers who used ADSL [3]. ADSL2 achieve higher bit rate than ADSL which is include more DSL services with 12Mb/s, the bandwidth or reach can be further increasing with ADSL2+ technology which is up to 24 Mb/s [4]. Therefore, the ADSL2+ network can support triple play services with higher data rate. Another require that will chose for this research is related to QoS which is that not a simple topic. The IP platform is robust and efficient, however is not suitable for the transmission of time-critical data streams. Delay time between packages and package loss are among the most prominent features of an IP network and that caused a problem for the audio and video communication. To prevent these problems will be implemented in higher protocol layers mechanisms to monitor and support the QoS measurement.

The rest of the paper is organized as follows. ADSL2+ technology which is new standardize and recommendation will be discuss in the section2 in this paper and a brief overview of the Triple Play architecture will explain in section 3. Details of the simulation platform and the performance results obtained are discussed in Section 4, 5. Some of the lessons learnt from the simulations are also discussed in this section. We finally conclude in Section 6.

2. ADSL2+ High Speed Interactive Application

ITU G.992.5 is an ITU standard, also referred to as ADSL2+ or ADSL2Plus. Commercially it is notable for its maximum theoretical download speed of 24 Mbit/s. ADSL2+ is capable of doubling the frequency band of typical ADSL connections from 1.1 MHz to 2.2 MHz. This doubles the downstream data rates of the previous ADSL2 standard (which was up to 12 Mbit/s), see fig.(1), and like the previous standards will degrade from its peak bitrate after a certain distance. [5]

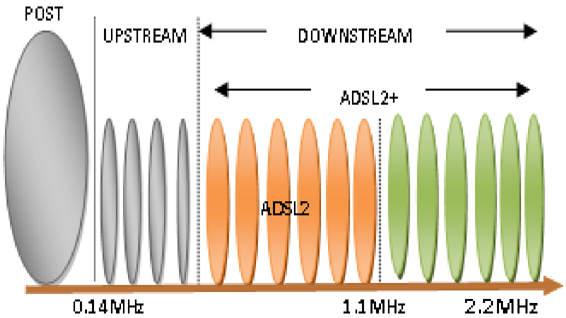


Figure (1): ADSL2+ doubles the bandwidth used to carry data. [6]

Furthermore, ADSL2+ and VDSL2 (Very-High-Bit-Rate Digital Subscriber Line 2) data rates make it possible to easily integrate voice; video and data services over a single telephone line, commonly denominated Triple Play Services see fig.(2). With all these technological developments, it is now practical and economical to simultaneously provide multiple standard and high-definition television channels (SDTV and HDTV) to the residential user. The latest technologies to emerge from the DSL family are ADSL2++. ADSL2++ is still in its infancy and is not yet supported by an appropriate standard. ADSL2+ however, is standardized and allows transmission of sufficient bandwidth for some video, voice and data services, over greater distances than VDSL, without the need for DSLAM relocation. As a result ADSL2+ is becoming the upgrade path for operators wishing to improve upon their standard ADSL service offerings. [7]

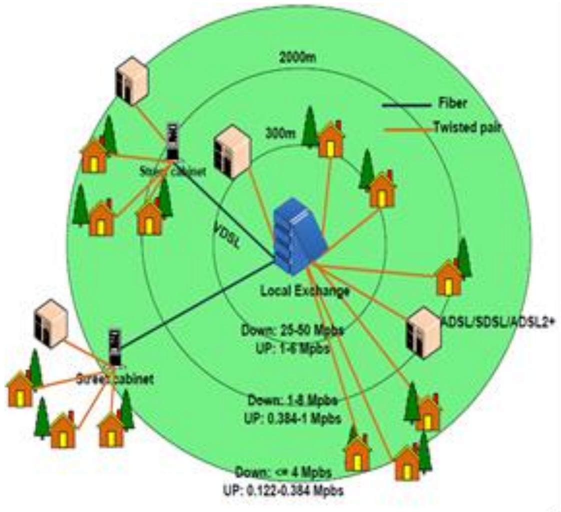


Figure (2): Network architectures for various forms of xDSL; note the xDSL bandwidth is dependent on distance from the local exchange/central office or the remote street cabinet. [7]

3. Triple - Play Architecture Overview

A Triple Play Architecture Overview (see Fig.3) consists of the head-end (Service Provider Network), the core network, the access network where end users reside and the equipments at the subscriber's home (i.e. CPE, DSL modem , the IP set-top box and so on). A Triple Play solution can distribute 50 to 150 TV channels over an IP network with voice over IP and high-speed Internet. Services (video, voice, and data) are sent from the IP head-end using an IP core network over an optical backbone network to the central office (CO). The CO relies the data to the access network (AN) in which digital subscriber line access multiplexers (DSLAMs). There we have distribution of the service (i.e. video or voice) which afterwards enters the subscriber’s house through the modem. For the video service in particular, an IP set-top-box (STB) is also used to unscramble the signal and display our movies on the TV set. [1]

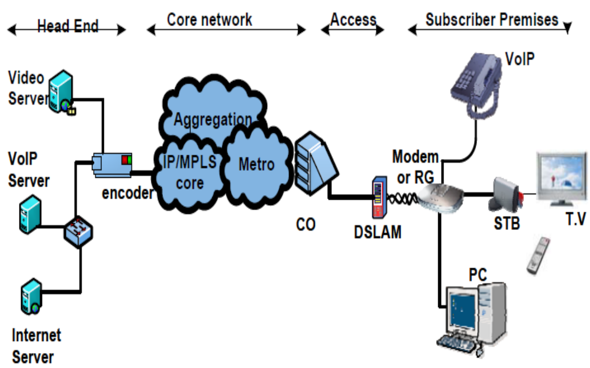


Figure (3): Basic Triple Play architecture

In the following paragraphs we briefly describe the basic components of a triple play network which is shown in figure (3).

The head-end: One or two main data centers house that consist the infrastructure servers, such as NMS, RADIUS, and log servers. Often collocated at these same data centers are servers for VoIP, IPTV, and VoD applications. These servers are connected to switches by use 100BASE-T and 1000BASE-T electrical connections, which are then fed into the core via optical fiber links.

The core network: Core networks are consisted of switching offices and transmission lines that connect central offices. The core network, which is a high capacity fiber backbone network, uses ATM or IP/Ethernet over SONET.

The DSLAM: DSLAMs are a terminate subscriber copper local loops and provide a DSL modulation service to the DSL Modem.

The customer household: is the network's demarcation point between service provider and customer. Services are delivered to the demarcation point, which may be a piece of equipment managed by the service provider called the Residential Gateway (RG), which is the Customer Premises Equipment (CPE). Most CPE are multi-play networks is complex pieces of equipment provided by the service provider that integrate a DSL modem and router. Sometimes only the DSL modem function is used.

STB: STBs take VoD and IPTV traffic coming from the network side and send it to a TV connected via composite, SCART, HDMI, or RF outputs. To let other devices connect to the broadband network, DSL Modem have several Ethernet ports to connect PCs, gaming consoles, Macs, and other devices in the home.

4. Simulated Model Design

In this paper, the OPNET Modeler 14.5 is used to simulate MPEG4 Video traffic with voice and data over the ADSL2+ network. Two scenarios with (85.5%-93%) and (25%) availability were tested and discussed in the following subsection.

4.1 Network Topology:

The network topology of the model is shown in Fig.(4). The whole network architecture is separated into three subnets: Local network, Aggregation network, and Backbone subnet.

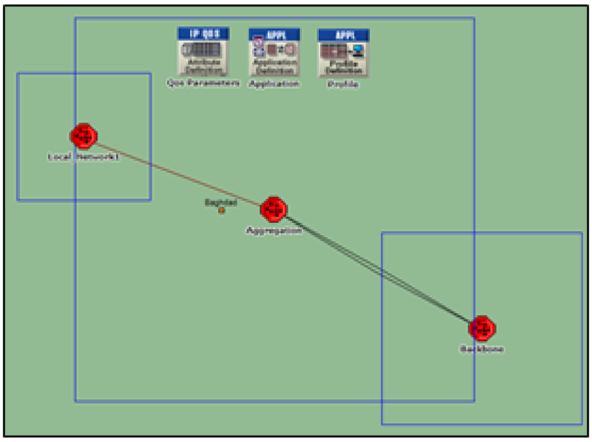


Figure (4): Model of Network Topology

In fig.(5) the "Backbone" subnet is designed with 4 servers configured to stream stored audio and video contents, HTTP and FTP. It contains a 100Mbps IP network and an access routers for both IP Multicast traffic load (R3, RP, and R4) and IP Unicast traffic load (DSR), these routers are connected to the switches (Source and PPP) which are divided traffic into VLAN throughout the network, one of the easiest ways of delivering a triple- or multi-play service is to allocate a single VLAN with multiple modes and allow all customers to share these VLAN. VLAN7 is used with packet traffic. This model can be used to study the basic performance of telecommunication network architecture for multi-service. Then these routers are connected to the BRAS router at Aggregation subnet through a 45 Mbps Digital Signal (DS3) wide area network (WAN) link. The approximate distance between the backbone subnet and Aggregation subnet is 14km, which corresponds to approximately 46.67ms propagation delay.

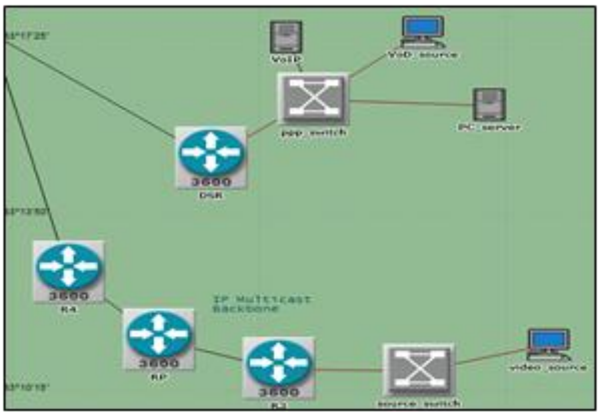


Figure (5): IP Backbone Subnet

In the "Aggregation" subnet, metro Ethernet network with Gigabit Ethernet links the capacity of this link was set to 1Gbps is designed when a switched layer between the DSLAM and the BRAS provided see figure (6).

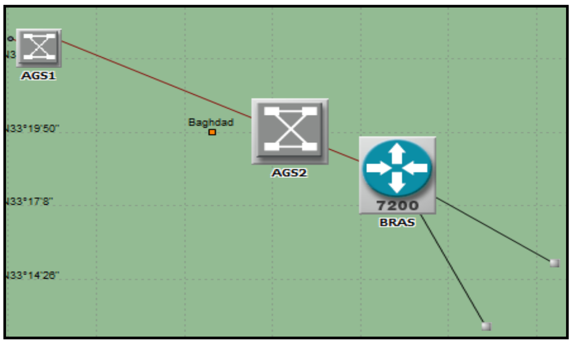


Figure (6): Aggregation Subnet

In fig.(7) see Local network subnet, a 10 Home_Network's subnets are integrated into DSLAMs over twisted pair links with data rate (downlink 24Mbps / uplink 6Mpbs). DSLAMs are also Layer2-aware and take Ethernet frames from the customer side and forward them to the BRAS on the network side. Each Home_Network see fig.(8) provided Residential Gateway (RG), which is the Customer Premises Equipment (CPE), connected to the RG one Set-Top Boxes (STB). Other devices in the home such as PC and Phone are connected to the broadband network by several Ethernet ports which provided by R router via 100baseT links.

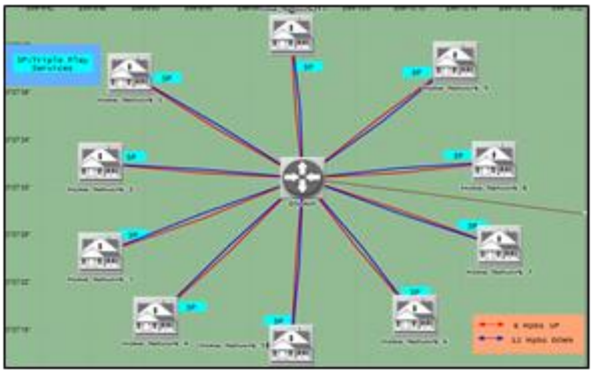


Figure (7): Local_Network Subnet

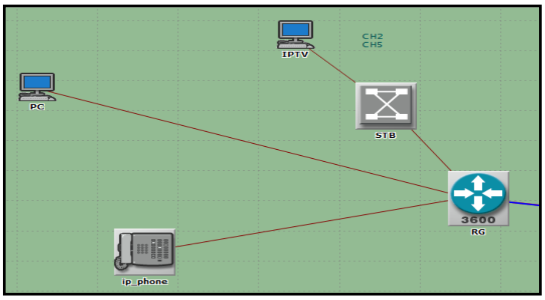


Figure (8): HOME_Network Subnet

4.2 Network Configuration Parameters:

Each object in the Triple Play Model (server, node, application and links) has a specific set of parameters. In general those parameters can be classified as follows:

Application Parameter: Application Attribute definition is used to specify/choose the required application among the available applications such as FTP, HTTP, Video, Voices, and Print etc. It is possible to create a name and give the relevant description in creating new application. In this paper define the following application.

- Voice Application: VoIP is a technology that allows users to make telephone calls over an IP data network (Internet or Intranet) instead of traditional PSTN (Public Switched Telephone Network). Therefore; VoIP provides a solution that merges both data and voice which gains benefits include cost savings, high quality and value added services. Voice application was simulated by setting up a VoIP application characteristic shown in table 1 between residential users and an Ethernet server located in the IP backbone network.

Table 1: Residential VoIP Application Characteristics

Encoder Scheme	Voice Frame Per Packet	DSCP	Compression Delay(s)	Decompression Delay(s)
G.711	1	EF	0.01	0.01

- Web browsing application was defined with different characteristics in table 2 for residential users. Residential users browse the internet to find images, socialize in Face book etc., search for information or even read an electronic version of a newspaper.

Table 2: Residential HTTP Application Characteristics

Http Specification	Object Size(Kbytes)	Number of Object (Object Per Page)	DSCP
Http 1.1	Constant(1) Media Image (0.5-2) Large Image (2-10)	1 40 5	BE

- File sharing /downloading: In order to simulate users downloading and sharing files, we set up an FTP application with the characteristics in table 3. Users request to download a file of 5MB, in random time windows, from a server which is located at the ISP's (in backbone network). Unfortunately it was impossible to simulate FTP service using a bigger file (e.g. 600MB), since then the simulation time would increase a lot.

Table 3: Residential FTP Application Characteristics

Download	Size File (MB)	DSCP
100%	5	AF13

- IPTV is (Internet Protocol Television) is a platform which delivers Internet TV, exploiting the architecture of an IP network, instead of radio frequency broadcast, satellite signal and cable television. For standard definition signal (SDTV), IPTV requires 2-4Mbps data rate, while high definition (HDTV) requires 7-9Mbps data rate using MPEG-4 coding. To simulate IPTV video, we use the video traces provided by Arizona State University can be shown in Table 4 [8], and an import video to OPNET using the instructions is given in [9]. After importing the video traces, we create the application configuration and profile configuration, which allows a server to provide streaming video services to the clients.

Table 4: IPTV Application Characteristics

Parameters	Matrix III
Resolution	**352x288**
Codec	MPEG-4 Part 2
Frame Compression Ratio	47.682
Minimum Frame Size (bytes)	8
Maximum Frame Size (bytes)	36450
Mean Frame Size (Bytes)	3189.068
Display Pattern	IBBPBBPBBPBB
Transmission Pattern	IPBBPBBPBBIB
Group of Picture Size	12
Frame Rate (frames/sec)	**25**
Number of Frames	180,000
Peak Rate (Mbps)	7.290
Mean Rate (Mbps)	**0.637**
DSCP	**AF33**

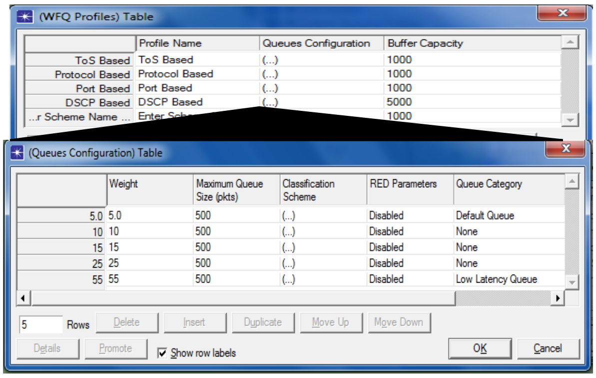
- Video on Demand (VoD) services are much more difficult to engineer into the network given the uncertainty of when and how many users would request this service simultaneously in table 5 shown VoD application characteristics that delivered to each user.

Table 5: VOD Application Characteristics

Frame Rate	Incoming Frame Size (KB)	Outgoing Frame Size(KB)	DSCP
30fps	34560	34560	AF41

Profile Parameter: Profile Attribute definition will be used to create user profiles, these profiles can be specified on different nodes in network designed to generate the application traffic. While configuring profiles applications that are defined in the application configuration are used. Many profile in this paper were used for the simulated model and these profiles are (ftp profile, http profile, voip profile, iptv_ch1 profile, iptv_ch2 profile, iptv_ch3 profile, iptv_ch4 profile, iptv_ch5 profile, iptv_ch6 profile, vod_1 profile, vod_2 profile, vod_3 profile).

IP QoS Parameter: The QoS Attribute Configuration object defines the following technologies: CAR, FIFO, WFQ, Custom Queuing, and Priority Queuing. Each technology contains a table in which each row represents one queue. Each queue has many parameters such as queue size, classification scheme, RED parameters, etc. To the author's knowledge there are only two research group that have tackled the area of QoS in triple play and publishing there result in [1] and [2], examined they performance of common packet scheduling techniques (PQ, WFQ and WFQ – LLQ) and they found the solution with the use of WFQ – LLQ scheduling techniques. Therefore, in this paper we set queue priorities with Weight for WFQ and Low Latency Queue (LLQ) profile. Fig.(9) shows adequate bandwidth is allocated permanently to the voice service (Queue 5, which is configured to be a LLQ) to providing high QoS for the subscribers through constant low packet latency. Also to guarantees that the Internet traffic flows are never excluded from the bandwidth usage (Queue1, which is configured to be a Default Queue). If the LLQ is empty, other queues are serviced according to the regular Weighted Fair Queuing mechanism based on their "Weight" attribute settings. In this scheme, lower weights are served first. We set the weights for FTP, Video and Voice are 10, 25 and 55 respectively. The buffer capacity is set in OPNET to be 5000. The Voice, Video and FTP traffic are set up in OPNET to be of a DiffServ Domain which is a set of routers implementing the same set of PHBs. The DiffServ PHB class selector offers three forwarding priorities (Expedited Forwarding PHB, Assured Forwarding PHB and Best Effort PHB). In the previous subtopic "Application configuration" attribute define for each application the type of services which is shown we set DSCP value with priority (HTTP/BE, FTP/AF13, IPTV/AF33, VoIP/EF and VoD/AF41).


(WFQ Profiles) Table

	Profile Name	Queues Configuration	Buffer Capacity
ToS Based	ToS Based	(...)	1000
Protocol Based	Protocol Based	(...)	1000
Port Based	Port Based	(...)	1000
DSCP Based	DSCP Based	(...)	5000
...r Scheme Name ...	Enter Sch...	(...)	1000

(Queues Configuration) Table

	Weight	Maximum Queue Size (pkts)	Classification Scheme	RED Parameters	Queue Category
5.0	5.0	500	(...)	Disabled	Default Queue
10	10	500	(...)	Disabled	None
15	15	500	(...)	Disabled	None
25	25	500	(...)	Disabled	None
55	55	500	(...)	Disabled	Low Latency Queue

Figure (9): IP QoS Configuration Parameter

4.3 Scenarios:

This paper provides two example scenarios. The execution process is for 500 second for each scenario to enable the OPNET simulator to simulate all the underlying layers.

The objective of these scenarios is to compare the performance of Triple Play applications over ADSL2+ in both case (93%-85.5%) and (25%) availability for access broadband networks at end users. In the first scenario when the network availability is 93%-85.5% (7-14.5% traffic in core links), many IPTV, data and voice connections are supported simultaneously to the 10 ADSL2+ household with guaranteed QoS. An immediate question is how many IPTV, data and voice connections can be supported simultaneously with guaranteed QoS in the case of previous scenario, this case however is not realistic and should be ignored. In other words, the second scenario used to examine how to efficiently utilize limited bandwidth network resources to provide Triple Play Services with guaranteed QoS which is a challenging problem. The arrangement of objects and the parameters of this scenario is the same as in the previous one but the difference is related to increased number of user to three time than in scenario one (40 ADSL2+ Home_network) but not all used full services at a home simultaneously, this case can be supported in OPNET Modeler by for example allocated only voices and data services to home and other home with video only which is shown in fig.(10). The network's performance was observed in this scenario by assuming that all network links had 75% load (25% availability) simulating high utilization in the core network.

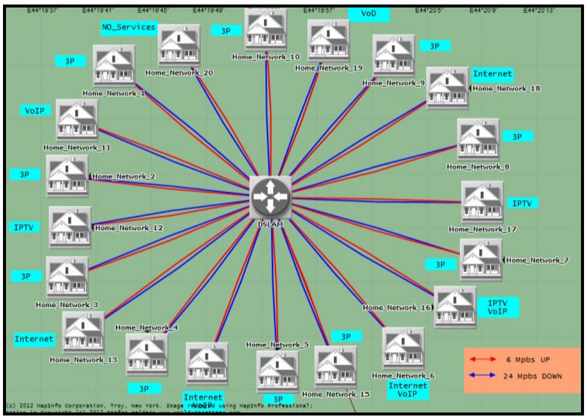


Figure (10): The numbers of the users change in scenario 2.

5. Results – Evaluation

This section introduces the results for the simulation scenarios presented in the section 4.3. Detailed results of each simulation scenario with analysis and discussion in the following sections we will display.

- IPTV and VoD are much demanded services in terms of bandwidth and latency. As we can observe, when there is available bandwidth 85.5%-93% in all networks, end-to-end packet delay is low averaging 60msec. However, when the network availability decreases, end-to-end delay increased, reaching a peak value of approximately 110msec which is shown in the fig.(11). For the IPTV and VoD services, packets must be arriving in order and within an acceptable time window see fig.(12), Loss is depicted as deviation from the blue line representing 30pkts/sec VoD when availability 25% (75% traffic load in core link) which is not large enough to prevent dropped downlink packets at customer side.

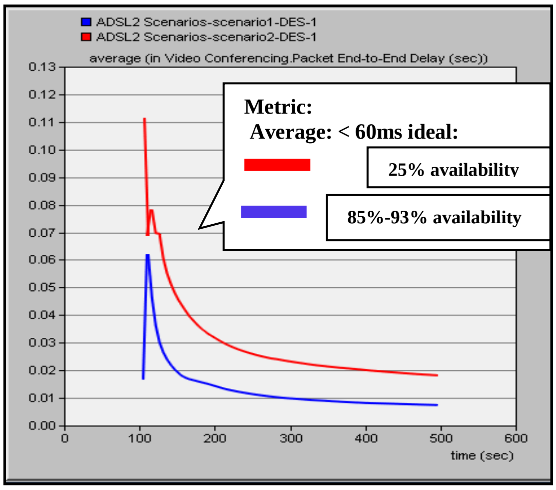


Figure (11): Video Packet Delay

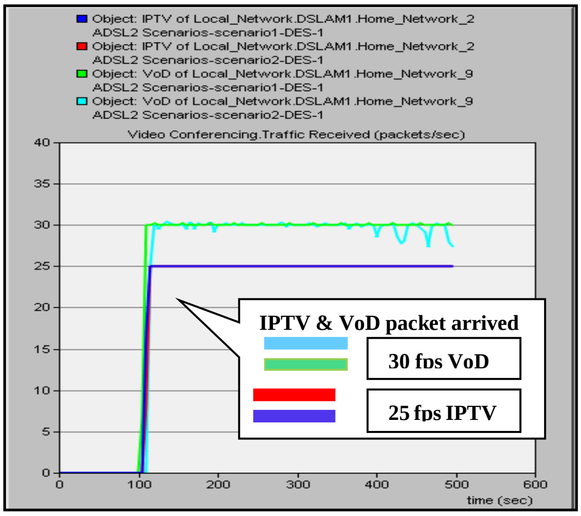


Figure (12): Video Packet Received

- VoIP application is very sensitive to jitter and end to end delay since it is a real time application. Various providers have specified a maximum jitter their (SLA 30ms maximum jitter) and total end to end delay must be 150 ms [10, 11] in order for conversation to be of acceptable quality. As it is observed in figs.(13 & 14), Jitter is zero with 25% availability but in case of 85%-93% availability with 10 homes user values are increased to 0.0035ms then drop to 0ms. End to end packet delay is stable at 41ms in both scenarios this mean though the numbers of the users change dynamically during the simulation scenario, still quality of voice is more acceptable since voice services is not sensitive to the bandwidth. ADSL2+ is mostly suitability to VoIP application; G.711 is the international standard for encoding telephone audio on a 64 kbps channel [12], [13]. The average throughput of voice call that use pulse code modulation (PCM) scheme with G.711 encoded data streams must equal 64Kpbs in fig.(15) shown these result.

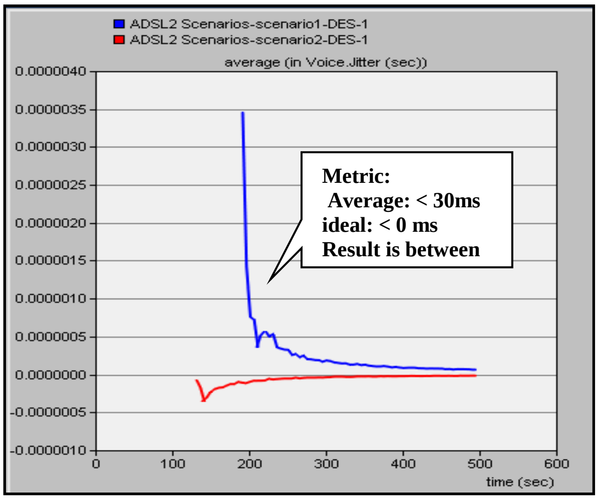


Figure (13): Voice Packet Jitter

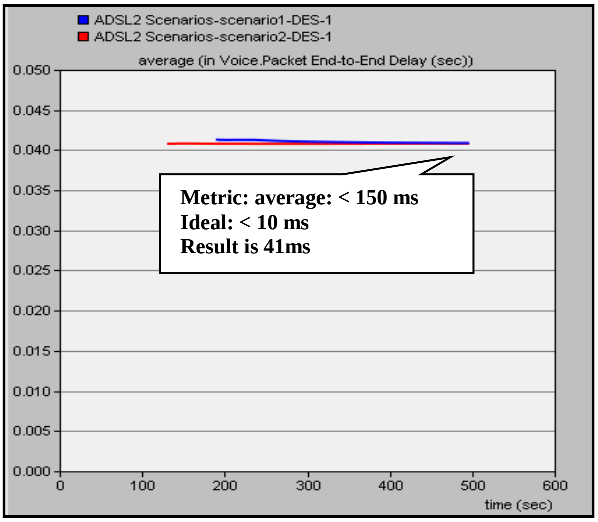


Figure (14): Voice Packet End to End Delay

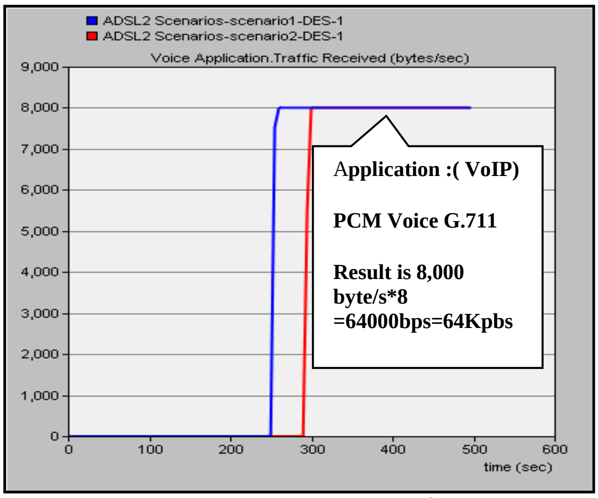


Figure (15): Average Throughput for Voice Packet

- FTP OPNET application profile was used to simulate a peer to peer connectivity or an application such as ITunes. According to the simulation results, a 5Mb file requires approximately 10 sec to be downloaded from a user, when the network is 85.5%-93% available. However, this value may be even doubled depending on the network availability. Also, average response time is 50% greater than when the network is 25% available see fig.(16). Since however the simulation was run with the FTP server located at the ISP premises, this value will increase, if the FTP server is located within another ISP or especially in peer to peer services where a file is shared among users worldwide.

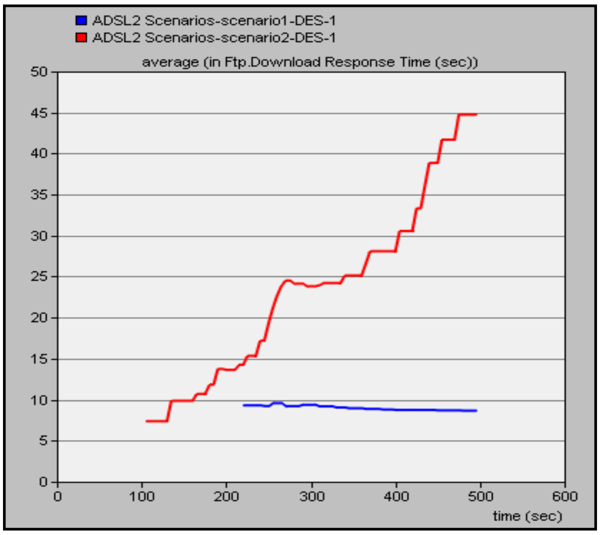


Figure (16): FTP Average Response Time.

- Fig.(17) illustrates the download response time for the HTTP application. As it is observed, an HTTP requires 150msec to be downloaded to the user terminal which performance is excellent for such service. When the network's availability decreases, download response time is increased to 2.2sec, but this will not be taken under consideration by the final user. However, if the server was to serve more users or was located to another ISP's premises, this value would increase more reaching probably a delay value into seconds. We should take under consideration, that HTTP is treated as a low priority application, since a user generally requires the HTTP to reach its destination without error and not to be received immediately by the recipients.

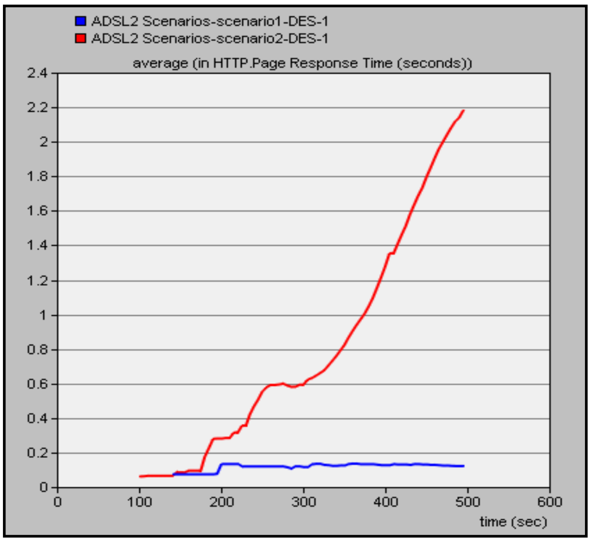


Figure (17): HTTP Average Response Time

- Figs.(18-19) illustrate the simulation results for Ethernet delay and IP packet drop, when the network's availability is 85%-93%, both Ethernet delay and IP packet drop is stable with a value of 0.16msec and 750pkts/sec respectively. However, delay is increased when the network has low availability. Ethernet delay increases when the core link is 75% utilized. The average value of Ethernet delay is almost doubled, while for certain time windows it can be 8 times greater. IP packet drop however increases significantly (4,500pkts/sec). Furthermore, IP packet drop is not stable having variations.

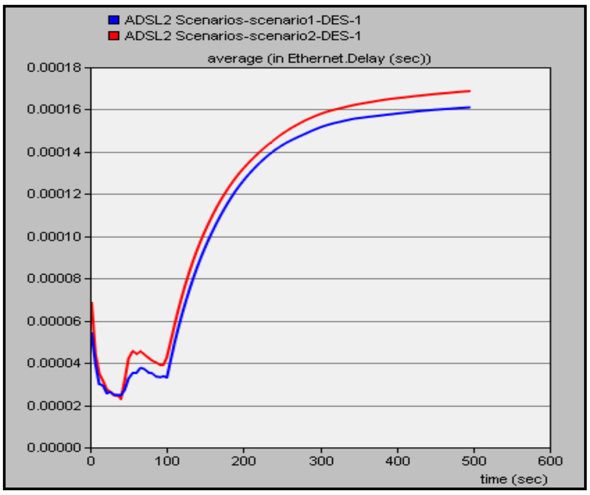


Figure (18): Ethernet delay

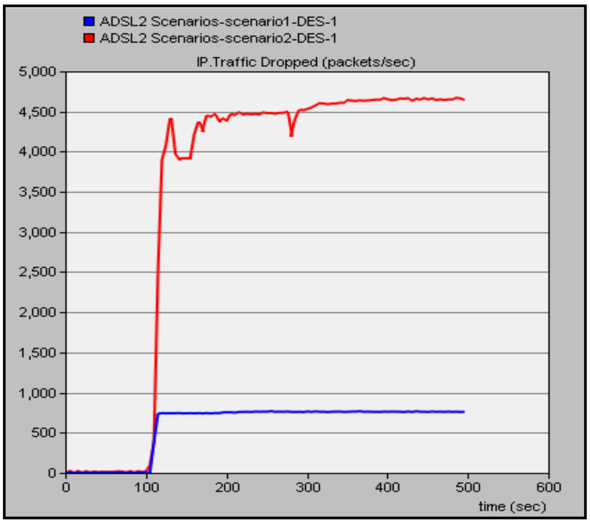


Figure (19): IP Traffic Drop

6. Conclusion

The most important point during this paper is to take up rate of DSL grows to supporting triple play services this is accomplished through design of simulation topology based on commercial triple play network architectures with ADSL2+ by using OPNET simulator. The conclusion can be classified as follows:

1. Using the fact of ADSL2+ that provides numerous improvements, including line diagnostics, power management, power cutback, and reduced traffic. It is found that from the result, the ideal ADSL2+

technology will use for eliminating last-mile bottlenecks, enabling global mass deployment of Triple Play services and allows carriers to efficiently compete with cable providers.

2. The simulation results verified what we expected. That the QoS requirements of voice and video are better satisfied with WFQ technique.
3. It is found that the link congestion must be avoided at all costs, and links should operate at 75% of their total capacity, in order to avoid triple play services degradation, then they should program a network upgrade, in order to increase their network availability and maintain QoS.

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جودة الخدمة و تحليل أداء خدمات التشغيل الثلاثي على ADSL2 + (خط المشترك الرقمي غير المتماثل 2 +)

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الخلاصة:

في هذه البحث يتم اختبار شبكة النطاق العريض ADSL2+ التي تدعم خدمات التشغيل الثلاثي (الصوت, فيديو و البيانات) على الميل الأخير. التطبيقات الأوليه المقدمه من الوسائل الترفيهيه هي الفيديو(الفيديو عند الطلب VoDوالإرسال المتعدد للفيديو IPTV) ,الصوت (الصوت عبر بروتوكول الإنترنت)، وأفضل جهد للبيانات (مثل تصفح الإنترنت، وتبادل الملفات / تحميل). تعرض المبادئ التوجيهية والتوصيات لجودة خدمة الأداء في هذه الورقة في سياق تأخير نهاية إلى نهاية، غضب، والتباين تأخير.ثم يتم اختبار الشبكه لحالات عندما توفر الشبكة 85.5٪ - 93٪ (الخدمات كاملة في كل المنازل في وقت واحد، وحركة المرور 7-5،14٪ في الروابط الأساسية) و 25٪ (ليست الخدمات كاملة في كل المنازل في الوقت نفسه، وحركة المرور 75٪ في الروابط الأساسية) والمقصد هو للوصول إلى تكنولوجيا ADSL2+ بلاعتماد على عمارة الخدمات التشغيل الثلاثي، وتنفيذها. نتيجة لذلك، يمكن للمزايا + ADSL2 تسهيل إدارة المحتوى المحلي وخفض تكلفة الوصول إلى المحتوى من خلال تقليل حركة المرور التحميل من الرابط.